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Electronic Musician

MARCH 2007

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Taxi's Michael Laskow Peers into the Future

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Reviews

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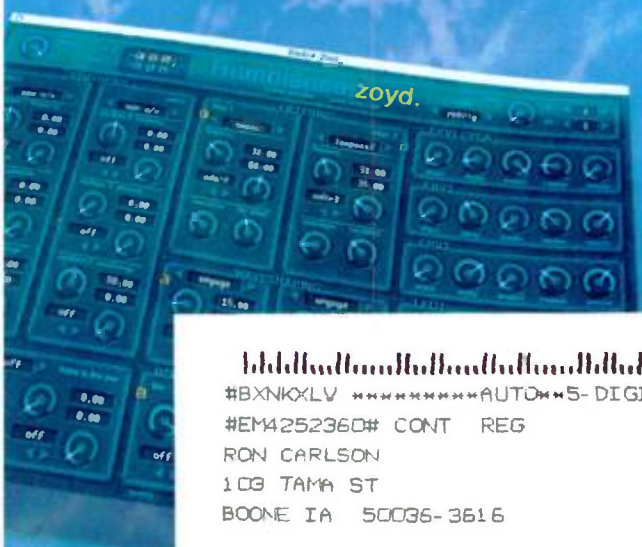

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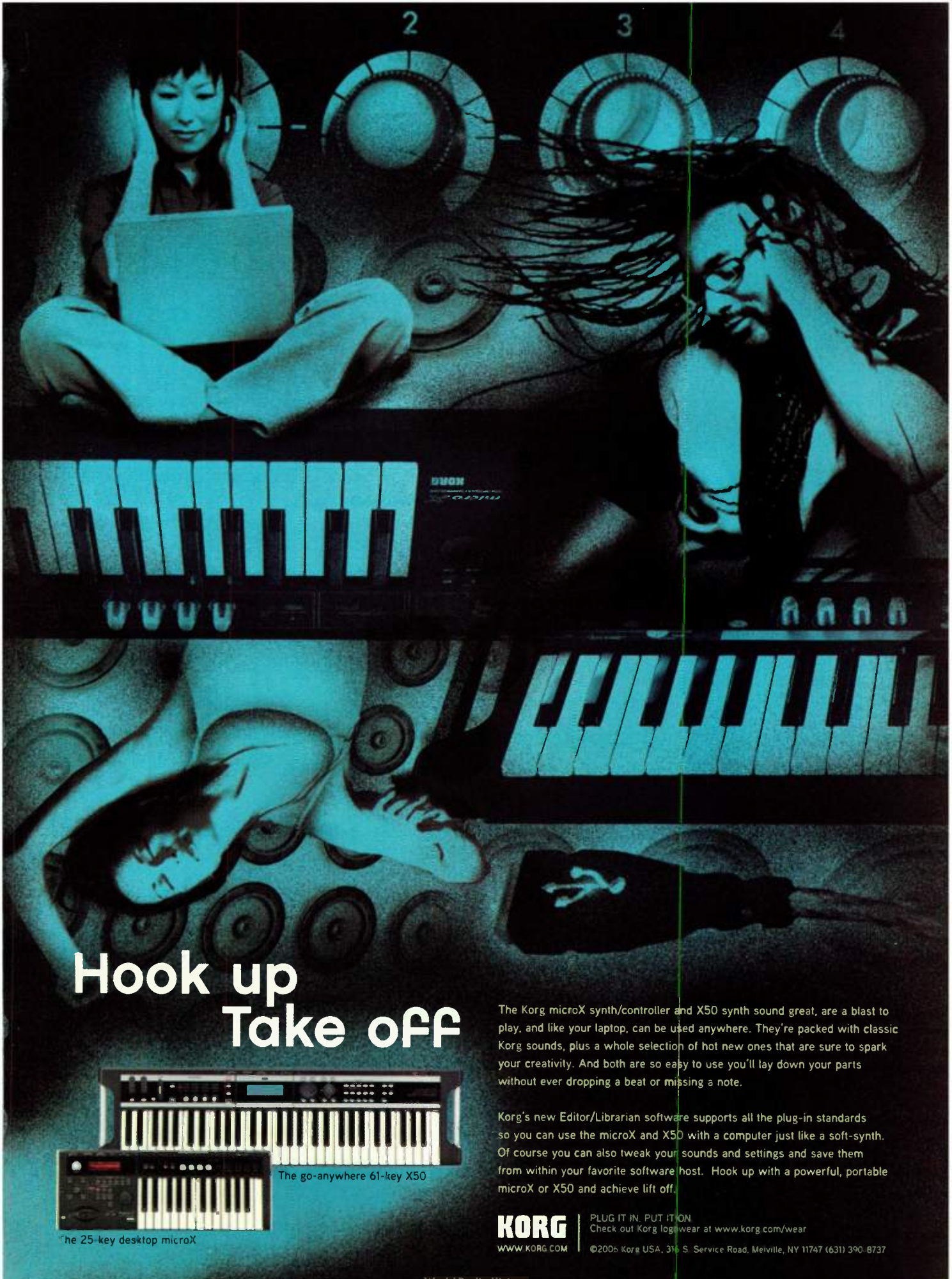

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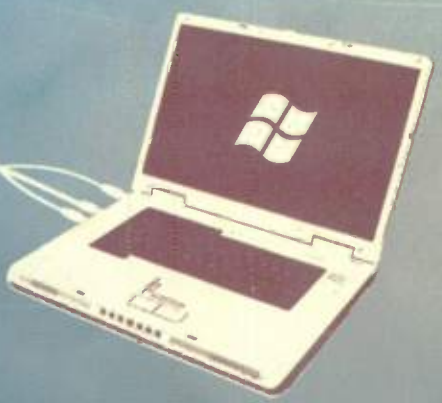

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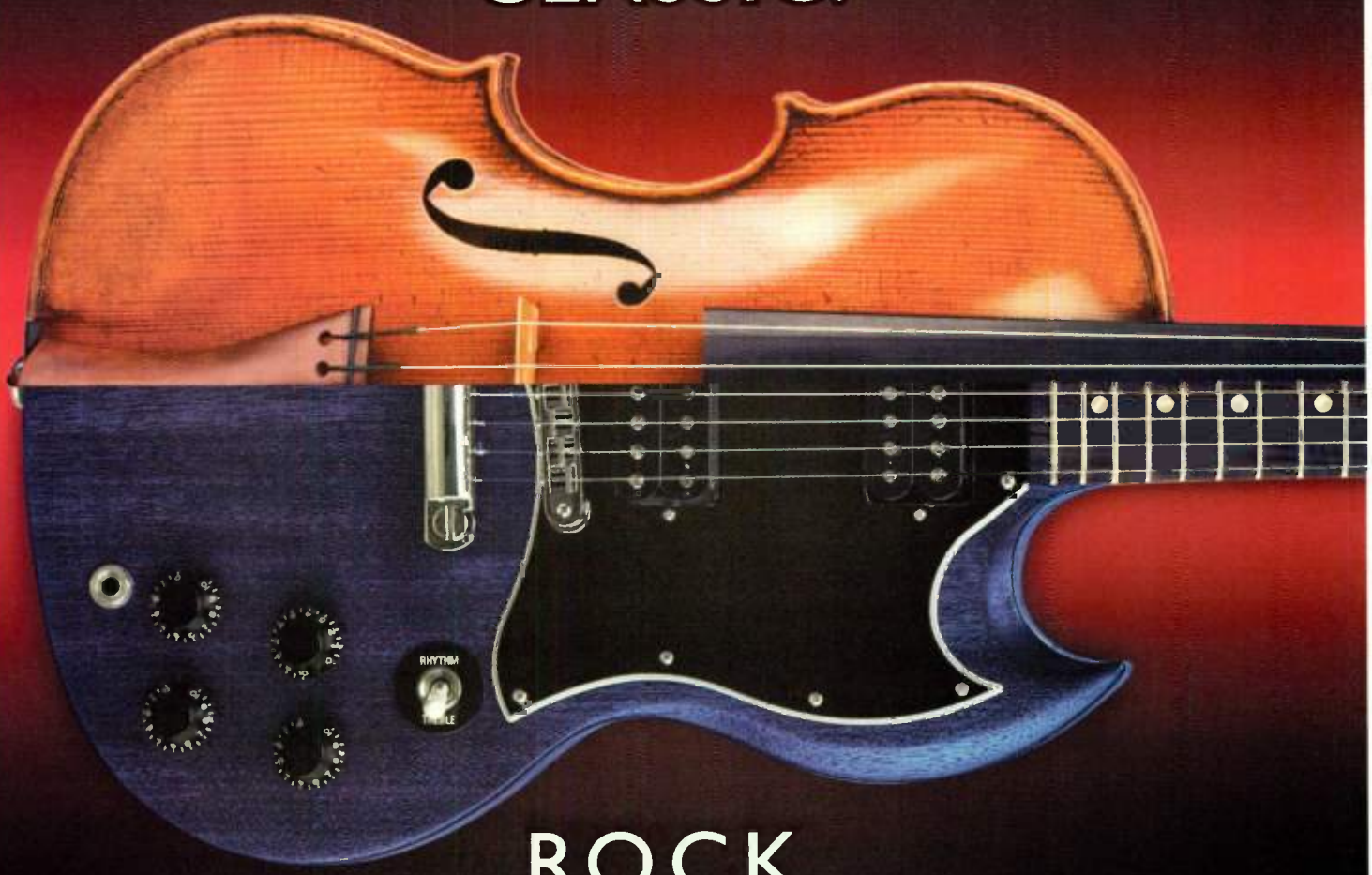

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Electronic Musician
March 2007

INSIDE

COVER STORY

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FEATURES

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Heil Sound PR 40 dynamic microphone

Audio Engineering Associates (AEA) The Ribbon Pre (TRP) microphone preamp

Arturia Analog Factory 1.2 (Mac/Win) software instrument

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E-MU

Podcasting Bread on the Waters

I've written a lot in the past few months about our wealth of new EM projects, including print publications, e-newsletters, and our Web redesign. I'm not sure how we manage to publish so many good products, given the size of our staff and budget, but somehow we keep creating projects, so I keep announcing them!

Among the latest EM products is a monthly Podcast, produced by Senior Editor Mike Levine. Called "EM Cast," it includes product news by Associate Editor Geary Yelton and an introduction to the upcoming print issue by yours truly. But the truth is, if that's all we were delivering, we wouldn't bother. After all, we give you product news in our eMusician Xtra e-newsletter, as well as in EM's "What's New" department. Nevertheless, it's nice to hear a human voice delivering the news, and in some cases, "EM Cast" goes into a bit more depth than print can.

JULIE BIRUM

What makes "EM Cast" very cool are its feature and "Talking Tech" interviews, both conducted by Levine. The debut "EM Cast," which came out this past December, focused on producer Ted Perlman, who has worked with the likes of Burt Bacharach, Bob Dylan, and Elton John. In the "Talking Tech" segment, studio tech–computer consultant Sergio Samayoa gives out some studio-setup tips. The January edition of "EM Cast" looks at New York–based producer-engineer Bob Power (whose credits include Erykah Badu, D'Angelo, and the Roots), while "Talking Tech" discusses how Zenph Studios digitally resurrects great piano recordings of the past. None of these interviews appears elsewhere; they are exclusive to "EM Cast." To check it out, click on the EM Podcasts link on our Web site at www.emusician.com.

We have also released a new print product designed specifically for musicians who are interested in using Steinberg's Cubase 4.0 digital audio sequencer. *Personal Studio Series: Mastering Cubase 4* is the second volume in a series that began with last year's popular volume about Digidesign Pro Tools LE 7. Like the Pro Tools volume, the new Cubase 4 edition includes sets of step-by-step instructions, drawn from Thomson Course Technology's newest book about Cubase 4, which is on how to use key features of the program. The text has been tweaked by the EM staff, and the EM art department has added color graphics that surpass those found in the book.

As in the previous volume, we have included a slew of short interviews with noteworthy producers and engineers, as well as a wealth of supplementary EM stories (some new, some reprinted from past issues) that go well beyond Cubase coverage. Many of these features give tips on how to use virtual instruments that will work with any host program that supports VIs, not just Cubase. So even if you aren't a Cubase user, you might want to check out *Personal Studio Series: Mastering Cubase 4*. It's available at our online bookstore, Mixbooks (which you can access from our home page), and at bookstores and newsstands that carry EM.

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Editor in Chief

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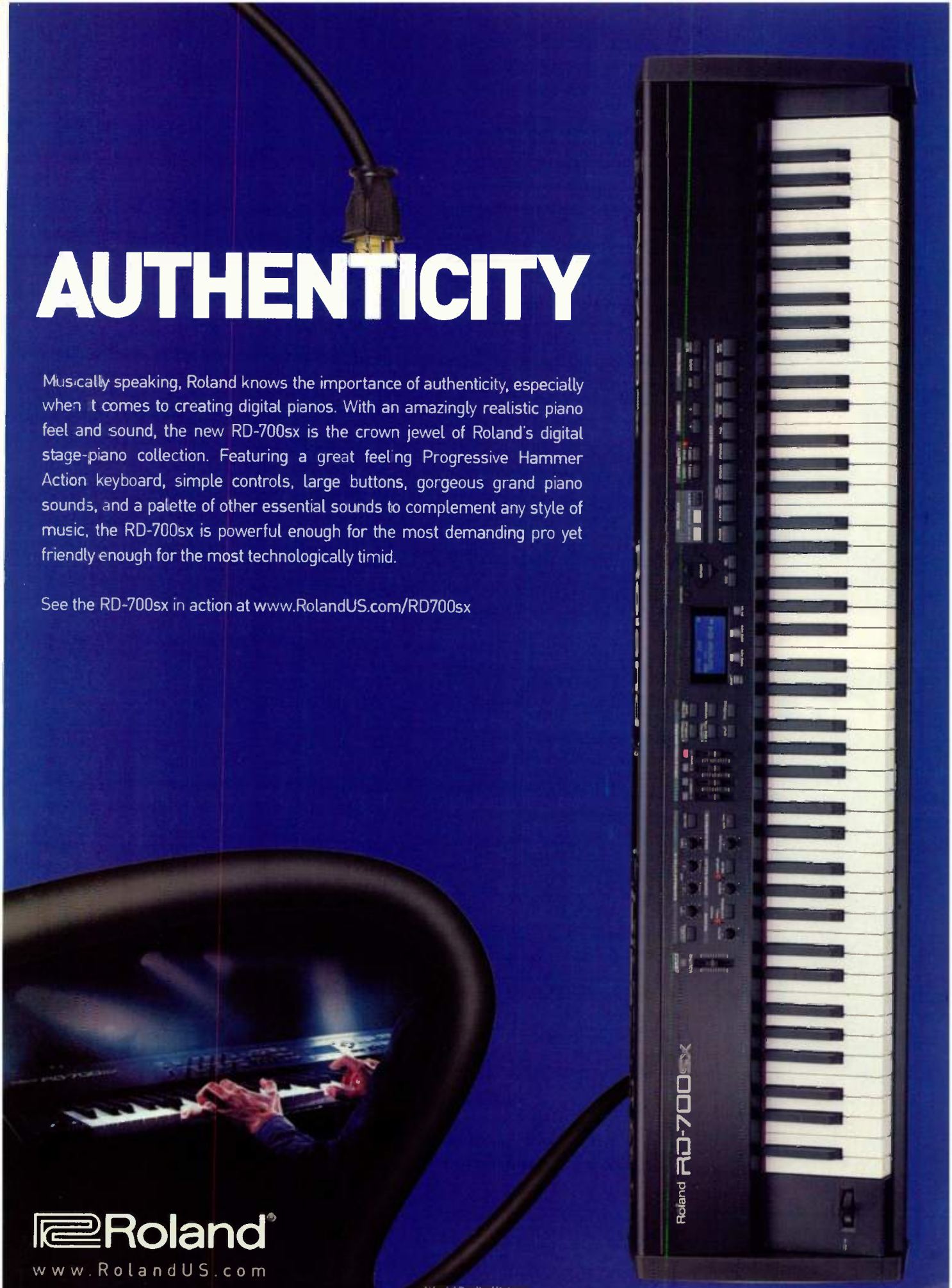

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LAURA WILLIAMS

Letters

Listen to This

Charles Dye's article on how to monitor a mix is brilliant (see "Monitoring for Success" in the December 2006 issue of EM). As he points out, staying in the control room sweet spot for too long can fool even the best ears. Long ago I realized that I could give myself a valid second opinion on how a mix was going by leaving it playing with the door open, going outside, and then noting my first impression as I came back in along the corridor to the control room. Charles identifies with this and several other techniques that cost very little (or nothing!) but give huge gains to the success of a mix. In these days of plug-in mania, it is good to be reminded that listening is what makes a mix sound great.

Richard Hallum
Music and Audio Institute of New Zealand

Mic the Amp

I wanted to chime in regarding Michael Cooper's feature article "Six-String Strategies" in the December 2006 issue of EM.

First, I found the insight very helpful, especially regarding the engineer's preference for single microphones when recording acoustic guitar. Second, I wanted to add an often-overlooked method for recording acoustic guitar, one that I tend to employ in my less-than-desirable home-studio space, especially since the room I record in is not treated and is not flattering in terms of ambience and reverb decay. That option is to play the acoustic guitar through an amp and then mic the amp.

Certainly, there are immediate naysayers to this option, but I wish to remind all that this is exactly the method employed on the first several Beatles albums. In fact, I practiced this method only after reading through Brian Kehew and Kevin Ryan's wonderfully in-depth book *Recording the Beatles*. Commenting on the recording of the Beatles' first efforts, Kehew and Ryan wrote, "John and George's A.C. Vox 30 amps are mic'd with Neumann U47 microphones. John and George both play their Gibson J-160E acoustic/electric guitars plugged in but not mic'd (though a bit of John's would leak into the vocal mic" (p. 355).

I own a Vox AC30 and I use either a Røde NT2000 or the smaller AKG C 1000 to mic it, and I get very satisfying results. For spaces with acoustic ills, this provides a certain relief while retaining the feel and sound of the acoustic guitar.

Shelby Smoak
Carrboro, North Carolina

Isn't It Grand?

In the October 2006 issue of EM, you compared virtual pianos to see which one sounds and plays the best (see "Software Eighty-Eights"). Although Synthogy Ivory and Native Instruments Akoustik Piano were the standouts, Akoustik was more appreciated because of its Steinway (it was even chosen for the 2007 Editors' Choice Awards). I have both Ivory and Akoustik in my studio, and I can say that I've come to the same conclusion myself.

However, I now also have the Italian Grand Expansion Pack for Ivory that came out just after the October issue. I spent a whole evening testing and comparing its Fazioli to Akoustik's Steinway, and after much debate, the Italian Grand came out as my preferred piano. The sustain feels more natural, and the Velocity and various parameters give me better playability. We shouldn't forget that Ivory allows much more tweaking, something the October issue didn't really consider when making "as few parameter adjustments as possible." Although Akoustik has a simpler interface, Ivory allows for much more control, so it becomes an advantage once you've gone through its interface. It didn't matter before the Italian Grand came out, but now it does!

André Favreau
Montreal, Quebec

Baffled by Samples

As a writer whose credits include more than a dozen articles and reviews in the pages of *Electronic Musician* (including a feature article on sampling grand pianos), I believe I have earned a certain amount of credibility with EM readers. When your virtual grand "shoot-out" appeared in the October 2006 issue, I had not had the opportunity to hear all of the libraries. Thus, like all the other EM readers, I was interested in the results. Now I'll be the first to admit that every musician has his or her own standards by which all samples are judged. Beyond that, I think we'd all agree that certain pianos

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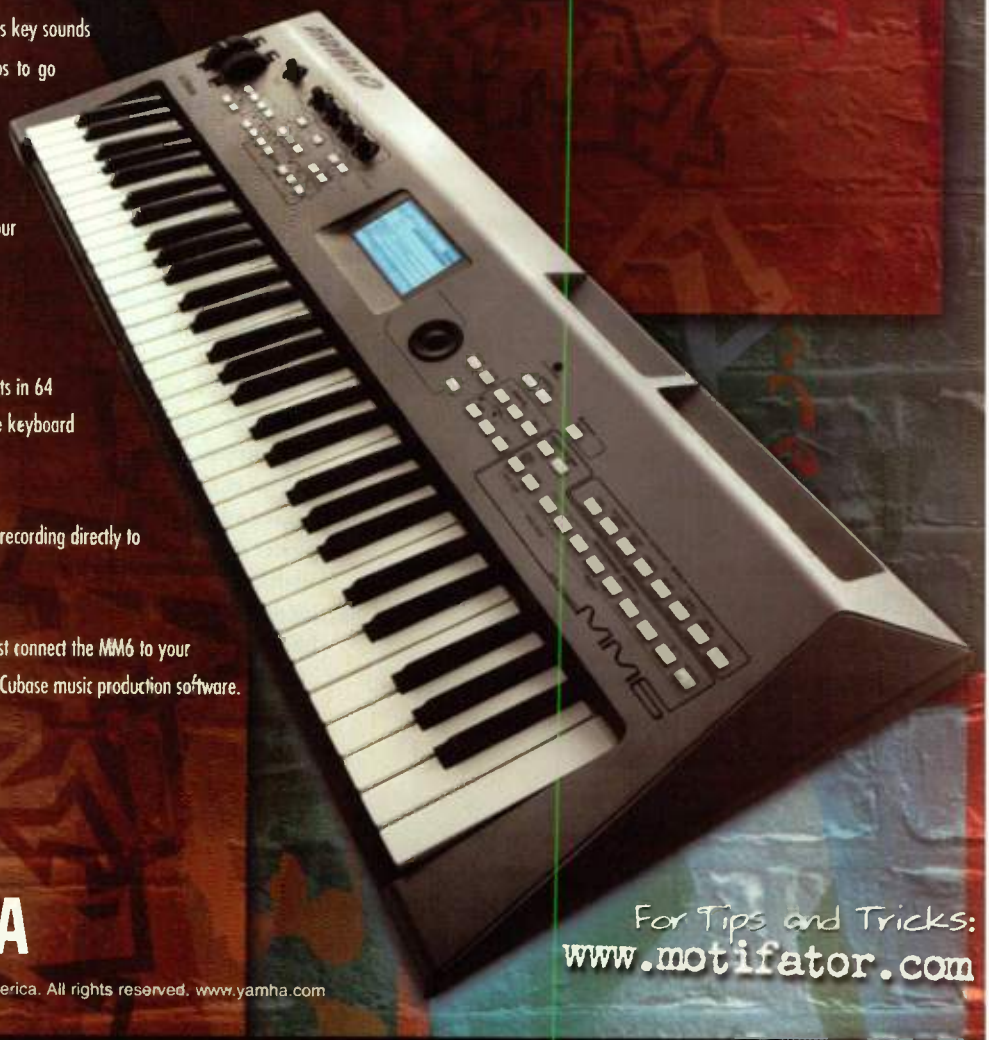

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Next Month in EM

COVER

FEATURES

USB Microphone Roundup
We compare the sound and features of several USB mics from Blue, Marshall, Røde, Samson, SE Electronics, and more.

Audio for Mobile Games
Produce sound for cell-phone games with this start-to-finish guide.

Production Values: Jerry Harrison
We discuss Jerry Harrison's post–Talking Heads career, focusing on the recent production work done in his personal studio.

COLUMNS

Making Tracks: Slice, Move, and Glue
We show you how to chop up and quantize multimiked live drums with Apple Logic.

Sound Design Workshop: Machine Tools
Use EM's tips and techniques to navigate SFX Machine Pro.

Square One: How Your DAW Does Math
Does your DAW treat audio as floating-point or as fixed-point numbers? What's the difference, and should you care?

Music Business Insider: Michael Petricone
We talk with Michael Petricone, VP of the Consumer Electronics Association, a trade group that has taken a strong stance against the Recording Industry Association of America's attempts to restrict digital technology.

. . . and much more

Letters

> Staying in the control room sweet spot for too long can fool even the best ears.

are more appropriate than others depending on what's being played. Who would judge a piano used to play roadhouse blues by the same standards as an instrument meant for classical applications?

All that aside, I would expect that EM would be particularly careful about choosing an overall "best piano." However, once I got my hands on those same virtual grands and had a listen for myself, I had to seriously question your results. While you can usually make a case for certain samples sounding better over different monitoring systems, the bottom line is that I don't believe you've given EM readers sound advice (sorry for the pun). I don't want to beat up anyone's product, as I'm sure a lot of work went into all the packages, but when you play back a sequence performed by a master pianist (in this case, Eric Genuis), it's quite easy to hear which instruments actually sound like a piano being played and not a sample.

One of the pianos in your shootout sounded so bad that I had to wonder if I had a software problem, but other people had heard the same things I had, confirming that indeed these pianos were almost unusable.

Adding to this mystery is just why that particular piano got the 2007 Editors' Choice Award for best sample player. I freely admit, I don't have the legendary golden ears that some musicians possess, but I'm absolutely certain that almost anyone who has ever played a real piano would agree that this one (from a company I have an enormous amount of respect for and whose many other products I use regularly) misses the mark. I expect EM to set the bar high for all soundware developers, but this particular set of circumstances has me baffled.

Jim Miller
Stratus Sounds

Jim—Thanks for your comments. We stand by our conclusions, which were reached in both the article and the 2007 Editors' Choice Awards through the consensus of a group that included performers with a broad range of experience. We felt that all of the pianos covered were worthy contenders.
—Len Sasso

We Welcome Your Feedback

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or email us at emeditorial@emusician.com.
Published letters may be edited for space and clarity.

You're Hired

Thanks for the well-written article about recording studio musicians in Nashville (see "Hired Guns" in the April 2006 issue of EM). A few years ago, a friend of mine who is a singer and songwriter hired a Nashville producer to record his songs, and the results were great. Michael Cooper has demystified the process for the rest of us.

Mark McCarron
Great Neck, New York

Error Log

January 2007, "2007 Editors' Choice Awards," p. 65. The final sentence should have read: "And as of this writing, VSL has just released a major upgrade to its user interface, which should make working with this superb collection even easier." EM

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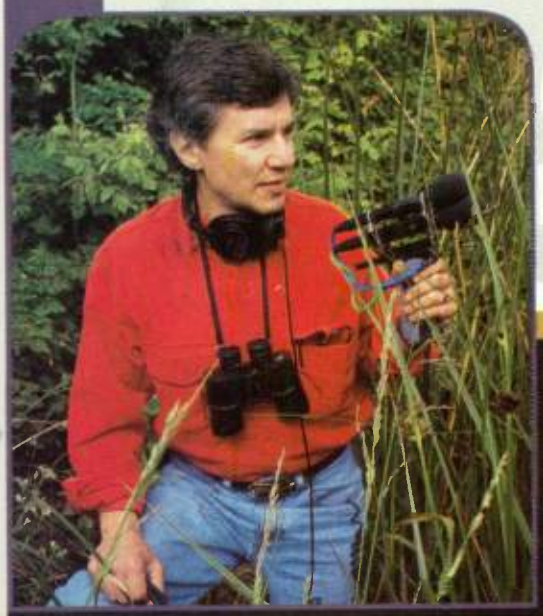

FRED MERTZ

EMspotlight

Bernie Krause Gone Wild

Composer and bioacoustician Bernie Krause points out that the world around us holds many surprises, many of which we would hear and see if we would only listen. He should know, having spent thousands of hours documenting it. In this exclusive interview with *Electronic Musician*, Krause talks about his field-recording techniques, the importance of acoustic ecology, and learning how to listen in the long term. By Gino Robair. emusician.com/em_spotlight

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

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EM Show Report

The 2007 Winter NAMM show is the biggest annual musical-instrument expo in the United States. Visit emusician.com for a report on the exciting new recording gear, music software, and electronic musical instruments unveiled at this year's show.

EM get on the bus

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

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By Geary Yelton

M-Audio MidAir and MidAir 37

MIDI cables are so 20th century, don't you think? When M-Audio (www.m-audio.com) debuted the MidAir 25 wireless MIDI keyboard, the company promised that new wireless systems were right around the corner. Now it's made good on that promise with the MidAir 37 ($299.95) and the MidAir wireless MIDI system ($149.95). Their receivers are identical to the MidAir 25's: both new products operate in the 2.4 GHz band and have an effective range of 30 feet. The pocket-size receiver connects directly to your computer's USB port and is powered by either USB or a 9V DC adapter.

The MidAir 37 is a 37-note keyboard controller that has nine assignable sliders in addition to all the MidAir 25's features, which include eight assignable knobs, MIDI Out, a sustain input, and Velocity sensitivity. The MidAir wireless MIDI system pairs a small MIDI transmitter with the receiver. The 9V battery-powered transmitter has a detachable MIDI input cable for connecting to any device that sends MIDI data. According to M-Audio, it is possible to operate multiple MidAir systems in close proximity. All MidAir products support Mac OS X and Windows XP or later.

Novation XioSynth 49

U.K.-based Novation (www.novationmusic.com) is now shipping the XioSynth 49 ($599.95), a larger version of its XioSynth 25. With 49 semiweighted keys and 7 Velocity curves, the 8-note polyphonic monosynth has 3 audio oscillators, 2 LFOs, and a resonant multimode filter. A function called X-Gator lets you control amplitude level using a 32-step sequencer to create automated rhythmic patterns. For hands-on control, an *x-y* touch pad supplements the spring-loaded pitch-bend and modulation joystick.

The XioSynth is more than a synthesizer. It's a USB 1.1 class-compliant audio/MIDI interface that has an XLR mic input with 48V phantom power, an unbalanced ¼-inch input, two unbalanced ¼-inch outputs, and a ¼-inch stereo headphone output. It also operates as a control surface for software instruments, with 16 MIDI templates to control a variety of software instruments and audio programs. Additional features include 200 rewritable patches, an onboard arpeggiator, a sustain pedal input, and a MIDI Out port. The XioSynth 49 runs off either USB, six AA batteries, or a 9V DC power supply.

Korg MR-1 and MR-1000

Two of the most exciting products announced by Korg (www.korg.com) in 2006 are now beginning to ship. The MR-1 ($899) and the MR-1000 ($1,499) are mobile digital recorders that incorporate 1-bit audio at very high sampling rates. The handheld MR-1 contains a 20 GB hard drive, includes a stereo electret condenser mic, and records at rates up to 2.82 MHz—64 times as high as a standard audio CD. The tabletop MR-1000 doubles those specs with a 40 GB hard drive and a maximum rate of 5.64 MHz. The MR recorders were designed for applications ranging from location recording to archiving final mixdowns and master recordings. Korg's MR recorders support numerous 1-bit formats, including DSD, DSF, DSDIFF, and WSD, as well as 16- and 24-bit BWF at rates as high as 192 kHz.

The MR-1000 has two XLR/TRS combo inputs with phantom power, XLR and RCA outputs, and a ¼-inch headphone output. It runs on AC power or AA batteries. The MR-1 has two balanced mic/line minijack inputs, a stereo minijack output, and a headphone minijack. It runs on AC power or a factory-installed lithium polymer battery. Both recorders have USB 2.0 connectivity and come with AudioGate (Mac/Win), a software application that converts 1-bit recordings to multibit WAV and AIFF formats and vice versa. AudioGate also performs functions such as gain control, fade-in and fade-out, and DC-offset removal.

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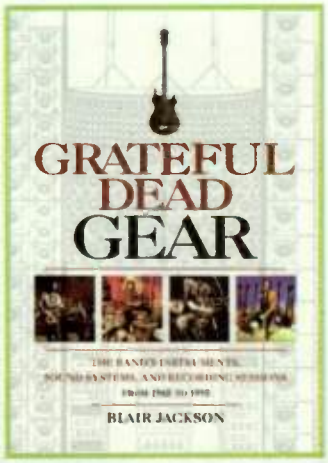
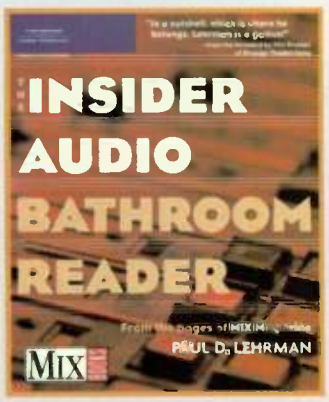
For the inside story on the Grateful Dead's instruments, equipment, and studio sessions, you need look no further than *Grateful Dead Gear* ($34.95), from **Backbeat Books** (www.backbeatbooks.com). Written by Blair Jackson, longtime authority on the Dead and a senior editor of *Mix* magazine, this large-format paperbound book covers the band's history from its humble beginnings in 1965 to its dissolution in 1995. He traces the Dead's instruments from stock Gibson, Guild, and Gretsch guitars through custom axes made by Steve Cripe and James Trussart. Jackson describes their keyboards, from Farfisas to Kurzweils, and explores the evolution of the Dead's recording techniques. You'll learn about the band's touring sound systems (from Altec Lansing systems designed for movie theaters to modern megawatt monstrosities with in-ear monitors) and the men responsible for running them. The 300-page book has more than 100 photos (many of them never published before), offering a one-of-a-kind technical perspective on the Dead's life and times.

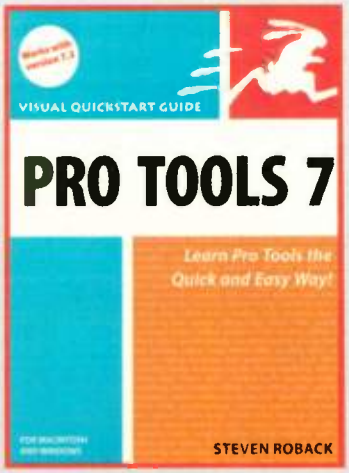


After more than a decade in print and straight from the pages of *Mix*, Paul Lehrman's monthly "Insider Audio" column is arguably that magazine's most popular read. It's almost inevitable, then, that **Thomson Course Technology PTR** (www.courseptr.com) would compile the best of those columns into a book. *The Insider Audio Bathroom Reader* ($24.99) is filled with insightful interviews, humorous anecdotes, heartfelt eulogies, questionable speculation, adamant opinions, and plenty of jokes about musicians. The 430-page anthology delivers an entertaining romp through the audio industry as seen from many angles, virtually all of them skewed by Lehrman's unique outlook. Beginning with a hilarious foreword by the Firesign Theatre's Phil Proctor, *The Insider Audio Bathroom Reader* tackles subjects ranging from politics and social issues to software upgrades and vintage recording gear, complete with new introductions and updated commentary by the author.



Another recent title from Thomson Course Technology PTR is *Sonar 6 Power! The Comprehensive Guide* ($39.99), in which author Scott Garrigus updates his authoritative tour of Cakewalk's flagship sequencing program. The 522-page book promises to pick up where the official software manual leaves off by taking a hands-on approach to mastering Sonar 6. Alongside discussions of Sonar's user interface and operations, Garrigus introduces the reader to new concepts and features such as AudioSnap, Active Controller Technology (ACT), and Session Drummer 2. Unlike previous editions, *Sonar 6 Power!* doesn't include a bundled CD-ROM; instead, you can download bonus chapters and additional content from the publisher's Web site.

Like other books in the Visual QuickStart series from **Peachpit Press** (www.peachpit.com), *Pro Tools 7: Visual QuickStart Guide* ($29.99) presents a step-by-step method for learning your way around the popular software program. Over the course of 648 pages, author Steven Roback explains what you can do with Pro Tools LE, a MIDI controller, a microphone, and an audio interface—and tells you how to do it. He describes how to set up your system, navigate the Mix and Edit windows, work with different kinds of tracks, manage audio and session files, enhance your recordings with effects, and master your final product. He devotes entire chapters to topics such as working with regions, editing MIDI, automating mixes, and using Pro Tools for video post-production. The book concludes with a 7-page glossary and an extensive index.



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Vir2 Instruments Acoustic Legends HD

Soundware developer Big Fish Audio (www.bigfishaudio.com) has spun off a virtual instrument brand called Vir2 Instruments. The first Vir2 product to ship is Acoustic Legends HD (Mac/Win, $299.95), a 19 GB collection of fretted-instrument samples recorded in 24-bit stereo at 96 kHz and paired with Native Instruments Kontakt Player 2.

In addition to seven steel-string guitars made by Gibson, Martin, Taylor, and others, Acoustic Legends HD supplies samples of two 12-string guitars, two nylon-string guitars, acoustic bass guitar, mandolin, ukulele, and banjo. Three DVDs furnish a variety of articulations and playing styles such as single notes played with picks and fingers, chords, mutes, and harmonics. Samples include fret noise and release layers for each guitar. For creating realistic rhythm-guitar strumming patterns, 6- and 12-string chord instruments provide a dozen chord types in various positions with up- and downstrokes, keyswitched and Velocity layers, and multiple takes per layer. You can even change the position of a virtual capo.

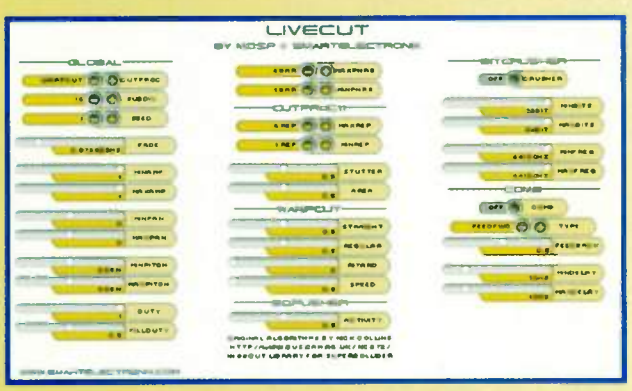
The multitimbral Kontakt Player 2 interface lets you control the EQ, reverb, and stereo width, and other controls vary by instrument. For external control, all onscreen knobs respond to MIDI Control Changes.

Download of the Month

mdsp LIVECUT (MAC/WIN)

LiveCut is a free beat-slicing effects plug-in with a mind of its own. You can try telling it what to do, but it doesn't pay much attention. LiveCut is Remy Muller's admirable attempt to implement, as a VST and AU effects plug-in, the analysis and synthesis algorithms developed by Nick Collins for his BBCut Library, which codifies the principles of automated breakbeat cutting for the software instrument SuperCollider.

Muller's work is distributed under the name "mdsp" by the Smartelectronix consortium of developers (www.smartelectronix.com). You'll find a lot of excellent free and inexpensive software there, including two other mdsp offerings: the multiview spectrum analyzer Fre(a)koscope and the third-place winner in the 2006 KVR Developer Challenge, Fire.



LiveCut uses probabilistic algorithms to cut a chunk of audio from a longer phrase and then repeat the cut, with various kinds of processing, for the duration of the phrase. Typically LiveCut inserts a fill involving quick repetitions at the end of the phrase. You select from three cutting procedures—CutProc11, WarpCut, and SQPusher—set some rate and range parameters that affect the cutting and processing, and then let LiveCut have its way with your audio. You might, for example, choose the CutProc11 algorithm, select 16th-note subdivisions, set the phrase-length range from 1 to 4 measures, and specify processing parameters such as volume and pitch-shift range. LiveCut would use the CutProc11 algorithm to make a cut that's an odd number of 16th notes, choose how many times to repeat the cut (called the block size) before making a new cut, choose a phrase length between 1 and 4 measures, and repeat the blocks for the length of the phrase, adding volume- and pitch-shifting.

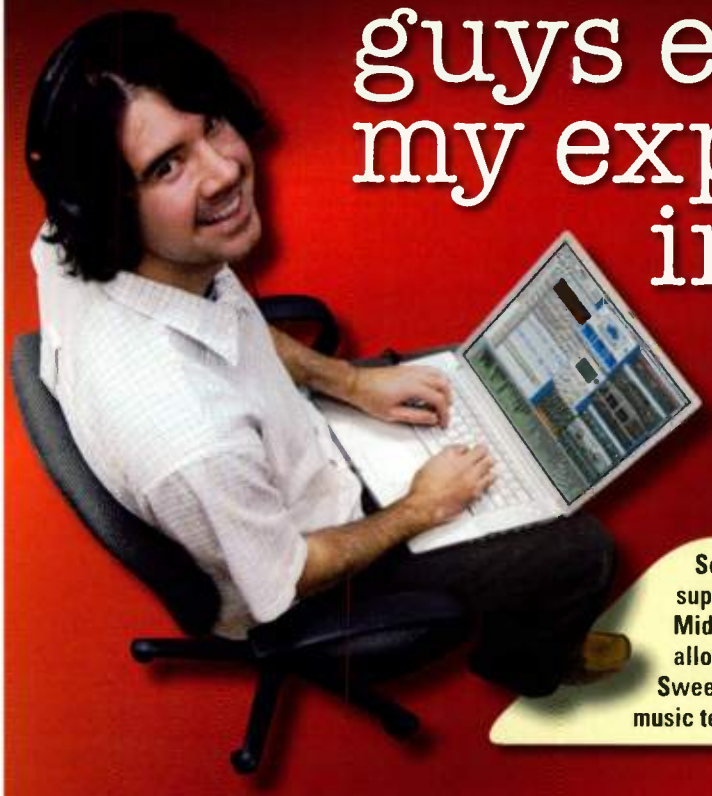
The outcome of LiveCut processing is unpredictable, but it produces interesting and musical results for a wide variety of material. Furthermore, you can pass individual tracks, mixes, and submixes through several instances of LiveCut, and you can use LiveCut as an insert or a send effect; something useful almost always comes out (see **Web Clip 1**). But as with all probabilistic effects, you may have to give it several tries before you get something you like. If you need to be in charge, LiveCut may not be the plug-in for you, but winging it has its charm.

EM WEB CLIPS

—Len Sasso

"Once again you guys exceeded my expectations in every way..."

— Miles



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Digidesign Mbox 2 Mini

The price of admission for entering the world of Pro Tools continues to plummet, thanks to a new product from Digidesign (www.digidesign.com). Delivering the smallest and most affordable Pro Tools LE system yet, the Mbox 2 Mini (Mac/Win, $329) is a compact USB-powered audio interface bundled with Pro Tools LE, a selection of audio applications, and dozens of useful plug-ins. The 24-bit, 48 kHz interface furnishes an XLR mic input with 48V phantom power, two unbalanced ¼-inch inputs that can handle line or instrument levels, a ¼-inch stereo headphone output, and two unbalanced ¼-inch line outputs. It also features zero-latency monitoring, input- and output-level knobs, and a built-in Kensington security lock, and it connects to your computer by means of a USB 1.1 port.

In addition to a full version of Pro Tools LE, the Mbox 2 Mini comes with pro-studio tools such as DigiRack and Bomb Factory plug-ins, the soft synth Xpand, and the Pro Tools Ignition Pack. DigiRack plug-ins include EQs, compressors, gates, multitap delay, reverb, and pitch-shift. Bomb Factory plug-ins include BF76 compression, Funk Logic Masterizer, and five others. The Pro Tools Ignition Pack features entry-level versions of Ableton Live, Propellerhead Reason, FXpansion BFD, IK Multimedia AmpliTube, and other programs, as well as the instructional DVD *Pro Tools Method One* and a one-year membership to Broadjam.com.

Synk Audio Musicbed DV

Now available from Synk Audio (www.synkaudiostudios.com) is Musicbed DV (Mac, $249), a soundtrack-generating application that comes with a music-library bundle. A unique set of controls lets you specify musical intensity, complexity, dynamics, and other parameters in real time to match the mood and timing needs of your audio project, whether it's a film score or a Podcast. You can choose from a library of musical themes and underscore your project with automation curves that determine how the music evolves and where hits occur along a timeline.

The latest version, Musicbed DV 1.6, features Seasons 1 and 2 stock music libraries with a royalty-free license, more than doubling the content of previous versions; content ranges from cinematic and orchestral styles to rock and down-tempo electronica. Version 1.6's music-generation enhancements include Keyframe Snapshots, which let you capture custom styles and textures, and new surround presets designed for diverse scenarios. You can specify whether adding new music to the timeline will overlay or overwrite existing music, and you can replace styles, moods, and instrumentation without affecting the timing or automation curves.

Sound Advice

Those piano-sampling perfectionists at **Synthogy** (whose products are distributed by Ilio, www.ilio.com) have been at it again. Now available is *Ivory Italian Grand* ($139), an add-on expansion pack for the virtual piano Ivory, version 1.5 and later. Featuring more than 2,000 samples of a hand-crafted 10-foot Fazioli F-308 recorded at the State University of New York's concert hall, the library furnishes more than 18 GB of all-new content. A dozen Keysets, each with up to 12 Velocity layers, soft-pedal samples, and release samples, have been arranged into 24 programs ranging from Cathedral Grand and Modern Jazz to Dark Abyss and Rockin' Studio 10 Foot. *Italian Grand* takes advantage of Ivory's new browser-style interface and furnishes new effects presets too. Like previous Ivory content, the library was recorded and programmed by Joe Ierardi, the sound designer responsible for the piano content in Kurzweil instruments such as the PC88 and K2600. EM

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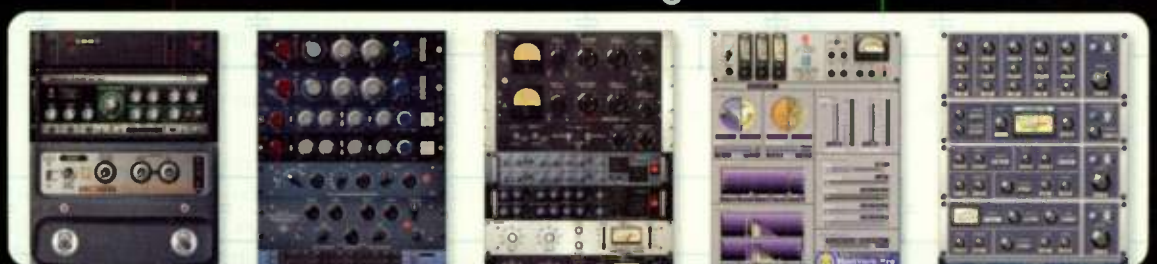

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Light Mic

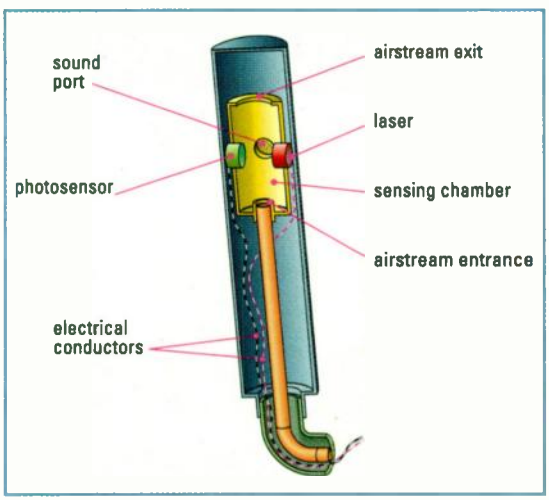
By Scott Wilkinson

A laser-based mic could eliminate distortion.

Microphones are among the most mature devices in the electronic musician's toolbox, and they all work according to the same principle: sound waves impinge on a diaphragm of some sort, causing it to vibrate in a pattern that is analogous to the acoustic waveform. The diaphragm's vibration is converted to an electrical signal whose waveform is analogous to the vibration (and thus to the acoustic waveform) by means of electromagnetic induction.

This process has just one problem: all diaphragms have a finite mass, which prevents them from changing their vibration instantaneously in response to an instantaneous change in the sound wave. Thanks to the immutable law of inertia, it takes a certain amount of time for the diaphragm to change its vibration to match the new sound wave, leading to distortion in the electrical signal.

California-based inventor David Schwartz has come up with an ingenious solution to this problem. Sound waves perturb particles, such as water vapor, suspended in a stream of air. The perturbations, which manifest themselves as changes in the density of the particles, are analogous to the acoustic waveform, just like a conventional diaphragm. Unlike a diaphragm, however, the particles in the airstream have far less mass and can thus respond to changes in the sound wave with virtually no delay or distortion.



CHUCK DAHMER

FIG. 1: In a particulate flow detection microphone, a stream of air infused with particles passes through a chamber within the outer casing near the top, where the particles are perturbed by sound waves entering from the back. A laser beam passes through the stream, which modulates its intensity, and a photosensor converts that modulated light into an electrical signal analogous to the original sound wave.

To convert the perturbations in the particle stream to an electrical signal, a laser beam passes through the stream to a photosensor on the opposite side. The changes in density correlate to changes in the transparency of the stream, modulating the intensity of the laser light at the sensor, which converts the modulated light into an electrical signal that is analogous to the original sound wave.

In his patent application, Schwartz describes one possible design for what he calls a "particulate flow detection microphone." As illustrated in **Fig. 1**, the stream enters the microphone at the bottom and travels through a tube into a sensing chamber near the top. Sound waves enter the chamber from one side, while the laser and photosensor are aligned on an axis perpendicular to both the stream and the direction of the sound waves. The stream then exits the chamber, replaced by unperturbed particles.

That last point is important—for the microphone to work properly, the airstream must be in constant motion so that the sound waves can affect "fresh" particles that have not already been perturbed. In a way, the airstream resembles a piano roll, representing the sound wave in a continuous strip rather than as a single object vibrating back and forth.

Schwartz has experimented with *nonturbulent flow* (technically called *laminar flow*), as well as with *turbulent flow*, and has found that both produce interesting results. In particular, a turbulent flow could be used to generate a "whiter," more random noise floor, or one in which the noise is mainly in high frequencies, where it has a less perceptible effect on the audio.

The beauty of this idea is that it eliminates all nonlinearities in the conversion from acoustic to electrical energy. As a result, distortion drops to zero, and the electrical signal is a far more accurate representation of the sound wave than any conventional microphone can manage. I'm excited by the potential of this invention, and I hope to see it become available as a commercial product someday. EM

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LAURA WILLIAMS

Joining the Reel World

By Eddie Ciletti

What to expect when buying a used analog tape recorder.

The best way to give your digitally recorded tracks an authentic analog tape vibe is by sending them through a reel-to-reel tape machine. For example, you can route a pair of tracks from your digital audio workstation to a stereo tape recorder while simultaneously sending the recorder's output back to your DAW.

Assuming that you are using a 3-head deck, the "analog processed" tracks will be delayed by the distance between the record and playback heads as determined by the tape speed. You can easily compensate for this delay by realigning the processed tracks in your DAW. (With a 2-head machine, you'll have to finish recording to tape, then rewind and play your tracks back into the workstation.)

Unfortunately, open-reel tape recorders are not being commercially produced these days, so you'll have to find one used. I recommend buying from a company that specializes in recorder restoration, rather than making a risky purchase on eBay: you'll pay more up front but be rewarded with a dependable machine.

Because buying a used recorder is a gamble, you'll have to invest some time to prove that the machine is worthy before playing an expensive test tape or a valuable master on it. What follows is a list of steps you can take to assess the operating condition of an open-reel machine and diagnose any problems along the way, as well as an overview of what's involved in setting up and maintaining a trustworthy analog tape recorder.

on your machine is missing from the manual or out-of-date relative to current tape availability.

Watch Your Azimuth

Once the recorder is proven safe and functional, it is time to address the routine aspects of tape machine maintenance. The azimuth is perfect when the record and playback heads are perpendicular to the tape and parallel to each other. That's hard to see with the naked eye, but there are several ways to check it.

Send pink noise from your workstation's multifunction generator to all of your tape machine's channels. Put the machine into record and monitor playback while listening to all channels summed to mono. The noise should sound bright and clear with no swishing. However, if it sounds swishy, then the heads are not precisely lined up. By applying a bit of thumb pressure to the tape, on either side of the heads, you can nondestructively manipulate the azimuth enough to exaggerate this effect. (I will discuss adjusting the azimuth in a moment.)

Keep in mind that we have not yet calibrated the playback, but have been testing other aspects of the machine's performance in order to determine where the main problems are. If you've gotten this far without issues, then the machine is safe and ready for the alignment tape. You should now remove all tapes from the area, power down the machine, and demagnetize it.

Playback Alignment

The remaining task before calibrating the machine is to choose the playback reference level. As tape, heads, and electronics have evolved, so too has the ability to record at higher levels, lowering the noise floor in the process. You don't have to peg the meters to record loud—they're expensive to replace. Simply pick an operating level based on the capabilities of your machine and the brand of tape you plan to use on it.

The reference level is specified in nanowebers per meter—185 nWb/m was an early popular standard. (A nanoweber is the quantity used to express the magnitude of magnetic flux.) The standard or reference level has changed over the years as tape formulations have improved by having greater headroom and lower noise. Just as 0 VU is a reference (and not the lack of signal), so too is 250 nWb/m.

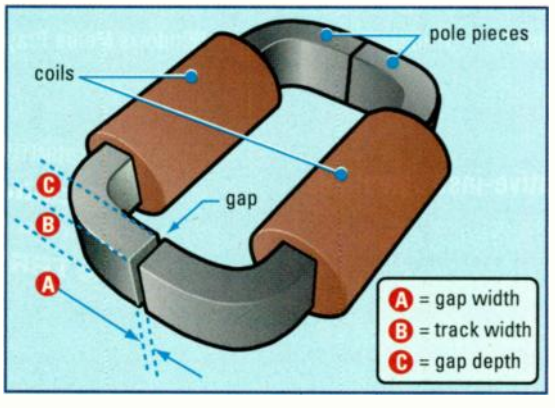


FIG. 4: Behind the polished head face is what determines head life. Performance and the amount of overbias for recording are determined by gap depth.

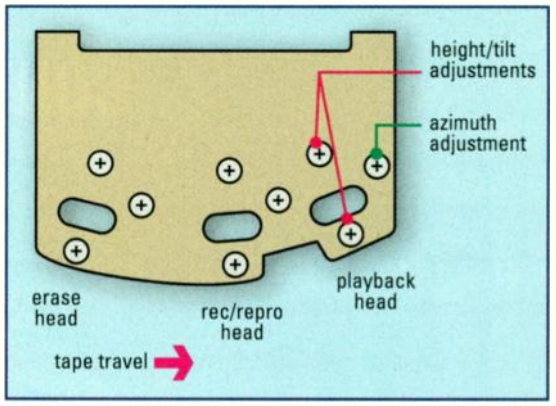


FIG. 5: A typical head-block assembly that shows where you would make adjustments for each of the heads. Notice that each head has height, tilt, and azimuth screws.

Elevated levels generally increment in 3 dB steps. The level on tape will be referred to as "+x over y," where "+x" refers to the number of dB over the reference level "y." For example, +3 dB over 185 nWb/m is 250 nWb/m. To avoid confusion, always know and state the reference level; don't just say "+6."

On narrow-format machines, level calibration should be done according to the manufacturer's specification (using the modern equivalent tape formulation). Internal noise reduction reduces the need to hit the tape harder.

Test tapes typically have a 1 kHz tone for checking playback level and a 10 kHz tone for checking high-frequency playback level. (A 1 kHz tone is in the middle of the audio spectrum and least affected by most anomalies.) Low-frequency response is typically adjusted after recording a tone sweep from 250 Hz down to 20 Hz. Because test tapes are recorded in full-track mono, do not adjust to the prerecorded low-frequency tones unless they have been "compensated for multitrack reproduction." (Full-track tapes will show exaggerated low-frequency response when played on a multichannel machine.)

Playback Azimuth

Tones at 8 and 16 kHz are provided on test tapes for coarse and fine adjustment of the azimuth. This adjustment optimizes the head's high-frequency response and compatibility with other machines. While adjusting azimuth, note any level discrepancies that may exist between 8 and 16 kHz. (A 10 kHz tone is used to set playback EQ, and the two frequencies on either side can be used to

LAURA WILLIAMS

Joining the Reel World

By Eddie Ciletti

What to expect when buying a used analog tape recorder.

The best way to give your digitally recorded tracks an authentic analog tape vibe is by sending them through a reel-to-reel tape machine. For example, you can route a pair of tracks from your digital audio workstation to a stereo tape recorder while simultaneously sending the recorder's output back to your DAW.

Assuming that you are using a 3-head deck, the "analog processed" tracks will be delayed by the distance between the record and playback heads as determined by the tape speed. You can easily compensate for this delay by realigning the processed tracks in your DAW. (With a 2-head machine, you'll have to finish recording to tape, then rewind and play your tracks back into the workstation.)

Unfortunately, open-reel tape recorders are not being commercially produced these days, so you'll have to find one used. I recommend buying from a company that specializes in recorder restoration, rather than making a risky purchase on eBay: you'll pay more up front but be rewarded with a dependable machine.

Because buying a used recorder is a gamble, you'll have to invest some time to prove that the machine is worthy before playing an expensive test tape or a valuable master on it. What follows is a list of steps you can take to assess the operating condition of an open-reel machine and diagnose any problems along the way, as well as an overview of what's involved in setting up and maintaining a trustworthy analog tape recorder.

No Pain, No Gain

When EM's editors asked me to write about analog tape machines, I was in the midst of restoring a mint-condition Fostex G16S, a ½-inch 16-track recorder. You might be wondering, why is it necessary to restore a mint-condition machine? As with any vintage item, time takes its toll on parts, often in very subtle ways. And few other pieces of audio gear are as finicky as vintage analog tape machines.

As you might expect, a 2-track deck will be somewhat easier—and consequently less expensive—to calibrate and maintain than a multitrack machine. It can be a challenge to get all 16 or 24 tracks working equally well at the same time. That's why I recommend buying a used tape deck from a company that specializes in machine restoration. No tape recorder that is 20 years old or more can be counted on to be fully functional all the time, no matter how nice it looks.

Of the original manufacturers that still exist, few can provide support *and* parts. However, a remarkable amount of support is available on the Web, including third-party vendors that are making parts for the most common machines (see the sidebar "Resources for Analog Recorders"). No matter which track format you choose, the goal is to be able to record with confidence and know that you are getting the best of what analog tape has to offer (unless, of course, you are looking for a "broken" sound).

Expect Obstacles

On a 24-track pro machine that uses 2-inch tape, three tracks take up ¼ inch of space (see **Fig. 1**). On a semipro, narrow-format machine, you will have four or eight tracks for every ¼ inch of tape. The more tape real estate each track uses, the better the overall audio fidelity.

As an example of the kinds of problems you may find in a used recorder, let's look at a Fostex narrow-format multitrack recorder.

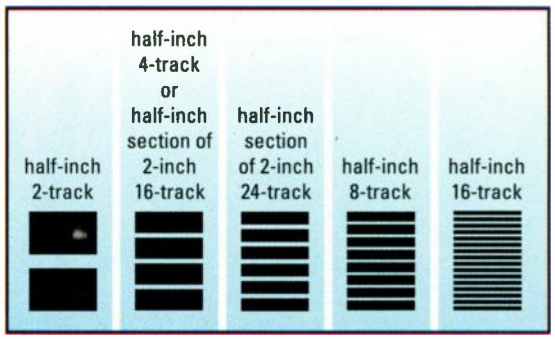


FIG. 1: This figure represents a ¼-inch cross section of tape. Wider tracks yield more signal, less noise, better low-frequency response at higher speeds, and more headroom across the frequency spectrum.

On the upside, a fully functional G16 or G24 has features like looping, built-in synchronization, and noise reduction. The transport is fast and responsive, and the front panel serves as the remote and offers metering. Squeezing up to 24 I/O cards into 19 inches was possible thanks to early surface-mount technology.

The downside is that surface-mount capacitors from that era are particularly prone to failure. So what may have started out as a $450 eBay deal could turn into a $2,000 project—either you're cool with that or you're not. But it's still a deal if you consider the original price of the recorder and the value of this machine in general.

Whether semipro or professional, nearly every multitrack is a similar investment. For instance, a 2-inch head assembly could easily run $2,500 per head, a true reality check if you paid as much for the recorder itself.

When service is required, narrow-format machines like those from Fostex, Otari, and Tascam can be shipped, though not necessarily to the manufacturer (with the exception of Tascam). On the other hand, a professional-grade 2-inch-tape machine by Ampex, Otari, Sony/MCI, or Studer requires that you either have it restored in advance, be willing to crate and ship it both ways to a repair shop, or be able to import a technician. These options depend, obviously, on your location and budget.

Nonetheless, the essential accessories for any tape machine include a service manual, a head demagnetizer, a dedicated set of nonmagnetic tools, and a test tape for playback calibration. A service manual will help you do the simple stuff, whether you're a geek or not. (Some manuals are actually a treasure trove of information.) At minimum, you'll need to know how to remove the head assembly so it can be shipped out for evaluation, restoration, and alignment.

Check Your Heads

Heads—electromagnetic devices that get signal to and from the tape—are *the* wear item on a tape machine. Like

RESOURCES FOR ANALOG RECORDERS

This list of third-party vendors should be helpful if you own an analog reel-to-reel deck.

Athan Online (www.athan.com) rebuilds motors and sells replacement pinch rollers.

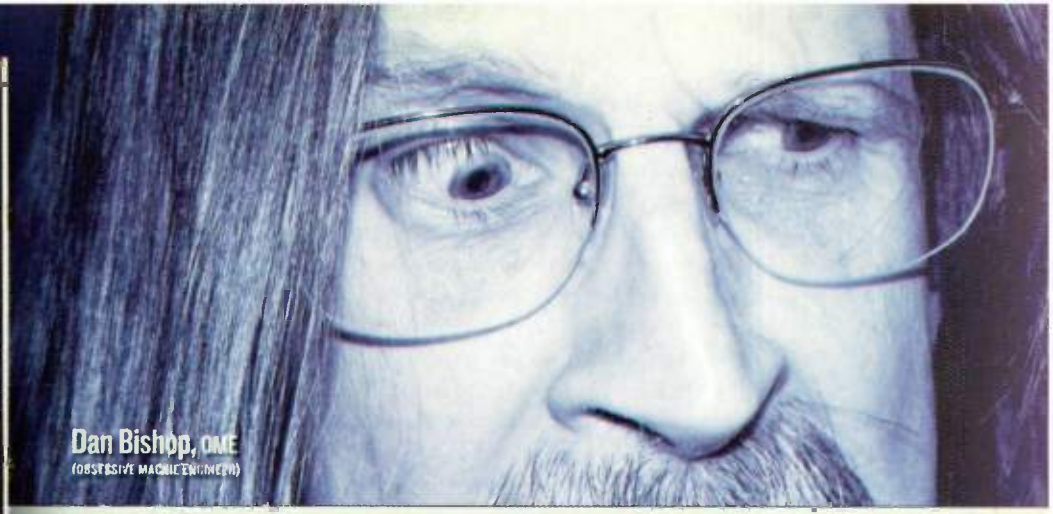
ATR Services, Inc. (www.atrservice.com) offers Ampex machine restoration, recording tape, test tapes, parts, and classes.

JRF Magnetic Sciences (www.jrfmagnetics.com) offers head restoration, heads, and test tapes and does custom work.

Magnetic Reference Laboratory, Inc. (http://home.flash.net/~mrltapes) makes and sells calibration tapes.

MDI PrecisionMotorWorks (www.precisionmotorworks.com) offers machine restoration, rubber parts, and rollers and rebuilds motors.

Terry's Rubber Rollers and Wheels (www.terrysrubberrollers.com) sells pinch rollers for professional, consumer, and specialty machines (such as the Maestro Echoplex and the Roland RE-201 Space Echo).



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NEW XDR2 MIC PREAMPS
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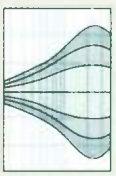

MORE MUSICAL EQ
Musical EQ isn't new to Mackie, but with our new EQ the three bands work better independently of each other, giving you more overall control.

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brake drums on your car, heads need to be resurfaced, contoured, and polished (the process is known as *lapping*) to maintain optimum performance. I can't emphasize enough that heads in good condition can minimize the more finicky aspects of tape head alignment.

To do any work in the head area, you will most likely need a hex key set. The recorder's country of origin will determine whether SAE (the United States) or metric (everywhere else) tools are used. You'll also want #1 and #2 Phillips screwdrivers, preferably in new condition, made from an extradurable alloy. Screws that are tight will require downward pressure so that the driver does not slip out and damage the screw head or the driver tip.

The capstan is the rotating shaft that maintains constant tape speed, and the pinch roller is the rubber tire that presses the tape up against the capstan. Know what a pinch roller costs and where to get one, and have a spare. Reel table clamps (for 10.5-inch reels) are becoming scarce, especially for machines like the Tascam MS-16. When the manufacturer cannot supply what you need, eBay (www.ebay.com) is a good place to search.

Basic Maintenance

It's important to have a good visual view of the head area when doing maintenance. That is especially true considering the amount of oxide old tapes can shed. For best viewing, the deck should be horizontal. Place a piece of white paper in front of the head stack to reduce glare (see Fig. 2). Clean the heads with an alcohol-dampened cotton swab, using 99 percent (anhydrous) isopropyl or denatured alcohol. Do not use rubbing alcohol, which is 30 percent water.

Note that pinch rollers can turn gummy due to many factors, not the least of which is age. Whether it is real rubber or synthetic, the pinch (and other rubber-clad rollers such as the tachometer) will react to chemicals in the tape as well as in various cleaning fluids. For this reason I recommend cleaning all "rubber" immediately after a recording project is completed—not when starting one—so that the tape chemicals will not have a chance to be absorbed by, and do damage to, the pinch roller material. This will make the cleaning process easier, require less chemicals, and minimize the aging process.

Because it is likely that you will be buying a new pinch roller, the manufacturer will recommend (and I suggest that you purchase) its product-specific cleaner. For example, Athan's pink cleaning solution is water based, while MDI PrecisionMotorWorks' Head, Red & Roll cleaner is a more volatile, quickly evaporating elixir. Both are equally effective at cleaning their respective rubber products and most likely will not do damage if applied to another manufacturer's products. That said, if there is any doubt, consult the manufacturer.

To clean original rubber parts, start with a cloth dampened (not soaked) with a water-based product such as Windex, Fantastik, or Formula 409 (which also works well on ceramic capstan shafts). Wipe a second or third time with a water-dampened cloth to remove any residual dirt and soap. Do not allow liquid to go down into the capstan shaft, or the bearing and motor will be damaged. Avoid using consumer-grade rubber cleaners.

Demagnetize

There is only one head demagnetizer to own: the R.B. Annis Model 115-S Han-D-Mag (see Fig. 3). A demagnetizer with a switch is dangerous and should be discarded or repurposed to the school science lab. Note that powering a demagnetizer up or down in close proximity to recorders and tapes can do more damage than residual magnetism from the recording process ever could. If the following exercise does not yield results, do not demagnetize your recorder without help from someone experienced in tape machine maintenance.

All tools should be tested and demagnetized before coming in contact with the heads. If a screwdriver can pick up a razor blade, one or the other is magnetized. (The residual magnetism on the heads is far less and can be measured only with a very sensitive magnetometer.) Plug in the demagnetizer at least 3 feet away from tools, tapes, and machines, then practice by slowly moving the demagnetizer toward the screwdriver and then slowly away. Do the same with the razor blade and then confirm that the screwdriver can no longer pick up the blade.

The tape machine must be powered down before demagnetizing the heads, or you'll risk damaging a preamp. Power up the demagnetizer away from the machine, then slowly position it toward the erase head, moving up and down to cover the top and bottom tracks. While maintaining the up-and-down motion, slowly move the demagnetizer away from one head and toward the next. Also demagnetize the surrounding components—guides and lifters—although these are not typically made of magnetic material.

BAKING

The sad reality is that the glue, also known as the *binder*, that secures the iron particles to the plastic tape absorbs moisture over time and eventually becomes more like rubber cement. Tapes with degraded binder will shed their oxide onto stationary surfaces such as heads, guides, and lifters.

The good news is that baking the tape at a low temperature (130 to 140 degrees Fahrenheit) eliminates the moisture and reactivates the binder. The length of time required to bake the tape depends on the tape width, type, brand, and condition and the number of reels being baked.

Tape baking is typically done before archiving previously recorded material. It is not recommended for recycling old tape for new projects. For more information, visit www.tangible-technology.com/tape/baking1.html.

Check Your Parts

Using a tape machine requires a certain amount of electromechanical awareness. Each component in the recording and playback process—from the iron particles in the tape to the preamp required by the playback head—contributes to noise. For that reason, all tape machines boost treble when recording and use an inverse equalization curve during playback. The standard curves are AES, NAB, and IEC (formerly CCIR).

While some machines have all of these options, others have none. You need to know which curve you're working with before buying a test tape. For example, semipro machines are typically fixed for IEC EQ. Pro machines running at 15 inches per second can be either NAB or IEC. At 30 ips, AES is the standard EQ curve. EQ adjustments compensate for tape, electronic, and mechanical variations.

In addition to optimizing the signal-to-noise ratio, EQ adjustments compensate for tape and day-to-day electromechanical variations. The most obvious compatibility issue is a tape recorded on one machine and played on another, but even when a tape "lives" on a single machine, the performance can vary with temperature throughout the day.

Rolling

Considering the high cost of a test tape, do not attempt a playback calibration until you are sure the machine is operating properly. Here are a few DIY tips to determine whether a machine is in good working order.

Thread up a noncritical tape—but one that doesn't need to be baked—making sure the tape is wound snugly around the reel hub before pressing Play (see the sidebar "Baking"). Careless threading can stretch tape as well as bend or break guides, tension arms, and rollers.

Before it will do anything, the machine must recognize the tape's presence by using either a mechanical arm and a switch or an optical sensor. Some machines have mechanical brakes, while others use an electronic system. Once the tape is threaded, the brakes may release and the transport controls may be activated, or you may be required to press the Stop button. In the case of the Ampex ATR-100-series recorders, you will need to nudge the reels while pressing the Stop button.

Press Stop, then Play, then Stop: the tape should move forward and come to a gentle halt with no spillage or tension arm dropping. Next, fast wind to the middle of the reel from either end and press Stop. The tape should smoothly come to a halt in about the same amount of time in either direction. Any suspicious or dangerous behavior is reason to investigate or to consult a professional.

Now press Play and pay closer attention to how the tape passes through the guides, over the heads, and around the capstan on its way to the take-up reel. There should be minimal up-and-down tape movement and no tape-edge curling at the guides. (Use reels that are not bent or warped, so the tape edge doesn't scrape on the flange.) Some manufacturers, such as Otari, provided reel table shims to compensate for reel thickness or poorly adjusted reel table height.

Several mechanical issues can affect and exaggerate tape path problems: uneven tension on either side of the capstan, how "square" the capstan and other rollers are relative to the deck plate (the surface of the machine), or a pinch roller that has lost its shape. (Shimming the capstan or adjusting anything other than head azimuth is not for the squeamish or impatient.)

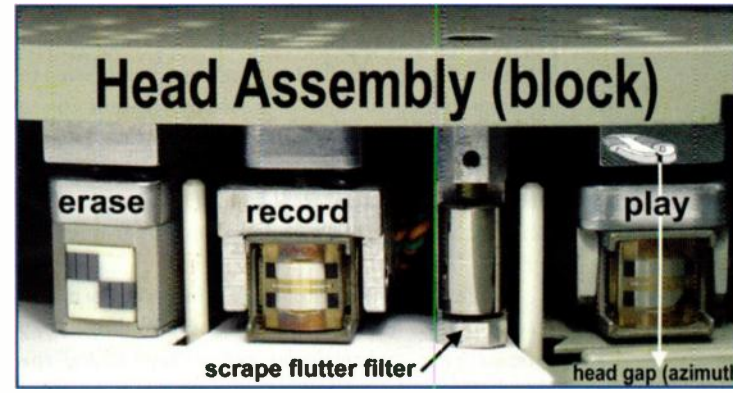

FIG. 2: A Sony APR-5000-series head assembly showing the head and scrape flutter filter locations, as well as azimuth orientation. Notice how the white paper under the erase and record heads improves visibility. Above the playback head is a miniature view of a single track. See Fig. 4 for a larger look.

Any of these issues can cause the tape to skew up or down, and when things are really bad, the tape will curl and migrate out of the guides. Consult the manual for tension-measuring tools, techniques, and procedures. Fortunately, part of the head-lapping process includes a full mechanical alignment of the head assembly.

Quick Record Confirmation

All machines have midband, 1 kHz playback and record level adjustments. Typically, narrow-format machines offer a minimal adjustment range for high-frequency record EQ. Do not use bias to manipulate the record EQ because its purpose is to minimize distortion.

While the tape is stopped, set the machine to monitor input and apply a 1 kHz sine wave, either directly from an oscillator or through a mixer, preferably to all channels at once. (Be sure to disable any built-in or external noise reduction.) At some point, the oscillator's level must be precisely known and set, but don't strive for perfection yet. For the moment, make adjustments until the meters read *roughly* 0 VU.

Assuming that the deck has three heads, simultaneous playback during record is possible. Most narrow-format multitracks have only erase and record heads, and so monitoring from tape is not possible without

first recording and then rewinding. This makes any part of the record alignment procedure a bit tedious and very time-consuming.

Put the machine into record and toggle between input and reproduction (playback mode) to confirm that all switches and relays are reliable. If the meters are not steady, the problem may be electronic (dirty switches and relays) or mechanical (funky tape, poor tape path). Continuing to toggle between modes is an exercise that may self-clean the switch or relay contacts in the short term. The alternative is to manually clean the contacts, if possible, or replace the parts.

Note that mechanical VU meters may also be part of the problem, especially if the previous owner loved to pin them in the red. There is a more transparent way to get additional saturation without slamming the meters: it's called *elevated level*, and it's achieved by calibrating with the playback tape to something less than 0 VU, then increasing the internal record level adjustment to compensate.

Switch the oscillator to 40 Hz while monitoring on headphones. A 40 Hz tone is great for finding funky pots, switches, and relays, as well as "tape rocks," the latter being a sign of damaged or used tape, magnetized heads, or bias-oscillator distortion.

One other potential noise source is the scrape flutter filter (see Fig. 2), a roller located between the record and playback heads designed to support the tape so it behaves less like a resonating guitar string. You can easily test it by applying a little bit of finger pressure. If the noise stops, the roller should be removed, disassembled, and lubricated with analog watch oil.

Record 100 Hz, 1 kHz, and 10 kHz tones, flipping between input and repro (playback during record). The tones should be steady, even if not perfectly aligned at 0 VU. If they aren't, check the tape for up-and-down wandering or curling in the guides, then apply a little drag to the supply reel with a finger. If that helps, try to localize the problem by gently applying finger pressure to the tape on either side of the record head and then the playback head. If the signal level increases or becomes stabler, there may be tension or mechanical alignment issues. The heads may also be worn, in which case they can often be relapped.

FIG. 3: The R.B. Annis Model 115-S (Short) Han-D-Mag has a curved tip for getting into hard-to-reach places.

Necessary Bias

Bias is a high-frequency signal that is applied to the erase head and mixed in with the audio signal on its way to the record head. Without bias, only the audio signal's peaks would magnetize and be captured by the magnetic tape particles, resulting in a very distorted recording. With bias, the particles are magnetized and oriented to capture the full dynamic range of the source material.

Bias optimizes the tape's sensitivity and minimizes distortion at mid frequencies and below. The optimum bias setting for minimum distortion reduces high-frequency sensitivity by a few dB, hence the term *overbias*. The recommended amount of overbias and the high frequency that is used for this adjustment is machine and speed specific and correlates to the size of the record head gap depth (see Fig. 4).

Bias is adjusted with a potentiometer, a variable capacitor, or digital controls. The adjustment location will depend on the make and model of the deck. The earliest tape machines had separate cards for repro/sync, record/input, and bias/erase adjustments. Later machines put all of the adjustments on one card. Access to the adjustments is through a pair of doors on pro machines and through removable panels on semipro machines. Consult the manual for more details.

Set the machine to input and apply a 10 kHz sine wave for 15 ips (or 20 kHz for 30 ips) so that the input level is 3 dB below 0 VU. Enter record and monitor repro while adjusting the bias control until the signal reaches the maximum level. If the meter pegs, then lower the audio oscillator's output level.

Once you find the maximum, further increase the bias by turning the bias level control clockwise until the signal is reduced by the specified amount as indicated in the manual. For example, if the recommended overbias at 15 ips is 3 dB, then continue to increase the bias past the peak until the 10 kHz level drops by 3 dB. If recording at 30 ips, you may use 20 kHz and the same amount of overbias, or use 10 kHz and overbias by half (1.5 dB). Consult a professional if the information on adjusting the bias

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on your machine is missing from the manual or out-of-date relative to current tape availability.

Watch Your Azimuth

Once the recorder is proven safe and functional, it is time to address the routine aspects of tape machine maintenance. The azimuth is perfect when the record and playback heads are perpendicular to the tape and parallel to each other. That's hard to see with the naked eye, but there are several ways to check it.

Send pink noise from your workstation's multifunction generator to all of your tape machine's channels. Put the machine into record and monitor playback while listening to all channels summed to mono. The noise should sound bright and clear with no swishing. However, if it sounds swishy, then the heads are not precisely lined up. By applying a bit of thumb pressure to the tape, on either side of the heads, you can nondestructively manipulate the azimuth enough to exaggerate this effect. (I will discuss adjusting the azimuth in a moment.)

Keep in mind that we have not yet calibrated the playback, but have been testing other aspects of the machine's performance in order to determine where the main problems are. If you've gotten this far without issues, then the machine is safe and ready for the alignment tape. You should now remove all tapes from the area, power down the machine, and demagnetize it.

Playback Alignment

The remaining task before calibrating the machine is to choose the playback reference level. As tape, heads, and electronics have evolved, so too has the ability to record at higher levels, lowering the noise floor in the process. You don't have to peg the meters to record loud—they're expensive to replace. Simply pick an operating level based on the capabilities of your machine and the brand of tape you plan to use on it.

The reference level is specified in nanowebers per meter—185 nWb/m was an early popular standard. (A nanoweber is the quantity used to express the magnitude of magnetic flux.) The standard or reference level has changed over the years as tape formulations have improved by having greater headroom and lower noise. Just as 0 VU is a reference (and not the lack of signal), so too is 250 nWb/m.

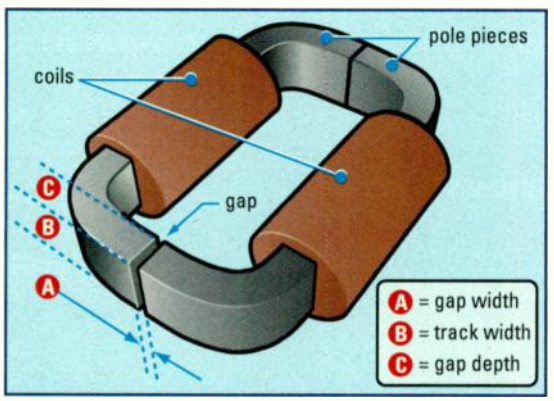

FIG. 4: Behind the polished head face is what determines head life. Performance and the amount of overbias for recording are determined by gap depth.

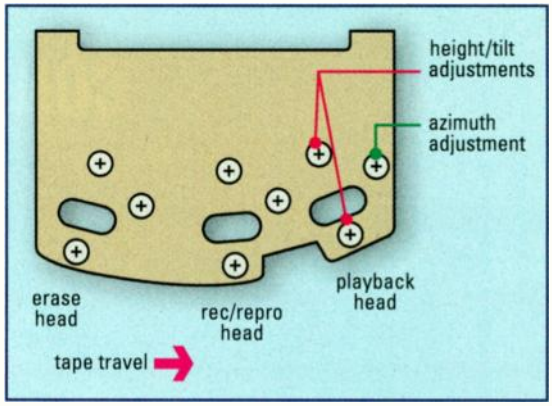

FIG. 5: A typical head-block assembly that shows where you would make adjustments for each of the heads. Notice that each head has height, tilt, and azimuth screws.

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On narrow-format machines, level calibration should be done according to the manufacturer's specification (using the modern equivalent tape formulation). Internal noise reduction reduces the need to hit the tape harder.

Test tapes typically have a 1 kHz tone for checking playback level and a 10 kHz tone for checking high-frequency playback level. (A 1 kHz tone is in the middle of the audio spectrum and least affected by most anomalies.) Low-frequency response is typically adjusted after recording a tone sweep from 250 Hz down to 20 Hz. Because test tapes are recorded in full-track mono, do not adjust to the prerecorded low-frequency tones unless they have been "compensated for multitrack reproduction." (Full-track tapes will show exaggerated low-frequency response when played on a multichannel machine.)

Playback Azimuth

Tones at 8 and 16 kHz are provided on test tapes for coarse and fine adjustment of the azimuth. This adjustment optimizes the head's high-frequency response and compatibility with other machines. While adjusting azimuth, note any level discrepancies that may exist between 8 and 16 kHz. (A 10 kHz tone is used to set playback EQ, and the two frequencies on either side can be used to

evaluate and get in the ballpark with the HF response.) Fig. 5 shows the typical locations where you would find adjustment screws.

Playback azimuth adjustment is tricky because the wavelengths of 8 and 16 kHz are so small. (The reason I suggested using pink noise during the record test is that any artifacts will be obvious to the ear.) The easiest, most accurate way to adjust record- and playback-head azimuth is by mixing all tracks from the test tape to mono (for the record and playback heads, respectively). The alternative is to route two neighboring tracks to either an oscilloscope or a mixer (summing to mono) while monitoring on a VU meter. (Each individual signal should read –6 VU and sum to exactly 0 VU.)

Record Alignment

If you correctly set the bias as described earlier in this article, you can now adjust the record level. Set the machine to monitor input, then apply a 1 kHz tone at the machine's reference level: +4 dBu or –10 dBV as required by either XLR or RCA connectors, respectively. (This translates to –18 dB Full Scale in the digital domain.)

Some machines have an Input Calibrate adjustment that may interact with the record level calibration. Enter record, switch to repro, and adjust the record level for 0 VU. Check input and tweak the Input Calibrate control if necessary. Once satisfied, set the oscillator for 10 kHz, confirm the input is still 0 VU, then monitor the repro head while tweaking the record EQ adjustment.

Next, on a 3-head deck, record a bass sweep from 250 Hz down to 20 Hz while monitoring the playback head. Align the low-frequency EQ until peaks and dips fall on equal sides of 0 VU, then select a low frequency that falls on 0 VU. Print that tone to tape and note it on the tape box. Include the bass sweep at the beginning of the tape if it will be used for a mix master.

On a 2-head deck, record the bass sweep, then rewind and check playback. Because many narrow-format machines have neither a bass EQ adjustment nor VU meters, all you can do is note the frequencies at which there are peaks and dips. Select a low frequency that falls on 0 VU, print that tone to tape, and note it on the tape box.

Extreme bumps above 0 VU can cause noise-reduction mistracking, which results in artifacts such as pumping. Having the heads lapped can cure exaggerated bass bumps on all machines.

Tape Storage: Tail Out

Tapes should be stored after being played to the end, which is known as *tail out*. This serves two purposes. As it plays, the tape is packed evenly and smooth so that dust and humidity won't damage the exposed edges.

In addition, louder audio signals will saturate the magnetic particles on the tape, making it easy to magnetize the surrounding layers, causing pre- and postecho. Postecho is more acceptable and is often masked by the original sound.

Reel Gone

Once you've done all the dirty work, it's time to play. Get to know the sonic characteristics of your machine. For example, record a kick drum while monitoring the playback head, slowly increasing the record level until you notice saturation. Then compare the input level—what's being sent to tape—with what's coming back. Saturation will affect the observed *and* perceived levels, and how much of it you want is your call. Don't forget that you can also send an entire mix to your "new" analog recorder.

Owning and maintaining a tape machine is a labor of love. The reward is knowing what your machine can do. **EM**

Eddie Ciletti has several tape machines in his audio cave, known as the Prototype, and he's secretly looking for a disc-cutting lathe.

Twelve Under a Hundred

By Len Sasso

From time to time, EM's editors search out cost-effective solutions for common problems. Most electronic musicians use software instruments to one extent or another, and most are on a budget. So we decided to see what's available in low-cost virtual instruments.

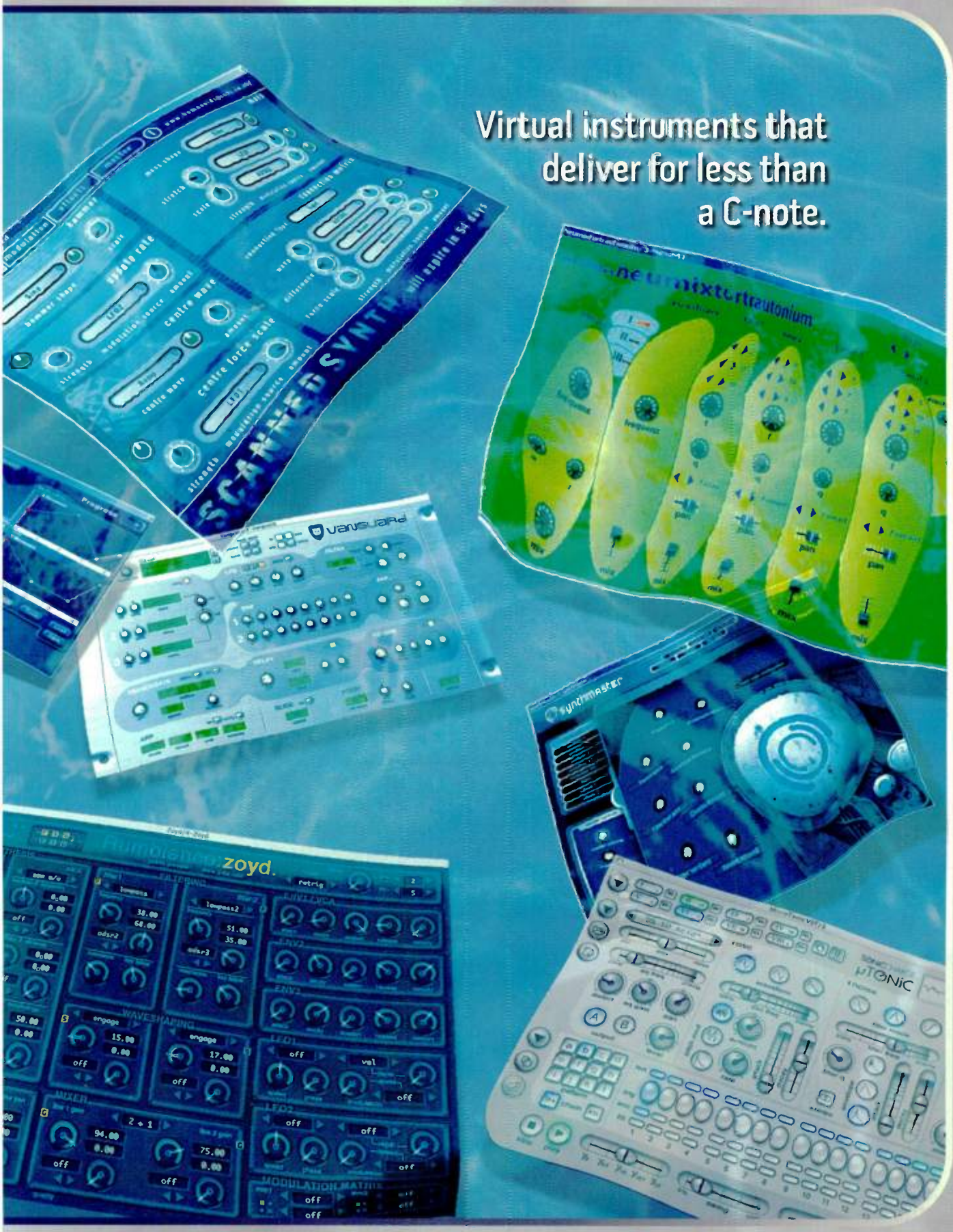
A quick trip to KVR Audio, the Synth Zone, Harmony Central, or the "What's New" and "Download of the Month" departments in back issues of EM reveals a huge number of software instruments. Prices range from free to stratospheric, and all platforms and formats are well represented. For this roundup, we decided to stick with synthesizers (no sample-based instruments) and limit the price to less than $100. Even at that, the field is huge, and we make no claim to having tried them all or picked only the best. Our goal was to present an interesting and varied collection of instruments, and that proved to be no problem. But many of the inexpensive soft synths we didn't cover are equally worthy.

The Field

Four of these virtual instruments are cross-platform, running under both Mac OS X and Windows. Seven of the remaining eight are Windows only and one is Mac only. That totals 11 Windows and 5 Mac synths. All of the Windows synths support the Steinberg VSTi plug-in format, and KV331 Audio SynthMaster also supports RTAS. All of the Mac instruments support the Apple AU format, and all but U-he Zoyd also support VSTi. SynthMaster also runs as a standalone application.

The synths run the gamut from commercial products with large preset libraries and complete manuals to experimental instruments that require some ingenuity to take full advantage of their features. Many come with little or no documentation, but you'll always find some presets and at least a few helpful hints to get you going.

Various synthesis methods are represented. Subtractive synthesis (sound generators followed by filters) is at the heart of the vast majority of software



DMITRY PARICH

and hardware synths, and that's also true here. In fact, every synth covered uses filtering at some point in the signal path. LinPlug Alpha 3 and ReFX Vanguard adhere closely to the classic analog subtractive model, although each has features not found on classic synths. Green Oak Crystal, KV331 Audio SynthMaster, and U-he Zoyd are hybrid synths, featuring other methods along with subtractive synthesis. They are also semi-modular, meaning you can rearrange their signal paths. Sonic Charge MicroTonic is a drum machine that uses subtractive synthesis for its sounds.

Six of these synths deviate significantly from the subtractive model. Nusofting Modelonia uses physical modeling of plucked-string and blown-bore instruments to generate its sounds. ConcreteFX Vocal combines oscillators with formant filters to create vocal-like sounds. Humanoid Sound Systems Scanned Synth VST uses a relatively new method called scanned synthesis. Mutagene Pocas and Progress Audio Soup are additive synths; Pocas adds a technique it calls chaotic particle modulation, and Soup adds vectored patch morphing. Rick Jelliffe's Neumixturtrautonium employs subharmonic synthesis.

ConcreteFX

Vocal ($46, Win, VSTi)

Although Vocal is designed to produce vocal sounds, its architecture and modulation routings give it a much broader reach. Vocal's signal path starts with two 3-oscillator sound sources; the oscillators play the same waveform, but you can tune them individually in semitones. You get standard synthesizer waveforms, noise, and two user-definable additive waveforms.

Each oscillator feeds its own 5-band formant filter, and the formant filters together with a separate noise generator feed a resonant multimode filter followed by an effects block. The multimode filter has 12 and 24 dB lowpass, bandpass, highpass, and band-reject modes. The effects block consists of chorus, phaser, stereo delay, and stereo spread in series along with tremolo and vibrato modulators.

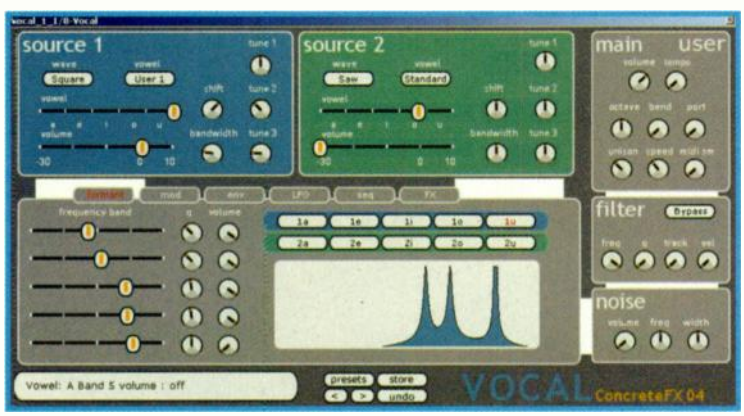


FIG. 1: You use Vocal's formant-filter editor to construct two sets of five formants.

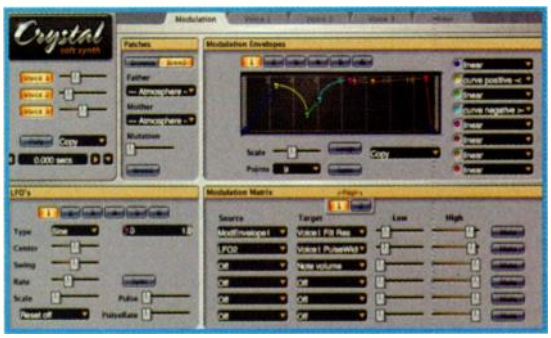


FIG. 2: Crystal has six breakpoint envelope generators, six LFOs, and a 12-row modulation matrix to set things in motion.

The formant filters are the heart of this synth. Each of the six formant-filter presets—Standard, Bass, Tenor, Ctenor, Soprano, and Alto A—consists of separate 5-band configurations for the vowels *A, E, I, O,* and *U.* A slider morphs between the vowels for the chosen filter preset, and you can modulate, automate, and remotely control (by means of MIDI) the slider. Two user configurations, which are saved with the preset, can be drawn from scratch or created by modifying one of the factory configurations (see Fig. 1).

For modulation, Vocal has six AHDSFR envelopes, six LFOs, and six step sequencers. You use a modulation matrix to route any of those sources to oscillator and formant-filter parameters. The step sequencers hold from 1 to 16 steps and can run at rates ranging from ¼ to 4 times the host tempo. You can shape the transition between steps from instant to linear to convex-curved. The latter shapes are more useful for formant modulation.

If you're thinking Vocal is only good for vocal sounds, a spin through its six factory preset banks will disabuse you of that notion. This synth is great for pulsing ambient pads, honking leads, and resonant basses. The presets also illustrate a wide range of vocal effects (see **Web Clip 1**). Check out the free demo to see what Vocal has to offer.

Green Oak

Crystal (free, Mac/Win, VSTi/AU)

In soft-synth terms, Crystal is a bit long in the tooth, and the pace of updates has slowed recently. But it is actively supported, and up against the new, new thing, it still sounds great. Furthermore, most of the bugs are out, and a significant preset library has evolved over the years.

Crystal layers three identical synths (called voices), each of which you can disable to save CPU cycles (see Fig. 2). Each voice consists of an oscillator and filter, with breakpoint envelopes for filter

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cutoff frequency and amplitude. The oscillators play standard single-cycle waveforms, multicycle sampled waveforms, and SoundFonts. You can also implement rudimentary forms of FM, ring modulation, hard sync, and granular processing. The oscillators feed resonant multimode filters, a saturation processor, and a waveshaper.

Crystal's mixer is one of its most powerful and unusual features. It incorporates a 4-way crossover and four feedback delay lines. Each voice has sliders for the amount sent to the crossover, each delay line, and the final output. You can use those to set up complex multitap delays, and the delay lines are capable of very short delay times for chorus, flange, and phase effects.

Modulation is another area where Crystal shines. In addition to the dedicated filter and amplitude envelopes, you can route six general-purpose envelopes to virtually any voice or mixer parameter. Each envelope can have as many as nine breakpoints, and each segment comes in a variety of shapes such as curved, pulsed, and spiked. A 12-row modulation matrix routes both the envelopes and six multiwave LFOs.

Programming Crystal is a little daunting, but three autoprogramming features circumvent much of the hard labor. As with many synths, you can randomize all settings, and with Crystal, that produces good-sounding results much of the time. You can also breed new presets that combine settings from two selected presets, which gives you a little more control. Finally, you can assign a modulator or MIDI controller to morph between two presets in real time. The breeding and morphing features, combined with a large collection of factory presets, mean you won't run out of fresh sounds anytime soon (see **Web Clip 2**).

FIG. 3: Scanned Synth VST offers controls for all elements of its string-scanning model.

FIG. 4: You can configure which eight parameters are accessed on SynthMaster's Easy page.

Humanoid Sound Systems

Scanned Synth VST (free, Win, VSTi)

Scanned synthesis is the conception of Bill Verblank, Rob Shaw, and Max Matthews. You can think of it as a type of dynamic wavetable synthesis in which the performer manipulates the shape of a physical object (shaking a thin metal sheet, for example) while a closed path (say, a circle on the surface of the sheet) is scanned at audio rates. Changes in the shape of the physical object determine the shape of the scanned waveform, and the scanning rate determines the pitch at which that wave is played back. Because the shape is constantly changing, the scanned waveshape and timbre constantly change, but those changes are independent of pitch.

Scanned synthesis has been realized both physically and with the aid of music programming languages like Csound. Scanned Synth VST is one of the first attempts to bring scanned synthesis to the desktop. The current version is free. A $99 version with an expanded feature set is planned for the future.

The original mathematics for scanned synthesis modeled a circular string from which masses were hung with springs and dampers. Scanned Synth VST follows that model, with controls for the model's physical parameters (see **Fig. 3**). You can modulate each of those physical parameters with LFOs or ADSR envelope generators, move the onscreen controls with the mouse, and assign each of them to a MIDI controller. Manipulating the physical parameters corresponds to the actions of the performer in the scanned synthesis model. The output of the model is fed to four familiar synth effects: flange, chorus, filter, and reverb. A Master page has controls for the amplitude envelope, portamento, and pitch modulation.

Because Scanned Synth VST's controls affect a mathematical model of a physical process, their aural effect is not easy to grasp. Tweaking a slider or knob

FIG. 7: Modelonia combines plucked-string and blown-horn physical models to create intriguing hybrid instruments.

touch, you can also feed the output of either resonator as a stimulus to the other, and simultaneously doing both produces many interesting cross-feedback effects. You can also adjust the mix of the string and horn resonators to model hybrid instruments not found in nature.

Although effects and modulation are not the name of the game, Modelonia does have two LFOs and an ADSR envelope generator, each of which can be routed

may have no effect or may produce a surprise akin to stepping on a cat's tail (see **Web Clip 3**). Most of the 70 or so factory presets are the edgy, evolving ambiences that you would expect from a physical model run amok, but some (Harpsidron, Fretless, and Slow String 2, for example) show that you can also get delicate, playable sounds. Electronic ambience is clearly the name of the game here.

Scanned Synth VST has a clever and useful randomize feature for which you can specify the degree of

Twelve Under a Hundred

to various sound-generating parameters. The output stage contains an EQ designed to simulate resonating bodies and a basic reverb geared toward emulating small rooms. Finally, there are individual tuning adjustments for each note between A3 and G♯6 for each resonator. Those are sometimes needed to compensate when the resonator's characteristics force it slightly out of tune for specific notes.

Modelonia ships with a bank of 128 factory presets along with 10 Sound wizards, which serve as good starting points for creating your own sounds. But tweaking the factory presets is probably the best way to get up and running on this synth. The best thing about physical modeling is that the sound is almost always organic, even if not precisely like an acoustic instrument (see **Web Clip 7**).

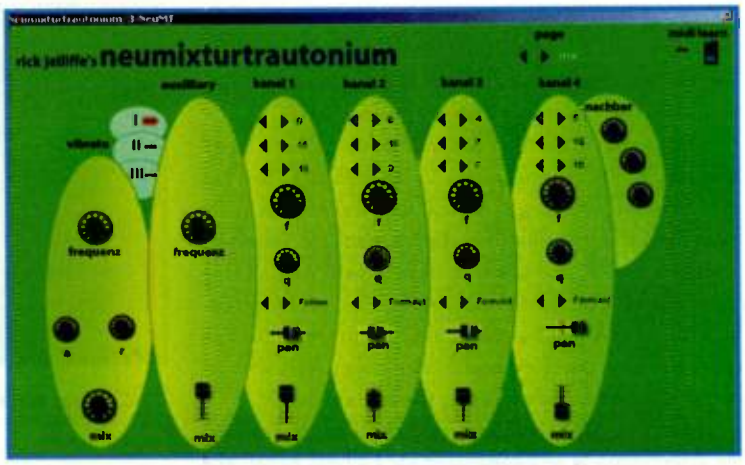


FIG. 10: Neumixturtrautonium has four subharmonic oscillators and uses the MIDI Mod Wheel to select one of three subharmonic configurations.

Progress Audio

Soup ($85, Win, VSTi)

Soup is an additive synth with an unusual approach that takes much of the tedium out of creating additive waveforms. It also implements robust, vectored sound morphing. You amass presets around the perimeter of Soup's Morph Bowl, then use the mouse, built-in automation, or MIDI to morph among the sounds (see **Fig. 8** and **Web Clip 8**).

Soup gives you three ways to create additive waveforms. With the standard method, common to most additive synths, you use a bar graph to set the level, tuning, and pan position of each of the up to 64 partials. You then use breakpoint envelopes to change those parameters over time. The second option is to resynthesize an additive waveform, including its evolution over time, from any WAV file at your disposal. The third option, unique to Soup, is called the Elements Editor. It incorporates five Bowls to shape characteristics of the sound: Tone, Tone Shift, Pan, Tuning, and Volume Envelope. As with the Morph Bowl, you add preset waveforms around the perimeter of each bowl, then position a central cursor to define the mix of those waveforms. This provides a very fast way to get complex, evolving waveforms.

Soup has six basic effects: reverb, feedback delay, distortion, chorus, multimode filter, and formant filter. Modulators include two LFOs and a breakpoint envelope generator. Those, along with various MIDI controllers, are routed to parameters using an 8-row modulation matrix. Soup is a very capable additive synth, but it's the patch morphing that makes it stand out from the crowd.

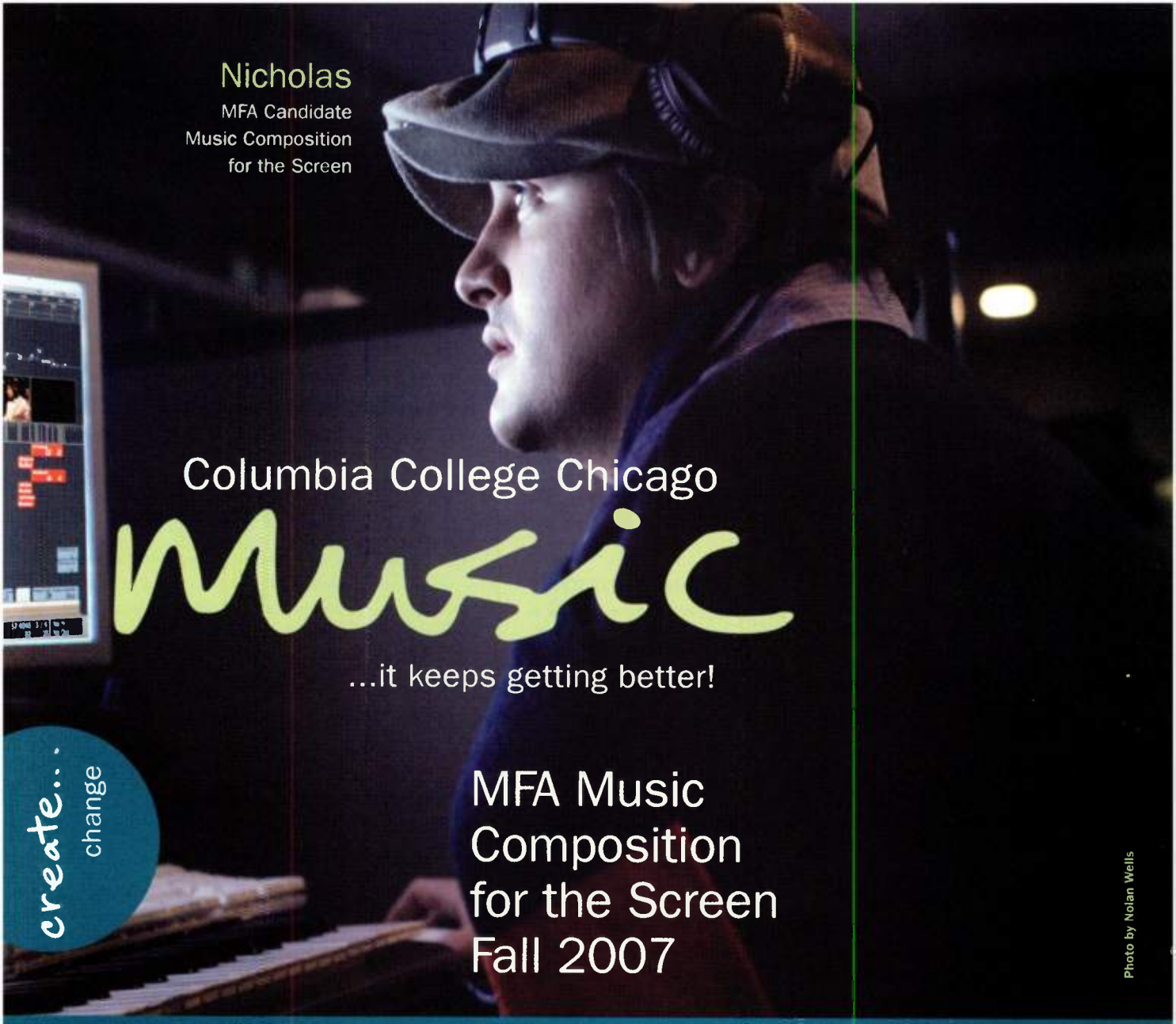
FIG. 9: Combining Vanguard's Trancegate, arpeggiator, and delay line produces lots of rhythmic variety.

ReFX

Vanguard ($89.99, Mac/Win, VSTi/AU)

With Vanguard, it's back to familiar territory—a 3-oscillator subtractive synth with two ADSR envelope generators, three LFOs, arpeggiation, gating, delay, and reverb (see **Fig. 9**). What distinguishes Vanguard is excellent sound quality, ease of use, and what it does with its basic assets. Three banks of 128 presets show that Vanguard covers the bases for classic analog sounds.

The oscillators have 31 waveforms, including 6 random forms ranging from the sample-and-hold-like R2D2 to white noise. Two of the waveforms, Sine AM and Sine FM, provide basic modulation timbres. Each oscillator has a dedicated triangle-wave LFO that you can route to tuning, pulse-width, and filter cutoff. Using more than one of the LFOs to modulate filter cutoff produces complex filter patterns.



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Scanned Synth VST has a clever and useful randomize feature for which you can specify the degree of randomization for different sections of the model. For instance, you can turn randomization off for the effects, modulator, and output sections and randomize only the scanning parameters. If you don't have a firm grasp of each control's function, that's probably as good a way as any to create new sounds. Scanned Synth VST is definitely worth a listen—it's different.

KV331 Audio

SynthMaster ($99, Win, standalone/VSTi/RTAS)

SynthMaster is the most modular of these synths. It has three oscillators and an audio input. They can be combined or split in various ways before being processed by three multimode filters, which can operate in series or parallel (see **Fig. 4**). Because of the audio input, you can use SynthMaster as an effects processor. The effects bus is unusually robust and includes a vocoder, 8-band parametric EQ, echo, tremolo, chorus, and reverb.

Each oscillator's output is the sum of eight user-configurable waves. For each wave, you select a waveform—sine, triangle, square, sawtooth, or white noise (first operator only)—a harmonic number, a frequency offset, and a mode—add, subtract, or multiply. For example, subtracting sawtooth waves with different phases produces variable-width pulse waves, and multiplying waves results in ring modulation. The combined output feeds a waveshaper and a 20-band EQ. Finally, each oscillator has its own amplitude and frequency envelope.

The filters' 23 modes include lowpass, highpass, bandpass, and band reject with various slopes and characteristics, including a fat-sounding analog-saturation emulation. The filters feed distortion units, which you can bypass. Each oscillator can feed its own filter, or a mix of the oscillators can feed all filters arranged in series or parallel.

FIG. 5: Alpha 3 is a classic analog-modeled subtractive synth with a huge preset library.

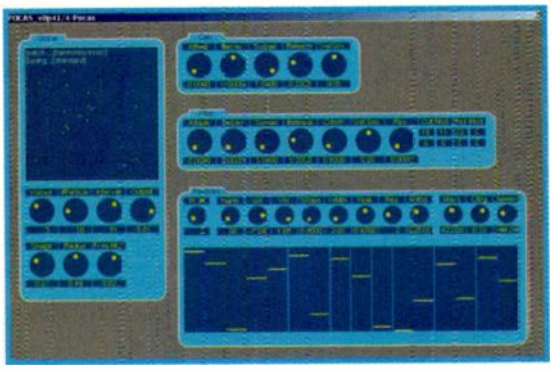


FIG. 6: In Pocas, each additive harmonic is represented by a moving dot in the two-dimensional Global display.

You can apply four general-purpose modulators either singly or in combination to most oscillator, filter, and effects parameters. Each modulator is an enveloped oscillator with DC offset. The oscillators can run at low-frequency or audio rates and offer a variety of waveforms including random and user-defined step sequences. Two additional tempo-synced MIDI LFOs generate MIDI controller data that you can route simultaneously to 16 different destinations. A highly programmable arpeggiator rounds out the complement of modulators.

SynthMaster presents a programming challenge; its eight tabbed views reveal dozens of sliders, knobs, and buttons. But the detailed manual and 148 tutorial presets will get you as deeply into this synth as you want to go. You can also mix and match settings from the hundreds of factory presets by copying and pasting complete tabs or subsections of a tab (see **Web Clip 4**).

LinPlug

Alpha 3 ($79, Mac/Win, VSTi/AU)

Alpha 3 is about as classic as it gets. Two oscillators and a noise generator feed a multimode filter followed by an amp. The filter and amp have dedicated ADSR envelope generators. For effects you get chorus. A little ring modulation, three LFOs, and a 7-row modulation matrix are thrown in for good measure (see **Fig. 5**). But if you're thinking ho-hum, think again. Alpha 3 is one of the best-sounding and easiest-to-program synths covered here, and it comes with a huge collection of presets to prove the point (see **Web Clip 5**).

Each of Alpha's two oscillators is actually a dual-waveform generator with a control for interpolating between the waveforms, and each waveform has its own octave-range setting. Thirty single-cycle waveforms provide a rich variety of starting timbres, and

morphing expands on those enormously. Beyond that, you can modulate waveform symmetry and amplitude. You can use the ring modulator to ring- or amplitude-modulate the second oscillator with the first.

Alpha's resonant filter has four modes: 12 and 24 dB lowpass, 12 dB bandpass, and 12 dB highpass. You can overdrive the filter as well as modulate its cutoff at audio rates from either oscillator or the noise source. The effect is similar to FM, and when combined with Alpha's ring modulation, it produces many of the rich, evolving timbres found in Alpha's ambient and pad presets. You can apply the filter envelope positively or negatively with an optional timed fade-out during the sustain period.

Much of Alpha's charm comes from its simplicity. Everything is well organized on a single panel, and except for a few of the finer points, you can almost dispense with the manual. If you have any experience with classic analog-synth programming, you'll have Alpha 3 speaking your language in no time.

Mutagene

Pocas (free, Win, VSTi)

Pocas ranks with Scanned Synth VST as one of the most unusual synths here. It is based on additive synthesis, but its implementation differs from that of any additive synth you may have encountered. Once you've chosen the number and spacing of the harmonics, the fun begins.

Each harmonic is represented by a particle floating in a two-dimensional window (see Fig. 6). You set each particle's mass, charge, and speed, and a variety of enigmatically named variables that determine how the particle's motion affects the sound. For example, a particle's position might modulate its pitch, its volume, or the cutoff frequency of a global lowpass filter, among other things (see Web Clip 6).

Global controls affect the pattern, speed, and overall pitch variation as well as ADSR envelope parameters for the resonant lowpass filter and output amplifier. Context menus offer preset harmonic patterns (sawtooth, square wave, and so on); 5, 20, and 100 percent randomization; and the mode of motion—particles orbiting or chasing each other.

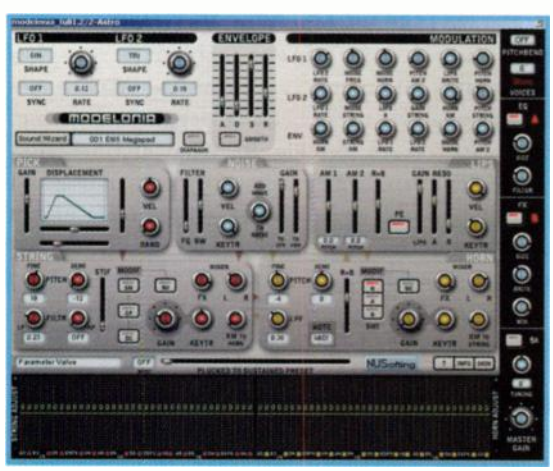


FIG. 7: Modelonia combines plucked-string and blown-horn physical models to create intriguing hybrid instruments.

FIG. 8: Soup morphs among any number of presets placed around the Morph Bowl in the center.

This synth has no documentation and few presets. It's definitely for the experimentalist, but a little perseverance will produce lush, evolving, additive ambiences.

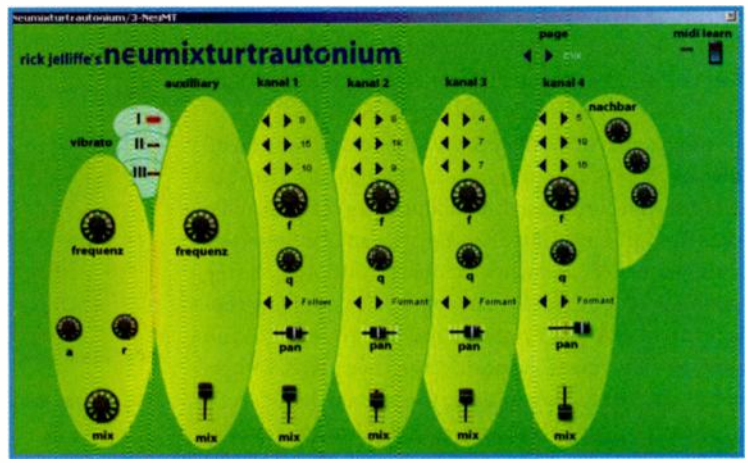
Nusofting

Modelonia ($79, Win, VSTi)

Modelonia starts with mathematical models of sound generation in acoustic instruments. The models involve a vibrating resonator to produce the sound and some form of stimulus to start the resonator vibrating. For resonators, Modelonia gives you string and horn models. For stimulators, you get a pick (applied to the string), vibrating lips (applied to the horn's mouthpiece), and noise, which you can apply to either resonator. The string and horn models differ in that the horn resonator is fed back to its stimulator to produce sustained sounds, whereas once plucked, the string resonator dies out over time.

The essential part of programming Modelonia is setting the characteristics of the stimulators and resonators (see Fig. 7). For example, you can control the shape and amount of the pick's effect on the string; the rate and amplitude of the lip vibration; the brightness and pitch of the horn resonator; and the stiffness, pitch, and EQ of the string resonator. In an unnatural touch, you can also feed the output of either resonator as a stimulus to the other, and simultaneously doing both produces many interesting cross-feedback effects. You can also adjust the mix of the string and horn resonators to model hybrid instruments not found in nature.

Although effects and modulation are not the name of the game, Modelonia does have two LFOs and an ADSR envelope generator, each of which can be routed



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Vanguard's multimode filter has resonant lowpass, highpass, and bandpass modes with 6, 12, and 24 dB per octave slopes along with several multiband configurations. The filter is followed by delay and reverb effects and an output section with overdrive distortion. You can route each of the two ADSR envelope generators to volume, filter cutoff, pulse width, and oscillator tuning.

The Trancegate and arpeggiator provide flexible, tempo-based modulation. The gate has 16 steps, variable attack and decay, separate stereo channels, and an amount control that balances between the gated and ungated signal. The arpeggiator has five modes—up, down, alternating, random, and play-order—and the rate and duration are set separately. You can generate a lot of action using the Trancegate, arpeggiator, and delay together (see **Web Clip 9**).

MANUFACTURER CONTACTS

ConcreteFX www.concretefx.com
Green Oak www.greenoak.com
Humanoid Sound Systems www.humanoidsounds.co.uk
KV331 Audio www.kv331audio.com
LinPlug www.linplug.com
Mutagene http://dsp.mutagene.net
Nusofting http://nusofting.liqihsynth.com
Progress Audio www.progressaudio.co.uk
ReFX www.refx.net
Rick Jelliffe http://extra.schematron.com
Sonic Charge www.soniccharge.com
U-he www.u-he.com

Rick Jelliffe

Neumixturtrautonium ($29.95, Win, VSTi)

Neumixturtrautonium is a variation of a variation of an early (possibly the first) electronic musical instrument, the Trautonium, invented by Freidrich Trautwein in the 1930s. German physicist and Trautonium virtuoso Oskar Sala made significant modifications, most notably the introduction of subharmonic synthesis, in the Mixtur-Trautonium. Neumixturtrautonium is a noble attempt to realize the Mixtur-Trautonium as a VSTi plug-in.

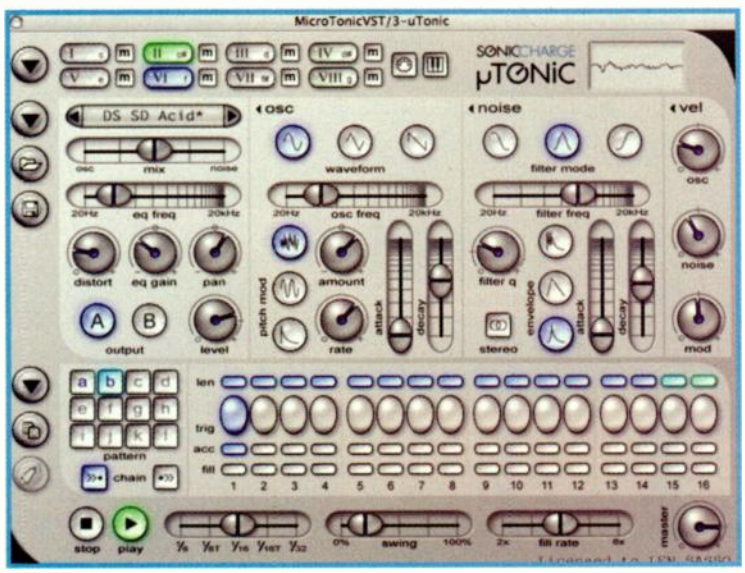

FIG. 11: MicroTonic is a classic synthesized drum machine featuring a 12-pattern 16-step pattern sequencer to play its oscillator and noise sound generators.

Subharmonic synthesis uses frequency dividers to create new waves whose frequencies are whole-number divisors of the source wave. The subharmonics have the same waveform as the original, typically a sawtooth. A combination of four subharmonics is called a mixture or bank, and in Neumixturtrautonium you select between three banks with the Mod Wheel. In each mixture, each subharmonic has its own filter, and you can switch the filters between formant and resonant lowpass modes (see **Fig. 10**).

Depending on the subharmonic levels and frequencies, a note played on Neumixturtrautonium might sound like a single note or a chord. Furthermore, just as with higher harmonics, some of the subharmonics may not be in the equal-tempered scale. Because subharmonics are harmonically complex (not sine waves) and lower in pitch, they dominate. In short, Neumixturtrautonium does not always play nicely with others, but you can, with some effort, retune the subharmonics. A basic reverb and pitch-shifter round out the signal path.

Neumixturtrautonium comes with no manual and only 16 presets, but the presets show that this instrument has an unusual sound palette. Variations range from ethereal string and choral sounds to mechanical-industrial effects (see **Web Clip 10**). It's worth your time to try it out, and the shareware fee is quite reasonable. You'd be hard-pressed to re-create its sound on another synth.

Sonic Charge

MicroTonic ($89, Mac/Win, VSTi/AU)

No plug-in library is complete without a drum machine. With the availability of excellent sampled drum libraries of every stripe, a high percentage of drum machine plug-ins are specialized sample players with built-in step sequencers. Synthesized drum machines are harder to find, and despite its modest price, MicroTonic is one of

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the most versatile. It also comes with a huge collection of drum patches and programs (patches for each pad with accompanying sequences).

MicroTonic is an 8-pad drum synthesizer with a built-in pattern sequencer capable of holding 12 patterns (see **Fig. 11**). Patterns have from 1 to 16 steps, and you can chain adjacent patterns together to create longer patterns (see **Web Clip 11**). Step size ranges from eighth to 32nd notes, triples are included, and you can use the Swing slider to shift even-numbered steps. Each step has accent and fill buttons; the fill button creates a flam at from one-half to one-eighth the step size. A context menu allows patterns to be copied, pasted, and automatically altered in various ways. You can export pattern chains as either MIDI or audio files. Exporting MIDI is especially handy for dequantizing your drum patterns.

MicroTonic's drum synth has eight pads, which can be triggered by MIDI with or without the pattern sequencer. You can, for example, create basic drum parts in the pattern sequencer, then spice them up with long, nonlooping MIDI parts. With the multioutput version of the plug-in, you can route each pad to its own output for individual processing.

FIG. 12: Zoyd's two oscillators are followed by filters, waveshapers, and a mixer whose signal routing can be reconfigured.

Each of MicroTonic's drum pads has an identical synthesis engine consisting of an oscillator section, a noise section, and a mixer for balance, EQ, and distortion. The oscillator section has a 3-mode oscillator (sine, triangle, or sawtooth), a bipolar pitch modulator with random, sine-wave LFO, or decay envelope as sources, and an AD volume envelope. The modulator is the key here, providing everything from naturally falling drum pitch to

rattle- and shaker-style percussion. The noise section feeds white noise through a multimode resonant filter. The level is controlled by an AD envelope with exponential, linear, and retriggering decay modes. Retriggering is useful for stuttering sounds like handclaps.

U-he

Zoyd (free, Mac, AU)

Zoyd is a prototype for the commercial (not free) plug-in synths subsequently released by U-he. Consequently, it suffers a few limitations such as no further development, monophonic operation, minimal documentation and preset library, limited optimization, and occasional unexpected clicks and pops. So why include it? Because it sounds different, is different, and offers a lot to play with.

Zoyd's sound generators are two oscillators and a noise generator. The oscillators have three waveforms and unique Morphology and Variance controls whose effect depends on the selected waveform. Waveform Saw o/e is the most interesting; Variance detunes the odd versus the even harmonics, and Morphology adjusts their volume (amplitude) ratio.

Zoyd's signal path splits after the sound generators and thereafter is somewhat configurable—you can choose from nine configurations. These arrange a pair of filters, a pair of waveshapers, and a mixer in various ways. For example, each oscillator can feed its own filter, after which the signal is mixed and sent through the waveshapers in series, or oscillator 1 can feed a waveshaper and filter in series, then be mixed with oscillator 2 to feed the second waveshaper and filter in series. The signals appearing at the mixer can be added, subtracted, or ring-modulated. Along the way, you can modulate most parameters with any of three ADSR envelope generators and two LFOs (see **Fig. 12**).

Being monophonic and aggressive-sounding, Zoyd is best for leads and basses. It can also have a decidedly vocal quality, which is evident in many of the factory presets (see **Web Clip 12**). You may not use this synth in every piece, but when you do you'll get the listener's attention.

More Is More

It's tempting to say that this roundup covers the Alpha to Zoyd of budget virtual synths, but in truth it barely scratches the surface. A little exploration will turn up dozens of picks in every category, and most of them have something unique to offer. Beyond that, they're usually underexposed and therefore not overused. A little time on the Web instead of leafing through your existing preset library might just turn up that perfect sound. **EM**

EM WEB CLIPS

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

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KEITH MARTIN

Beat Crazy in Atlanta

By Mike Levine

Producer and keyboardist LRoc thrives in hip-hop's hottest market.

As I pulled up to the ramshackle house in the Atlanta suburbs, I was sure that I had the wrong address. This couldn't be where producer, keyboardist, and Grammy-winning songwriter LRoc, who's worked with everyone from Usher to Mariah Carey to Nelly to LL Cool J, had his studio. No way. I was expecting something more modern and well kept up—maybe even a little flashy.

But a quick check of my notes confirmed that this was indeed the location that LRoc's management had meant to send me for the interview. So I walked to the door and rang the bell, still half expecting whoever answered to look at me quizzically when I asked for LRoc.

But when the door opened, it was LRoc who answered. He greeted me amiably, and then motioned for me to follow him through a living room area equipped with a huge flat-screen TV and a Sony PlayStation console. We then went down the stairs to the basement studio that he calls "dakitchen." It has professional acoustic treatment on the walls, a vocal booth, and a small live room. He explained that he purposely situated his studio in this nondescript house (he lives elsewhere) so that potential burglars would have the same reaction of "There's no way there's anything valuable in there" that I had.

In fact, there was plenty of choice gear—enough to pique the interest of any self-respecting burglar. But my mission was not one of criminal intent; I was there to spend the day watching LRoc work and to gain insight into his production techniques.

Background Check

LRoc, whose real name is James Elbert Phillips, was born in Liberia. As a child there during the '70s, he studied classical piano. However, his musical imagination was stimulated not by

Bach or Mozart but by the sounds of American pop music that he heard on the radio. "I would hear those records, like Parliament Funkadelic and Prince," he recalled. "I learned a lot from them."

Although his classical training played a role in his musical development, he wasn't what you would call a model student. "When I left music class, that book went into the piano bench and didn't come out until the next session," he remembered. "But I did play while listening to records. I practiced by listening to artists like Stevie Wonder, so my ear and my improvisational skills improved more than my reading."

KEITH MARTIN

FIG. 1: LRoc tapping rhythms into his Akai MPC3000 groove sampler. Behind it are his Korg Triton and his Fender and Sadowsky basses.

LRoc taught himself to play the bass guitar by "listening to the radio and listening to records. It's a simple instrument. I picked it up, and I loved it a lot. Classical piano was more my parents' idea."

LRoc's family immigrated to the United States and settled in the San Francisco Bay Area when he was 16. There, he continued listening to funk bands such as Cameo, the Time, and the Brothers Johnson, but his musical tastes broadened, encompassing artists like Count Basie, Thad Jones, the big-band stuff, Chaka Khan, Chick Corea, and John Patitucci.

Herbie Hancock was also a huge influence. "He was a great musician, but he still could go and do some great-sounding commercial stuff," said LRoc. "So was Bernie Worrell of Parliament Funkadelic; he was classically trained but played the funk. The kind of musicians I gravitated toward were skilled and able to play the simplified music without compromising their artistic integrity." That concept was to become key to LRoc's later success.

LROC: A SELECTED DISCOGRAPHY

Producer, Songwriter, Instrumentalist

Mariah Carey, "Say Somethin'" (So So Def remix featuring Dem Franchize Boyz) (Island Def Jam, 2006)

Chingy, "Pullin' Me Back," *Hoodstar* (Capitol, 2006)

Dem Franchize Boyz, "My Music," *On Top of Our Game* (Virgin, 2006)

Janet Jackson, "So Excited," "This Body," *20 Y.O.* (Virgin, 2006)

LL Cool J (featuring Jennifer Lopez), "Control Myself," *Todd Smith* (Island Def Jam, 2006)

Bow Wow, "Big Dreams," "Caviar," "Do What It Do" (no sample), "Do You," "Is That You," "Mo Money," "Fresh AZIMIZ," *Wanted* (Sony BMG, 2005)

Jermaine Dupri, "Gotta Getcha," *Jermaine Dupri Presents: Young, Fly, and Flashy*, vol. 1 (Virgin, 2005)

Nelly, "Grillz," *Sweatsuit* (Universal Records, 2005)

Bonecrusher, "Back It Up," "For the Streets," "Lock and Load," "Peaches and Cream," *AttenChun!* (Arista, 2003)

Lil Jon and the East Side Boyz, "Get Your Weight Up," *Part II* (TVT Records, 2003)

Songwriter, Instrumentalist

Ice Cube, "Go to Church," *Laugh Now, Cry Later* (Lench Mob Records, Inc., 2006)

Mario (and Mario featuring Juneville), "Boom," *Turning Point* (J Records, 2004)

Usher, "Yeah," *Confessions, Special Edition* (LaFace Records, 2004)

Lil Jon and the East Side Boyz, "Diamonds," "I Don't Give a F***," "Knockin' Heads Off," "Nothin' On," "Ooh Na Na Naa Naa," *Kings of Crunk* (TVT Records, 2002)

Soldiering in the Studio

A stint in the army followed, and LRoc, stationed in Germany, found plenty of musical opportunities. He set up a MIDI studio in his barracks, in which he had an old Yamaha sequencer, a Commodore 64 computer running C-Lab software, a Yamaha DX7, a Sequential Prophet 600, and a Yamaha RX15 drum machine. "I was doing a lot of sessions, and everybody started calling me a great producer," he said. "So all of a sudden, I was a producer, I guess."

LRoc's passion was clearly more musical than military. So when his enlistment ended in 1989, it was off to Atlanta, a city with a budding music scene that would become one of the hottest in the country a decade or so later. He immersed himself in the city's gigging and recording circuit and steadily built his reputation. In 1994 he joined a band called the Chronicle. It was an all-instrumental improv band that was huge in the area. "We were like the southern [band] Roots. Hip-hop, funk, jazz, DJ—a killer band," said LRoc. "It was incredible. It was spontaneous. We

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"It's all manipulation; it's how you manipulate people to move."

always made up stuff onstage. We did that for six years and sold out every show. It was crazy."

As the Atlanta hip-hop scene grew bigger and more influential, LRoc became the regular keyboardist in Lil Jon and the East Side Boyz. The group was one of the main innovators of "crunk," a subgenre that developed out of the club scenes in Atlanta and Memphis and that has become a major factor on today's national hip-hop scene. Working with Lil Jon, LRoc won a songwriting Grammy in 2005 for his innovative keyboard melodies on Usher's R&B megahit "Yeah." LRoc's biggest break came when he was signed to a production deal by hip-hop mogul and producer Jermaine Dupri and his company So So Def. That signing opened a lot of production doors for LRoc, who now splits his work time between his own setup and Dupri's slick SouthSide Studios.

Inside the Beats

When I arrived at LRoc's studio, he was producing *beats*, the rhythm-track loops that are the foundation of hip-hop music. He explained that the songwriting process of today's hip-hop and R&B is different from that of other forms of popular music. A song often starts with a beat that's composed by the producer, and then the singer or rapper writes his or her raps and hooks around that.

"I have different artists I'm working on stuff for now. I tend to work on six or seven things at once. I'll work a little on one beat—get it to the point where it feels comfortable," he explained. "I might have 20 different sequences in one file. And then, when the writers get here, I start to track stuff. I just come up with a lot of ideas. Sometimes I change the tempo based on how the hook flows; I don't like to put it in stone yet. I like to have the option to mess with the tempo—to change certain things. Once we've got a hook, then I track into [Digidesign] Pro Tools."

Although LRoc often starts the beats on his own, he likes to have the artists' input as the process develops. He said about several of his current rap clients, "Usually when they hear something that they like, they start rapping. And if I see that the response is great, we'll run with it and then track it," he said. "If they don't like anything, I have to start something fresh while they're here. We usually do both."

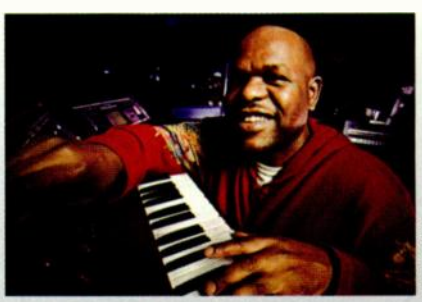
MPC Front and Center

Although he records to a Pro Tools HD system, the most important piece of gear in LRoc's studio is his vintage Akai MPC3000 groove sampler. "This is the brainchild here," he said, pointing to the tabletop device. "Most of my drums come from here" (see **Fig. 1**).

He explained that he slaves it to Pro Tools but uses its sequencer for programming his basic rhythm parts (see **Web Clip 1**). The MPC, which has had many incarnations over the years, is huge among hip-hop producers. "I've got a signature sound that a few of us, including Lil Jon, have been using for years. It hasn't failed us very often; we've had a lot of hits. For different timbres, I use the MPC and the [Roland SP-] 808."

LRoc likes the sounds in the SP-808, although

THE VIEW FROM DAKITCHEN



KEITH MARTIN

FIG. A: LRoc tweaks his Minimoog at dakitchen.

If you're sitting in the mix position of the control room at dakitchen, LRoc's studio, his Minimoog (which has obviously seen heavy use) is on your extreme left (see **Fig. A**). In front of it is a rack containing a number of outboard synth modules, including a Roland V-Synth XT and Fantom XR, an E-mu Mo Phat and Proteus 2000, and an Oberheim Matrix 2000.

A Roland SP-808 Groove Sampler is on a rolling stand to the left. A desk in front features an old Korg Triton, some patch bays, a PreSonus Central Station monitor controller, an Apple monitor for LRoc's G5 (which runs his Digidesign Pro Tools HD setup), and his most important piece of gear—an Akai MPC3000.

Dynaudio BM15A active monitors are on either side of the Apple monitor, and KV Audio EX 10 monitors on speaker stands are farther out to the side. A KV Audio EX 2.2 subwoofer is on the floor. On the right is a rack with his Pro Tools hardware and mic pres.

When he records vocalists, LRoc uses a Soundelux U-195 mic. "It's not too rich, but it fits everything I'm doing," he said. "If I need something that's high end, then I rent. I haven't decided yet what other mics I'm going to get, but the Soundelux has been working fine—it's an old faithful."

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he wishes it had multiple outputs rather than just a stereo pair. (He plans to buy an SP-909, which does have individual outs.) He uses the SP-808 mostly for percussion parts, but the MPC is definitely his go-to device. He likes the way its converters add subtle dirt to a sound. "I sample a lot of my drum sounds into the MPC," he said. "It's not clean, but it's punchy."

LRoc has been dabbling with a few virtual instruments, such as Native Instruments Battery, which he uses for certain drum sounds. He has the entire Native Instruments suite of soft synths. "I've got those in there," he said, pointing to his Apple Power Mac G5 running Pro Tools, "and on my laptop."

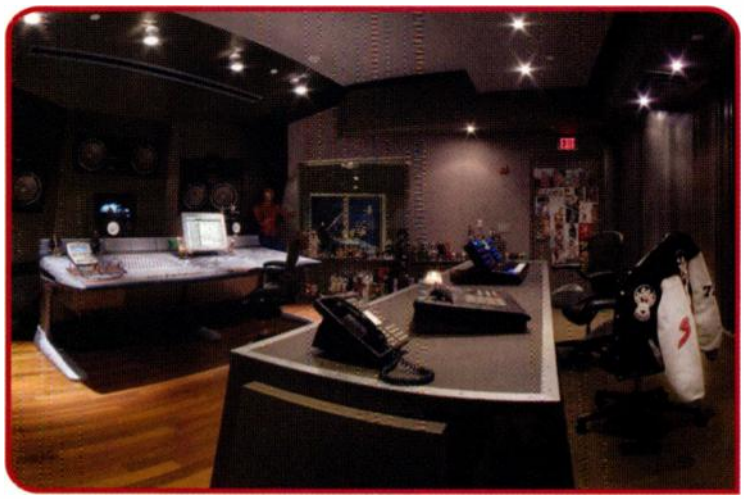

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FIG. 3: When he's not at dakitchen, LRoc produces, composes, and plays keyboards at Jermaine Dupri's well-appointed SouthSide Studios.

Because he began producing when outboard gear reigned supreme, LRoc has been cautious in adopting virtual instruments. But he's now making a big push toward the virtual world. He told me about going to a music store after not having gone to one for a while. "I was like, 'Hey, where are all the keyboards?'" he said laughing. "I realized then that it's a new day. But I love it. I was hesitant at first, but then I started hearing some of the other producers at So So Def, and they had everything on virtual instruments. And I said, 'Hey, did your drums come from that?' They said, 'Yes,' and I said, 'Okay.'"

KEITH MARTIN

FIG. 2: Although he has soft instruments in his setup, LRoc still relies a lot on outboard gear such as the Roland V-Synth XT.

An Apple MacBook Pro laptop provides LRoc with a portable setup for working on beats when he's not at his own studio. Unlike his G5 desktop at dakitchen, on which Pro Tools is the main sequencer, his laptop features Apple Logic Pro, which he's just getting the hang of.

He has more jobs than he has time for, so his laptop helps him maximize productivity. He uses it to work on tracks at home and if he has extra time when he's at SouthSide, where he goes almost daily. "Working with Jermaine, I have to be there a lot. So that doesn't give me much time. That's what prompted me to get Logic [for the laptop]. There's a lot of downtime there. If I go there at 3 or 4 p.m., I might leave at 5 or 6 a.m. While Jermaine plays some video games or shoots ball, I go in and create some loops."

LRoc told me that he plans to spend some time in the near future becoming more proficient in Logic and with his virtual instruments. "My focus is on getting everything integrated and working better," he said.

Beat Making

According to LRoc, his inspiration for creating beats is often triggered by just messing around with his gear or by hearing a particular sound on a synthesizer or sampler. "A lot of times a sound will inspire what I do, too," he said. As an example, he triggered a cool-sounding vocal patch from his Roland V-Synth XT module (see Fig. 2) that sounded like a choir singing in Latin. "I just bought this," he said about the V-Synth XT. "It has a lot of great synth sounds in it. I use that sound in order to get a basic ghetto beat."

He sequenced a melody that featured the V-Synth, and then wove in a slow drum part on the MPC. The Latin vocal sample inspired the creation of this groove.

engineering and keeps the studio gear set up and running efficiently. After they worked on beats a while more, LRoc said that there was a mix session scheduled that day over at the SouthSide studio for Bow Wow's new CD. LRoc had helped work on the album, and he wanted to attend. "Sometimes they need me to add something," he said, "but I doubt it on this album."

The mix for the CD was down to the wire. "The album has to be turned in tomorrow," he explained, "because Bow Wow has a single out with Chris Brown that's called 'Shorty Like Mine.' Jermaine released it a couple of months ago, and it's blown up. When that song was released, we hadn't started the album. So we had to rush and finish it."

After a detour to LRoc's favorite Mexican restaurant for a quick bite, LRoc, Alexander, and I headed down the highway to Dupri's SouthSide studio complex. I asked LRoc if he ever mixes any of his own music, and he said no. "I know how I like it to sound. But as far as how those guys get the frequencies and the placements . . . it's tough understanding all that stuff, like how they use all the outboard, the formulas for delays, the right kind of delay and effects. That's my green area. I have to admit that everything I create is done with a team. I just master what I do and slowly learn the other stuff." But he does want to develop his mixing skills. "Eventually, I'd like to become really good at it," he told me.

Fun and Games

When we arrived, we found out that the mix session wasn't happening. There was a big huddle going on in the main control room concerning the Bow Wow album. The label wanted to release only a "clean" version that contained no swearing so that the CD wouldn't get an "Explicit" sticker. Bow Wow was apparently not happy with how the editing necessary to make it "clean" affected the final product. There was much discussion about what to do, especially considering that the album was due to the label the next day.

Not being involved with that aspect of the Bow Wow project, LRoc took this opportunity to show me around SouthSide. The main studio is a super-high-end Pro Tools–based room, with a large Digidesign Icon console in the control room (see **Fig. 3**). LRoc showed me several side rooms that are production and composing facilities featuring gear setups fairly similar to those at dakitchen, including MPCs and KV Audio EX 10 systems with subs. The EX 10s sit on speaker stands and are basically like P.A. speakers. With the subwoofer, they're capable of cranking out the bass and simulating club conditions. Apparently, KVs are ubiquitous on the Atlanta hip-hop production scene. LRoc had the same KV speaker system in his studio, and he said that Lil Jon has it as well.

it's how you manipulate people to move *[he laughed]*. It's just those little factors. When I'm making a track, I'm thinking about all of those things. So it becomes second nature.

The composing rooms have Dynaudio BM15A active monitors, just like the ones LRoc has at dakitchen. "Jermaine and I write on the Dynaudios and some Samsons, and once we get something we like and he wants to listen, we go to KVs," said LRoc. He said he sometimes does the same thing at his studio. "Especially when I'm doing a rough mix, then I use the KVs for reference. I go back and forth."

As impressive as all SouthSide's gear was, the studio's most astonishing feature was its indoor full-court basketball court. When people working at the studio have downtime, they can relax and play hoops. People less inclined to the hardwood can amuse themselves at a full-size billiard table or with video games. Clearly, Dupri has spared no expense to make SouthSide a comfortable working environment.

LRoc showed me one of the live rooms, and it featured an isolated room off to its side with a grand piano. He sat down at it and started playing fluently. "I've been practicing lately. I made a conscious effort to get my chops back up, because I've gotten lazy just with simple stuff. But I want to start doing more stuff and getting back to some playing."

Later, sitting and talking in one of SouthSide's production rooms, I asked LRoc about how his Liberian background had influenced him musically. "I listened to African music in general," he told me. "I listened to stuff that you could hear on the radio all the time. I was playing a lot of those songs by ear. And in bands. They've got different kinds of rhythms [than American music]. I understand other kinds of percussive rhythms and feels; I learned a lot from that, and I still apply it."

I asked LRoc what he hopes to accomplish in the near future. He said that he'd like to be able to produce more beats without a specific artist in mind, and then sell them. "I'm working on my skills of just doing tracks and getting them placed. If I can do that, then my income will increase," he said. "When I'm selling tracks that I made on my laptop, I'll be the happiest. I'll just get my little M-Audio bag and go on the plane, compose on the way to where I'm going, and when I get there, it'll be done."

In addition to increasing his proficiency in Logic and with his virtual instruments, LRoc said that the biggest challenge presented by producing beats strictly from his laptop is to successfully duplicate the sounds he gets in the studio. "I can create a song on it, but my signature—my drum sound—isn't the way I want it yet. I'm still working on that, and that's key." He knows that having killer sounds is crucial to staying successful in the hip-hop production game. "Sonically, it's got to beat so hard," he said, "because the competition will." EM

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He sequenced a melody that featured the V-Synth, and then wove in a slow drum part on the MPC. The Latin vocal sample inspired the creation of this groove.

LRoc explained that many southern hip-hop and R&B songs are set at relatively slow tempos. "A lot of the crunk music was at 82 to 85 bpm," he said. "The common thread in all those songs is the hi-hats—the high *t-t-t* sound. You slow it down to get those 16th-note hi-hats, and then everything is built around that."

Southern hip-hop, he said, is often defined by its drums. "The 808, the boom. The hi-hats, the handclaps. You don't hear too many snares; you hear more claps."

LRoc said that songs with a "South-style" groove typically have a sonic palette similar to "bass music," which originated in Miami. The 2 and the 4 are emphasized less than in some other types of hip-hop, and that has a different effect on somebody listening or dancing to it. "In the up-north stuff, you have snares poppin' in your face," he said. "It makes you nod your head like that, on 2 and 4. Whereas when you've got a hi-hat riding like this [South style], and you've got a 2 and a 4 meshed in and not standing out, it makes you move differently. It's all manipulation; it's how you manipulate people to move *[he laughed]*. It's just those little factors. When I'm making a track, I'm thinking about all of those things. So it becomes second nature.

"If I'm doing a South beat, we have a kind of formula: a lot of hi-hats and handclaps. I'll start with a handclap, just getting the 2 and the 4—the pulse," he explained (see **Web Clip 2**). Although he follows convention to a certain extent, he likes to add his own touches to his beats. "Sometimes I'll put in the hi-hat to start the beat, but then I'll take the hi-hats out and start putting other kinds of percussion in there."

LRoc is clearly confident enough in his abilities that he'll push the envelope a little without worrying that the result will turn off his clients. When I first arrived, he played me a beat that featured a New Orleans–style horn section combined with hip-hop rhythms that he'd put together for a client who was originally from Louisiana.

Although he likes to be creative, he does have some sonic staples that he knows will work. "I have certain sounds that I like to use on a lot of my stuff. It makes it easy. A lot of times clients want a song like a song that's already out. So I use certain sounds that are really distinctive from that song."

Bass in Your Face

Later on LRoc was working on another beat, and I noticed that it had no bass part in it. I asked him if he was just waiting to add the bass. "A lot of times in the South stuff, this will be the bass," he said, referring to the 808-type kick (aka the "boom") coming from his MPC.

LRoc said that he often transposes that kick sound to match the song's chord changes. He usually tunes it to the root of the particular chord, but sometimes he tunes it to the fifth. "The drums are harmonic too," he said. "Subliminally harmonic. It's amazing how tuning a drum will make a song feel better. It just changes it." He also likes to subtly tune the snare. "If you pitch it up, it makes it a little bit tighter, and it just feels better."

On the songs that have a synth bass, LRoc often uses the Spectrasonics plug-in Trilogy. "It's just a beast," he observed. But as a longtime bass player, LRoc hopes to inject more real electric-bass parts into his music. "That's for my next phase. I use certain instruments, and I go in phases."

Analog World

From his years as a performing keyboard player, LRoc has become very proficient on the Minimoog and uses it a lot in his productions. "I can pretty much look at the knobs and tell what the sound is," he said. "I've programmed on the spot so many times for so many years onstage, doing the same songs with no presets." He said he often turns it on and starts playing a bunch of effects "wild," that is, not along with any song (but with a click). He'll have Pro Tools on, and he records the results. "I record myself tweaking and looking for stuff," he said (see **Web Clip 3**). "And in that process, I go back and hear a lot of stuff that I can use." Later, he'll take out pieces here and there and throw them into a song.

According to LRoc, his analog experience, both with synths and with recording gear, has been very helpful in his production work. It gives him a perspective that younger producers, who work only "in the box," don't have. "It really helps, you know—coming from being in the world of splicing tapes, rewinding, and fast-forwarding," he said. "Can you imagine what a kid would think of that today? 'Rewind? I got to wait for that?' But like I said, just having that experience of both worlds is definitely a plus when it comes to sonics and just about everything."

On to SouthSide

Later that afternoon, LRoc's engineer, Vincent Alexander, arrived. Alexander helps LRoc with

engineering and keeps the studio gear set up and running efficiently. After they worked on beats a while more, LRoc said that there was a mix session scheduled that day over at the SouthSide studio for Bow Wow's new CD. LRoc had helped work on the album, and he wanted to attend. "Sometimes they need me to add something," he said, "but I doubt it on this album."

The mix for the CD was down to the wire. "The album has to be turned in tomorrow," he explained, "because Bow Wow has a single out with Chris Brown that's called 'Shorty Like Mine.' Jermaine released it a couple of months ago, and it's blown up. When that song was released, we hadn't started the album. So we had to rush and finish it."

After a detour to LRoc's favorite Mexican restaurant for a quick bite, LRoc, Alexander, and I headed down the highway to Dupri's SouthSide studio complex. I asked LRoc if he ever mixes any of his own music, and he said no. "I know how I like it to sound. But as far as how those guys get the frequencies and the placements . . . it's tough understanding all that stuff, like how they use all the outboard, the formulas for delays, the right kind of delay and effects. That's my green area. I have to admit that everything I create is done with a team. I just master what I do and slowly learn the other stuff." But he does want to develop his mixing skills. "Eventually, I'd like to become really good at it," he told me.

Fun and Games

When we arrived, we found out that the mix session wasn't happening. There was a big huddle going on in the main control room concerning the Bow Wow album. The label wanted to release only a "clean" version that contained no swearing so that the CD wouldn't get an "Explicit" sticker. Bow Wow was apparently not happy with how the editing necessary to make it "clean" affected the final product. There was much discussion about what to do, especially considering that the album was due to the label the next day.

Not being involved with that aspect of the Bow Wow project, LRoc took this opportunity to show me around SouthSide. The main studio is a super-high-end Pro Tools–based room, with a large Digidesign Icon console in the control room (see **Fig. 3**). LRoc showed me several side rooms that are production and composing facilities featuring gear setups fairly similar to those at dakitchen, including MPCs and KV Audio EX 10 systems with subs. The EX 10s sit on speaker stands and are basically like P.A. speakers. With the subwoofer, they're capable of cranking out the bass and simulating club conditions. Apparently, KVs are ubiquitous on the Atlanta hip-hop production scene. LRoc had the same KV speaker system in his studio, and he said that Lil Jon has it as well.

The composing rooms have Dynaudio BM15A active monitors, just like the ones LRoc has at dakitchen. "Jermaine and I write on the Dynaudios and some Samsons, and once we get something we like and he wants to listen, we go to KVs," said LRoc. He said he sometimes does the same thing at his studio. "Especially when I'm doing a rough mix, then I use the KVs for reference. I go back and forth."

As impressive as all SouthSide's gear was, the studio's most astonishing feature was its indoor full-court basketball court. When people working at the studio have downtime, they can relax and play hoops. People less inclined to the hardwood can amuse themselves at a full-size billiard table or with video games. Clearly, Dupri has spared no expense to make SouthSide a comfortable working environment.

LRoc showed me one of the live rooms, and it featured an isolated room off to its side with a grand piano. He sat down at it and started playing fluently. "I've been practicing lately. I made a conscious effort to get my chops back up, because I've gotten lazy just with simple stuff. But I want to start doing more stuff and getting back to some playing."

Later, sitting and talking in one of SouthSide's production rooms, I asked LRoc about how his Liberian background had influenced him musically. "I listened to African music in general," he told me. "I listened to stuff that you could hear on the radio all the time. I was playing a lot of those songs by ear. And in bands. They've got different kinds of rhythms [than American music]. I understand other kinds of percussive rhythms and feels; I learned a lot from that, and I still apply it."

I asked LRoc what he hopes to accomplish in the near future. He said that he'd like to be able to produce more beats without a specific artist in mind, and then sell them. "I'm working on my skills of just doing tracks and getting them placed. If I can do that, then my income will increase," he said. "When I'm selling tracks that I made on my laptop, I'll be the happiest. I'll just get my little M-Audio bag and go on the plane, compose on the way to where I'm going, and when I get there, it'll be done."

In addition to increasing his proficiency in Logic and with his virtual instruments, LRoc said that the biggest challenge presented by producing beats strictly from his laptop is to successfully duplicate the sounds he gets in the studio. "I can create a song on it, but my signature—my drum sound—isn't the way I want it yet. I'm still working on that, and that's key." He knows that having killer sounds is crucial to staying successful in the hip-hop production game. "Sonically, it's got to beat so hard," he said, "because the competition will." EM

EM WEB CLIPS

Mike Levine is an EM senior editor.

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Train REX

By Jim Aikin

Quantizing MIDI tracks to REX files in Cubase 4.

Using REX files is a great way to add prerecorded beats to almost any project. One advantage of REX files, which are available from many sound developers, is that you can change tempos without compromising audio quality.

To play a REX file with a compatible plug-in, load the file into the plug-in, and then extract the file's MIDI data to a MIDI track in your sequencer using a utility command in the plug-in. (Check the plug-in manual for specifics.)

This data extraction adds two important capabilities to REX-file playback. First, you can reshuffle the individual hits in the REX file or change the feel of the beat by editing the MIDI data. Second, using your sequencer's groove template features, you can turn the MIDI data into a groove template to which you can quantize other MIDI tracks. That is especially useful when you quantize a bass synth to a beat; the rhythm will sound tighter if the bass locks to the drums (see **Web Clip 1**). Quantizing the bass to regular 16th notes might not do the trick, because the REX-file beat might have a more flexible feel.

Cubasics

But what if you don't have a REX-file player? Like some other sequencers, Steinberg Cubase will import REX files directly into audio tracks. The timing of the original file is preserved, and you can change the tempo. You can easily drag individual hits within the REX file forward and backward in time or to other tracks for separate effects processing.

You can't, however, extract the timing information from the REX file so as to quantize MIDI tracks to it. Cubase will create groove templates from audio beats that have been sliced apart using the program's own Hitpoints feature. (For most purposes, using this feature is as good as using REX files, though Cubase's slices lack the overlapping tails found in many REX files, as seen in **Fig. 1**.) But Cubase can't create a new quantization template from an imported REX file.

If you have Propellerhead ReCycle, you can use it to extract a REX file's associated MIDI file, and then use that as a groove template in Cubase 4. Fortunately, there's an easy work-around for those who don't have ReCycle (see "Step-by-Step Instructions" on p. 68).

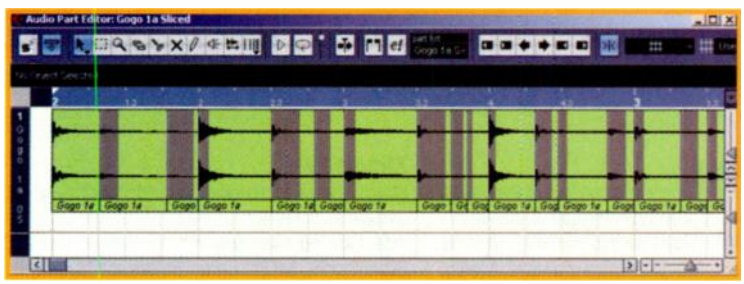


FIG. 1: REX files contain some extra audio at the end of each slice so that there won't be gaps when the tempo is slowed down. When a file is displayed in Cubase at its original tempo, the gray areas indicate overlaps between one slice and the next.

REX to MIDI

First, choose Audio File as the file type to import using the Import submenu of the File menu. In the dialog box, select the REX file you want to use.

Set the left and right locators in the transport to the start and end of your imported REX file. If your project already has some tracks recorded, mute all of the tracks except the one with the REX file, and then choose Audio Mixdown from the File menu's Export submenu. In the Export Audio Mixdown dialog box, make sure that you've checked both of the boxes in the Import Into Project area at the bottom. This will produce a "vanilla" audio track that should sound exactly the same as the REX file.

Double-click on the part containing the new audio to open the Sample Editor, click on the Hitpoint Edit button, and adjust the Hitpoint Sensitivity slider so that Hitpoints show up on all of the attack transients. Then go to the Hitpoints submenu of the Audio menu and select Create Groove Quantize From Hitpoints.

Try Before Buying

Before discarding the new audio part, try out your new groove template to make sure that the quantize template has enough points but not too many. Open up a Key Editor window for a MIDI part that you want to quantize, select the template in the Quantize menu if necessary, and look at how the notes relate to the light gray grid.

If there are notes that fall midway between the grid lines, don't quantize them to the grid. Instead, drag-select a few bars of notes, shift-click on all of the notes you don't want to quantize, and then press the Q key to quantize

STEP-BY-STEP INSTRUCTIONS

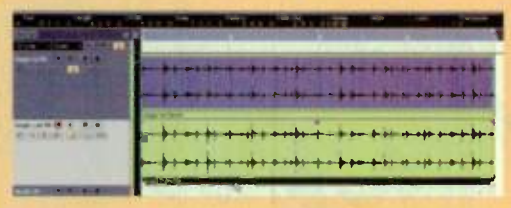
1

Find a REX file that you like, select an audio track, and use the Import submenu of the File menu to insert it into the track.

2

After soloing the track and placing the locators, use Audio Mixdown to bounce the beat.

3



Now you have two versions of the same beat, one REX and one plain audio. Here, I've dragged the new audio track up so that it's directly beneath the REX track.

4

Open the audio part and create Hitpoints using the Sample Editor.

5

While still in the Sample Editor, select Create Groove Quantize From Hitpoints from the Hitpoints submenu of the Audio menu.

6

Check out your new groove by quantizing a MIDI part to it.

those that are still selected. Presto—you've just quantized a Cubase 4 MIDI track to a REX file.

You can quantize start times of the slices in a REX file in the Audio Part Editor, as if they were MIDI events. You can easily add swing to a straight-time beat (or vice versa). Or you can import several REX files and extract groove templates from them, and then quantize one file to the groove of another. Using a Latin groove as the source for the template and applying it to a trap-set beat can yield an evocative feel. EM

Jim Aikin writes, teaches, and plays music in Northern California. You can visit him online at www.musicwords.net.

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Sounds in Motion

By Mitchell Sigman

Three ways to animate your vector synth.

Vector synthesis was invented by Sequential Circuits and first appeared in the company's Prophet-VS synthesizer in 1986. Subsequent classic hardware vector synths include the Yamaha SY22/TG33 and the Korg Wavestation series.

The Prophet-VS has a joystick that controls the mix of four oscillators, the volume of each oscillator being proportional to the joystick's distance from one corner of its travel. Vector synths usually provide one or more means for automating the movement of the joystick, and that can produce anything from subtle timbral shading to radical morphs.

Virtual Vectoring

Vector synthesis is making a comeback in the virtual world. Arturia's Prophet-V combines the classic Prophet-5 analog synth with the Prophet-VS. Korg has reissued the Wavestation in virtual form in its Legacy Collection. Other virtual synths that incorporate vectoring are Apple Logic ES2, Native Instruments FM8, and U-he Zebra 2. Zebra 2's vector implementation is impressive, offering four *x-y* pads and an intuitive MIDI Learn function. Teaming a vector soft synth with a MIDI *x-y* controller such as the Korg Kontrol or one of the Novation ReMote series gives you hardware control of vectoring in the digital domain.

I'll use Arturia Prophet-V for my examples, but you can adapt them to any of the soft synths just mentioned. **Fig. 1** shows the Prophet-VS mode of Prophet-V. The joystick in the Mixer section mixes the output of the waveforms chosen for the four oscillators to its left. Clicking on the numerical next to an oscillator label pops up a window displaying a graphic of the selected waveform, and moving the joystick to the corner for the same oscillator allows you to audition the waveforms as you scroll the numerical.

FIG. 1: The Mixer section of the Envelopes tab has a 5-segment envelope for automating the vector in Arturia Prophet-V.

Having chosen four waveforms, you can use the onscreen joystick to mix them in any proportion. You can also use the Mixer section of the Envelopes tab of the Modulation area to set up a 5-segment envelope automating the vector. You set up the automation by dragging the numbered squares to different positions, clicking on the Envelope button below the Mixer, and setting the four knobs to the desired times for the segments of the envelope (see **Web Clip 1**).

Clever Crossfades

Vector envelopes work well for sustained notes and chords but not for rapidly played notes, because the vector envelope retriggers every time a note is struck. And vector envelopes can be tricky to program, especially if you want them to follow the music rhythmically. An elegant alternative is to record MIDI controller data in the context of a song using your hardware joystick (see **Web Clip 2**). That allows the vector contour to evolve over many notes, but you do sacrifice each note having its own contour. The musical context will determine the preferred method.

If you don't have a joystick or other hardware *x-y* controller, you can use two standard sliders or knobs. First, map the controller's MIDI output to the soft synth's vector. In the Prophet-V, right-click (PC) or Command-click (Mac) on the onscreen vector control, and enter the MIDI Control Change numbers either manually or using MIDI Learn.

Once you have the hardware configured, record note passages into the sequencer as usual. Vector-manipulated sounds work well for quick 16th-note arpeggio or ostinato passages and long, sustained chordal motifs. Because there will be a lot of motion in the sound, don't blow through rapid chord changes; you'll have plenty of color with more-straightforward, simple parts.

Next, create a new track in your sequencer for the same MIDI channel and use that track to record the joystick controller data. You could overdub the controller data on the music track, but having a separate track will simplify editing. You can draw in automation as an alternative to recording the controller data, but recording usually yields a better feel (see **Web Clip 3**). EM

EM WEB CLIPS

Mitchell Sigman is a Los Angeles–based musician. He plays keyboards in the classic '80s synth-pop band Berlin.

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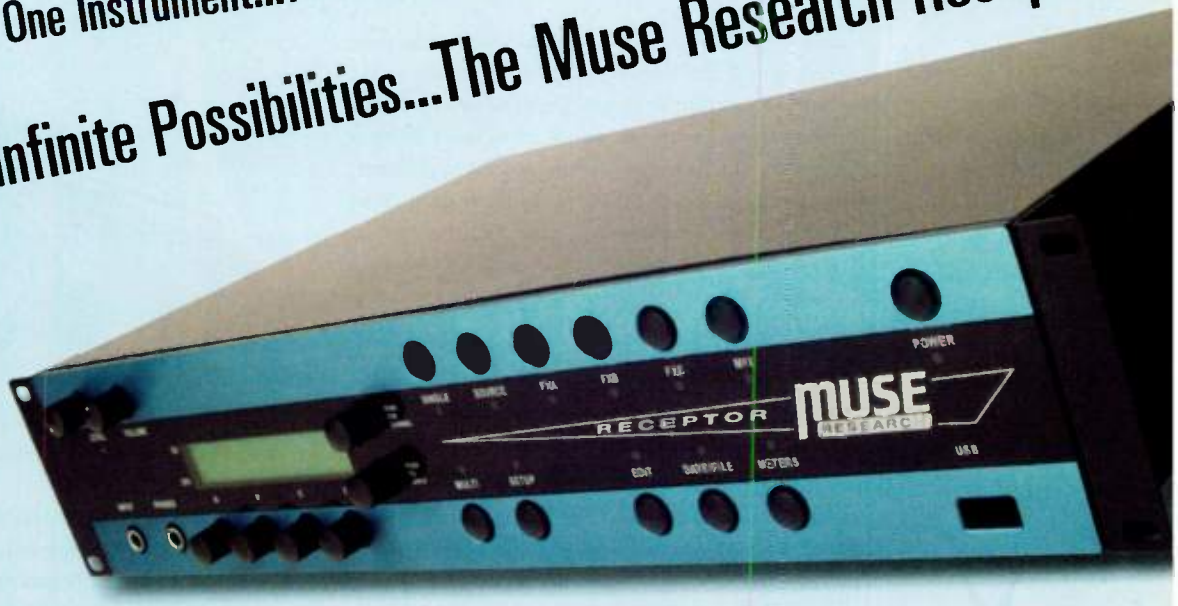

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Receptor: A New Way to Play.

Movin' On Up

By Jim Aikin

Upgrading your computer's audio hardware.

The audio input/output (I/O) that's built into your computer is fine for playing games and maybe for watching movies. But once you begin to produce your own music on a computer, you'll soon want to step up to a better-quality audio interface. At that point, you'll probably face a bewildering number of options in a market that's overflowing with competing products. In this column, I'll sort through the confusion and start you off on the right path. Note that I will discuss only interfaces—the peripheral components that convert analog audio into digital audio and back. Your computer might also have built-in speakers, but studio monitors are a topic for another column.

First off, what's wrong with your computer's current built-in audio hardware? Well, it might not have enough inputs and outputs to meet your growing needs. Furthermore, it's likely to be a bit noisy, because the manufacturer probably didn't spend much on the audio components. A little noise on playback is bearable, but any that is added to the audio signal on input will be recorded into the track. You'd have to spend hundreds of dollars on noise-removal software to get rid of that noise, and it makes more sense to spend your money on a better audio interface and get lots of other features in the bargain.

Audio Specs

Manufacturers love to toss out impressive-sounding specs when they introduce audio interfaces. But what do the numbers really mean?

FIG. 1: The Edirol UA-1EX costs less than $100 and provides stereo input and output at up to 24-bit, 96 kHz. It connects to the computer via USB.

When a stream of analog audio (such as a signal from your mixer) enters the interface, it's *digitized*—that is, it's converted to a stream of numbers. The *sampling rate* determines how many numbers are generated each second, and the *bit resolution* (also called the *bit depth*) determines how accurate each number is. More is better with both of those specs—up to a point.

Audio CDs store audio with 16-bit resolution, and their sampling rate is 44,100 samples per second, or 44.1 kHz (sometimes these specs are shortened to "16/44.1"). All audio interfaces on the market today record and play back audio with those CD-quality specs, and most offer 24-bit, 96 kHz operation. A 24-bit signal will be slightly cleaner, and a recording made at 88.2 or 96 kHz will have a more sparkling high end.

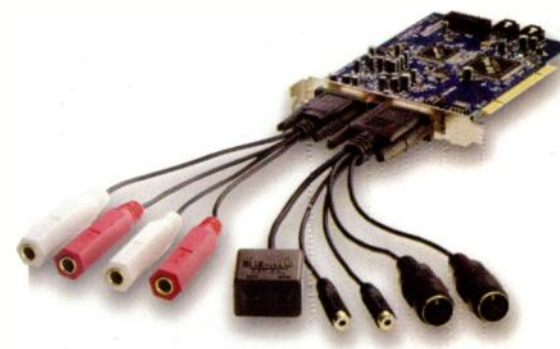

FIG. 2: The E-mu 0404 is an affordable 2-in, 2-out interface that has zero-latency monitoring and onboard DSP.

Although 32-bit, 192 kHz interfaces are available, when you record audio at a higher rate and resolution, your computer has to work harder and you'll quickly eat up space on your hard drive. If you have an older computer or are interested mainly in doing home demos and honing your craft, 16-bit, 44.1 kHz is more than adequate, even though your interface might be capable of more.

Connections

Other reasons to upgrade your computer audio hardware include having more inputs and outputs and having the kind you need. Interfaces with 4, 8, or more ins and outs are readily available. Unless you're planning to record a full band and capture each instrument on a separate track, a basic 2-in, 2-out interface such as the Edirol UA-1EX will be more than sufficient to get you started (see Fig. 1).

Many interfaces have built-in microphone preamps (or *mic pres*), which are usually associated with 3-hole XLR jacks. The purpose of a mic pre is to boost the low-level signal coming from the mic to the same level as the rest of your audio. Professionals often use dedicated hardware mic pres because they also impart a subtle but pleasing quality to the sound. When you're

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Chris Pitman- Guns N' Roses

My favourite synth of all time is now a soft-synth thanks to the wonderful people at Arturia. As with all their soft-synths Arturia have given us a meticulously crafted software version of the Classic Jupiter-8 with some fabulous extra features to bring us into the 21st century. This is a must have for all of us JP8 lovers.

Howard Jones

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initially building a computer audio system, though, you should be able to get by with an interface that has its own mic pres.

Many interfaces have dedicated headphone outputs. If you're planning to record with other musicians, having two headphone outs, each with its own volume knob, may be desirable.

Also desirable is *zero-latency monitoring.* If you're listening to your own performance while you record (as opposed to listening to already-recorded backing tracks, which all interfaces will allow), and if the audio signal you're generating passes through the computer before coming back out to the headphones, you'll become aware of a small time lag. A time lag of this sort is called *latency.* Latency is less of an issue today than it was a few years ago, but even when using newer gear, some musicians find it distracting. An interface that passes its input(s) directly back to the headphone/monitor outputs at the same time it's being sent to the computer will remove the distraction caused by latency.

In addition to analog I/O suitable for connection to a mixer, most interfaces include some form of digital audio I/O. This is usually in the form of stereo S/PDIF connectors (either optical or coaxial). Some interfaces have 8-channel ADAT digital audio I/O. Digital audio connectors are used for noise-free signal routing to other digital devices.

Many audio interfaces have one or more MIDI In and Out jacks. You can use these jacks to connect your computer to a keyboard or some other type of MIDI controller for sequencing. If you plan to do any MIDI sequencing and don't have a MIDI interface yet, an audio interface that has MIDI will be very useful.

The Computer Hookup

Ten years ago, audio interfaces usually connected to the computer via a circuit board that was plugged into a PCI or NuBus slot inside the computer. Some manufacturers still build PCI-based interfaces, but the more common ways to connect an interface to a computer today are via USB and FireWire. Yet another connector, the PC Card slot, is gaining in popularity.

USB and FireWire interfaces tend to be 1-box solutions. An interface that uses the PC Card slot has a second box or a breakout cable, which connects to the card that slides into the slot. The same is true of most PCI interfaces (see **Fig. 2**).

All of these hookups work fine for audio, as long as the manufacturer's driver software is solid. A *driver* is a small program that you install when you buy the interface. The driver is required for the computer to "see" the interface as an audio device. How can you tell whether the driver is trouble-free? Check the manufacturer's user forums for discussions of problems and solutions, and then cross your fingers and hope for the best.

Some interfaces can receive electrical power via the USB or FireWire connection. That is convenient, especially if you're carrying a laptop around to various recording locations and don't want the hassle of plugging in an extra power supply. If you do this, keep the laptop plugged in, because an audio interface will drain the battery quickly. Also, not all laptops provide power to their connectors. Before you buy, check with the interface manufacturer to learn about compatibility with laptops. And if you need multichannel audio, stay away from USB 1.1—it's slower than the others (many older laptops have USB 1.1).

FIG. 3: The Mackie Onyx 1220 mixer connects to your computer via an optional FireWire card that installs inside the mixer's chassis.

Multipurpose Boxes

Many manufacturers have started building audio interfaces into other types of hardware. If you shop around, you'll be able to find MIDI keyboards, mixers (such as the Mackie Onyx 1220; see **Fig. 3**), and control surfaces that supply their own computer audio I/O. Depending on your needs, this type of all-in-one product may be a great option. At present, most of the audio I/O found on MIDI keyboards is basic, with no frills. You can buy a mixer, however, with many channels for conventional mixing and also many simultaneous channels of audio streaming to and from the computer.

Speaking of bundles, you'll often find free software extras included with some interfaces. These might include digital audio sequencers such as Steinberg Cubase LE or Ableton Live, audio editors, and other types of music software. The bundled software can be a terrific incentive if you're just getting started, but be aware that the versions of well-known programs that are often bundled with hardware are older or limited in some way.

There's never been a better time to upgrade your computer's audio I/O. If you do a little online research and read EM faithfully, you'll have no trouble finding the interface that's right for you. EM

Jim Aikin is a music-technology expert, a cello teacher, and a hobbyist computer programmer. You can visit him online at www.musicwords.net.

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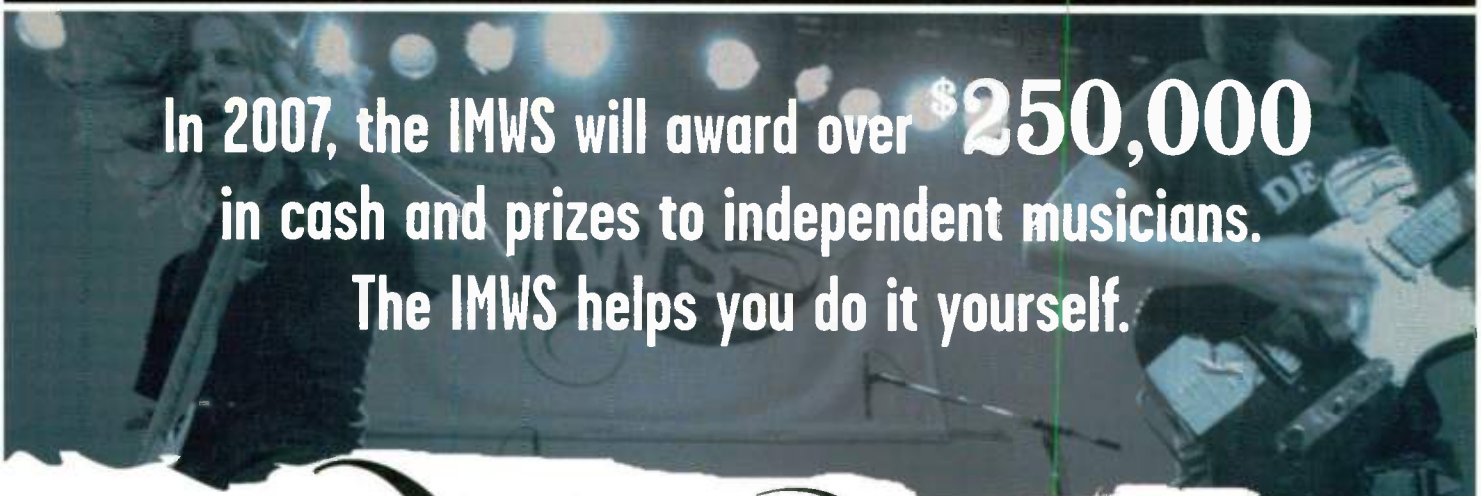

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Sebastian St. John, Rose's Pawn Shop

Q&A: Michael Laskow

By Mike Levine

Gazing into the future of the music business.

Predicting the future is never easy, but trying to anticipate what it will bring can help a lot as long as you really know the subject that you're speculating about. So when I was looking for somebody to forecast which opportunities and challenges independent musicians will face in the next five or ten years, I turned to Michael Laskow (see **Fig. 1**). As the founder, president, and CEO of Taxi (www.taxi.com)—an organization that helps unsigned bands, artists, composers, and songwriters get label deals and find placement for their songs and compositions—Laskow is particularly attuned to the circumstances that recording musicians encounter.

Initially a recording engineer, Laskow has worked with artists such as Eric Clapton, Neil Young, and Crosby, Stills, Nash, and Young, to name just a few. In his years working at Taxi, he has dealt frequently with record labels, music publishers, music supervisors, and production music libraries, as well as with other similar entities. In short, he has been living and breathing the music business for the past 32 years.

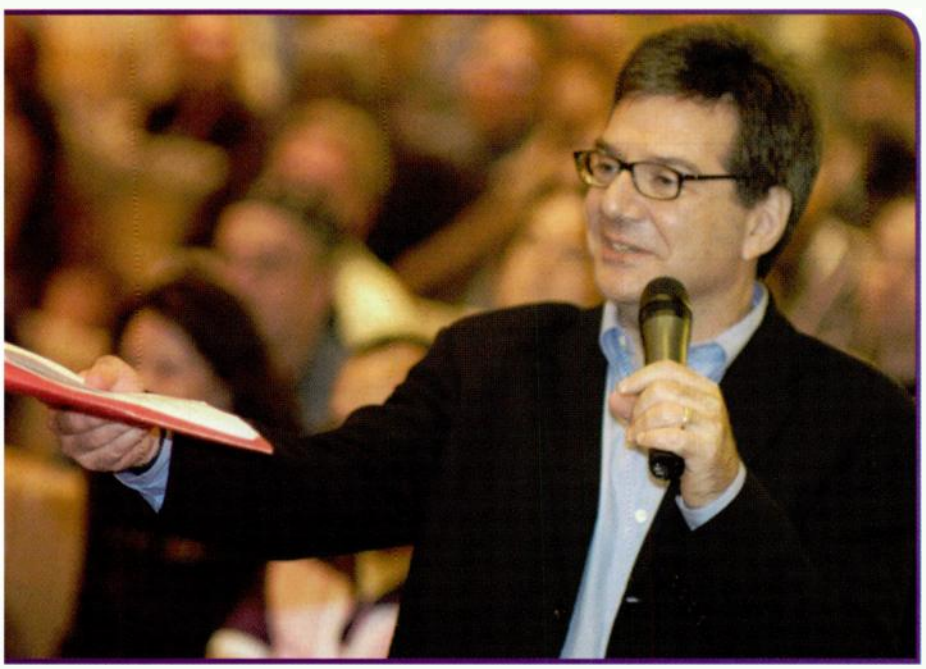

FIG. 1: Michael Laskow sees the emergence of a musician "middle class" in the not-too-distant future.

JIM DIMODICA

Where do you see the music business going in the next five to ten years?

Things are going to change. The major labels simply won't be able to operate the way they do now. However, the whole music-revolution thing isn't what it appears to be. Musicians are so used to hating major record labels, and it's easy for them to think, "Screw the major labels; I'm going to do the independent thing." But truthfully speaking, most musicians don't really hunker down and do the hard work and the heavy lifting, so they stay right where they are. They might build a page on MySpace, but they'll be one of millions. They'll probably put their money into recording a CD, have it pressed, and make it available for sale online—all of which are good ideas. But then they stop right there, so that in the end, all those independent-music revolutionaries don't end up any better off than before their CD was produced. They are somewhat invisible today, and most of them will be relatively invisible tomorrow.

How do you see things changing? Do you think that technology will alter the delivery system for music?

The delivery system has already been changed. Downloading and streaming are certainly easy enough to do. In a perfect world, people could go to a Web page one time and fill out a survey, answering items such as "These are acts that I like," "These are acts that I've liked in the past," and "These are genres that I listen to." They could subscribe to services that filter the music for them and deliver it to them in the order that they want. So they would be able to get whatever they want, wherever they are, whether it's tethered to a wire or untethered via satellite.

Into a portable unit of some sort?

Yeah. I think that someday in the not-too-distant future, you will be able to access your own personal jukebox of music that will reside somewhere else on a hard drive, so that you don't have to schlep a device with you. Why should you have to carry an iPod or a cell phone

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that has a hard drive with you? Why can't you just have your music and be able to access it wherever you are? Let's say that all the music is stored on a central hard drive; if I go to your house for a party, I could punch in my PIN code and pull up "Laskow's disco party mix" on your home stereo. Or if we're doing a road trip together, I could pull up my list in your car.

Do you think that the interaction of video and music will become more important as Internet video delivery becomes even faster?

That will be determined by demographics. It will be very important for the 13- to 24-year-olds; we're seeing strong evidence of that fact demonstrated on YouTube. Kids don't necessarily want to watch some $300,000 video made by a major label. Now they think it would be much cooler to shoot their own video, whack it together in iMovie, and throw some band's song underneath it.

You don't need to go to a $300-per-hour, 48-track digital studio.

How do you foresee that musicians will be marketing themselves and their music in five or ten years?

The key word is *marketing*. It used to be that you could just create great music, and maybe an A&R guy would find you because you were talented. But now, there's so much clutter out there. I came to a realization reading in bed two weeks ago: I was reading *The Long Tail* by Chris Anderson [Hyperion Books, 2006], which is an excellent book. My realization was that the type of people who got the attention of record companies in the past year or five years ago, built a following, got signed, and maybe even had a big success will probably be the same ones who will succeed in the future—because they work hard at it.

So even though technology is changing, only a tiny percentage (like about one-tenth of 1 percent) will ever have what I call a huge success—that is, one that puts hundreds of thousands or millions of dollars in their pocket. What I'm hoping will come about as a result of the Internet and other delivery systems, though, is the creation of a musician "middle class."

Can you expand on that concept?

The advent and continuous growth of high-quality, inexpensive recording gear for the home-based recording musician and engineer has created an abundance of good music that works for all kinds of things, not just hit records. Film and TV placements are the most obvious. You don't need to go to a $300-per-hour, 48-track digital studio with a 15-foot-long SSL sitting in the room to record great-sounding tracks that are just fine for film and TV.

So musicians who have a knack for composition, some recording gear, and decent engineering chops could be part of that musician middle class?

I think that we're 60 percent of the way there already. You can take somebody with a decent little home studio and reasonably good skills using that studio, and they can earn $150,000 a year creating music for film and television. It's absolutely possible. I know because I've seen it with my own eyes. I'm not talking about a scenario where you get a song placed in a movie and make $50,000—that's not likely to happen. But it's definitely possible to work diligently and get dozens of songs placed each year in production music libraries. And over a five- to ten-year period, those songs multiply to become hundreds of songs in several different libraries, and this thing just snowballs over time.

How would you recommend that musicians prepare themselves in order to adapt to the changing musical landscape?

Technology has dramatically improved a musician's ability to create and distribute music. But technology hasn't changed a musician's ability to market their music. So if I were a musician, I would sit down and decide what my goal is: Do I want to be a rock 'n' roll star and fly around on a private jet and have a briefcase filled with millions? Or do I want to be able to make my own music, do what I love to do, and earn my living that way? Once you determine what your goal is, then make a list of steps. Most musicians are creative people and tend to skip those businesslike functions.

Figuring out a career path isn't that hard. What is hard is making yourself do the hard work and the heavy lifting. Don't rely on the technology to do the work for you. Yes, it can help you make better music. Yes, it can help you get that music out to more people. But you have to know how to do the marketing. EM

Mike Levine is an EM senior editor.

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MAKEMUSIC Finale 2007 (Mac/Win)

A significant upgrade to a powerful notation program.

By Peter Hamlin

Music-notation software Finale has been around since 1988, an eternity in personal-computing years. Over time, this sophisticated and feature-packed program has become an invaluable tool for musicians who work with music notation. It's impossible to give a comprehensive assessment of such an extensive program, but I can offer a general idea of how Finale works, describe some of its distinctive features, and discuss some significant upgrades from previous versions (you can read reviews of Finale 2003 and 2004 online at www.emusician.com).

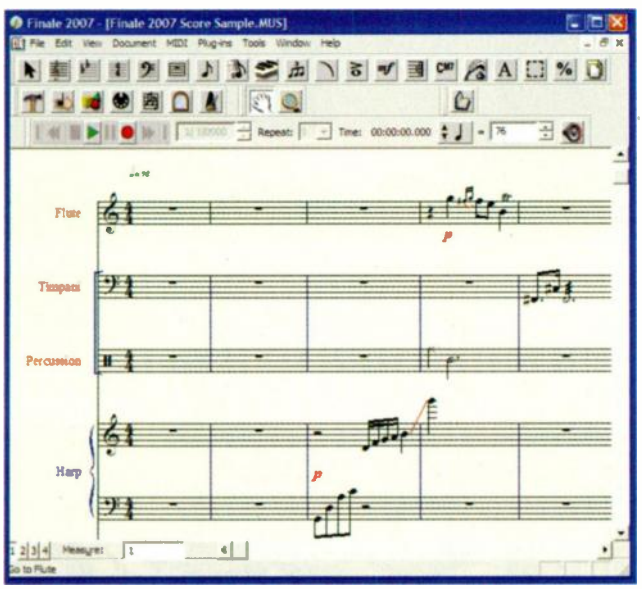


FIG. 1: Finale's work space is dominated by the score. Above the score are tools to enter and edit the various musical elements. You can customize how the tools appear in palettes and remove the icons of those tools you don't use.

Where to Begin?

Finale is a huge program, but it isn't difficult to learn if you approach it one step at a time, learning what you need to know when you need it. A series of excellent tutorials takes you through the program piece by piece.

You get started with Finale by running the Setup wizard to establish the staves, transpositions, key and time signatures, and other basics. When you're ready to enter music, you can choose one of several note-entry tools for that purpose.

GUIDE TO EM METERS

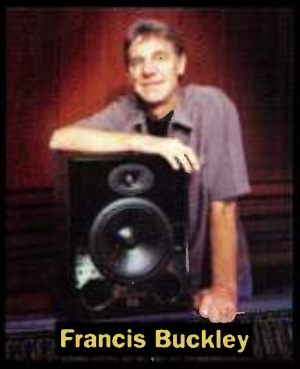
5 = Amazing; as good as it gets with current technology
4 = Clearly above average; very desirable
3 = Good; meets expectations
2 = Somewhat disappointing but usable
1 = Unacceptably flawed

Specifications tables for EM reviews can be found at www.emusician.com/specs.

"We Switched"

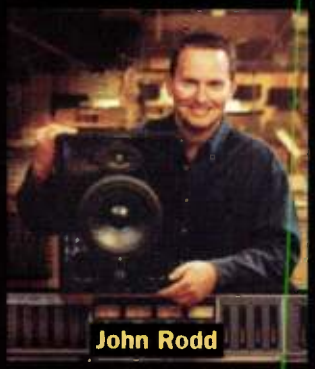
Grandmaster Flash

Producer / DJ / Recording Artist



Francis Buckley

Multi-Platinum Grammy-Winning
Engineer / Producer

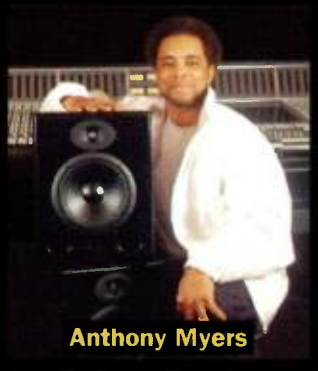


John Rodd

Orchestral Scoring Recordist
Music Scoring Mixer

"Prince Charles" Alexander

Multi-Platinum Grammy-Winning
Engineer / Producer

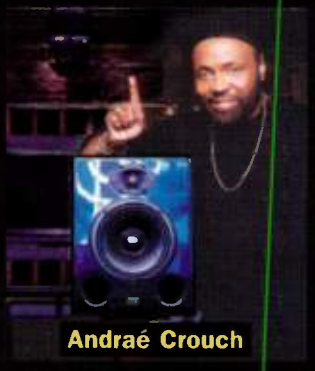


Anthony Myers

Engineer / Producer
Sound Designer

Gerhard Joost

Multi-Platinum Engineer
Mixer / Producer



Andraé Crouch

Multiple Grammy-Winning
Gospel Singer / Producer / Pastor

Lionel Jarvis

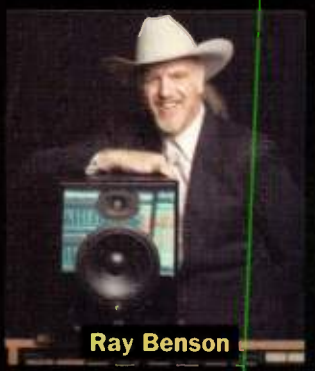
Multi-Platinum Musician
Musical Director

DJ Ron G

Remixer / DJ

Steve Pageot

Grammy-Winning Producer
Musician / Composer / Engineer



Ray Benson

Multiple Grammy-Winning
Singer / Songwriter / Producer

Michael Jay

Emmy-Winning Composer / Engineer
Editor / Studio Consultant

Studio Precision™ Biamplified Direct Field Monitor

Audio Quality: 5.0 Features: 5.0 Ease of Use: 5.0 Value: 5.0

Cons: None *—Electronic Musician*

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Another method is Step-Time Entry, which lets you play notes or chords on your MIDI keyboard while selecting the rhythmic values on your numeric keypad (pressing 7 for a whole note, 6 for a half note, 5 for a quarter note, and so on). You can choose from two fast and powerful tools for Step-Time Entry: Simple Entry and Speedy Entry. Their functionality overlaps, but they are different enough that you may prefer one over the other.

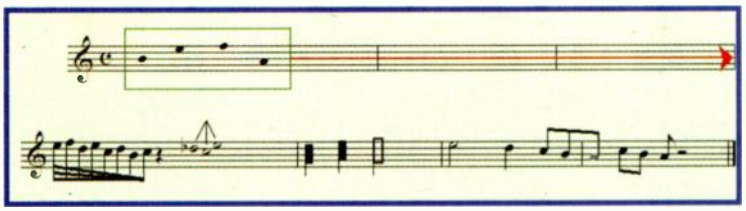


FIG. 2: Special tools allow you to customize your notation. Some examples are indeterminate notation (top staff), feathered beaming (to indicate a slowing of tempo), tone clusters, and beaming across the bar line (bottom staff).

You can play music in real time using the HyperScribe tool, a technique that offers several quantization options so that Finale accurately transcribes your playing. Another alternative is MicNotator, which allows brass and woodwind players to transcribe monophonic performances using a microphone along with Simple Entry, Speedy Entry, or HyperScribe. You can also import a MIDI file, a scanned printed score, or one of several notation file formats.

All the Right Tools

Finale is organized into 30 tools, each with a particular function (see **Fig. 1**). In addition to the note-entry tools, the program has tools to enter or edit time signatures and key signatures, add articulations, insert dynamic markings and other expressions, format the score, and even microedit a passage to create nontraditional notation (see **Fig. 2**).

Space limitations for this review prevent me from discussing all of Finale's tools and their functions, but suffice it to say that you'll use some tools frequently and others (such as the one for creating *ossia* passages) less often. Finale gives you tools for creating chord symbols and fretboard charts, early-music notation, lyrics—the list encompasses just about any music-notation symbol or situation you can imagine (see **Web Clip 1**). The beauty of Finale's organization is that you can concentrate on the tools you need and ignore the others. You can even customize the tool palettes to show only the tools you'll use.

The Smart Shape tool is available for entering slurs, trills, dynamic hairpins, octave symbols, glissandos, and more. The Smart Shape tool's curves and lines are "smart" in the sense that they adapt their shape and size automatically, stretching and shrinking to accommodate changes in the music and automatically adjusting to span the next system of music if needed. Smart tools such as glissandos and dynamic hairpins are also automatically reflected in the score playback.

Another longtime feature of Finale is Metatools. Metatools are macros that let you assign customized keystrokes to enter a particular notational symbol. For example, when using the Articulation tool, the default Metatool for an accent is the A key, and the Metatool for staccato is the S key. If you click on a note while holding those keys, you instantly create the articulation. You can also lasso-select a group of notes while holding the appropriate key to enter many articulations at once. Entering symbols in this way is extremely fast and efficient.

One of my favorite features in Finale is Scroll view. Though you can view a score as it will appear on the page (Page view), Scroll view allows you to see it as a continuous staff flowing across the screen from left to right. I haven't seen this option in many other notation programs, but it's useful, especially during the early stages of composition.

Request an Upgrade

Since EM's last review of Finale, the program has gained several valuable features. It has long been available for the Windows and Macintosh platforms, and Finale 2007 offers versions that run natively on both Intel- and PowerPC-based Macs.

An important addition to Finale 2006 was the inclusion of Garritan Personal Orchestra, which greatly improved the playback quality of scores (see **Web Clips 2 and 3**). GPO sounds more realistic in Finale 2007 because Finale's Human Playback features have been expanded to better reflect tempo changes, dynamics, and articulations indicated in the score. Human Playback also supports a wide range of performance techniques, and many of them (such as bends, glissandos, and harmonics) are idiomatic to each instrument. Although the default settings work well, Finale has detailed options for customizing Human Playback if desired.

Another feature added in Finale 2006 was Studio

PRODUCT SUMMARY

MAKEMUSIC Finale 2007

notation software

$600 (academic price, $350)

FEATURES	5
EASE OF USE	4
DOCUMENTATION	4
VALUE	4

RATING PRODUCTS FROM 1 TO 5

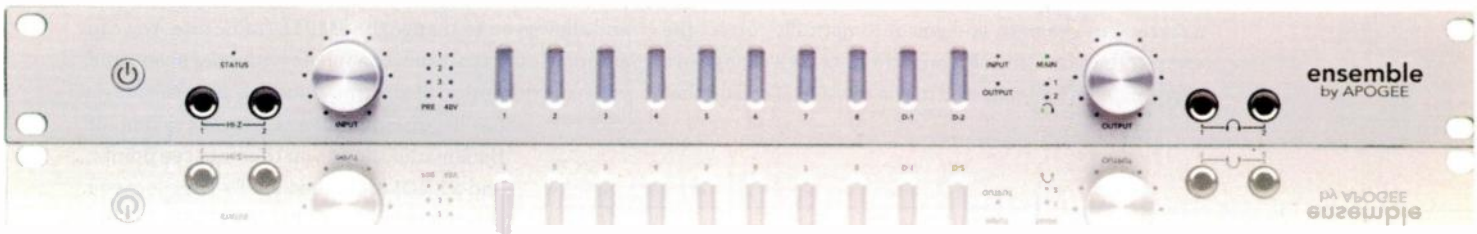
PROS: Extensive features. Valuable updates. Supports wide range of notation styles. Offers alternate techniques for using many functions. Good technical support.

CONS: Some rarely used customization features are difficult to use.

MANUFACTURER
MakeMusic, Inc.
www.finalemusic.com

Apogee Quality, Apple Integration & Sweetwater Expertise.

The Native Dream Team.



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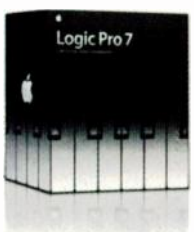

Apple Logic Pro 7 – Everything you need for creative audio production. Once your audio reaches your Mac, you need a powerful, flexible application to turn your musical vision into reality. That's where Logic Pro comes in. With more than 30 software instruments, over 70 effects and distributed audio processing, Logic Pro is the choice of musicians, engineers and producers the world over.

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view, offering a mixer and other functions that resemble a sequencer's controls. Also, a TempoTap feature lets you adjust the tempo for lifelike performances (see Fig. 3).

Finale 2007's most significant new feature is Linked Parts. Thanks to Linked Parts, you don't need to create separate files for each orchestra part; the parts are created automatically and integrated into the same file with the score. Make a change in the score, and that change is automatically reflected in the appropriate part. Make a correction in a part, and you automatically correct the score at the same time. If you want to make a change in a part that won't be reflected in the score, you can do that too. A color code helps you keep track of which items in the part are linked to the full score. Finale's implementation of Linked Parts makes creating, editing, and managing parts effortless and trouble-free.

Finale 2007 also introduces integrated onscreen video. Before this version, you could work with video clips only by syncing to an external editor. Now an integrated video window makes the process much more convenient. Finale supports a wide range of video formats and allows you to change the SMPTE frame rate. You can automatically synchronize a movie with your score, and you have the option of starting the video after the music has begun or vice versa. Finale's system of Bookmarks allows you to create cue points, and a MIDI tool called Fit To Time lets you select a region of the score and specify either its length or end time. Fit To Time then changes the tempo to arrive at those times, ensuring that the music matches the desired locations in the video.

Finale 2007's Latin Percussion plug-in allows you to instantly create Latin-flavored rhythmic passages. Simply select a region of your score, bring up the plug-in, and select from a list of Latin styles including several varieties of Afro-Cuban, bolero, cha-cha-cha, mambo, and salsa. A percussion section with authentic rhythms, performance practices, and notation conventions is instantly created in the selected region (see **Web Clips 4 and** 5).

Although the version of GPO that comes with Finale is extensive, you can purchase expansions of the library. You can buy the full Garritan Personal Orchestra for $189 and Garritan Jazz and Big Band for $239. In addition, Garritan's General MIDI Collection ($139) and Marching Band ($199) should be available by the time you read this.

Is Anything Wrong?

Although Finale 2007 doesn't give me much to complain about, I still have a wish list. First, the index of the PDF manual could be easier to navigate; when searching for information, I often wish for a better way than scrolling through several screens of the 2-column index. On the other hand, clicking on an item in the index jumps to the appropriate page. Also, a few customization features are unnecessarily complicated to use. The tools for creating custom key and time signatures and playback controls called Executable Shapes are very powerful, but the process of creating them

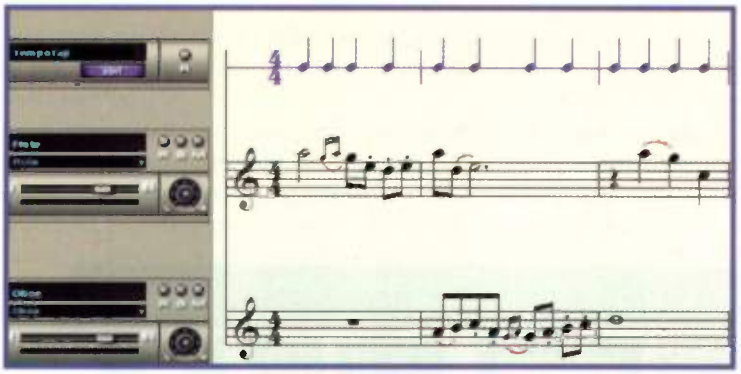

FIG. 3: Studio view combines traditional music notation with faders and panners reminiscent of a sequencer. The top staff is used with the TempoTap feature that lets you conduct tempo changes by tapping a MIDI or QWERTY key.

could be somewhat smoother and more streamlined.

Additionally, it would be wonderful if Finale could implement automatic rehearsal letters. Currently, rehearsal letters are created manually, which means that if you change the music in a way that puts the rehearsal letters out of order, you have to edit them all by hand. A system that would automatically keep rehearsal letters in sequence even when you moved them would be a welcome addition.

Standing Ovation

About 95 percent of the time that I work on a score, I pay no attention to Finale. The program has become as natural a part of the compositional process as my piano and pencil. The only time I think about Finale is when I don't know how to notate something and I have to think a bit or dig into the documentation to do what is needed. In such a situation, I especially appreciate Finale's many options for controlling almost every aspect of a score's appearance.

Finale is a mature, sophisticated, flexible, and powerful music-notation program. If you already use Finale, I strongly suggest that you purchase the 2007 upgrade. And if you're in the market for a notation program, give Finale a serious look.

EM WEB CLIPS

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

Some less commonly used functions are hidden, however. You'll need to read the manual to discover, for instance, how to adjust the Velocity sensing of a patch or set the keyboard to MIDI Local Off mode. These functions and a number of others are accessed from the front panel via button combinations.

Around the back, you'll find a strain-reliefed input for the lump-in-the-line power cord, stereo analog audio inputs (RCA phono, unfortunately), and a multipurpose pedal input (see **Web Clip 1**). You can attach either a footswitch or a sweep pedal. USB, MIDI, and a headphone output complete the connector complement.

Voicing

There's no denying that the SH-201's voice architecture lacks some of the refinements of instruments costing three times as much. But it has all of the necessities and includes a surprising number of extras. The factory presets cover all of the usual analog-modeling bases, from cutting bass to ethereal pads by way of chimes and pulsing arpeggios. Overall, the SH-201 sounds rich and lively, with lots of presence. It can cut through on top of a dense drum track or fill in the spaces in a mix with subtle warmth.

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Ring modulation and oscillator sync are included. By selecting the external audio input as the oscillator "waveform," you can ring-modulate it from the other oscillator and process it through the voice filter and effects. One of the very few areas where I felt the SH-201 could be improved was the pitch-envelope depth. It maxes out at one octave, which is not enough for truly hair-raising synced oscillator sweeps.

The synth engine's resonant multimode filter may not be quite as fat sounding as the filters on high-end synths, but it's very good indeed. Depending on the patch, some stair-stepping will be audible when the cutoff knob is turned, as its data output is not being smoothed.

The LFOs provide a choice of seven waveshapes. Each LFO has two selectable destinations, not including panning, unfortunately. The SH-201 has stereo

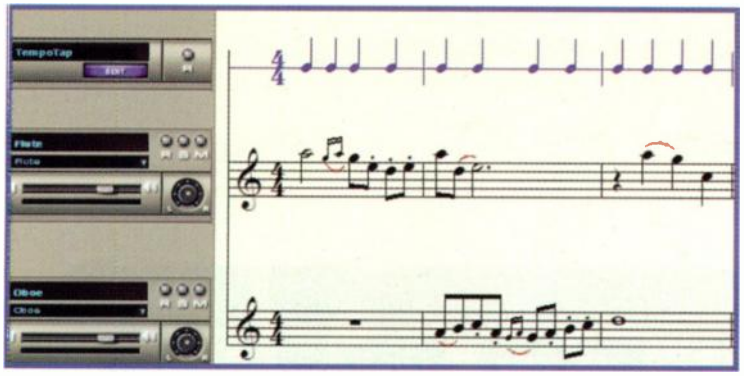

FIG. 3: Studio view combines traditional music notation with faders and panners reminiscent of a sequencer. The top staff is used with the TempoTap feature that lets you conduct tempo changes by tapping a MIDI or QWERTY key.

could be somewhat smoother and more streamlined.

Additionally, it would be wonderful if Finale could implement automatic rehearsal letters. Currently, rehearsal letters are created manually, which means that if you change the music in a way that puts the rehearsal letters out of order, you have to edit them all by hand. A system that would automatically keep rehearsal letters in sequence even when you moved them would be a welcome addition.

Standing Ovation

About 95 percent of the time that I work on a score, I pay no attention to Finale. The program has become as natural a part of the compositional process as my piano and pencil. The only time I think about Finale is when I don't know how to notate something and I have to think a bit or dig into the documentation to do what is needed. In such a situation, I especially appreciate Finale's many options for controlling almost every aspect of a score's appearance.

Finale is a mature, sophisticated, flexible, and powerful music-notation program. If you already use Finale, I strongly suggest that you purchase the 2007 upgrade. And if you're in the market for a notation program, give Finale a serious look. EM WEB CLIPS

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

FIG. 1: Terratec Producer has updated the Axon AX 100 guitar-to-MIDI converter hardware and developed a software editor for it.

TERRATEC PRODUCER Axon AX 100 mkII

Guitar-to-MIDI conversion takes another step forward.

By Orren Merton

By the time Terratec Producer acquired the Axon AX 100 from Blue Chip Music, the formerly cutting-edge guitar-to-MIDI converter needed updating. A new and improved version featuring numerous cosmetic and functional enhancements was released in 2006. Immediately evident is a redesigned hardware case, and on the front panel, a stereo headphone output with a dedicated level control has replaced the original's monophonic ¼-inch input (see **Fig. 1**). An internal soundboard has also been added.

Firmware updates accommodate independent tunings for each string, allowing for alternate guitar tunings such as Open G and Drop D, no matter how your guitar is tuned. They also give you the ability to save setup data (including tunings) with each of the 256 presets. Further updates let you specify destinations (the internal soundboard, MIDI Out, or both) and give you the ability to send Program Changes on a MIDI channel separate from the channels that carry note data. Perhaps the most exciting addition, however, is Terratec's fully featured Axon software editor. As a bonus, the Axon AX 100 mkII ships with Steinberg Cubase LE.

Nuts and Bolts

A guitar-to-MIDI converter detects the pitch and dynamics of a guitar performance and uses that information to generate MIDI data. This process requires a guitar that's equipped with a hexaphonic pickup system. Because only a handful of guitars come standard with hexaphonic pickups, aftermarket systems are available from Axon, Roland, RMC Pickups, Graph Tech, and others. Hexaphonic pickups send signals from individual strings to the Axon through a cable with 13-pin connectors. The Axon then translates those signals into a stream of MIDI data for each string. Because hexaphonic systems also

Such flexible splitting allows you to create some amazing fretboard setups.

send analog audio signals from the magnetic pickups to the Axon, it can simultaneously transmit MIDI and audio signals.

All too often, the process of detecting and converting a guitarist's performance introduces triggering inaccuracies and tracking latency. To combat such problems, according to Terratec, the AX 100 mkII features an early-recognition system that evaluates the initial picking impulses from the string and immediately generates MIDI data defining pitch, dynamics, and note length from those impulses. Terratec claims that this technology makes the AX 100 mkII the fastest and most accurate guitar-to-MIDI converter on the market today.

The AX 100 mkII's internal soundboard offers 500 General MIDI– and XG-compatible sounds, enabling the Axon to function as a true guitar synthesizer. I was unimpressed by the sounds, however; though adequate, they're not much better than a computer's GM-compatible QuickTime sounds. But clearly the Axon's controller and conversion abilities are its real strength.

Ins and Outs

The single-rackspace AX 100 mkII's front panel contains the 13-pin guitar input, the aforementioned headphone output and level knob, a 2 × 16–character LCD, and a Display Contrast knob. The front panel also has a power switch and a complement of buttons for programming the unit.

The outputs are on the Axon's rear panel (see Fig. 2). It has two unbalanced ¼-inch audio outputs for the internal soundboard and ports for MIDI In, Out, and Thru, as well as jacks for two expression pedals and two footswitches. You can use footswitches to control Hold functions, the internal sequencer, and the arpeggiator. The power connection for the included 12 VAC wall wart is next to those outputs. On the far right is an unbalanced ¼-inch output for routing the guitar's magnetic-pickup signal to a guitar amp or mixer.

I connected my guitar to the AX 100 mkII with a 13-pin cable and then connected the unit's MIDI In and Out to my Mac's audio/MIDI interface, an RME Fireface 800. The Axon communicated flawlessly with Apple Logic Pro 7, but not with the Axon's software editor (which I'll discuss later). When I connected the Axon's MIDI to my Korg Kontrol49 USB keyboard, however, the Axon successfully communicated with the software editor as well as Logic Pro. I dearly wish that the AX 100 mkII had its own USB port, which might eliminate such connectivity issues. Terratec won't comment on future products or updates, but the company is aware that USB is a highly requested feature.

FIG. 2: The Axon AX 100 mkII's rear panel houses all of its audio, MIDI, footswitch, and pedal connections.

PRODUCT SUMMARY

TERRATEC PRODUCER Axon AX 100 mkII

guitar-to-MIDI converter
$699

FEATURES	5
EASE OF USE	3
DOCUMENTATION	3
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Fast and accurate. Flexible splitting capability. Onboard arpeggiator and step sequencer. Excellent software editor.

CONS: No USB. Disappointing onboard sounds.

MANUFACTURER
Terratec Producer/Synthax (distributor)
www.synthax.com

Doing the Splits

One of the Axon's key features is its flexible splitting capability. A String Split lets you divide your guitar's strings into two groups, allowing you to trigger a bass sound with the bottom two strings, for example, and a piano sound with the upper strings. You can place the split between any strings you want. You can also create a Fret Split, which divides the length of your fretboard into two areas. Like String Splits, Fret Splits can be set up anywhere on the fretboard. A Pick Split divides the picking area between the bridge and the highest fret into as many as three sections.

You can combine the three different split types into as many as 12 possible combinations. Each split can have its own instrument assignment, MIDI program, volume, transposition, Pitch Bend quantization, attack time, and even MIDI Control Change (CC) values. Each string can also be transposed separately; unlike the alternate MIDI tuning feature, which allows different transpose values per string, each split's transposition parameter affects all six strings.

Such flexible splitting allows you to create some amazing fretboard setups, in which three picking zones can control various synth parameters using MIDI CC messages, different strings and fret positions can play different presets, and so on. You can transpose or quantize certain instruments without transposing or quantizing others, which allows natural pitch bending.

The AX 100 mkII provides four modes for quantizing Pitch Bend data. On and Off modes enable and disable quantizing for a given split; On mode quantizes in semitone steps, and Off mode continuously responds to bends. Trigger mode works like Off mode but affects only the current split zone. My favorite quantization mode was Auto, which quantizes Pitch Bend when you play chords but has no effect when you play monophonically.

I really love the Axon's advanced split and quantization capabilities. Unfortunately, you can't set the Axon to quantize only note attacks, which would have allowed bends to occur after a specified attack time. That feature would have been great for catching errant fingering and still allowing natural bends. Nevertheless, quantization was especially useful with instruments for which string bends sound unnatural.

Pedal to the Metal

The AX 100 mkII delivers far more than just MIDI conversion and splits. By attaching a momentary switch (which is included), you can trigger the Axon's arpeggiator, step sequencer, and various Hold modes. For example, when you press the switch, play a chord, and release the switch, the Axon will hold that chord on its own MIDI channel so you can accompany yourself on other channels. You can also use the switch to activate a sequencer pattern or to layer two presets.

The unit's full-featured arpeggiator responds to whatever notes you play for as long as you press the Hold switch. You can select from 7 arpeggiator modes (Assign, Reverse, Recycle, Up, Down, Up/Down, and Random) and 32 rhythms. Half the rhythms are factory presets, and half are user definable. You can also select each arpeggio's number of notes, number of repeats, octave limits, and tempo—either fixed or synchronized to MIDI.

The Axon's sequencer can store a maximum 32 patterns and 8 tracks. As soon as you trigger a sequence with the Hold switch, the Axon plays a preprogrammed pattern or an entire track's worth of patterns on a separate MIDI channel. Trying to program the sequencer from the front panel alone would be extremely tedious, but using the editing software makes it fairly straightforward.

Virtual Fretboard

The Axon divides its functions, parameters, and settings into separate modes, menus, and submenus, all of which you can reach using the front-panel buttons. The unit does its best to make every mode and submenu as accessible as possible, but editing such a deep device using 12 buttons and a rather small LCD screen is never ideal.

Fortunately, Terratec has developed AX 100 Editor, an application that ships with the device and runs under Windows XP and Mac OS X on PowerPC and Intel Macs (see Fig. 3). It furnishes tabs for each of the Axon's modes, and each tab places all of that mode's submenus onscreen at the same time. The editor also features very handy graphical user interface elements such as virtual fretboards, graphs, and pull-down menus.

After using the Axon for several months, I couldn't imagine using it without the editing software. Its intuitive layout makes editing the

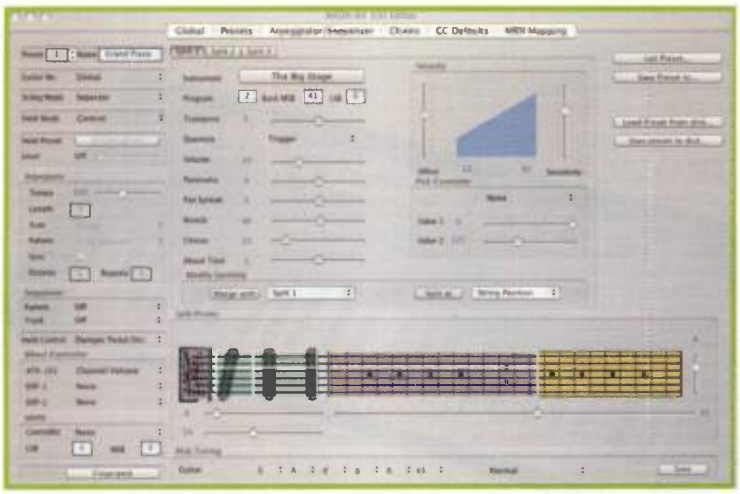

FIG. 3: AX 100 Editor gives you complete access to all the hardware's parameters and provides a GUI that's much more intuitive than the unit's front panel.

unit a breeze. I only wish that the sensitivity control on the editor offered metering like the LCD does.

Get Your Ax On

The Axon AX 100 mkII is a very compelling product. Its guitar-to-MIDI conversion is as fast and accurate as anything available. Whenever I use it with my custom Koll Tornado guitar equipped with Graph Tech's Ghost Modular Pickup System, the AX 100 mkII accurately converts my performances into MIDI data (see **Web Clip 1**). Its amazing breadth of features makes it ideal for live performance.

Along with the Axon's power comes complexity, but the excellent editing software helps to minimize the learning curve. Some guitar-to-MIDI converters offer simpler operation but fewer features, and others offer better built-in sounds but less flexibility. If you want a guitar-to-MIDI converter with impressive performance and a ton of features, the AX 100 mkII is the one for you.

EM WEB CLIPS

Orren Merton is the author of Guitar Rig 2 Power! *(2006) and other books for Thomson Course Technology.*

FIG. 1: The Roland SH-201 is an analog-modeling synth with 49 full-size, Velocity-sensing keys and enough knobs and switches to keep even the most hyperactive synthesist busy.

ROLAND SH-201

Analog-modeling synthesis and audio I/O in one.

By Jim Aikin

If you're shopping for a great-sounding, versatile, yet inexpensive analog-style synthesizer, put the SH-201 near the top of your "gotta check it out" list. This little monster would be ideal for aspiring keyboard players, performing DJs who need both keys and an external filter for processing mixes, and home-studio owners in search of a 1-box solution for computer audio I/O and MIDI master-keyboard chores.

The SH-201 is an amazing value and a wonderful analog-modeling keyboard.

The Web pages I checked, for both Roland and online retailers, neglect to mention how many notes of polyphony the SH-201 has. Given the modest price, I was wondering if it might be monophonic. But when I got it out of the box and plugged it in, I found that it's 10-note polyphonic—one note for each finger. Though not multitimbral, it can play two patches at once in Split or Dual mode.

The Physical Package

Don't be fooled by appearances: at first glance, the SH-201 looks and feels almost like a toy. It's extremely light, to the point where I'd worry about it tipping off of an unstable keyboard stand at a gig. The front panel is labeled in big, high-contrast letters, and although there are plenty of LEDs to show the current settings of switches, there's no LCD at all (see **Fig. 1**).

The 4-octave keyboard feels light and snappy. It senses Velocity but not Pressure. You can transpose its range up or down by up to three octaves using Octave switches. For left-hand control, Roland's standard pitch/mod paddle is augmented by its D-Beam sensor. The knobs are big and easy to grab, and they feel solid, not cheesy. Ditto for the sliders.

Given the absence of an LCD, Roland was obviously going for a 1-knob, 1-function panel layout, in which nothing is hidden. To a great extent, the company succeeded. The oscillators and LFOs (two of each) are selected using buttons that appear on graphical "file tabs," allowing one set of controls to do double duty in each section. This arrangement is instantly understandable.

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

Jenna Drey – TAXI Member – www.jennadrey.com

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

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

Who Hears Your Music?

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

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

A Record Deal With Lots of Zeros!

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

Madonna, Bowie, Jagger, and me!

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

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Some less commonly used functions are hidden, however. You'll need to read the manual to discover, for instance, how to adjust the Velocity sensing of a patch or set the keyboard to MIDI Local Off mode. These functions and a number of others are accessed from the front panel via button combinations.

Around the back, you'll find a strain-reliefed input for the lump-in-the-line power cord, stereo analog audio inputs (RCA phono, unfortunately), and a multipurpose pedal input (see **Web Clip 1**). You can attach either a footswitch or a sweep pedal. USB, MIDI, and a headphone output complete the connector complement.

Voicing

There's no denying that the SH-201's voice architecture lacks some of the refinements of instruments costing three times as much. But it has all of the necessities and includes a surprising number of extras. The factory presets cover all of the usual analog-modeling bases, from cutting bass to ethereal pads by way of chimes and pulsing arpeggios. Overall, the SH-201 sounds rich and lively, with lots of presence. It can cut through on top of a dense drum track or fill in the spaces in a mix with subtle warmth.

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The synth engine's resonant multimode filter may not be quite as fat sounding as the filters on high-end synths, but it's very good indeed. Depending on the patch, some stair-stepping will be audible when the cutoff knob is turned, as its data output is not being smoothed.

The LFOs provide a choice of seven waveshapes. Each LFO has two selectable destinations, not including panning, unfortunately. The SH-201 has stereo

effects, but no dynamic panning. In addition to the programmed modulation routings, LFO 2 is always available for vibrato from the modulation paddle. When an LFO is synced to internal or external clock, its rate knob changes the rhythmic subdivision, but of course there's no data display to show what the setting is, only a blinking LED. So unless you have the editor software hooked up, you'll have to use your eyes and ears.

Additional LFO parameters are accessible only from the editor program. Here, you can set a fade-in time for each LFO or click on the Key Trigger button to set it to polyphonic operation. Other advanced parameters, such as LFO rate modulation, are not provided, however.

The effects section of the SH-201 is not fancy, but it's functional. The reverb and delay line each have only two controls—time and depth—from the front panel, and the overdrive has only an on/off button. The editor software enhances the picture considerably: the overdrive has an amount knob in software, and the delay line has seven parameters and can be used for stereo chorusing. The reverb has a generous 12 parameters, including predelay, density, diffusion, and so on. In Split and Dual modes, the upper and lower voices share the same effects.

Given the depth of the voicing possibilities, it comes as a surprise that the SH-201 can store only 32 factory and 32 user presets. You can create and save an unlimited number using the editor and librarian, however, so the shortage of onboard memory may not be an issue for many users.

The Computer Connection

If you're just getting started in the world of computer-based recording, you may be delighted to learn that the SH-201, a USB cable, and a basic digital audio and MIDI recording and editing package are all you need to produce music (other than monitors or headphones, of course). The SH-201 comes with a CD-ROM that contains ASIO audio drivers for Windows, and it's also

PRODUCT SUMMARY

ROLAND SH-201

analog-modeling synthesizer
$599

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

RATING PRODUCTS FROM 1 TO 5

PROS: Sounds terrific. Provides computer audio I/O. Real-time processing of external audio. Friendly panel layout. Cool arpeggiator.

CONS: Onboard patch memory is skimpy. Some parameters can be edited only with the included software.

MANUFACTURER
Roland
www.rolandus.com

compatible with Intel Mac audio.

There are at least three scenarios in which you may want to use the SH-201 with a computer. First, it can be your audio interface: analog audio arriving at the SH-201's external inputs can be recorded into an audio track. The SH-201 operates strictly at 44.1 kHz but can sample at 16-bit or 24-bit. You can also record the sounds generated by the synth itself onto an audio track for subsequent mixing.

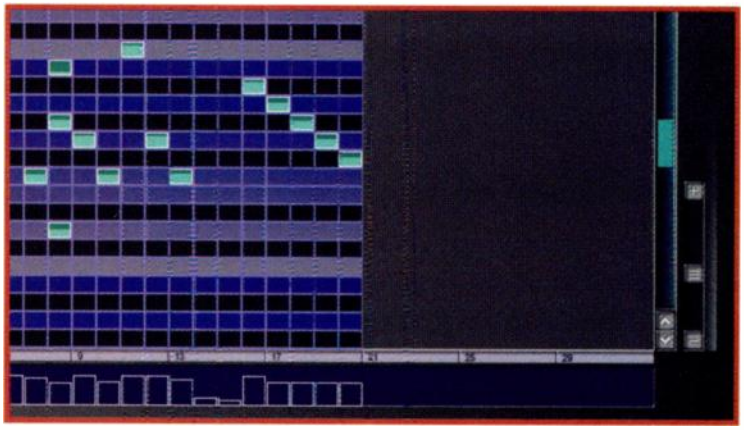

FIG. 3: Notes can be entered and edited graphically in the arpeggiator page in the SH-201's editor program.

Second, the computer's audio output can be processed through the SH-201's external audio filter, which has cutoff and resonance knobs and a couple of mode buttons. The output of this filter is then passed on directly to the SH-201's analog audio outputs. This hookup allows you to run a laptop at a gig, for instance, and manipulate the sound of the laptop from filter knobs in real time. As an alternative, the computer's audio can be routed through the voice filter and effects.

The same options are available for processing external analog audio in real time. There's even a center-cancel button for impromptu karaoke sessions.

Third, if you're using the SH-201 as a tone module for MIDI sequencing, you can load its editor program as a pseudo VSTi plug-in. This lets you edit a sound as part of your song file and have the sound be reloaded (including any edits) the next time you load the song. It's not a complete VSTi implementation, because the editor's knobs can't be automated as VSTi controls. But knob moves can be recorded as MIDI data from the front panel, so that isn't a big issue.

The method for getting the editor to show up in a Windows sequencer is buried in the Quick Start information on the CD and is hard to find. Using Steinberg Cubase, I had to copy the editor's DLL file manually into the Cubase VST Plugins folder, because the editor's installer didn't take care of this. (Adding the default folder to Cubase's list of places to look for plug-ins also works.)

FIG. 2: The included software editor for the SH-201 provides access to a number of parameters not available from the instrument's front panel. Its amenities include a MIDI Message data field (upper right), which shows the System Exclusive or Control Change message being sent for each edit. These messages can be copied and pasted into your sequencer.

Editor and Librarian

The editor software has a clean, easily understood design (see Fig. 2). It also allows you to edit numerous parameters that are not accessible from the hardware's front panel or that are accessible only via button combinations and possibly a quick trip to the manual.

The added LFO and effects parameters were mentioned earlier. In addition, you can name your patches, set the keyboard-split point, set pitch-bend depth, balance the levels of the lower and upper tones in a dual or split preset, pan the tones separately, reassign the mod paddle so that it controls LFO depth for some destination other than pitch, and more. The biggest benefit of using the software is that it provides graphical editing of arpeggio patterns.

The librarian is not integrated with the editor. Both can be run at once, but after making changes in one, you'll need to click on a button or two to get them back in sync with one another and with the hardware unit. This is annoying, but again, not a big deal. Don't look for advanced database functions in the librarian. You can add comments to patches and copy them from one bank to another, but not much else.

Arpeggiator and Recorder

From the SH-201's front panel, you can turn the arpeggiator on or off, change the tempo, and select any of 32 preset factory patterns. After launching the editor,

you can freely edit the patterns in a graphical display (see Fig. 3). Patterns are polyphonic and can be up to 32 steps long. Velocity and duration can be edited for each note.

What's especially cool is that if you select a normal arpeggio mode, such as up/down, the pattern will be interpreted by choosing the nearest note from the phrase. Though this feature is difficult to describe in words, it lets you create pulsing chord patterns and arpeggios that have a few chords thrown in (see **Web Clip 2**).

The phrase recorder has no interface in the editor. It can record up to eight measures of notes and/or knob gestures and then loop them for playback. Eight phrase memories are provided; these are global to the SH-201, not stored per patch. A metronome beeps quite loudly during recording, so you won't want to record phrases on the fly during a performance. But storing a few knob-only phrases ahead of time and recalling them at a gig would be a great way to produce complex sonic changes while you play, as if by magic.

The manual doesn't explain how to erase a previously recorded phrase, but it turns out there are two ways to do it: you can hold down the Cancel button while in Record mode and keep it held down for the entire length of the phrase, or you can change the phrase length before starting to record. Entering Record mode without changing the length gives you overdub recording. Unfortunately, cool phrases can't be archived externally using the librarian software.

As a synthesizer, the SH-201 is not revolutionary in its design. It sounds terrific, but so do a lot of other keyboards. (To be fair, most of them are more expensive.) What sets the SH-201 apart are its easy-to-play front panel, the near-seamless integration with a computer, and the ability to run external audio through its filters in live performance.

Given the depth of its voice programming, the fact that only 32 user memory slots are available is a sticking point. But realistically, 32 may be enough for a live set, and the included computer editor and librarian will let you design and store sound banks until the cows come home. Don't be fooled by the price, the lightweight chassis, or the nonsexy graphics on the panel. The SH-201 is an amazing value and a wonderful analog-modeling keyboard.

When he's not writing about music technology in his PC-based home studio, Jim Aikin plays cello, writes interactive fiction, and wishes there were more hours in a day and more years in a lifetime.

GET IN THE ZONE

AND STAY THERE

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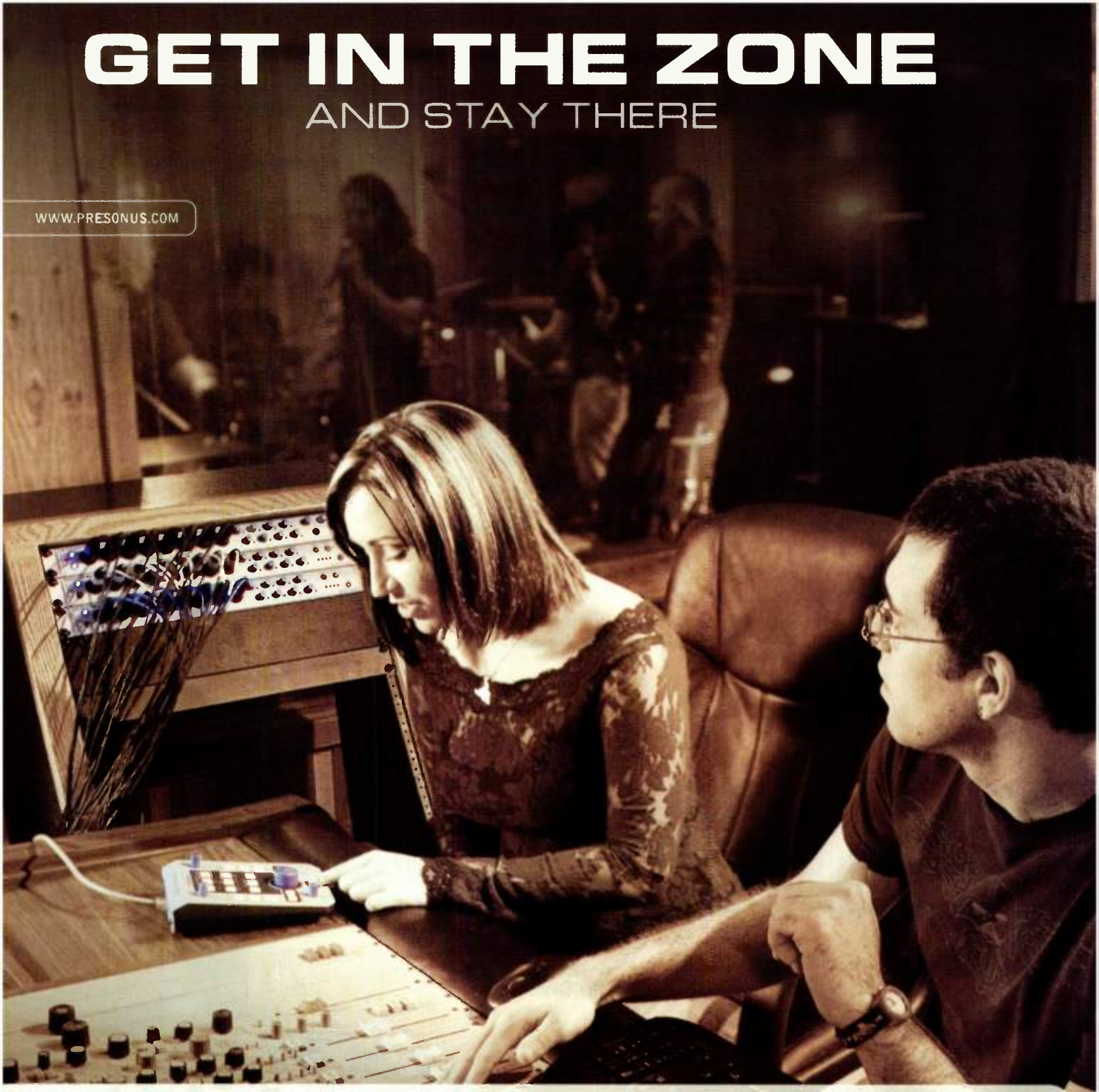

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FIG. 1: The Boss RC-50 is a real-time audio looper housed in a floor processor with seven footswitches, plenty of hands-on controls, and 24 minutes of stereo recording time.

BOSS RC-50 Loop Station

A new Boss box throws us for a loop.

By Larry the O

Numerous musical genres employ looping, and numerous hardware and software tools supply looping functionality. One of the latest hardware devices for looping is the Boss RC-50 Loop Station. It improves on the RC-20XL by adding impressive features such as three separate mono or stereo loops (called *phrases*) that can play sequentially or simultaneously. It also offers tap tempo and time-stretching functions, various phrase-playback options, USB connectivity, and 24-bit, 44.1 kHz audio sampling.

Simply Hardware

The RC-50 is a floor unit dominated by seven footswitches (see Fig. 1). Three switches select which phrase is active, and the other four control record, play, stop, undo, redo, and tap tempo functions. Above the footswitches is the control panel, and on its left side are a 2-character patch-number display and a 2 × 16–character LCD, which provides visual access to the RC-50's menu structure. Just to the right of the displays are manual controls that include a data wheel, menu-navigation arrows, and Name, Write, and Exit buttons. In the Play Mode section are two important buttons for selecting Single or Multi mode and Loop Sync (more about those later).

The control panel's right half hosts a row of level pots for controlling phrases, the Rhythm Guide track, inputs, and the master output. Below each phrase-level pot is a button and LEDs that indicate when reverse or a one-shot operation is active for the phrase. Below the input-level pots are buttons for Auto Record, Input Mode, and Overdub Mode. LED indicators show the status of the Loop Quantize feature and the selected overdub mode.

The rear panel hosts a pair of unbalanced ¼-inch inputs for instruments, an XLR mic input (with a

The RC-50 sits squarely in the middle of the price continuum.

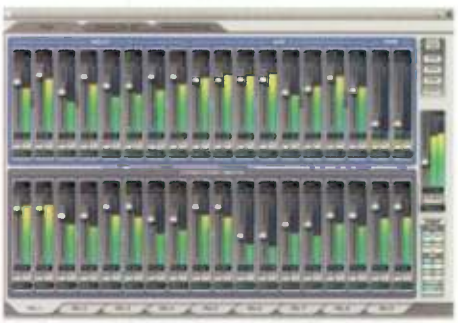
FIRESTUDIO™

26x26 FireWire recording system with DSP mixer/router

Optional MSR
(Monitor Station Remote)

Talkback, dim, mute & mono
3 modes: track, surround, mix
Speaker/input source switching
Dual headphone amps

Control Console
DSP mixer/router
(Included)



PreSonus ProPak Complete Software Suite
(Included)

Cubase LE
Reason Adapted
Amplitube LE
BFD Lite
Drumagog LE

Plus over 2GB of plug-ins, virtual instruments, drum loops and samples!

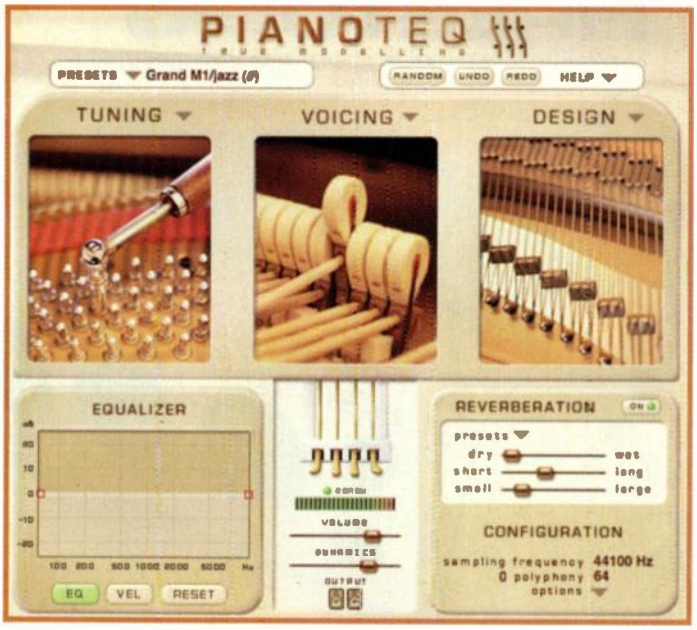


FIG. 1: Pianoteq's control panel is laid out in three horizontal sections: preset management and help (top), physical-modeling panels (middle), and pedals and playback controls (bottom).

MODARTT Pianoteq 1.0.3 (Mac/Win)

The first physical-modeled virtual piano is outstanding.

By Len Sasso

Modartt Pianoteq 1.0.3 marks a milestone in physical-modeling technology. From the first note, there's no doubt that this is a very playable piano with all the special controls that only physical modeling can bring. You can create a 30-foot grand with your own unison tuning, octave stretching, and string-harmonics spectrum; you can shape the physical characteristics of the soundboard and control individual as well as global resonances; and you can create hybrid instruments that sound organic if not quite acoustic.

Modartt, short for Models and Data for Arts and Technology, is a collaboration between piano technician turned mathematician Philippe Guillaume and software designer Julien Pommier. Guillaume created the model, and Pommier put it on your desktop as a VST and AU plug-in for the Mac and a VST plug-in for Windows. A standalone version is planned for the future.

Physical modeling is not new; physical-modeled virtual string and wind instruments get better with every generation. But the mechanical complexity of a grand piano presents a different level of challenge. Pianoteq is more than a first shot at this challenge, and it's likely to be around for a long time.

For Ex Sample

The biggest question regarding Pianoteq is, how does it compare with state-of-the-art sampled-piano virtual instruments? "Software Eighty-Eights" in the October 2006 issue of EM (available online at www.emusician.com) compares six of the best, with the conclusion that there are some very playable sampled pianos. I used the same MIDI files used for the Web Clips from that article to play

Pianoteq can hold its own against top-of-the-line sampled pianos.

several Pianoteq presets. You'll find them in **Web Clip 1** for this review, and they make an interesting comparison with the sampled-piano clips.

A couple of things stand out right away when comparing Pianoteq with a sampled piano. No matter how closely miked, there is always some room ambience in piano samples, and when it doesn't get in the way, it adds to the realism of the virtual instrument. With Pianoteq, no recorded samples means no room ambience. Pianoteq includes a reverb to fill the void, but the result is a little less alive—a problem that might be mitigated with a high-end convolution reverb.

The second issue is that Pianoteq doesn't reproduce the sound of a particular name-brand piano. You may get close to the sound of a Steinway Concert D or a Bösendorfer 290, but it's not going to be a dead ringer. Modartt has modeled four historical pianos as part of the Keyboard Instrument Virtual Restoration project (KIViR) and provided those as Pianoteq presets. You can expect more in the future, and some current models may be released as well.

Make Mine Pineapple Mint

What you do get with a physical-modeled instrument is the ability to customize every aspect of its sound. You also get finer resolution of some performance parameters. For instance, you're not limited to a small number of Velocity zones, as you are with even multigigabyte sample libraries. With Pianoteq, each of the 127 MIDI Velocity gradations has its own effect.

Half pedaling is another example. If your sustain pedal sends out more than two values (127 for on and 0 for off), Pianoteq will reproduce the effect of partially pressing the damper pedal on a real piano. Because progressive sustain pedals are hard to come by, and standard foot controllers don't feel right for sustain, it would be nice if a second controller could be used to set the damper amount when pressing the sustain pedal.

Pianoteq's Harmonic pedal is yet another example of the advantage of physical modeling. This fourth pedal, which couldn't be implemented in the real world, lets you play staccato notes with damper-pedal-down resonance. It's like having the dampers raised for all notes except the ones played, an unnatural but altogether pleasing effect.

Most of Pianoteq's controls are devoted to tweaking the physical model. Those are separated into Tuning, Voicing, and Design controls and are located behind panels in the center of the graphical user interface (see **Fig. 1**). Clicking on a category name uncovers the sliders and pop-up menus for its settings (see **Fig. 2**). The Presets and Help menus, along with Random, Undo, and Redo buttons, inhabit the top of the panel. The EQ, reverb, pedals, and output controls are located at the bottom. Hovering the mouse over any control reveals a help pop-up that you can also use to make MIDI remote assignments. MIDI remote is easy to set up and almost essential for tweaking this instrument.

The Tuning, Voicing, and Design settings have the greatest impact on Pianoteq's sound, and they give you a lot of latitude to create off-the-wall as well as realistic piano presets. The Random button, which affects only the settings in these sections, provides a great way to survey the kinds of sounds Pianoteq can deliver. Because there are no filters, envelopes, LFOs, and other synthy components, you always get a percussive, keyboardlike sound, but its timbre and note duration can vary widely (see **Web Clip 2**).

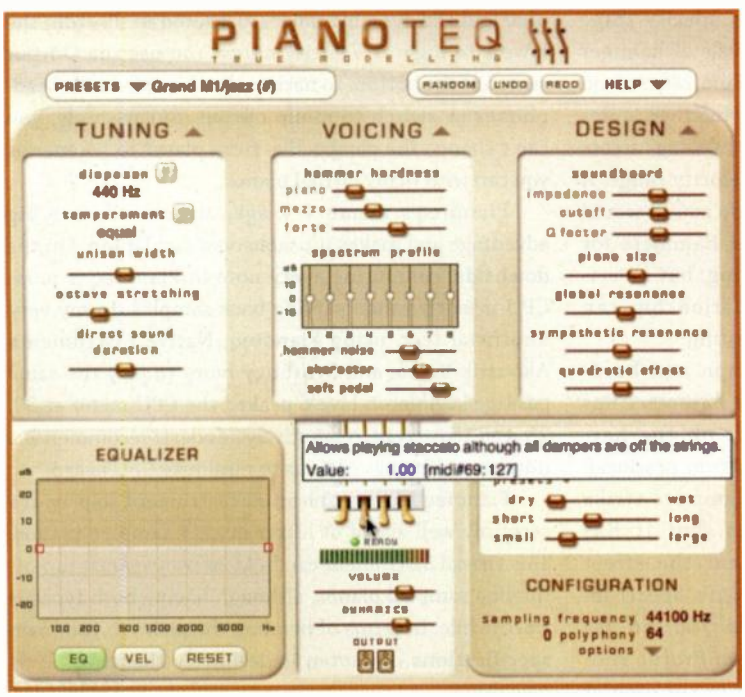


FIG. 2: Pianoteq's three physical-modeling panels open to reveal tuning, voicing, and piano design settings. Handy rollover help boxes describe the controls and enable MIDI remote assignments.

A Different Tune

The controls in the left panel set the tuning reference pitch (Diapason), temperament, unison detuning for multistring notes, octave stretching to compensate for string inharmonicity, and duration of the direct versus the decay portion of the sound. The unison and hammer settings influence the duration of the direct sound, but you can use the Direct Sound Duration control to compensate for that manually.

You get 15 choices of reference pitch, ranging from a semitone below to a semitone above the standard 440 Hz tuning for the A above middle C. Five temperaments are supported: Equal (standard), Zarlino (based on harmonic thirds and fifths), Pythagore (based on harmonic fifths), and two common Baroque tunings—Mesotonic and Well-tempered.

The Unison Width control sets the amount of detuning between the lowest and highest string for each note. The lowest bass notes have only one string, so the control has no effect. It has the greatest effect on the notes above the break, which have three strings. Some unison detuning is desirable, and settings to the left of center tend to make the piano a little lifeless. Settings far to the right produce honky-tonk out-of-tune pianos.

Tuners use octave stretching to compensate for string inharmonicity—short strings, especially, have harmonics that are slightly sharp. You can set octave stretching independently of piano size, but for realistic pianos, less stretching is needed for longer pianos. (Piano size is set in the Design section.)

Many Voices

You use the Voicing section to specify the hardness of the hammers and the amount of hammer noise, the harmonic spectrum of the strings and how much it varies from note to note, and the effect of the una corda (soft) pedal. Each of these adjustments has a profound effect on the piano's sound, but the Spectrum Profile controls are the most unusual.

You can specify three different levels of hammer hardness: Piano, Mezzo, and Forte. Pianoteq then scales the hardness settings across the MIDI Velocity range. A natural piano sound would have harder hammers for louder playing, but inverting the relationship can prove interesting.

The shape and hardness of the hammers has some effect on the harmonic spectrum produced when the hammers strike the strings, but it has nowhere near the effect that Pianoteq's Spectrum Profile does. You can use the Spectrum Profile sliders to boost or cut each of the first eight harmonics by as much as 10 dB. Boosting the first harmonic (fundamental) and cutting the seventh makes for a much less dissonant sound. Cutting the even and boosting the odd harmonics gets you closer to a hollow, square-wave sound.

By Design

The controls in the Design section affect the physical design of the piano. The Soundboard settings affect the length of a note's natural decay (Impedance), the intensity of the higher harmonics (Cutoff), and the decay of the higher harmonics (Q Factor).

The main impact of Pianoteq's Piano Size control is on the inharmonicity of the strings. As mentioned earlier, shorter strings have slightly sharp harmonics. That manifests itself when playing notes an octave apart, especially in the higher ranges, because the higher note will be out of tune with the second harmonic of the lower note. Octave stretching is used to compensate, but you can minimize the need for octave stretching by lengthening the piano.

The Global Resonance slider controls the resonance of the piano harp, soundboard, and cabinet. The Sympathetic Resonance slider affects the resonant vibrations of open (undamped) strings when other notes are played. The Quadratic Effect slider affects the level of frequencies at twice the normal harmonic frequencies that are produced for loudly played notes.

Out and About

In addition to the EQ, reverb, and pedal controls, the output section at the bottom has sliders for output volume and the dynamic range, measured in dB from the lowest to highest Velocity notes. You use the Output switch at the bottom to narrow the stereo field for headphones or switch to mono output. Surprisingly, you can't change the perspective from player to listener as you can with many virtual pianos.

Pianoteq's memory usage, almost nil, is a big advantage and makes it usable on a fast laptop. On the downside, computing every note in Pianoteq is more CPU intensive than playing back samples. In my very unofficial test, using Pianoteq, Native Instruments Akoustik Piano, and Synthogy Ivory to play the same passage in Ableton Live 6 peaked the CPU meter at 30, 23, and 15 percent, respectively. Modartt recommends a number of settings options to minimize CPU usage.

Pianoteq is a very pleasant instrument to play. It's certainly well ahead of lesser-quality sampled pianos. This virtual instrument can hold its own against top-of-the-line sampled pianos, although having both types is worthwhile. In terms of bending the piano to your own specifications, Pianoteq is definitely the new thing.

EM WEB CLIPS

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

PRODUCT SUMMARY

MODARTT Pianoteq 1.0.3

virtual piano
approximately $322

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

RATING PRODUCTS FROM 1 TO 5

PROS: The only physical-modeled virtual piano. Excellent sound. Clearly organized, minute control of all aspects of piano design. Well documented, including helpful tutorials.

CONS: EQ and reverb could be higher quality. Controls are small and sometimes difficult to hit with the mouse.

MANUFACTURER
Modartt
www.pianoteq.com

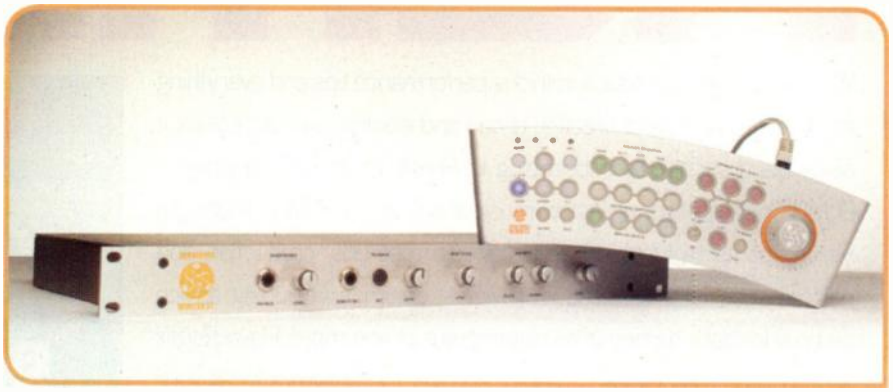


FIG. 1: The Monitor ST gives you remote-controlled input and speaker switching, talkback and cue functions, and more.

DANGEROUS MUSIC Monitor ST

Monitoring solutions for the discerning ear.

By Eli Crews

For years, people in the recording industry have been predicting the death of the console. With mixing taking place inside computers and summing boxes, and moderately priced outboard preamps and EQs in abundant supply, the large-format console has been diminishing in popularity in small-to-midsize studios for some time.

What has really driven the nail into the mixing desk's coffin, though, is the advent of the monitor-management system, which offers functionality formerly found only in a console's center section. Dangerous Music, already a leading force in high-end summing boxes, has a new product in this category: the Monitor ST Stereo Monitor Control System (see **Fig. 1**). Composed of a main unit and a wired remote control, it offers functions such as input and speaker switching, cue and talkback, and headphone amplification.

Outside the Box

The Monitor ST unit itself is a 1U-rackmount, brushed-aluminum box with the distinctive Dangerous logo screened in soft orange on the far left. On its face are ¼-inch jacks for an external talkback mic and headphones, and rubber-lined knobs for headphone, talkback mic, Main-to-Cue, Aux-to-Main, Aux-to-Cue, and Input 4 levels. (The Input 4 control lets you adjust the level of a CD player or other source that's plugged through Input 4, facilitating accurate A/B comparisons.) Also on the front panel is the internal electret talkback mic.

On the back of the unit (see **Fig. 2**), you'll find two 25-pin D-sub connectors. One is for the four main stereo inputs and the other for the four stereo outputs, which are meant to feed power amps or powered speakers. The connectors are wired according to the Tascam standard,

There's a button labeled PPI, which stands for Producer Pacification Indicator.

so if you happen to have cabling around for a DA-38 or DA-88, you might be good to go for your Monitor ST I/O. If not, expect to have to make or buy such a harness before you can get audio flowing through the Monitor ST. (Dangerous recommends Redco cables [www.redco.com] for this purpose.)

The rear panel is also home to a pair of female XLR jacks for the stereo Aux input. The other jacks on the back are a mono XLR Slate output, a stereo unbalanced XLR for the powered-cue output, an XLR jack for an additional talkback remote switch, an RJ-45 jack for the included remote cable, and a DC-in jack for the hefty lump-in-the-line power supply.

Remotely Speaking

The remote, which is included with the Monitor ST, is a sleek crescent shape about 12 inches in length. Other than the various level controls on the face of the rack-mounted unit, all business takes place at the remote, which crams an incredible amount of functionality into a compact package.

Starting at the right, the volcano-shaped volume control, an absolute position encoder, has 32 detented positions by which you can achieve extremely accurate volume consistency, down to a 0.02 dB error margin. This is accomplished using the Computer Controlled Analog Stepped-Attenuator Gain System, which, according to the Dangerous literature, is far superior to the VCA (voltage-controlled amplifier) or DCA (digitally controlled amplifier) circuits widely used in similar remote volume controls. A minor trade-off is that if you're standing near the rack unit, you can hear the mechanical noise of the relays switching when you turn the volume knob. However, this noise is in the room only and has no effect on the actual audio that's passing through the unit.

FIG. 2: In order to fit all of the I/O capabilities onto the back panel of the Monitor ST, D-sub connectors were used for the main inputs and outputs.

In addition to the exquisite volume control, the remote offers switching between the four stereo line inputs (five including the Aux input, which can be blended with any of the other four inputs) and the four stereo line outputs, speaker mutes, speaker solos, a dim

function, mono summing, and a talkback button. The buttons are color coded by functionality and glow a gorgeous soft green, orange, red, or blue when selected.

The Setup button allows you to change certain parameters, such as level offsets for inputs and speaker feeds. You can designate whether or not the subwoofer output (Output 4) is included when you select Main, Alt 1, or Alt 2, and whether or not the subwoofer gets a 57 Hz lowpass filter on any output set.

Also included is a programmable gain boost to bring any –10 dBV input up to +4 dBu, and the Dim switch and the Aux input can be set (independently) to engage automatically when you press the Talkback button (more about this later).

Lastly, there's a button labeled PPI, which stands for Producer Pacification Indicator. Think of this as a visual placebo, which can trick a producer's ears into hearing any change they want you to make.

Main Event

Dangerous Music has built its stellar reputation by using the highest-quality components in uncompromising designs, so I was very excited to "hear" my first piece of Dangerous gear. The quotation marks are there because the whole philosophy behind the Monitor ST is that you don't hear the unit itself; it is designed to pass your audio through its circuitry completely unscathed.

I A/B'd it, as much as that was possible, against my Amek Einstein console's center section, as well as against a widely available and less expensive monitor-management unit. Because doing A/B comparisons required plugging and unplugging speakers, it was only over longer periods of listening that I could make real qualitative assessments. I did conclude that the Monitor ST gave me a much cleaner, deeper picture of my reference mixes than the less expensive unit. When I compared the sound of the Monitor ST with my console's center section, the differences were subtler. That is to say, they both sounded great.

On a functional level, the Monitor ST trumps most high-end consoles' monitoring features, with the amount of control over input and output levels, the subwoofer-management controls, and the absolute position encoder. Even people who won't be ditching their consoles anytime soon, like myself, can benefit from the Monitor ST.

PRODUCT SUMMARY

DANGEROUS MUSIC Monitor ST

monitor control system

$1,899

FEATURES	4
EASE OF USE	4
AUDIO QUALITY	5
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Superb audio quality throughout all signal paths. Many "extra mile" features. Expandable to 5.1 capabilities.

CONS: Plastic RJ-45 jack on remote cable prone to breakage. Headphone jack on front will only mirror Cue output on back. "Momentoggle" functionality of buttons not defeatable. Clicking sound audible from rack unit during volume changes.

MANUFACTURER
Dangerous Music
www.dangerousmusic.com

Yakety Yak

The talkback features are extremely flexible. The talkback mic on the rackmount unit is a highly sensitive omnidirectional electret, which picks up your voice even when you aren't facing it. It picked up regular speaking voices from all the way across my 25-foot-long control room. This is mostly a good thing, but it did cause me to have to ask band members hanging out in the control room to be quiet a little more often than usual. If you have the unit mounted out of range of the built-in mic (like in a closet), there is a high-impedance external mic input, which you can drive with a common dynamic mic and inexpensive impedance-matching transformer. Plugging into this jack disables the front-panel mic.

The Talkback button itself is one of the many "Momentoggle" buttons on the remote. This term describes the following behavior: when the button is "pressed," or pushed for less than a half second, the switch is latching, meaning it stays on when you let go; if the button is "held," or pushed for more than a second, it functions as a momentary switch, and deselects itself when it is released. It is actually quite clever of the Dangerous designers to implement such a feature, and it surely took some engineering feats. The ability to press the Monitor ST's Talkback button once and be able to talk as long as you want without having to hold it down is greatly appreciated.

It does take some getting used to, though, and I would love the ability to disable the Talkback button's Momentoggle capabilities when needed. I often found that I had inadvertently left the button latched on, and band members were frequently saying less-than-kind things about their fellow musicians in the next room.

Next on the list of cool features is the talkback remote jack, for which I wired up a simple switch at the end of a mic cable, as specified in the manual. This is a welcome addition for most sessions, because a producer or bandleader can communicate with the sequestered musician without having to ask the engineer to repeatedly press the button. One last great feature is the addition of the Slate output, which lets you either send the talkback signal to a loudspeaker in the live room or print it directly to disk or tape.

Cue It Up

The Cue output is an amplified, stereo signal meant to feed a passive attenuator box. I was a little worried about sending this unbalanced signal to my headphone amps, into which I had previously been feeding a balanced,

line-level signal over 60 feet or so of cabling. My fears were unfounded; once I got the gain staging right with the Cue level control, my headphone amps were happy to receive (and reamplify) this signal, with nary a hum or buzz.

The Cue output can be fed whatever is going to the control room monitors or whatever is hooked up to the Aux input, or a blend of both. My only small beef with the Cue system is that the headphone jack on the front of the unit mirrors whatever is being sent out the Cue output on the back of the unit. It would be great to be able to monitor what is going to the speakers on headphones without also sending that out to the talent.

ST for Me

I have thoroughly enjoyed the addition of the Monitor ST to my studio. The precision of its volume control; the clean, clear audio path; and the extensive input, speaker, and cue feature sets have all enhanced my daily studio experience.

There were a couple of features that I wished were on this unit, such as digital inputs and a Dangerous-caliber D/A converter. It turns out that the company was thinking along similar lines, because both of those features will be part of a 1U expansion unit for the Monitor ST that's due out later this year (another expansion unit, the Monitor SR, which adds 5.1 surround monitoring, is already available).

In a market flooded with inexpensive solutions to console-free monitoring problems, the Dangerous Music Monitor ST stands out as a truly professional alternative. You'll pay a few extra bucks for it, but quality like this is, in the long run, well worth the price. Especially given the Monitor ST's expandability to 5.1 (and beyond), it is a true investment in every sense of the word, which may just be the best praise any piece of gear can garner.

Eli Crews often records, mixes, and masters (lowercase) dangerous music at New, Improved Recording, his studio in Oakland, California (www.newimprovedrecording.com).

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The rear panel hosts a pair of unbalanced ¼-inch inputs for instruments, an XLR mic input (with a

The RC-50 sits squarely in the middle of the price continuum.

phantom-power switch), and a stereo ⅛-inch minijack input labeled Aux for receiving a signal from an iPod or a CD player (see **Fig. 2**). The RC-50 also has a Center Cancel function available for the Aux jack (intended to remove a prerecorded song's lead vocal) and a Flat Amp Simulate function for the Aux and mic inputs. Flat Amp Simulate provides high- and low-frequency EQ boost for occasions when the RC-50 is playing through a guitar amp instead of a full-range sound system.

There are also two pairs of unbalanced ¼-inch outputs (main and sub—each input or phrase can be directed to either) and a ¼-inch stereo headphone output. Two ¼-inch TRS phone jacks accept as many as four foot controllers (one jack accepts two footswitches, and the other takes either an expression pedal and a footswitch or two footswitches). In addition, you get a USB port and MIDI In and Out ports.

The RC-50 is powered by an uncomfortably large wall-wart power supply. For some products, Roland has used the superior lump-in-the-line approach, which would have been welcome in the case of a transformer as sizable as the RC-50's.

Dancing in Circles

The RC-50 can hold a maximum 24 minutes of 24-bit, 44.1 kHz stereo audio, or a maximum 49 minutes of mono. An RC-50 patch contains all configuration settings, as well as audio content, for the three phrases. Up to 99 patches can be stored. Whether you consider the available recording time to be plentiful, merely adequate, or not enough will depend on how many patches you want to store and how long the phrases are.

You record a loop by stepping on a Phrase Select button to choose the loop location and then stepping on the Rec/Play/Overdub switch. Stepping on the switch a second time exits the operation as the unit continues to play. Hitting it one more time with the same phrase selected drops the unit into Overdub mode, in which you can either layer additional material on top of the existing phrase or replace it. To stop playback of a phrase, you must step on the Phrase Select switch and then the Stop switch.

This system results in a bit of a dance, with your foot darting between the Rec/Play/Overdub switch on the far left and the Phrase Select switches on the right. I would have preferred a system in which one control lets you choose between Record/Overdub and Play modes, and the Phrase Select switch engages the chosen function. Still, I got used to the RC-50's system.

Similarly, to stop playback of all phrases, you have to step on the Phrase 2 and Phrase 3 Select switches simultaneously. Again, it worked out most of the time, though I occasionally did it wrong in the heat of the moment. A work-around would be to plug in an external footswitch and assign it to All Start All Stop, but you should be able to assign a footswitch to All Start All Stop from the RC-50 itself; you already have enough footswitches to deal with.

Recording loops live is not the only available option for getting sound in and out of the RC-50. Its USB port lets you import or export loops as WAV files from your computer.

Modus Operandi

The RC-50 offers two basic operation modes: Single, in which phrases play sequentially, and Multi, in which they can play simultaneously. The unit has too many playback options to mention them all in this review. In Single mode, when phrases play sequentially, you can switch phrases immediately or at the end of the currently playing phrase. You can set each phrase to fade in when you start it or fade out when you stop it. In Multi mode, a function called Simul-Start allows several phrases to begin playback at the same time. Loop Sync forces all phrases to loop for as long as the longest phrase; with Loop Sync off, phrases loop independently.

You can set any phrase to play back as a one-shot or in reverse using buttons on the front panel. Because pressing them is difficult when you're playing an instrument and the unit is on the floor, those functions are prime candidates for external footswitch control. You can also set each phrase's volume and panning, but with the same limitation—you'll need to use either your hands or external foot controllers.

FIG. 2: On the RC-50's rear panel are two ¼-inch inputs, an XLR input with phantom power, a stereo minijack input, two ¼-inch main outputs, a stereo headphone output, two ¼-inch sub outputs, two foot controller inputs, a USB port, MIDI In and Out, a power switch, and a wall-wart connection.

Tempo is a crucial concept in the RC-50. Every phrase is recorded within the context of a tempo, but the phrases can have several different tempo relationships. The RC-50's tempo functions take advantage of its excellent real-time time-stretch processing (see **Web Clip 1**). Real-time processing allows the RC-50 to have a Tempo Sync function that forces all phrases to play back at the same tempo, regardless of each phrase's recorded tempo (if the tempos are too disparate, of course, audible artifacts are unavoidable). Each patch has a tempo stored with it, but you can set each phrase to either follow the current tempo or maintain its stored tempo.

Time-stretching also makes tap tempo possible. The RC-50 can derive its tempo from an incoming MIDI Clock, but without that, the initial tempo will be based on the phrase's length, which must be defined before recording.

The RC-50 furnishes a Rhythm Guide track, which is essentially a preprogrammed drum machine. It works fine for rehearsing but has several problems for performance. For one thing, the sounds are so-so (see **Web Clip 2**). And although a number of patterns are provided, the Rhythm Guide's complete lack of programmability makes it difficult to avoid monotonous repetition. In addition, though you can route the Rhythm Guide to the sub outputs instead of the main outputs (which can be useful as a reference for a drummer), you can't direct it to the headphone output only, which would be desirable for using it as a click track that the audience cannot hear. Another problem is that a new patch's default setting has the Rhythm Guide turned on, which I found somewhat annoying.

Getting Loopy

The RC-50 is aimed mostly at musicians who want to work in a one-man-band environment, whether for songwriting or for performing solo in a bandlike context. However, I am most interested in live looping, in which all the material is generated live and then manipulated in real time. Devices for this sort of activity range from the Boss RC-20XL and Line 6 DL4 Delay Modeler to the high-powered (and more expensive) Looperlative LP-1. A hardware looper's cost has a lot to do with the number of loops you can record, the number you can play simultaneously, the total recording time, and the unit's manipulation capabilities. The RC-50 strikes a good sweet spot, thanks to its three simultaneous loops and processing functions such as reverse and tap tempo, and it sits squarely in the middle of the price continuum for loopers.

I had a lot of fun with the RC-50, though it definitely has a learning curve. No one should expect to get the best out of the RC-50 without reading the manual, which is good. I really appreciated the RC-50's sound quality, which is well beyond that of most looping devices, including the RC-20XL. With multiple loops, level matching became one of the most important parameter settings in my patches. Because I was creating all material on the fly, I used only a few patches.

For my purposes, the RC-50 does have a few shortcomings. One is that it lacks pitch processing; it is a common trick to step on a footpedal to get half-speed playback that affects both pitch and tempo. And though most looping is tempo oriented, the RC-50 defaults to having Tempo Sync turned on, which imposes its definition of the tempo. It is not difficult to disable Tempo Sync for each phrase (though global disable isn't available), but if you use the default, you must remember to do it every time you start creating a new patch. You could set up your own patch to use as a template and then copy that every time you want to create a new patch, which would allow you to save any settings you use regularly. However, that work-around effectively renders the RC-50's clear-patch function useless.

I also heard a very brief ducking of the sound whenever I kicked out of Record mode. Between that and the default Tempo Sync setting, it was extremely hard for me to get a smooth loop ending when using sounds that were continuous or had long decays (see **Web Clip 3**). The easy fix is to always leave at least a tiny bit of silence at the end of the phrase you want to loop, but that was quite an annoying compromise.

Considering the number of important real-time RC-50 functions that are available only using manual front-panel controls, external controllers are quite important. The RC-50's jacks are a good start, but the only MIDI Control Changes that the RC-50 can receive are CC 1, CC 80, and CC 81, strictly limiting the extent to which MIDI can supplement the other controls.

If Roland ever offers a software update, one nice feature would be the ability to designate one phrase to remain unaffected by patch changes. Such a scheme would allow that phrase to continue playing as you call up other material in different patches and then switch between the patches. Routing the Rhythm Guide to the headphones only would be another helpful addition.

Not to Repeat Myself . . .

Looping is a broad and esoteric musical area. Though the RC-50 is unquestionably a strong product, its level of excellence will vary with your approach. It is a very good tool for live looping, though not the best (but not the costliest, either). For rehearsing, songwriting, and one-man-band-type performances, it is especially robust. Nonetheless, its complexity makes it challenging to master. No matter what your looping style, however, the RC-50 Loop Station is a useful and versatile tool, especially for the price. And for many applications, it is simply the best choice for the job.

EM WEB CLIPS

Larry the O is quite taken with live looping with live looping with live looping-ooping-ooping . . .

PRODUCT SUMMARY

BOSS RC-50 Loop Station

phrase sampler/looper
$689.50

FEATURES	4
EASE OF USE	3
AUDIO QUALITY	4
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Three simultaneous loops. Powerful tempo options. Many playback features. Excellent sound quality.

CONS: Difficult to loop long sounds smoothly. No half-speed playback. A few unfortunate default settings. Rhythm Guide cannot be directed to phones only.

MANUFACTURER
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FIG. 1: The Electro-Harmonix 2880 Super Multitrack Looper lets you record up to four tracks of loops, apply some special pitch, tempo, and reverse effects, mix them down, and then start all over again.

ELECTRO-HARMONIX 2880 Super Multitrack Looper

This digital looper takes up where recorders leave off.

By Orren Merton

A traditional 16-second digital delay line is at the heart of the Electro-Harmonix 2880 Super Multitrack Looper (see **Fig. 1**), but this device offers much more than digital delay; it records four independent mono tracks with integrated mixing and looping. You use a fifth, stereo mixdown track to bounce and then retrieve the mono tracks for reuse. The 2880, with its optional foot controller, is aimed at the desktop studio as well as the performing guitarist.

The 2880 gets its name from its ability to store 2,880 seconds of audio on a 256 MB CompactFlash card; however, the unit comes with a 128 MB card. A 2 GB card can hold 23,040 seconds, which pencils out to a maximum loop length of 64 minutes. (The maximum loop length equals the card capacity divided by six, which accounts for the 2880's four mono tracks and its stereo mixdown track.)

Controls, Ins, and Outs

The top of the 2880 houses its sliders, buttons, and knobs. Each track gets its own pan knob and volume slider. The volume sliders serve a dual purpose: during playback, they act like traditional track volume faders. In Overdub mode, they act as feedback controls, attenuating previously recorded audio as new material is overdubbed. With the slider at maximum, overdubs pile up with no attenuation.

A built-in metronome provides an optional click. A Quantize feature activates a 4-beat prerecord count-in and rounds your recordings to the nearest bar. You can sync the metronome to incoming MIDI Clock or use its internal clock and Tempo slider, which has a 30 to 240 bpm range. Once something has been recorded, a handy LED indicates when the Tempo slider is set to the original recording tempo.

All inputs and outputs are mounted on the back and right side of the 2880 (see **Fig. 2**). Quarter-inch TS jacks feed the left and right stereo channels. Associated trim knobs control the input level. An Aux In stereo minijack is designed

The 2880 offers an excellent combination of features and usability.

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for line-level signals from CD players, iPods, mixers, and so on. You cannot record the signal at this jack, but it is handy for playing along with pre-recorded material.

The left and right outputs are unbalanced and are served by ¼-inch TS jacks. There are no output level knobs, but the stereo TRS headphone jack does have a headphone volume knob. Two MIDI jacks for sending and receiving MIDI Clock, a jack for the included wall-wart power adapter, and a USB connector for interfacing the flash card to your computer round out the picture. You can read and write to the card from the computer.

FIG. 2: The back of the 2880 houses all of the input and output connections except for the USB connector and the CompactFlash card slot, which are on the right side.

Infinite Loop

You initiate a new recording on the 2880 by first pressing the New Loop button to enter Record Ready mode, and then pressing the Record button to start recording on track 1. When you're finished, you either press the Record button again to stop or press the Play button to enter Overdub mode, which plays back track 1 while you record on track 2. You then continue overdubbing, switching record tracks at will until you press the Record button again, which stops everything. Once you've recorded some tracks, you can return to overdub recording or, using the Punch In feature, do standard replace recording for any segment of any track. Although the 2880's four tracks are mono, you can configure them in pairs or as one stereo pair and two mono tracks.

It took me a bit of time to get the hang of recording and overdubbing. I kept expecting Record rather than Play to put me into Overdub mode. Even after I got used to it, I found that playing an instrument while pressing the buttons on the 2880 was clumsy, and I used the foot controller whenever I could.

Once material has been recorded, the Tempo slider functions as a varispeed control, affecting both the pitch and playback speed of the recording. Pressing the Octave button halves the speed and drops the pitch an octave. Pressing the Reverse button reverses the audio. The Tempo slider and both buttons apply to all tracks, so you cannot, for example, decide to reverse only track 2. You can, however, trick the 2880 into reversing one track by recording in Reverse mode and then turning Reverse mode off for subsequent recordings. A similar trick works for pitch-shifting one track.

One If by Foot

For a looper to be really useful in a live performance, it needs to be controlled by foot, and Electro-Harmonix developed an excellent companion foot controller for the 2880 (see **Fig. 3**). The 2880 Foot Controller ($189) is a sturdy metal unit with a footprint similar to that of the company's other footpedals. It has six metal footswitches arranged in two rows of three, with an LED accompanying each switch. You connect the Foot Controller to the 2880 with a standard guitar cable; it does not require a special cable or separate power.

The Foot Controller has a footswitch for each 2880 feature you need to access during live loop creation, and they are logically arranged. Footswitches to control loop creation are in the bottom row: New Loop, Record, and Play. Footswitches for special effects are in the top row: Track Select, Reverse, and Octave. The switches are sufficiently far apart that you don't need to worry about hitting the wrong one.

Mixdown

The 2880 offers two mixdown modes: Normal and Constant Tempo. Normal mode mixes the four tracks to a fifth, stereo mixdown track. Press the Mixdown button and then the Record button, and the audio from the four tracks will be recorded to the mixdown track. Any adjustments you make to the track faders and pan knobs during mixdown get incorporated in the mix.

The number of mixdowns is unlimited. You can fill four tracks with audio, mix them down, record four new tracks, mix those with the original, and so on (see **Web Clip 1**). Mixdown makes the 2880 far more flexible than a typical 4-track looper.

Constant Tempo mode lets you do some special

PRODUCT SUMMARY

ELECTRO-HARMONIX 2880 Super Multitrack Looper

multitrack digital loop recorder
$698

FEATURES	3
EASE OF USE	3
AUDIO QUALITY	4
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Stores loops on standard CompactFlash cards. Includes USB for easy computer integration. Varispeed and reverse functions. Great sound quality. Excellent integration with optional foot controller for live performance.

CONS: Loop length determined by first recording. No 24-bit recording option. Ships with minimal 128 MB CompactFlash card.

MANUFACTURER
Electro-Harmonix
www.ehx.com

varispeed and reverse effects during mixdown. It locks the tempo of the mixdown track but lets you use the Tempo slider and Reverse button to change the speed and direction of the four tracks being mixed down. I used that to create a varispeed, reversed mix, which I used as a psychedelic backdrop when recording normal audio on the other tracks.

Looper's Delight

I mostly used the 2880 as a guitar looper, first plugging the effects send of my THD Flexi-50 into the 2880, then trying my Koll Tornado plugged in directly. In both cases, the sound of the 2880's uncompressed, 16-bit, 44.1 kHz audio was very good. I found the 2880 easy to operate and fun to use as both a live-performance tool (especially with the optional foot controller) and a studio loop-construction device.

Although not entirely in the spirit of tape loops, it would be nice if the tracks had independent lengths, instead of having the first recording determine the length of the entire loop. I understand Electro-Harmonix's decision to record audio at 16 bits to maximize storage time, but I would have appreciated a 24-bit option. Those issues are relatively minor, though. The 2880 might not be the most feature-rich looper on the market, but it offers an excellent combination of features and usability. Anyone looking at standalone loopers should seriously consider the 2880, and don't overlook the Foot Controller.

FIG. 3: The well-designed optional 2880 Foot Controller makes live looping a breeze.

EM WEB CLIPS

Orren Merton is the author of Guitar Rig 2 Power! *(Thomson Course Technology, 2006) and* Logic Pro 7 Power! *(Thomson Course Technology, 2004).*

QUICK PICKS

HEIL SOUND

PR 40

By Eli Crews

When it comes to live sound, few people possess the engineering pedigree of Bob Heil. With credits including the Who, the Grateful Dead, and Jeff Beck, and innovations such as the Heil Talk Box and the reinforcement rig for the Quadrophenia tour, the former theater organist and ham-radio enthusiast has firmly cemented his place in rock 'n' roll history. So when Heil Sound recently released a trio of dynamic microphones, a lot of ears in the audio industry perked up.

The biggest and most expensive of the three new mics is the PR 40 ($375). It has a 1.12-inch diaphragm in a housing that weighs just under a pound. The mic comes with the useful SM-3 mic clip, which includes a Teflon bushing that tightens around the shaft of the mic using a thumbscrew. (The spider-style SM-2 shockmount is available for an extra $100.)

My review mic arrived in a red wooden box that was a bit on the flimsy side, but the mic now ships in an aluminum case. The PR 40 is assembled and tested in the United States and comes with a one-year warranty.

Although it looks like a side-address condenser mic, the PR 40 is a front-address (end-firing) moving-coil dynamic. Its frequency response is from 28 Hz to 18 kHz (–3 dB), which is a wider range than your average dynamic mic's.

Even with its extended low range, the PR 40 has a marked lack of boominess. Heil Sound says this is due to the reduction of the proximity effect, despite the PR 40's having a supercardioid pattern, which typically yields an artificial-sounding low-frequency boost when a mic is placed too close to a sound source. As a result, the PR 40 is ideal for close-miking. In addition, the off-axis rejection of the tight polar pattern offers good isolation when, for instance, you are tracking a full band in one room or are miking drums and want to reduce the bleed from the cymbals and hi-hat.

The Heil Sound PR 40 is a large-diaphragm dynamic mic with a wide frequency response and a supercardioid polar pattern.

Big Sound

I used the PR 40 and its sibling, the PR 30 ($299), in my studio for a couple of months and was impressed with the performance of both. (The PR 30 has a 1.5-inch diaphragm, and the low end of its frequency response is measured at 40 Hz.) The PR 40 captured the sounds of drums, trumpet, trombone, electric guitar, electric bass, and vocals with flying colors and sounded particularly good in the low end when used on a kick drum. I even liked it as a room mic on drums and as an ambient mic on a 4-piece horn section.

The only time I wasn't totally sold on the mic was in front of a male vocalist with a high, reedy voice. The PR 40's presence rise between 2.5 and 8 kHz and its lack of proximity effect didn't flatter this particular singer's voice. For another male vocalist who was singing in a lower register, however, the PR 40 proved its mettle as a strong vocal mic.

Standout applications for me were on a large marching bass drum struck with a mallet and on electric bass through a 4 × 10 cabinet. The low end captured in those situations was large without being overwhelming. In both instances, the PR 40 yielded just the right amount of attack, making the instruments come to life in the control room monitors.

Built to Last

My lasting impression of the PR 40 is that it sounds a lot like my favorite large-diaphragm dynamic mics but with an unmatched presence and clarity. I feel certain this mic will find its way into many studio and live-sound setups.

With its rugged build, wide frequency response, and affordable price, the PR 40 is a solid deal for anyone looking to expand their dynamic mic collection. I, for one, plan to use the PR 40 for years to come.

Value (1 through 5): 4
Heil Sound
(distributed by TransAudio Group)
www.transaudiogroup.com

AUDIO ENGINEERING ASSOCIATES (AEA)

The Ribbon Pre (TRP)

By Myles Boisen

EM ★★★★★ hot pick

Addressing the unique needs of ribbon mics, Audio Engineering Associates (AEA) has released the Ribbon Pre (TRP; $965), a 2-channel solid-state JFET mic preamp that offers a whopping +84 dB of output gain delivered by a minimalist, low-noise circuit design. With +63 dB of gain at the main gain stage and a secondary control offering an additional +21 dB, the TRP can supply sufficient clean gain for any conceivable ribbon-mic usage. The unit's noise is rated at an impressively low –130 dB EIN (A weighted), and its frequency response is equally remarkable, with

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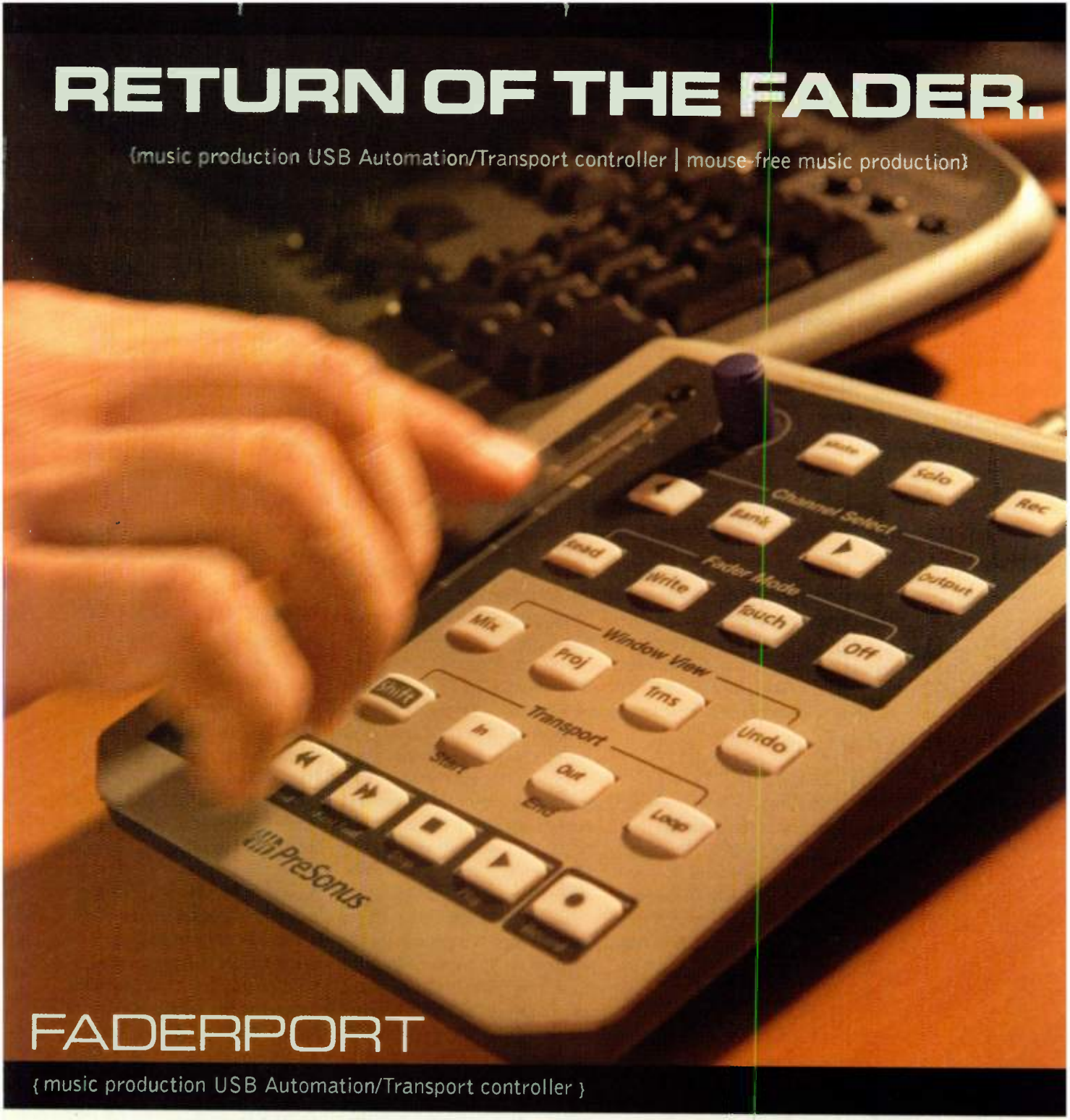

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- Mac and Windows compatible

a rated bandwidth of 300 kHz (–3 dB down at 6 Hz and 300 kHz).

Input impedance can have a noticeable effect on a microphone's performance, particularly with regard to output level, noise, transient response, and frequency response. The ½U TRP's singular design uses a high-input impedance (>30 kΩ) to provide an optimal match for the differing output resistance of various vintage ribbon mics. The preamp also provides DC coupling for superior bass response.

Another unique feature of the TRP is that it does not provide 48V phantom power. One reason for this is that most ribbon mics don't need phantom power (the new phantom-powered ribbons excepted) and can in some cases be damaged by it. Eliminating the 48V phantom supply also contributes to the sonic purity of the device by reducing the number of components in the audio path. This means that AEA's preamp cannot be used with solid-state condenser mics. If connected to an external +48V supply, phantom power is shorted at the TRP's inputs by a built-in protection circuit.

Go with the Gain

Each channel has its own set of controls, arranged symmetrically. The Gain control ranges from +6 to +63 dB, stepped in 5 and 6 dB increments. The Level control is continuously variable and detented at 12 o'clock, which is calibrated at +2.5 dB. Going clockwise from the detent, the Level control boosts the gain up to +21 dB.

When rotated counterclockwise, Level acts as a trim control; at its extreme position, it mutes the audio. Polarity-reverse switches are located to the left of the Gain controls, and 12 dB per octave low-cut switches are to the right of the Level controls. Three LEDs (green for nominal signal, yellow for 0 to +20 dBu, and red for overload warning at +20 dBu) indicate output gain for each channel.

On the rear panel, each channel has a balanced +4 dBu XLR in and out jack and an unbalanced ¼-inch output with 6 dB less level (–2 dBV). The power socket is a 7-pin DIN connector. There are no ¼-inch instrument DI inputs. The TRP has a threaded hole on the bottom of the chassis, and two units will fit on a standard universal rack-mounting tray.

Cutting Through the Mix

Although I am a big fan of ribbon mics, I rarely use them for recording vocals; typically (and even with modern ribbons) it is too difficult to get adequate high end and sufficient level with them. But my first studio experience with the TRP changed all that.

In preparation for recording vocalist Julia Shirar of the band Pillows, producer Kent Sparling and I auditioned a number of mics. We finally settled on the Royer SF-1, a flat-response ribbon that dealt well with Shirar's prominent midrange. But even with an assortment of expensive solid-state preamps at my disposal, I still wasn't satisfied with the sound we were getting. Then along came the TRP. Right away I was amazed at the difference that this preamp made on her vocal tracks.

Although I had previously struggled to get this singer to cut through, the TRP positioned her effortlessly in front of the mix with clarity and detail. Here was all the smoothness I had come to expect from a top-flight ribbon mic, along with silky highs that needed little EQ boosting to cut through electric guitar and drum tracks. Transients were crisp and realistic, and I had no problem getting sufficient level to analog tape.

In later sessions I used the TRP with an assortment of ribbon mics on violin, trumpet, and cornet. The tones I got were big and detailed, bringing dazzling new dimensions to some of my favorite mics. The TRP also worked well with one of my trusted tube microphones.

The Ribbon Pre is a 2-channel preamp designed for use with ribbon mics. It offers +84 dB of gain.

Sold

I thought I had sworn off buying new preamps, but the AEA TRP is clearly a must-have to supplement my mic closet. It is an excellent midpriced, 2-channel preamp that works wonders without breaking the bank.

A staggering 84 dB of gain makes the TRP one of the most potent preamps on the market and one of the few optimized for ribbon mics. And it can be used with dynamic and tube condenser mics, increasing its value and usefulness for studio owners.

Value (1 through 5): 5

Audio Engineering Associates (AEA)
www.wesdooley.com

ARTURIA

Analog Factory 1.2 (Mac/Win)

By Len Sasso

Beginning with its Moog Modular V in 2003, Arturia has released a steady stream of virtual instruments emulating classic hardware synths. Analog Factory 1.2 ($249) combines the synthesis engines from all these virtual instruments with a collection of 2,000 of their best presets and a full-featured preset librarian. An additional $50 will get you the Reloaded version with 1,000 more presets. (Note that Analog Factory does not play sounds sampled from Arturia's other synths; it reproduces them.)

Analog Factory gives you much of the best of Arturia's line of synths at a fraction of the cost. It's a compact, ergonomic package that runs as a standalone on the Mac and Windows platforms; as a VSTi, AU, and RTAS plug-in on the Mac; and as a VSTi and RTAS plug-in on the PC.

It's Not Memorex

To date, Arturia has reproduced six vintage synths: the Moog Modular, the Yamaha CS-80, the Moog Minimoog, the ARP 2600, the Sequential Prophet-5, and the Sequential Prophet-VS. The two Sequential synths are combined in Arturia Prophet-V, which enables you to create hybrid patches using modules from both synths. Analog Factory has a generous selection of presets from

Analog Factory's control panel shows the Preset Manager on top, with the keyboard and synth controls on the bottom. You can hide either view.

each synth in the Arturia line, although surprisingly, there are no hybrid Prophet-5/Prophet-VS presets.

It's hotly debated whether any software instrument exactly reproduces the sound of the hardware synth it is designed to emulate. EM put a number of virtual instruments to the test and got mixed results. Three Arturia models were covered: the ARP 2600V, CS-80V, and Minimoog V (see "The Lost World Rediscovered" in the June 2005 issue of EM, available online at www.emusician.com).

With Analog Factory, a more important question is whether it accurately reproduces the sounds of the other virtual instruments in the Arturia line. I was able to make that comparison for all of them except the Minimoog V, and Analog Factory nailed it every time (see **Web Clip 1**). There are some discrepancies in effects-amount settings, but you can easily fix them.

One from Column A

Analog Factory's Preset Manager makes it easy to find and audition sounds. There are three browser filters: Instrument, Type, and Characteristics. The word *filter* is something of a misnomer, because selections in each category are additive. Adding to selections makes sense for Instrument, because Instrument categories don't overlap—no preset belongs to two instruments. For Type, adding or filtering could work: if you select Pad and Organ, you probably would want all organ pads. But if you select Bass and Guitar, you likely would want to see all guitar presets and all bass presets. For Characteristics, you probably would want the selection to be refined as you choose more categories—Acid, Bright, and Short as opposed to Acid or Bright or Short. But whether it's filtered or additive is a small issue, and finding your way in the Preset Manager is easy.

You can further restrict the Preset Manager to show only your favorite presets (so designated by a checkbox to the right of the preset listing) or only presets that you've created. In both cases, the other categories are still operative, so you can find, for example, all the EFX presets that you have created and marked as favorites. For even greater convenience, you can assign presets to eight snapshot buttons that line the top left of the keyboard, and those assignments persist between sessions.

Tweak of the Week

Arturia gives you access to 15 parameters for each preset. Eleven of those are fixed, whereas four, called Key Parameters, change from preset to preset. Arturia could have taken the easy way out and assigned the Key Parameters on a synth-by-synth basis, but instead it has picked the four most relevant parameters for each preset. That gives you lots of programming flexibility.

The fixed parameters are also well chosen. They include filter cutoff and resonance (both of which you always want on an analog-synth model), LFO rate and amount, effects amount for the chorus and delay, and the ADSR amplitude envelope parameters. Not always having access to filter-envelope parameters is something of a handicap, but not all of the synth configurations allow it.

Analog Factory gives you a lot of cluck for your buck. It's especially roadworthy because it is CPU efficient and easy to use. If you like vintage sounds, you should give this synth a listen. EM

Value (1 through 5): 3
Arturia
www.arturia.com

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NEW

Liquid Technology

The Focusrite Liquid Mix is another Focusrite first and a true one-of-a-kind. Based on the same technology that brought the audio world the acclaimed Liquid Channel, Focusrite now brings Liquid Technology directly into your DP5 studio environment at a much more accessible price. Focusrite Liquid Mix offers emulations of 40 compressors and 20 EQs with a FireWire Hardware controller. Thanks to its built-in DSP, you can have up to 32 channels of modeled vintage or modern compression and EQ with one Focusrite Liquid Mix, with no impact on your native DP5 processing resources. Plus you can control Liquid Mix channels from directly within Digital Performer using Liquid Mix plug-in windows, which keeps your Focusrite processing seamlessly integrated with — and saved with — your DP projects.

The MOTU experts at Sweetwater can build the perfect portable DP5 rig for you. We'll help you select the right components to build a powerful system that lets you take your workflow anywhere, and we can even install, configure and test the entire system for you. Why shop anywhere else?

Waves native processing

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

Ultimate expandable synth

Ultimate Soundbank Plugsound Pro puts a complete "rompler" right inside Digital Performer 5, giving you all the essential sounds you need for your mobile music. Sharing the same UVI engine as the MOTU Ethno Instrument, Plugsound Pro gives you 8 GB of instruments and loops plus powerful effects and performance features, including the innovative drag and drop of audio or MIDI between the plug-in and Digital Performer. An ever-expanding range of add-on sound libraries include retro organs, classic synths, cutting-edge sound FX and textures, modern drum loops, and a just-released collection of must-have retro keyboards.

Legendary drummers

Submersible Music Drumcore delivers access to twelve world-class drummers, such as Terry Bozzio, Matt Sorum, Sly Dunbar and Zoro. The perfect tool for songwriters and composers who need drums quickly in a multitude of styles. Features include an Audio and MIDI librarian (quickly find that perfect groove), "GrooveSets" (for easier songwriting), MIDI instrument (loaded with each drummers' MIDI drumkits) and the "Gabrielizer" (groove generator). Simply drag-and-drop from Drumcore to your Digital Performer 5 tracks or Clippings window.

Komplete control

For DP5 users who want it all: Reaktor5, Kontakt2, Guitar Rig 2 software, Absynth4, Battery3, FM8, B4II, Akoustik Piano, Elektrik Piano, Vokator, Spektral Delay and Pro-53 in a unified interface with hands-on control — Native Instruments KOMPLETE 4 and KORE put an infinite universe of sound at your finger tips. Every preset included in NI KOMPLETE 4, more than 8,500 in total, has been preconfigured and categorized in KORE with searchable musical attributes and hands-on controller assignments. This seamless integration of software and hardware turns Native Instrument's award winning synthesizers and samplers into tactile instruments.

Authorized Reseller

www.sweetwater.com **(800) 222-4700**

Sweetwater

Hit the road with your Dual Core MOTU Studio

Compact MIDI controller

Digital Performer 5 gives you unprecedented control over your MIDI and audio tracks. And what better way to take advantage of this hands-on control than the M-Audio O2, a 25-key ultra-thin USB MIDI controller that goes anywhere you want to make music! Small enough to fit in a bag along with your laptop, the slim-line O2 is perfect for throwing down bass lines, programming drum patterns, triggering effects and tweaking virtual studio parameters - anywhere, anytime. It's easy to assign controllers to match DP5 or your virtual instruments. Plus, the O2 has five internal setup locations so you can switch from application to application. It also integrates with Enigma editor/librarian software for unlimited setup creation and storage.

Professional pad controller

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

Advanced waveform editing

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

Control room monitoring

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

Portable hands-on control

AlphaTrack by Frontier Design Group combines intuitive tactile controls in a compact and highly portable package, with native support for Digital Performer. Ride a 100mm, touch-sensitive, motorized fader with true 10-bit resolution for smooth and precise level control. Three touch-sensitive encoders let you adjust your DP track and plug-in parameters while the 32-character backlit display shows detailed feedback in response to your touch. Map your favorite shortcuts from the Commands menu to AlphaTrack's user-programmable buttons. Jog and shuttle with the touch of your fingers using the integrated Scroll Strip. Slide one finger across the Scroll Strip surface and your project's timeline scrolls in response. Drop a second finger onto the strip, and now your fingers control DP's shuttle speed. Zoom through your project with two fingers, then just lift one finger and slide to quickly set the precise position you want. AlphaTrack is powered entirely though its USB connection so it makes the perfect companion to your portable MOTU recording rig.

Accurate monitoring

The Mackie HR-Series Active Studio Monitors are considered some of the most loved and trusted nearfield studio monitors of all time, and with good reason. These award-winning bi-amplified monitors offer a performance that rivals monitors costing two or three times their price. Namely, a stereo field that's wide, deep and incredibly detailed. Low frequencies that are no more or less than what you've recorded. High and mid-range frequencies that are clean and articulated. Plus the sweetest of sweet spots. Whether it's the 6-inch HR-624, 8-inch HR-824 or dual 6-inch 626, there's an HR Series monitor that will tell you the truth, the whole truth, and nothing but the truth.

Hands-on control for the studio

Imagine the feeling of touch-sensitive, automated Penny & Giles faders under your hands, and the fine-tuned twist of a V-Pot™ between your fingers. You adjust plug-in settings, automate filter sweeps in real-time, and trim individual track levels. Your hands fly over responsive controls, perfecting your mix — free from the solitary confinement of your mouse. Mackie Control delivers all this in an expandable, compact, desktop-style design forged by the combined talents of Mackie manufacturing and the MOTU Digital Performer engineering team. Mackie Control Universal brings large-console, Studio A prowess to your Digital Performer desktop studio, with a wide range of customized control features that go well beyond mixing. It's like putting your hands on DP itself.

Hest Makes Best

By Mike Levine

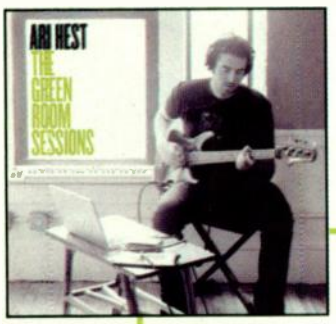
A stellar EP recorded in GarageBand.

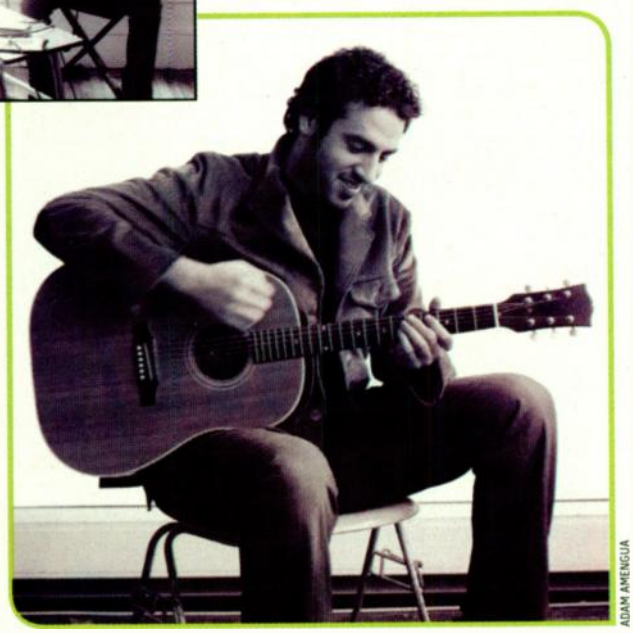
After years of touring and making several indie CDs, singer-songwriter Ari Hest's career was on the rise. He signed with Columbia Records and cut a CD's worth of tracks with vaunted producer Mitchell Froom. But then, the label suddenly soured on the project. "I wasn't sure if Columbia was going to put it out, and I just didn't know what the future held for me and the label," Hest says.

"I was thinking that I'd better make something positive out of the situation," he recalls. So he decided to record new material on his own, figuring that if it all fell apart with Columbia, he'd have something to put out himself.

Hest's plan had a couple of potential pitfalls. First, his only engineering experience had been recording demos for the Froom project. Second, his studio gear was minimal: a Mac G4 running Apple GarageBand, a Shure SM57 mic, and an M-Audio Fast Track USB audio interface.

He initially thought about switching to Apple Logic Pro for the project but decided against it. "I'm not a very computer-literate person," he explains, "and I didn't have a lot of money to spend on this either. So I thought, 'All right, I've got this computer, it has a program in it, everything sounds all right—I'm just going to continue like this.'" One factor in his favor was that his studio, located in his Brooklyn apartment, has just one window, which faces the living room rather than the street. Thus, noise wasn't an issue.





ADAM AMENGUA

The Green Room Sessions/Ari Hest

RIFFS

Ari Hest

Home base: Brooklyn, New York
Sequencer of choice: Apple GarageBand
Microphone: Shure SM57
Web site: www.arihest.com

The songs Hest recorded blend elements of rock, folk, and pop. He sang all the vocals and played acoustic guitar, electric guitar, and electric bass. The only drum sounds were loops from GarageBand. He overcame his lack of engineering experience through trial and error, creativity, and musical savvy. "I had to figure out how to make each instrument distort or not—depending on what I wanted. And I kind of played with the GarageBand plug-ins, which are better than you would think," he says. His biggest frustration with the program was its limited punch-in features.

Hest got surprisingly good results on his vocals and acoustic guitar through the SM57. He also did a superb job of mixing, considering his lack of experience and that he was listening primarily on a pair of Sherwood TV speakers—not exactly studio reference monitors.

Soon after Hest had finished up ten songs, a management change at Columbia brought him back into favor. The label put out six of Hest's self-recorded songs on an EP, *The Green Room Sessions*, last year and will soon release the CD that Froom had produced (*The Break In*, which is due out this month).

Mastering revealed one flaw with Hest's mixes: his TV-speaker monitors had caused him to overly hype the bottom end. But the mastering engineer was able to reduce the tubbiness, and the sound of the EP is impressive. The understated Hest is modest about his accomplishment. "I definitely feel lucky," he says. "I certainly worked hard at it. But I know it's not supposed to be that easy to come up with something that sounds good."

Despite his obvious engineering aptitude, he has no plans to move up to a gaudier sequencer anytime soon. "I'm actually afraid to graduate to a program that would require me to figure out things that I'm not accustomed to," he says. You can't argue with success. EM

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