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JULY 2006

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REVIEWS

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A PRISM BUSINESS MEDIA PUBLICATION

OUTSPOKEN MIX ENGINEER DAVE PENSADO

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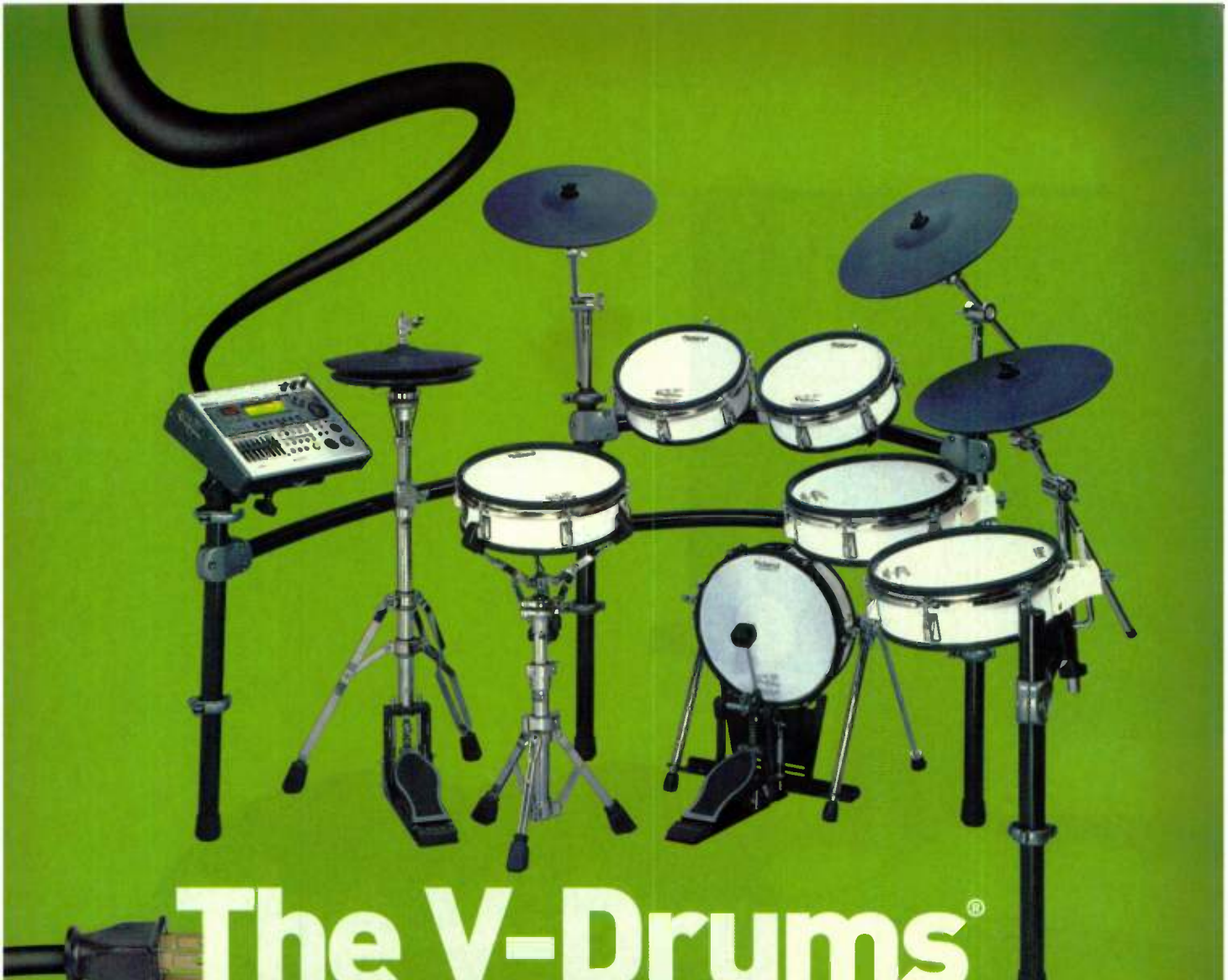

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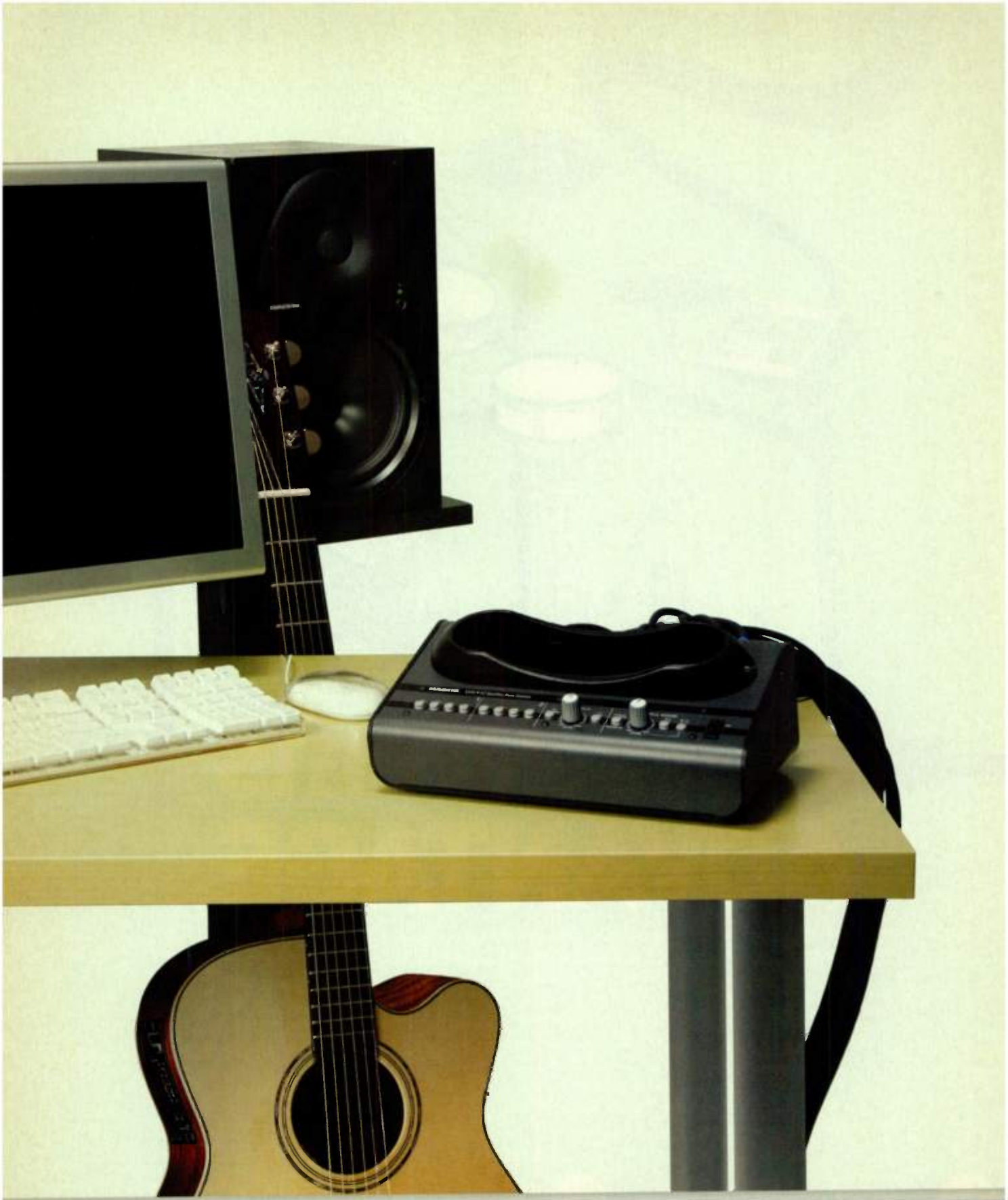

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Electronic Musician
July 2006

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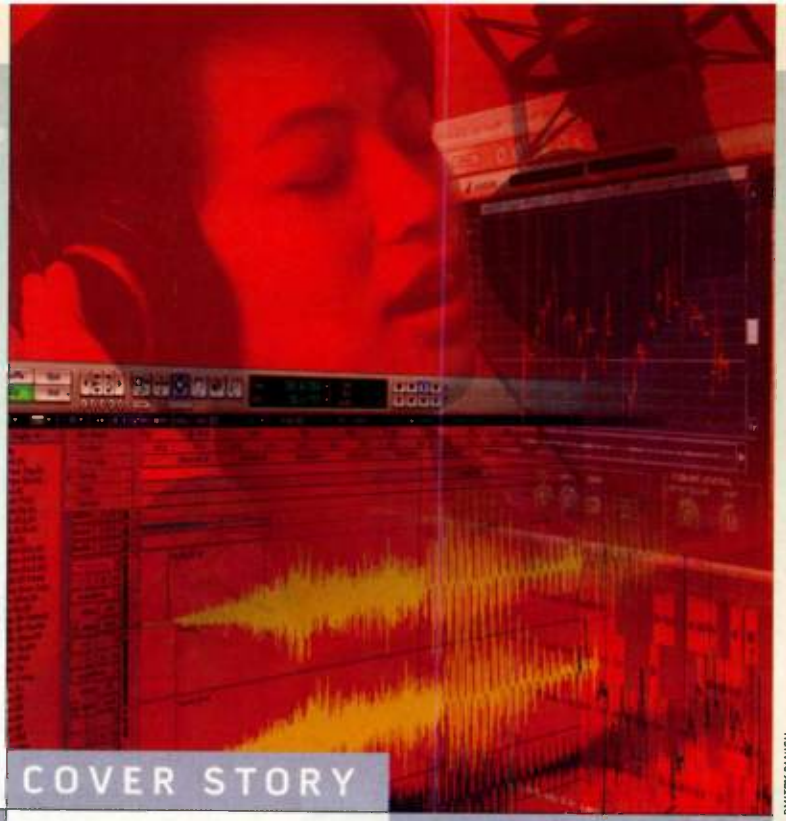

DMITRY PANICH

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33 LORD OF THE RINGTONES

Designing MIDI ringtones and audio real tones for cell phones can be a good business for personal-studio owners who understand the technology and the market. Here's the information you need to take advantage of this opportunity. By Luke Holden

42 VOCAL MAGIC

Six top producer-engineers discuss their techniques and tricks for desktop vocal production, including comping, pitch correction, dynamics control, and more. By Mike Levine

59 PRODUCTION VALUES: MIXING DOWN AND SPEAKING UP

Dave Pensado has mixed for such major artists as Destiny's Child, Christina Aguilera, and Beyonce and was one of the mixing engineers on Mary J. Blige's new album, *The Breakthrough*. In this provocative interview, Pensado offers controversial opinions on mixing and talks about working on the Blige project. By Paul Tingen

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The Man Behind the Curtain

People behind the scenes often have a much greater influence on the development of our industry than we realize. Such a person was Korg USA president Michael Kovins, who passed away on May 2 at age 57 after a long battle with leukemia.

Mike Kovins wasn't a product designer. He didn't play on hit records, although he was a good trumpet player and understood music production. As vice president of marketing at Unicord (Korg USA's predecessor), Kovins was key in bringing the Korg Polysix and Mono/Poly to the U.S. market. He later led the U.S. introduction of the legendary Korg M1 synth and helped bring the Triton, the Oasys, and many other products to these shores. But the real reason I bring up Kovins's work is to point out his deep dedication to music education and his efforts to grow the music-technology industry.

Kovins supported music education with a passion. His interest preceded his work at Korg, but he started making a real impact when he became a driving force behind the creation of SoundTree, the educational division of Korg USA. SoundTree not only sells music technology to educators, but also sets up music labs, offers extensive technology training for teachers, and provides free lesson plans and other aids.

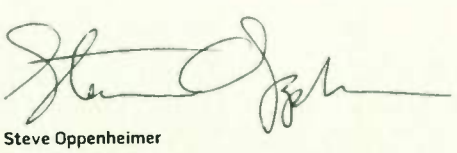
JULIE BIRUM

To help grow the overall market for music-technology products, Kovins cofounded the International Association of Electronic Keyboard Manufacturers (IAEKM), an organization that brought together a variety of influential companies in the music-technology field, not just keyboard manufacturers. IAEKM's purpose is to encourage manufacturers to share information to better market music-technology products and to get more people to make music with electronics. Convincing competing companies to cooperate was hard work, but Kovins persisted.

Kovins then helped convince IAEKM's members that a good way to get more people involved with making music was to reach out to the next generation by bringing music technology to the schools. Ten years ago, with backing from IAEKM and NAMM, he helped found the Technology Institute for Music Educators (TI:ME), a nonprofit organization dedicated to training music teachers in the use of technology. The Berklee College of Music recognized Kovins with its Golden Clef award for Lifetime Commitment to Music Education.

I met Kovins around 1990, at a Winter NAMM show. Later, when I started planning a music-technology magazine for teachers, he was one of the first people I approached for advice on the subject. Always a busy man, he repeatedly made time to discuss my ideas and offer suggestions. In 2003 the EM staff launched *Music Education Technology* magazine, which has been very well received by U.S. educators.

Mike always worked hard to make the industry better for all of us and to improve our children's education, and without his encouragement and his company's support, MET would not be as strong as it is. You can make a donation in his memory to the Leukemia & Lymphoma Society (www.leukemia.org), or if you are interested in music education, ask the folks at TI:ME (www.ti-me.org) how you can help.



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LAURA WILLIAMS

Letters

There's Much to Gain

Orren Merton ran into the same problem that I did when he reviewed the MOTU Traveler (see the February 2006 issue of EM). MOTU lists the unit's microphone gain range as +73 dB at full gain with the –20 dB pad disabled, but the highest gain shown on the front panel of my unit is only +53 dB. Unless the meaning of the word *pad* has changed completely, engaging the pad reduces the available gain to +33 dB, rather than increasing it to +73 dB as described in the Traveler manual.

A gain range of +73 dB should support a ribbon mic, especially a modern design such as the Royer R-121 that Merton used. But as he discovered, +53 dB won't be quite enough. I, too, accepted the published gain figure at face value, even though the Traveler could not raise the signal from my vintage RCA BK-5B above the hiss. A dedicated submixer with higher gain solved the problem, but your readers should be aware that if they plan to use this otherwise excellent piece of gear with unpowered ribbon microphones, they might be disappointed.

Robert J. Spear
Ithaca, New York

MOTU director of marketing Jim Cooper replies: Robert, we haven't experienced any issues with gain and ribbon mics, so your categorical dismissal regarding the Traveler and ribbon mics strikes me as an overgeneralization. It is more likely that individual mileage may vary, depending on the specific ribbon-mic model used and the source material being recorded.

For the gain range, we are thinking of the difference between these two states: pad engaged with trim set to zero (–20 dB) and pad disengaged with trim set to +53 dB. The difference (range) between these two states is 73 dB. We have been careful to refer to this figure as "gain range" and not simply as "gain." In fact, page 5 of the Traveler manual says: "The total gain range—from when the pad is enabled and the trim is turned all the way down to when the pad is disabled and the trim is turned all the way up—is 73 dB."

> There is a real problem with spending $10,000 (or more) on a box that "does it all."

Oasys or Mirage?

As a keyboard player and studio owner, I was very interested in reading Geary Yelton's review of the Korg Oasys in the February 2006 issue of EM. Kudos to Yelton for the extremely comprehensive and well-written review. I don't doubt that the Oasys is a marvelous piece of gear, and I certainly wouldn't kick it out of my studio if someone would like to give me one. I take issue with only one comment, which is in the conclusion and reads: "Commercial recording studios that don't own one will be at a real disadvantage."

That statement seems like quite a leap. It might have been better if Yelton had said, "Someone who wants to sample, record, sequence, add processing and effects, and play keyboards/synths with only one piece of equipment will be at a disadvantage if he or she doesn't buy the Oasys." How much, after all, am I really adding to my competitive advantage if my studio already has a Steinway grand, a collection of synths, a Pro Tools | HD rig, and a bunch of high-end dedicated effects, processors, sequencers and samplers (in hardware and software)?

Aside from that, there is a real problem with spending $10,000 (or more) on a box that "does it all." Does anyone remember another similarly priced item called the Kurzweil 250? I can get you one for a few hundred bucks now. Eventually, various components of the Oasys will become obsolete, underpowered, or by current standards painfully slow, and not all at the same time. There is a list a mile long of manufacturers who promised their stuff would never become obsolete, and guess what? It did.

The Oasys looks as though it would be a lot of fun to use on a gig, assuming that you want to lug around something that bulky. But a must-have for a well-outfitted commercial recording studio? Sorry, I'm not buying it (or the Korg, for that matter).

David Sparr
via email

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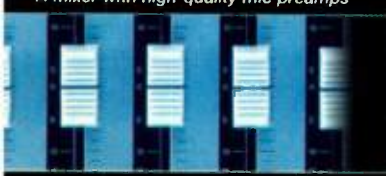

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Professional producer-engineers share their favorite ribbon-microphone applications.

FEATURES

Master Class:
Apple GarageBand

Macintosh guru Bob "Dr. Mac" LeVitus focuses on advanced tips, tricks, and techniques for the hugely popular program GarageBand.

Production Values:
Dweezil Zappa

EM gets the inside scoop on Dweezil Zappa's new CD, *Go with What You Know*. Other topics include Dweezil's summer Tour de Frank and the makeover of Frank's studio.

COLUMNS

Making Tracks:
Stronger Strings

Noted arranger Steve Skinner discusses his techniques for arranging string parts using a sequencer and sample libraries.

Sound Design Workshop:
Extreme Resonance

EM looks at the strange and unpredictable sounds that result from sending audio through a lowpass filter that has its resonance set at a high level.

Square One:
Using FM for Fun and Profit

Learn about the uses of various frequency-modulation synthesis parameters.

. . . and much more

Letters

EM associate editor Geary Yelton replies: Thank you for your letter, David. I have reconsidered my position—I should have said that commercial studios owning an Oasys will have an advantage, rather than saying that those that don't own one will be at a disadvantage. If a studio's prospective clients are inclined toward using electronic instruments, and all of its other gear and services meet their needs, a truly great synthesizer will be additional incentive. Although the Oasys is a machine that seemingly does it all, I most appreciate its capabilities as a synthesizer—as a versatile source of interesting timbres. The Kurzweil 250, though a groundbreaking instrument, was a relatively closed system that had a predictably limited life span. In its time, though, the K250 helped attract clients to many studios and was well worth the investment. I expect that the Oasys will be useful and respected for many years to come, but only time will tell. I'm glad you enjoyed the review, and I'm grateful you took the time to write.

Burn Them In

The September 2005 issue of *Electronic Musician* ran a review of our Proline 750 headphones, written by Michael Cooper. The first time I read the article was when I received the issue, and based on Cooper's comments, I knew there was a problem with the headphones he had received for the review. The 750s have a driver that's made of titanium, which is a very accurate but stiff metal. A "burn-in" period (24 to 48 hours of continuous play) really warms up these headphones, brings out the mids, and rounds out the high end so it isn't harsh. I realized that Cooper had been sent a pair that had not been burned in. The review had already been printed, and there was nothing we could do about that, but it was a personal issue to me because I, and many professionals, love these headphones. I wanted Cooper to hear what they should sound like, so I sent him another pair, burned in. Please print our letter so that readers of your magazine can have the most accurate information we can offer.

Cathy Kelly
CEO, Ultrasone of America

EM reviewer Michael Cooper replies: Ultrasone told me that the Proline 750s' new manual now mentions the need to burn in the headphones before using them, something that was never documented or relayed to me during the time I was writing the review.

The burned-in Proline 750s that Ultrasone recently provided sounded much warmer, smoother, and more balanced than the two sets of units I tested and wrote about in my review. They still sounded a little scooped in the midrange band, making vocals and guitar, for example, sound thin and cutting at times. But the spectral balance and stereo imaging had improved dramatically compared with what I remember hearing while using the review units. The Proline 750s' transient response is outstanding, and the phenomenal bass extension that these cans offer makes them an excellent reference for checking the bottom end of a mix. On the downside, I still feel the fit is uncomfortably rigid. However, my overall impression is that these are high-quality headphones that are worthy of consideration for professional use. EM

We Welcome Your Feedback

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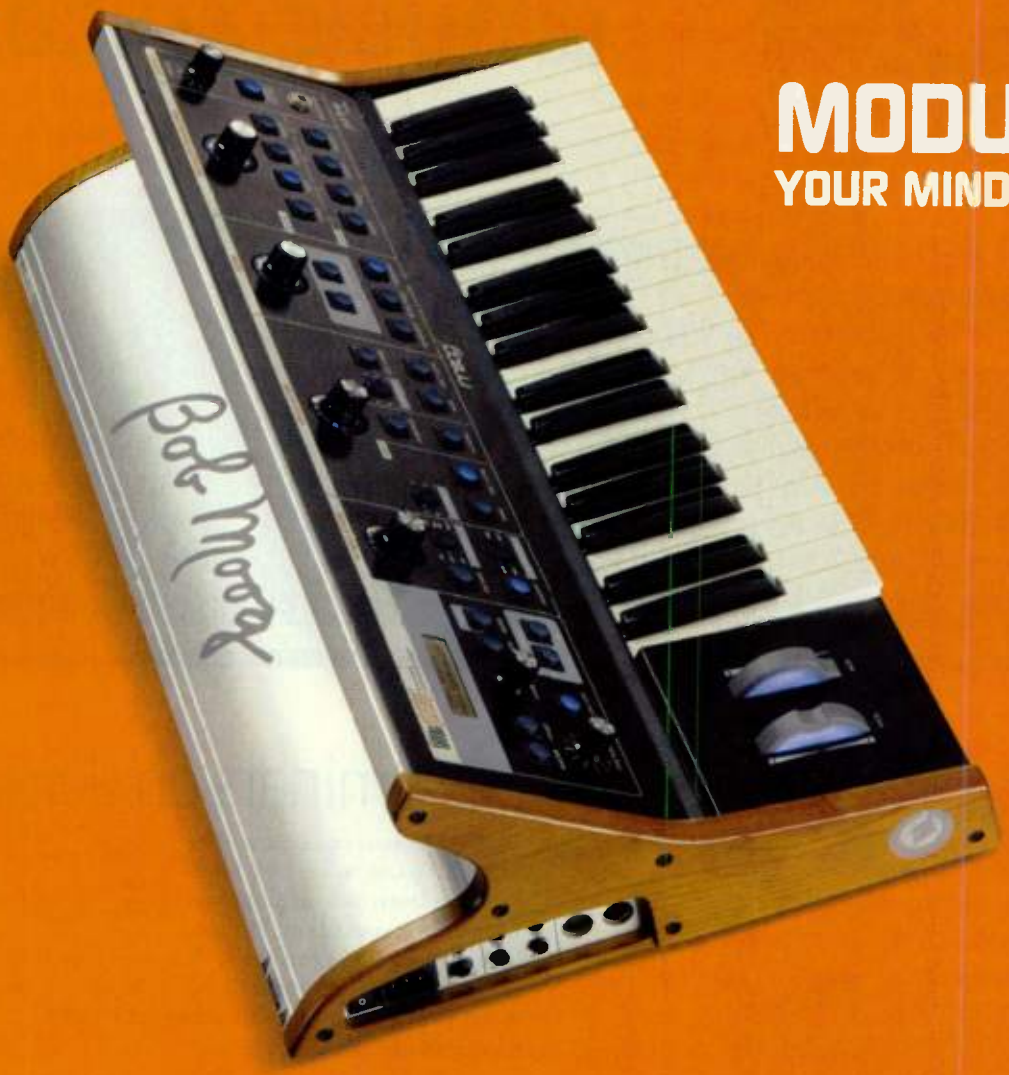

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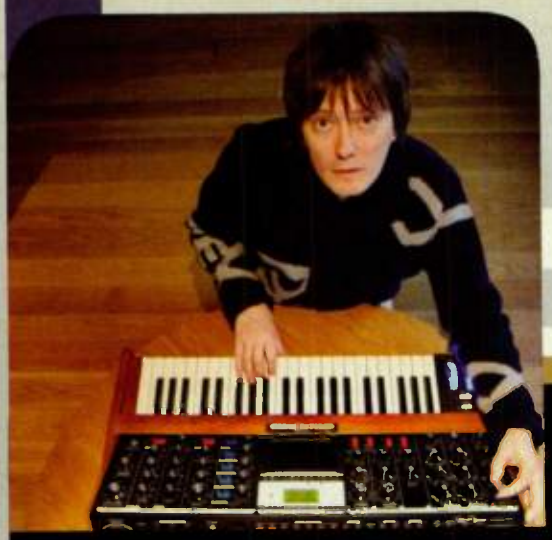

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EMspotlight

Roger O'Donnell: The Truth from the Voyager

Although Cure keyboardist Roger O'Donnell can use any synth he wants, the recent Moog documentary inspired him to focus on just one—the Moog Voyager—for his latest solo project, *The Truth in Me* (99 X Out of Ten, 2006). The result is a collection of songs that shows the sensuous and moody side of the Voyager. In this exclusive interview, O'Donnell talks about synth programming, songwriting, and his collaboration with singer Erin Lang. 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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TASCAM

By Geary Yelton

Open Labs MiKo

Open Labs (www.openlabs.com), maker of the OpenSynth NeKo, has begun shipping the MiKo ($2,499 and up), a smaller keyboard workstation based on a single- or dual-core 64-bit AMD Opteron processor. By combining a well-equipped computer, a MIDI keyboard and controllers, an audio interface, and an assortment of software, the MiKo lets you compose, record, sweeten, mix, and master your productions, and then burn them to disc or distribute them on the Web.

Like the NeKo, every MiKo is a custom configuration. The base model has 37-note and QWERTY keyboards, a 15-inch color touch screen, a dual-layer DVD-RW drive, 512 MB of RAM, four PCI slots, user-assignable Alpha and DJ control surfaces, and a 24-bit, 96 kHz audio interface with four inputs, six outputs, and two mic preamps. Connections include audio I/O, FireWire, USB 2.0, Ethernet, and 802.11g wireless.

You can order or install 4 GB of RAM, a 250 GB hard drive instead of the standard 80 GB drive, a second hard drive, or the optional OpenPad trigger pad array. The MiKo comes with Windows XP, Open Labs' Musical Operating System, Karsyn, mFusion, Cubase LE, and multiple software instruments, effects plug-ins, and sound presets. An optional Windows application called Mimik ($199) lets you sample any MIDI sound source automatically, effectively cloning other instruments.

MOTU Digital Performer 5

Digital Performer 5 (Mac, $795; $195 for upgrade), an updated version of the audio and MIDI sequencer from MOTU (www.motu.com), adds features such as Record-Safe input monitoring, enhanced count-off, customized click patterns, automatic voice allocation, and the ability to play instruments from your computer keyboard. You can organize tracklists into folders and subfolders that you open and close with a click. The toolbar contains four new audio-editing tools: Trim, Slip, Slide, and Roll. The waveform editor offers real-time pitch correction and direct editing of audio file tempo maps. For film composers, DP5 can display visual cues on a video monitor and display and export streamers, flutters, and punches. The Meter Bridge lets you monitor all audio signal paths at a glance.

Bundled with DP5 are six CPU-efficient instrument plug-ins. PolySynth is inspired by the Roland Juno-106, and Modulo is a versatile dual-oscillator subtractive synth. NanoSampler lets you create instant sampler instruments by dragging audio files from other windows. BassLine is a single-oscillator, Minimoog-style monosynth, and Proton is a 2-operator FM synth. DP5 also includes Model 12, a programmable drum module with hundreds of factory samples. Previous Digital Performer users will appreciate the ease of adding instrument and MIDI tracks simultaneously.

Native Instruments Kore

Kore (Mac/Win, $569), from Native Instruments (www.native-instruments.com), combines hardware and software in an integrated host system that runs any VST or AU plug-in. It operates either as a standalone or as a single plug-in within any AU, DXi, RTAS, or VST host program.

The new KoreSound format organizes routing and layering data and embedded MIDI files as well as combinations of sounds and effects from various plug-ins according to their musical attributes. Double-clicking on a sound in the Kore browser will load its associated plug-in. Thousands of presets for Native Instruments software are included, and you can categorize presets from any software instrument or effects plug-in. Kore's software gives views of a virtual rack containing instruments and effects, and a mapping editor facilitates assigning splits and layers graphically. Standalone mode simplifies using software instruments onstage with your computer. Kore also simplifies moving sounds and setups from one sequencer or computer to another.

Kore's controller is an audio and MIDI interface as well as a control surface. It has analog stereo in and out, coaxial S/PDIF out, a ¼-inch stereo headphone jack, MIDI In and Out, a USB 2.0 port, and inputs for two footswitches. Eight touch-sensitive knobs offer five times the resolution of MIDI controllers. When you touch a knob, its assigned parameter and value are displayed. Matching plug-in parameters are assigned to the same knobs across Native Instruments' entire product line, and you can easily assign knobs to any parameter that responds to MIDI Control Changes.


X2
Sample
Twist
Morph
Generate
Transform
EmulatorX2
E-MU's new Emulator X2 is a 24-bit/192kHz Software Sampler for PC that gives you the sonic tools to create any sound imaginable. Whether playing and tweaking sounds from E-MU's diverse expansion library, importing your favorite sounds (Giga, Akai, REX2 and more), or creating your own samples, Emulator X2 is an all-in-one sampler, synthesizer and sound module that lets you twist, morph, generate and transform your sounds like nothing else.
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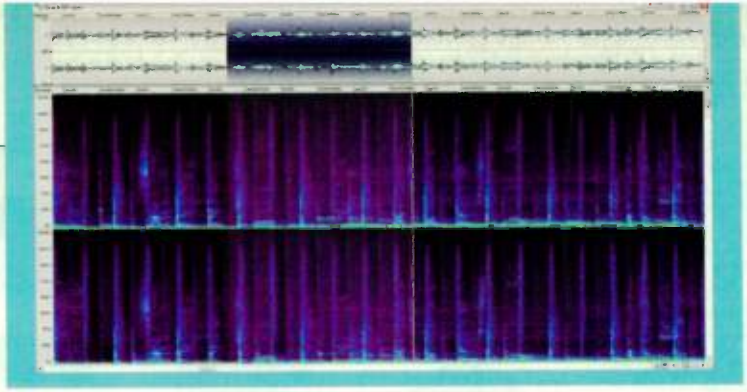
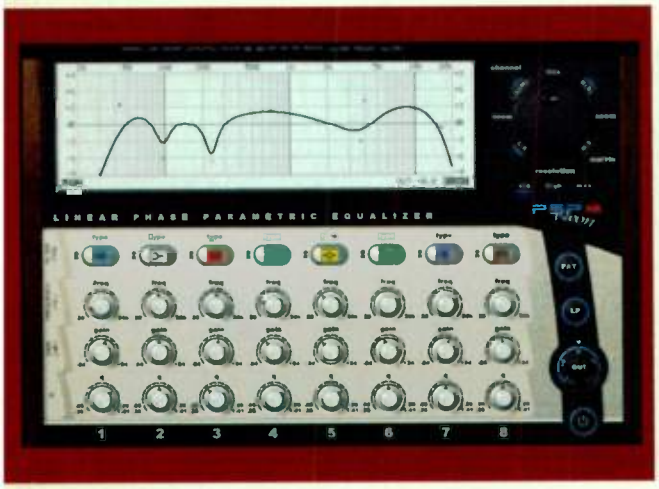


Steinberg WaveLab 6

Steinberg (www.steinberg.de) is shipping WaveLab 6 (Win, $699.99; discounted upgrades available for previous WaveLab owners), the latest version of its surround-capable multitrack audio-editing and mastering program. WaveLab 6 offers dozens of new and enhanced features, such as the sophisticated Spectrum Editor, Dirac time-stretching and pitch-shifting, and the ability to remotely control any WaveLab command with MIDI. The Audio Montage, which enables simultaneous edits across several tracks, has new editing, batch-processing, and clip-handling functions. Metering and monitoring tools include support for mastering engineer Bob Katz's K-System and the new Loudness Distribution window.

Spectrum Editor gives you a sonogram view and allows you to remove unwanted frequencies and noise, process selected frequencies with plug-ins, and cut and paste background ambience from one file to another. To counteract level variations caused by plug-ins and external hardware, SmartBypass automatically compensates for changes in perceived loudness. Additional enhancements include sampling-rate conversion up to 384 kHz and the ability to import files larger than 2 GB.



PSPaudioware PSP Neon

PSP Neon (Win, $149), a new product from Poland's most prolific purveyor of processor plug-ins, PSPaudioware (www.pspaudioware.com), is a linear-phase equalizer with eight independent bands, each assignable to any of seven filter types. The software uses frequency-domain, fast-convolution-based linear-phase algorithms, resulting in CPU efficiency and 100 percent phase accuracy. Even if you turn off Linear Phase mode to minimize CPU cycles, you can still use the plug-in's analog-modeled filters in Infinite Impulse Response mode.

In addition to the standard edition, there is PSP Neon HR (Win, $299), a high-resolution version of the program that operates at sampling rates as high as 192 kHz. It offers stereo and mid-side operation, independent control and processing of each channel, and PSPaudioware's exclusive FAT mode. PSP Neon HR requires an iLok USB key. Both plug-ins support DirectX, RTAS, and VST formats.

Key Changes

Registered owners of Kontakt 2 (Mac/Win) can download the much-anticipated 2.1 update from **Native Instruments** (www.native-instruments.com). Kontakt 2.1 gives you shorter loading times, less memory usage, better multiprocessor support, and a lower processor load. Its Search function has been improved, and you can minimize and customize the view. Other features include new effects, RTAS compatibility, and additional import formats . . . **Steinberg** (www.steinberg.net) has released a software development kit that updates the Virtual Studio Technology (VST) plug-in format to version 2.4. VST 2.4 supports 64-bit audio, double-precision floating-point processing, and Intel-based Apple Macintosh computers . . . After years of being available only for the Mac, Altiverb for Windows should be available soon, according to developer **Audio Ease** (www.audioease.com). It should be out by the time you read this . . . **FabFilter** (www.fabfilter.com) has updated all three of its plug-ins—Twin, Volcano, and One—for Windows and Mac OS X. Improvements include new functionality, updated VST and AU support, bug fixes, and support for Intel-based Macs. The updates are free for current users . . . **Applied Acoustics Systems** (www.applied-acoustics.com) is offering free updates for two of its software instruments. Ultra Analog 1.0.2 and String Studio 1.0.1 enhance their stability in host sequencers and extend their authorization grace periods to 15 days. Ultra Analog also has better filter response, and the behavior of String Studio's Termination model has been corrected . . . Prophet-V owners can download a free update from **Arturia** (www.arturia.com). Prophet-V 1.1 fixes a few bugs and adds Universal Binary support to the Mac version.

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Eventide Anthology II

Anthology II (Mac/Win, $1,195) from audio hardware and software maker Eventide (www.eventide.com) is a TDM collection that introduces six plug-ins to supplement the previous Anthology suite's nine. Anthology II adds two channel strips, two 48-bit double-precision equalizers, a phase-alignment tool, and a new Harmonizer. Existing Anthology owners can get the new plug-ins for $299. Upgrade pricing is also available for Eventide Reverb, Octavox, MassivePack, H3000 Band Delay, and Clockwork Legacy owners.

The Ultra-Channel channel strip has a gate, a 5-band EQ, a de-esser, a stereo delay, and a micropitch shifter, as well as an Omnipressor compressor/limiter. With a gate, five bands of EQ, and a compressor/limiter, E-Channel offers maximum CPU efficiency for a greater number of instances when tracking.

The versatile EQ65 Filter Set duplicates an analog filter bank with third-order lowpass and highpass filters and two variable bands of either notch or peak filtering. EQ45 has second-order lowpass and highpass filters and four bands of parametric EQ with continuously variable bandwidth and frequency.

The Precision Time Align plug-in corrects phase-alignment problems in separate tracks and is especially useful for dealing with multiple-mic setups used to record drums. Quadravox features four voices of Harmonizer pitch-shift, with individual delay and pan for each voice.

Download of the Month

PSYCHIC MODULATION RECEPTION 1.0 (WIN)

Imagine your latest masterpiece as the soundtrack for a vintage sci-fi thriller viewed on a 1952 10-inch TV with rabbit ears. That's what Psychic Modulation's Jack Resweber was aiming for when he designed Reception 1.0, a free VST effects plug-in for Windows. You'll find a variety of free and modestly priced VST instruments and effects at the Psychic Modulation Web site (www.psychicmodulation.com). You can buy a bundle of all current and future plug-ins for $49.95.

Reception combines feedback delay, distortion, ring modulation, and lowpass filtering with enigmatic controls and an undisclosed signal path. The trick is to forget about what's going on under the hood and just play with the TV-style controls. If your plug-in host allows you to assign MIDI controllers to the numericals and buttons, all the better—the controls are set up for real-time tweaking.

The Color and Tint numericals and the little vertical slider to their left affect delay-line modulation. Low numerical settings produce phasing, whereas high settings yield chunky discrete-delay sounds. The Contrast and Bit numericals control downsampling and bit-depth reduction, respectively. The Bright numerical sets the lowpass filter cutoff.

Buttons at the top of the TV toggle the effect on and off, introduce low-level noise, and add ring modulation. The vertical slider to the right of the Ringmod button controls the frequency of the modulating sine wave. The Warning button momentarily cranks the delay-line feedback to maximum. The Noise and Warning buttons are especially effective when assigned to a MIDI switch, button, or key and played in rhythm (see **Web Clip 1**).

My favorites among Psychic Modulation's other offerings are Paradigm Shifter 3 and Minimal. Paradigm Shifter combines filtering and delay, and it offers step sequencers for each effect's parameters. It can produce some truly off-the-wall results. Minimal is a 4-track synthesized drum-and-bass machine. Each track has a step sequencer and can be routed to its own output (host permitting). Minimal's simplicity belies its power. Finally, honorable mention must go to phOne, a synth designed especially for telephone and Morse code sounds.

EM WEB CLIPS

—Len Sasso

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DV Toolkit™ 2 option also also available, featuring a collection of powerful post production tools.

SoundTech LightSnake

Although you'll find no shortage of methods for getting a guitar signal into your computer, perhaps the simplest and most direct would be a cable with a ¼-inch phone plug on one end and a USB connector on the other. That perfectly describes the LightSnake USB Intelligent Instrument Cable ($69.99), a unidirectional "sound card on a cable" from SoundTech (www.soundtech.com). The LightSnake is a 15-foot, heavy-duty shielded cable containing a 16-bit A/D converter with a 48 kHz sampling rate and an embedded signal booster.

Just plug in your guitar, bass, or keyboard, and you can record audio sequencer tracks or play through your virtual guitar-amp software. The LightSnake is USB 2.0 device compliant; no drivers are necessary for Windows or Mac (an ASIO driver is in development). To minimize latency, an included ¼-inch splitter lets you send a signal to your computer and to an amp or mixer simultaneously. The LightSnake features Host Side Data Loss (HSDL) noise reduction and emits a luminous green glow when connected.

M-Audio Sputnik

Although the Sputnik ($699.95) is the first microphone from M-Audio (www.m-audio.com) to exceed the $400 price barrier, it is designed to reproduce the sound of mics costing considerably more. The Sputnik is a large-diaphragm tube condenser mic that lets you select cardioid, figure-8, or omnidirectional polar patterns. It is M-Audio's third lollipop-style mic, and it has a 3-micron-thick evaporated-gold-coated Mylar diaphragm housed in a solid brass body with a polished nickel finish. A tuned brass backplate ensures optimal on- and off-axis frequency response and pattern symmetry, according to M-Audio. Two switches enable an 80 Hz highpass filter and –10 dB attenuation.

In addition to imparting the vocal sound associated with tube condenser mics, the Sputnik offers maximum versatility for recording almost any instrument. Its Class A cascode amplifier circuit has a high output impedance and a discrete transconductance design that incorporates a handpicked, military-grade 6205M pentode vacuum tube wired as a triode. The dedicated power supply allows cable runs as long as 200 feet. Each mic also includes a 7-pin mic cable, a shockmount, a cloth bag, and an aluminum flight case.

SoundToys Native Effects

SoundToys (www.soundtoys.com), a developer of TDM plug-ins for Pro Tools, has bundled six of its most popular plug-ins and made them available for Pro Tools LE and Pro Tools M-Powered. Native Effects (Mac, $495) is a suite of RTAS effects plug-ins designed to emulate the warmth of analog processing. Each features MIDI synchronization, multiple modulation types, and a collection of presets.

EchoBoy is a virtual analog and digital delay plug-in with intricate rhythmic capabilities. It offers 30 types of repetition, from lo-fi chorused delays to tape-echo simulations. FilterFreak gives you rhythmic resonant filtering that syncs to tempo. PhaseMistress generates stereo phase-shifter effects and simulates virtually every popular hardware phaser. For time- and pitch-stretching tasks, Speed offers Simple, Graphic, and Calculator modes. Inspired by the Eventide H3000, Crystallizer combines granular reverse-echo slicing and pitch-shifting. Tremolator not only produces tremolos and gating effects, but also supplies programmable waveshape and rhythm editors to customize your sound. EM

...I've done a lot of shopping around to see who's got the best prices, and it always seems like it's you guys...

Rey Hernandez aka DJ Spinn

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Insane in the Membrane

By Scott Wilkinson

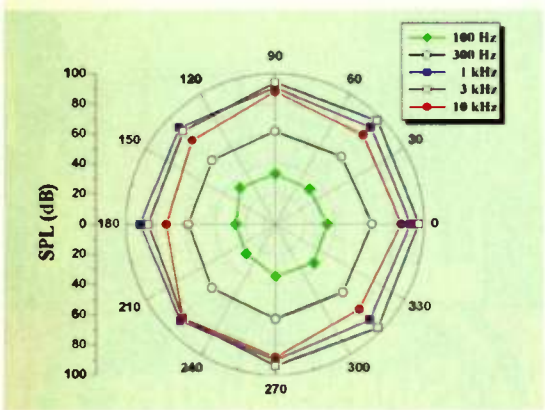
A new approach to thin-membrane speaker drivers.

Speakers haven't changed much over the past few decades, and speaker technology is one of the most mature and stable technologies available to the electronic musician. In particular, the radiation patterns of various drivers are well known and difficult to manipulate, which has led designers to try many different configurations of cones, domes, ribbons, and horn-loaded compression drivers to create speakers that have optimized directional characteristics.

One approach to controlling the directivity of a single driver is to construct it from a thin, flexible sheet of plastic that can be shaped in various ways. However, this presents two problems: finding a type of plastic that responds to electrical signals, and then finding a way to permanently attach electrodes to the plastic.

Koh Seok-keun, president and CEO of South Korean startup Plasma & Ionbeam Corporation (www.plasma-ion.com), may have a solution to both problems. In the 1980s, when he was working on his Ph.D. in the United States, Koh began working on drivers made from thin sheets of plastic. He used a type of plastic called polyvinylidene fluoride (PVDF), which exhibits piezoelectric properties. A material that has such properties can be made to vibrate when exposed to an electric current, but he was unable to reliably affix electrodes to the plastic film.

Koh dropped his research for a time and instead worked at the Korean Institute of Science and Technology on other projects, including the study of ion-assisted reaction (IAR) surface-modification technology. This technology immerses a material in a plasma gas at temperatures exceeding 10,000° Celsius and irradiates it with low-energy ion beams to change the characteristics of the material's surface.



COURTESY P&I CORP.

FIG. 1: A cylindrical plastic-film speaker exhibits an even radiation pattern at all frequencies.

IAR was initially used to improve the efficiency of heat exchangers in air conditioners, but Koh soon realized that the process could also be used to change the adhesion characteristics of PVDF to allow platinum electrodes to remain affixed to it. Specifically, the surface becomes hydrophilic—that is, it exhibits an affinity for water molecules—which plays an important role in bonding the plastic to the metal atoms in the electrodes.

COURTESY P&I CORP.

FIG. 2: A semicylindrical midrange/tweeter driver could be mated with a conventional woofer to cover the entire frequency spectrum while radiating the mids and highs evenly along the vertical axis.

Koh established P&I Corporation in 2000 to commercialize the IAR process for various applications, including plastic-film speakers. As with other planar-speaker designs, such as electrostatic panels, the P&I plastic-film speakers can't reproduce frequencies below 50 Hz, but they are said to match the sonic performance of conventional speakers above 300 Hz.

There are many applications for plastic-film speakers. They can be made transparent and placed in front of computer monitors or TV screens; they can even be made into banners or kites with graphics or text printed on them. The flexible plastic film can be shaped and molded as needed; for example, it can be formed into a cylinder with a very uniform radiation pattern (see Fig. 1), allowing it to be incorporated into a light sconce and other architectural design elements.

Of most interest to electronic musicians is how this technology might be applied to studio monitors. One of P&I's prototypes is a speaker that has a conventional woofer and a semicylindrical PVDF midrange/tweeter (see Fig. 2). Given the uniform radiation pattern of a cylindrically shaped film driver, such a design could allow the mids and highs to be evenly dispersed throughout the vertical axis. Alternatively, the PVDF driver could be mounted the other way to disperse the sound horizontally.

Koh anticipates that plastic-film speakers could be seen in commercial products as early as this year. I look forward to seeing how this technology might be applied to benefit electronic musicians everywhere. EM

New

bigfish audio
professional sound libraries

Platinum Essentials 2

$69.95
WAV/REX/Apple Loops

We snatched Keith outta Snoop's studio to finish this sequel. Loaded with more Hip Hop and R&B construction kits, "Platinum Essentials 2" contains all the sounds you've been askin' for, completely broken down. From bangin' beats and nasty bass samples to wicked synth and vicious sounds, nobody's puttin' it down west coast style like Clizark.

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Neo Soul 2

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WAV/REX/Apple Loops

Sultry producers/songwriters Josquin des Pres and Bernard Tortelli deliver a sequel that rivals the original. You can't help but feel G'd up and Classy with these sensual grooves. With drumloops, guitar, bass, synth, piano, percussion, organ, strings, horns and more, these construction kits give you the perfect blend of 70's soul, New millennium Hip Hop, Jazz and R&B.

Nu Metal City

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WAV/REX/Apple Loops

A head-banging collection of construction kits covering a variety of styles, including Nu Metal, Metalcore, Death Metal, and Industrial Rock. Inspired by the music of such artists as Slipknot, Korn, Anthrax, and Nine Inch Nails, *NU METAL CITY* features live and programmed drums,screamin' guitars, electric bass, synths, turntable FX, and atmospheres. If you're looking for butt-kicking loops with a hardcore attitude, welcome to *NU METAL CITY*.

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Off the Hook Hip Hop: Dirty South

$99.95
WAV/REX/Apple Loops

Bling Bling and country grammar... this is as dirty as it gets. From the legendary "Off the Hook" series, these construction kits flow with some serious southern Hip Hop flavor. 2116 loops and sounds including drums, synths, electric bass, organs, guitars, Rhodes, strings, percussion and more. Plus a ton of bonus material! With enough crunk juice to fill your pimp cup twice, you can ride with these tracks from St. Louis to New Orleans and back.

Chill:

Downtempo Loops and Beats

$99.95
WAV/REX/Apple Loops

This loungin' set of construction kits draw from jazz, hip hop, dub, funk, soul, ambient, and pop to create a whole new downtempo experience. Drums, guitars, keys, electric and acoustic bass, flutes and more gel somewhere between 60 and 110 bpm. Sometimes a little jazzy, sometimes a little old school, but always the perfect combination of reclined head bobbin' lazy pleasure.

Raging Guitars

$299.95
Kontakt Player/Plug-in
(VSTi/RTAS/AU/DxI2)

Virtual Instrument

This is the guitar onslaught you've been waiting for! A guitar virtual instrument done right, with over 11 gigs of multiple amps and levels of distortion, mono and stereo files. Loops that can be time stretched to match the tempo of your tune automatically, Sustained notes, chugs, hammer-ons, bends and lots more, so you can create your own guitar parts.

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Topographic Tracks

By Pete Prown

The Tangent's remotely recorded prog-rock opus.

Throughout sessions for the Tangent's latest album, singer-keyboardist Andy Tillison kept the Yes classic *Tales from Topographic Oceans* (Atlantic, 1973) right by his mixer. "I didn't want to copy it, but I frequently put it on for inspiration," says Tillison. "I wanted to make a record with the same kind of depth. So while *A Place in the Queue* still sounds like a Tangent piece—with Moogs, jazz piano, sax, and rock grooves—we tried to think outside the box, just like Yes did in 1973."

The Tangent is part of the renaissance of progressive rock, alongside favorites such as the Flower Kings, Porcupine Tree, and Marillion. *A Place in the Queue* (Inside Out Music, 2006), the band's third CD, was recorded in several home-studio locations, including the south of France, the United Kingdom, and Sweden. Though the finished work is seamless, the group had some studio challenges caused by the fact that its members were living in different countries during recording. Tillison says, "There were a lot of logistics to overcome, like getting musicians from one place to another or sending CDs and hard disks around in the mail. With our first album, the band members had literally never met each other face-to-face!

"But the Internet really brings this band together. For example, bassist Jonas Reingold will cut a part in Sweden [he uses Apple Logic Pro software], and 15 minutes later I've inserted his take into the mix and am listening to it here in France. Or Guy Manning [acoustic guitar, vocals] and I might use a VNC [a remote-desktop-control system] so we can work on a mix together and simultaneously alter onscreen parameters with our mice—and he's in England. As progressive musicians, we are seizing upon this technology, just as our forefathers seized upon the synthesizer and Mellotron."

COURTESY THE TANGENT

A Place in the Queue/the Tangent

RIFFS

The Tangent

Home base: Aveyron (France), Leeds (United Kingdom), Malmo (Sweden)

Sequencers of choice: Apple Logic Pro, Steinberg Cubase SX

Favorite mic: Studio Electronics SE1

Web site: www.thetangent.org

Both Tillison and Manning use Steinberg Cubase SX, which they run on PCs. Tillison says, "We tracked all the keyboards and most of the acoustic guitars, vocals, and Theo Travis's sax and flute through a Studiomaster 16:4:2 mixer and an M-Audio Delta 44 interface. I've been using Steinberg software for 20 years, and Cubase still offers the best solution for keyboard players. For us, the MIDI part is so important, and I've always found Steinberg's implementation of that to be superb."

Tillison shares keyboard duties with pianist Sam Baine, and the two use the best of vintage hardware and digital software in their rigs. "A '72 Minimoog is my pride and joy—you can hear it especially on the title track and 'In Earnest,'" Tillison says. "I also have a newer Minimoog Voyager which, unlike its predecessor, actually stays in tune. All the organ sounds on *Queue* are created using the Native Instruments B4 plug-in, controlled by the drawbars and keys of my old Roland VK7 organ. Most acoustic piano parts are from Steinberg's The Grand plug-in."

Tillison didn't use dedicated studio monitors when mixing. "I mix on Wharfedale hi-fi speakers—I don't own any professional monitors at all," he says. "Most people listen on regular stereo speakers, so that's what I mix on. I also use techniques such as old-fashioned stereo panning, instead of any elaborate stereo-doubling effects. That helped us make, in my opinion, prog rock that sounds punchy and direct. But in the end, it's more about the tracks you lay down, rather than how you mix them." EM

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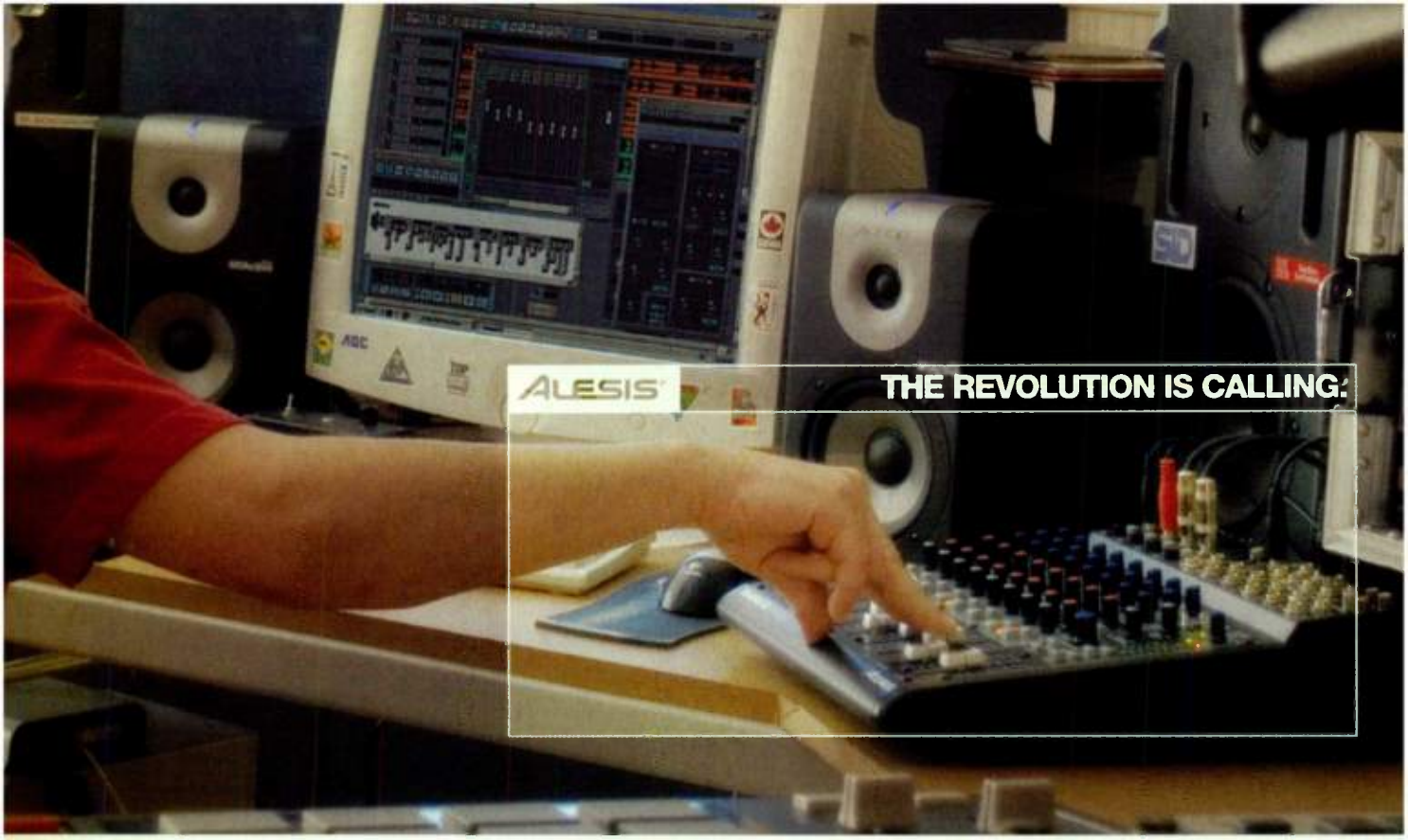

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Lord of the Ringtones

By Luke Holden

A guide to programming ringtones for fun and profit.

Since the late 1990s, the emergence of the mobile-telephone ringtone market has brought numerous employment opportunities to musicians, programmers, and composers familiar with MIDI sequencing and audio editing. Ringtones were first introduced as a consumer product in 1998 and shortly thereafter became a common feature on mobile phones. Early ringtones were monophonic, typically playing the melody of a popular tune. But soon mobile-phone manufacturers advanced the audio capabilities of their products, and the polyphonic ringtone was born in 1999.

Companies such as Beatnik, Faith, Nokia, and Yamaha began to create mobile MIDI solutions that allowed for more-elaborate-sounding ringtones consisting of an ever-expanding polyphonic note count (see the sidebar "Contact Information" for a list of companies mentioned in this article). Additional features followed, including support for Pitch Bend and controller information such as modulation, as well as text, vibration and LED synchronization, graphics, and real audio.

New mobile handsets arrive on the market at the rate of dozens per month, yet developer support is notoriously poor in this industry, and handset spec sheets are often horribly inaccurate. As a result, understanding the different formats required for different phones and manufacturers can be an onerous task. In this article, I will give an overview of the ringtone market and offer some suggestions on how you can get your foot in the door. And in case you get the call to provide ringtones for a client, I will provide some programming tips that you should find useful.

Here's the good news: nearly all ringtones are created by freelance programmers. Rarely will companies hire programmers as full-time employees. This has several obvious benefits, such as

allowing the programmer to work from home, around his or her own schedule. Most interaction is through email, and files are usually transferred via FTP. However, it's rare for a full-time mobile-media programmer to make a living solely on ringtone programming. In most cases, film and TV composers, game-audio programmers, and audio editors will create ringtones to supplement their existing income. And keep in mind that mobile-content providers typically aren't interested in original music, so your composition skills will not be a big asset.

From the Top

Programming polyphonic ringtones today is similar to designing game audio in the early days of PCs. Due to the CPU, memory, and bandwidth limitations of modern handsets, ringtone files must be kept clean, efficient, and small. Moreover, no single standard format exists for ringtone file distribution. The most popular polyphonic formats in the United States today are Qualcomm's CMX (Compact Media Extensions), Yamaha's SMAF (Synthetic music Mobile Application Format), and Nokia's SP-MIDI (Scalable Polyphony). (See the table "Format Free-for-All" for details on the formats mentioned here.) Though nearly all modern mobile phones can use Standard MIDI Files, playback will vary due to polyphony, MIDI-controller, and file-size restrictions. The volume levels of General MIDI voices in mobile phones also vary considerably across formats.

Phones created by different manufacturers have vastly different audio capabilities as well. This has a direct and crucial impact on any particular model's ability to produce a convincing ringtone. Therefore, a content provider might need to distribute optimized subformats, which have different specs for file size, polyphony count, audio capabilities, and so on. For example, there are dozens of SMAF handset models in the U.S. market, and exploiting their individual potential would require several different versions of an SMAF ringtone. Short of creating multiple versions, a programmer might simply aim for the lowest common denominator.

FIG. 1: Nokia Sound Converter is used to convert a Standard MIDI File into a file in Nokia's SP-MIDI format. It can also determine which channels of the file will be dropped if a phone can't support the file's full polyphonic note count.

To keep up with the industry, a potential ringtone programmer must have a working knowledge of the various formats, a familiarity with the variety of phones on the market, and an excruciating attention to minor details when creating MIDI files—things that are not always necessary when working as a gigging player, composer, or producer. Companies are aware of these issues, and it is common for a programmer to receive some training from the client. Moreover, there are a number

FORMAT FREE-FOR-ALL

Many consumers wonder why their ringtones don't sound as good as their friends'. One reason is that the quality of a phone's tone generator can vary greatly from handset to handset. In the United States, this is hugely dependent on what type of network the phone is supported by. American wireless networks generally use one of two underlying technologies: CDMA (Code Division Multiple Access) and GSM (Global System for Mobile communications). Programming polyphonic ringtones is substantially different when dealing with each technology.

The table below shows the main specs for each of the major polyphonic formats currently in use in the United States.

FORMAT	DEVELOPER	NETWORK TYPE	MAIN PROVIDERS	HANDSETS	POLYPHONIC NOTE LIMIT	AUDIO SUPPORT	IMAGE SUPPORT	TEXT SUPPORT	VIBRATION TRACK	LED
CMX	Qualcomm	CDMA	Alltel, Sprint, Verizon	virtually all	72	up to 32 kHz	yes	yes	yes	yes
SMAF	Yamaha	GSM	AT&T, Cingular, T-Mobile	most LG and Samsung	72	up to 16 kHz (in MA-3)	yes	yes	yes	yes
SP-MIDI	Nokia	GSM	AT&T, Cingular, T-Mobile	all Nokia	24	no	no	no	yes	no
Standard MIDI File (SMF)	n/a	GSM	AT&T, Cingular, T-Mobile	most Motorola, Siemens, and Sony Ericsson	16	no	no	no	no	no

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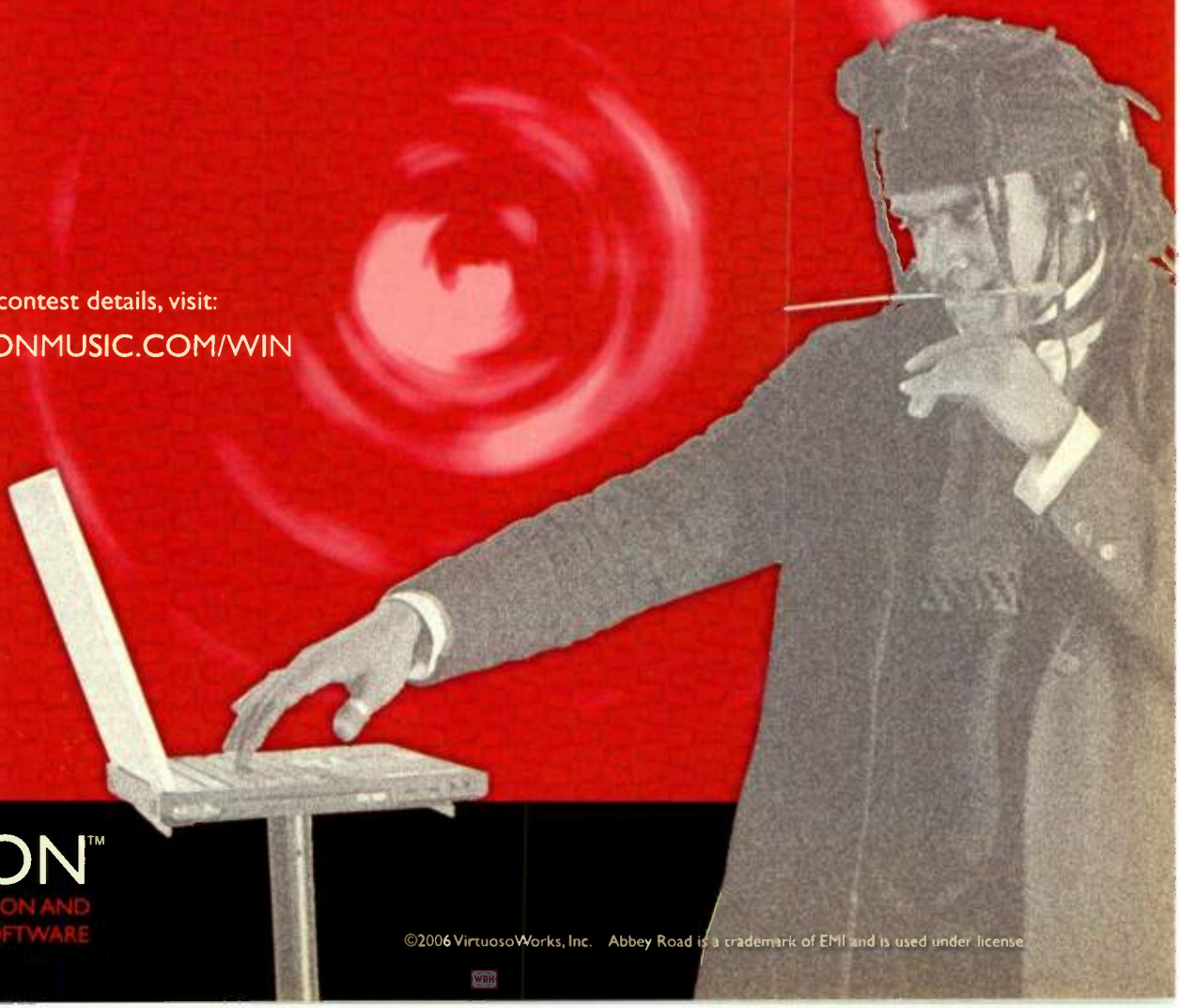

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of readily available authoring tools—for example, Nokia Sound Converter, which is part of Nokia PC Suite (www.nokia.com/support/phones/6230)—to help the ringtone programmer on his or her way (see Fig. 1).

Note that the myriad free-ringtone sites found on the Net usually offer just a single, unoptimized MIDI file, regardless of the handset to which it's being ported. (It's doubtful that many of the companies offering such files have cleared the rights for their titles.) This is one of the fundamental differences between those ringtones and ringtones purchased from major ringtone companies like Modtones, Yamaha, and Zingy.

Getting in the Game

To get started as a ringtone programmer, try contacting the content providers listed in the sidebar "Contact Information" (or feel free to email me at lholden@yamaha.com). These are the major parent companies in the ringtone business, and most distribute via a variety of consumer mobile applications.

If you're asked to submit work for evaluation, you'll most likely receive several clips to transcribe. The clips will probably vary drastically in style to test your breadth of musical diversity. Often the content provider will ask for several versions based on differing polyphonic note count. These will show the content provider your musical intuition. The accuracy of the transcription is what's most important to the content company. A "generic" mix (that is, one not fully optimized for a specific format) is acceptable, but incorrectly transcribed notes and rhythms will disqualify you for any ringtone programming work.

CONTACT INFORMATION

Content Providers

9 Squared www.9squared.com
AG Interactive www.interactive.ag.com
Electronic Arts Mobile Games www.ea-mobile.com
Faith, Inc. www.faith-inc.com/service/mobile.html
Hands-On Mobile (formerly mForma) www.mforma.com
InfoSpace, Inc. www.infospaceinc.com
Moderati www.moderati.com
Modtones www.modtones.com
Sony BMG Music Entertainment www.sonymusicmobile.com
Tone Player www.widerthan.com/americas
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Zingy www.zingy.com

Developers' Resources

Beatnik www.beatnik.com
Nokia www.forum.nokia.com/main
Qualcomm https://cmx.qualcomm.com
Yamaha www.smaf-yahama.com

FIG. 2: Because it contains the same sound chip used in many phones, the Yamaha MA-3 synthesizer is useful for previewing your ringtones as they will sound on the actual handset hardware.

Always make sure your MIDI file is clean and well organized. Those evaluating your work will look for orderly arrangements and clearly marked tracks free of any unnecessary controller data. Content providers want their ringtones created by programmers who are quick and reliable. Ensuring that your initial transcriptions are accurate and your MIDI files clean is the best way to land a ringtone programming position.

Start Your Engines

To get started programming, you'll need a sequencer that can export MIDI format 0 and 1 files—most any will do. Be careful that your file doesn't contain any stray SysEx messages, inserted either by you or by your software; these messages can cause the phone to begin playing erroneous notes, fail to loop, or even restart. The next requirement is a good set of ears—every polyphonic ringtone starts with an accurate transcription of the musical piece you're programming. Ringtones vary from hip-hop to classical to TV themes and practically every genre in between, so you should also be familiar with a variety of musical styles.

Initial transcriptions should contain only basic MIDI messages such as Master Volume and General MIDI System On. Channel messages should be limited to Program Change, Pan, and Volume. These messages should be consistent across tracks and inserted at identical times in the header of the track. Although modulation and Pitch Bend data is generally acceptable, some aging handsets have issues with these message types.

One of the most difficult parts of producing ringtones is that direct monitoring is not possible: what end users will hear on their handsets is not likely to be as robust as the sounds you get while working with your native sequencer in your studio. Though emulators exist for SP-MIDI and CMX formats, they are software based and not exactly trustworthy. For this reason, try to monitor using a poor-quality tone generator or a Yamaha MA-3 box (see Fig. 2). The MA-3 contains the actual MA chip

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that is commonly found in GSM network phones made by LG, Samsung, and other manufacturers. In fact, the MA series of chips, which includes the MA-2, 3, 5, and now 7, is based on scaled-down versions of the Yamaha FB-01 4-op chip. (For more information on the MA chips and SMAF creation, visit www.smaf-yamaha.com.)

You should also monitor on low-cost loudspeakers to better emulate the phone's frequency response. If you do use your main monitors, use EQ to cut everything below 300 Hz sharply, and forget about anything above 10 kHz.

Just for Effect

Another difficulty for ringtone programmers is the near-total absence of DSP effects to enhance the inherently weak GM patches found on cell-phone synthesizers. Reverb, chorus, or delay must be added manually. For instance, you can take a guitar melody line, copy it to a second channel, and detune the two using Pitch Bend to create a chorus effect. You'll get a nice chorus by tuning one of the tracks up 200 cents and the other down 200 cents. Expand those values to create more-intense chorus. Practically any and all patches, aside from acoustic pianos, can benefit from this technique.

To add reverb to a melody line, copy the channel's data to a blank channel, then time-shift the second line by a 32nd note. Reduce the second channel's volume to about half that of the original (these values will vary depending on the specs of the format you are programming for). Repeat this process for more reverb time. Using a breathy patch like a flute or ocarina for the time-shifted channels can add the airiness typical of reverb tails.

You can create delay effects in a similar manner. Again, copy the original track to a blank channel. Then time-shift the second line by an eighth note and reduce the second channel's volume to about half that of the original. Repeat this process multiple times to increase the "feedback" of the delay. Unlike for reverb, use the same patch for all delayed copies.

To enhance the sound of a ringtone melody line, you might need up to five tracks. Start with a saw lead (GM patch 82) or a harmonica (GM patch 23) as the melody's main patch. Then create a chorus effect using the technique described earlier. Next, double the melody one octave down to give it more body. Finally, add at least one delay channel to give the line depth and space.

Obviously, this creative method of DSP simulation will eat up channels quickly, and given that you have only 16 channels to work with, you need to decide which parts receive this special treatment. Certain formats, including SMAF and SP-MIDI, may require

a dedicated vibration track that will use up another channel. This track is used to synchronize kick drums or bass notes to the vibrating motor of the device, which can help the user "hear" bass that the phone can't actually reproduce. By inserting specific Bank and Program Change messages in the header of those tracks (MSB 121, LSB 6, and GM patch 125 for SP-MIDI format), the Note On and Note Off commands become routed not to play a sample at a specific pitch, but to trigger the device's motor to vibrate the handset. The vibration track's note Velocities and Channel Volume should all be set to minimum. Insert the note D2 in your vibration track to trigger the device's motor. (See the developer's supplementary documentation on the company's download pages.)

Keep in mind that in nearly all cases, transcription is only half the task for ringtone programming. Creating the files in the formats that the myriad modern handsets require is also typically done by the programmer. To stay current with new phones, services, and trends, bookmark the Phone Scoop site (www.phonescoop.com) and check back often.

Keeping It Real

In the United States, recent technological advances have brought increased network speeds and widespread availability of handsets that are real tone capable. A great example of this caliber of handset is Verizon's LG VX8100 (see Fig. 3). This popular multimedia phone has MP3- and CMX-format capabilities in addition to connection speeds fast enough for video downloads and full-track audio content. Thanks to its miniSD card slot, users can store a large amount of data, thrusting phones such as this into competition with standalone MP3 players and iPods.

FIG. 3: Verizon's LG VX8100 is a modern multimedia phone that supports MP3 and CMX formats in addition to video downloads.

The predominant format for audio playback on mobile devices is, not surprisingly, MP3. Many handset manufacturers now include MP3 support for their newer models, including Motorola, nearly all of whose mobile devices have adopted MP3. Other phones will play audio via SMAF or CMX format.

If you've ever downloaded real tones, you've probably noticed that the clips are quite short, rarely exceeding 12 seconds or so. This is because many handsets still have limits on the size of files they can store, and download speeds are not universal across handsets and networks, even on the same carrier.

Since real-tone production work is usually handled in-house by content providers, real tones represent the dark, ominous cloud preparing to ruin ringtone programmers' sunny polyphonic day.

Mobile Content's Future

Let's face it: polyphonic ringtones have hit their peak in the United States, and the initial boom and demand for programmers is gone. But even though demand has diminished, there still is money to be made in the polyphonic market. Smaller carriers will be slow to support newer, more expensive handsets, and for the foreseeable future, "free" phones given to consumers in exchange for contractual agreements will probably be limited to polyphonic ringtones.

One can look to Japan and Korea to see the future of ringtones. Current offerings in those countries consist of full-track downloads, handsets with incredible stereo imaging, and polyphonic limits approaching those of professional synthesizers. The 3G networks in place allow for quick data retrieval and robust multimedia downloads.

Games are another widely adopted form of mobile entertainment, and seem to be the perfect relief for many who endure bus and subway commutes. Mobile-game companies also have a need for composers, sound designers, and mobile-device contractors. Electronic Arts Mobile Games and Hands-On Mobile (formerly mForma) are two leading mobile-game developers in the United States.

Other areas of mobile-content development include GPS/location-based services, video downloads, up-to-the-minute news and weather info, adult content, and many of your favorite TV shows condensed for mobile viewing. Channels such as Verizon's V-Cast will spearhead this new wave of mobile content.

Specialty niche applications are also beginning to appear. One example is Yamaha's Musician's Companion, which replaces your metronome, pitch pipe, and chord reference book with an all-in-one mobile application for your cell phone (see www.yamaha-wireless.com for more information). Some handsets are even equipped with hardware such as FM tuners and medical devices such as diabetes testers.

In Japan, consumers have begun paying for a subway ride or a bottle of soda at a convenience store with a swipe of their phone past a scanner. Those attending a sporting event or concert can have a virtual ticket stored in their phone and available at the wave of their hand. From such examples, you can see that the cell phone is becoming central to the Japanese lifestyle. Some of these services will inevitably appear in America in the years to come, but until then, ringtones will continue to dominate the mobile-content market. EM

Luke Holden is product manager and marketing manager of the Wireless Content Department of Yamaha Corporation of America. He also runs Moebius Sound and Recording (www.moebiusrecording.com), a Southern California–based recording and production facility.

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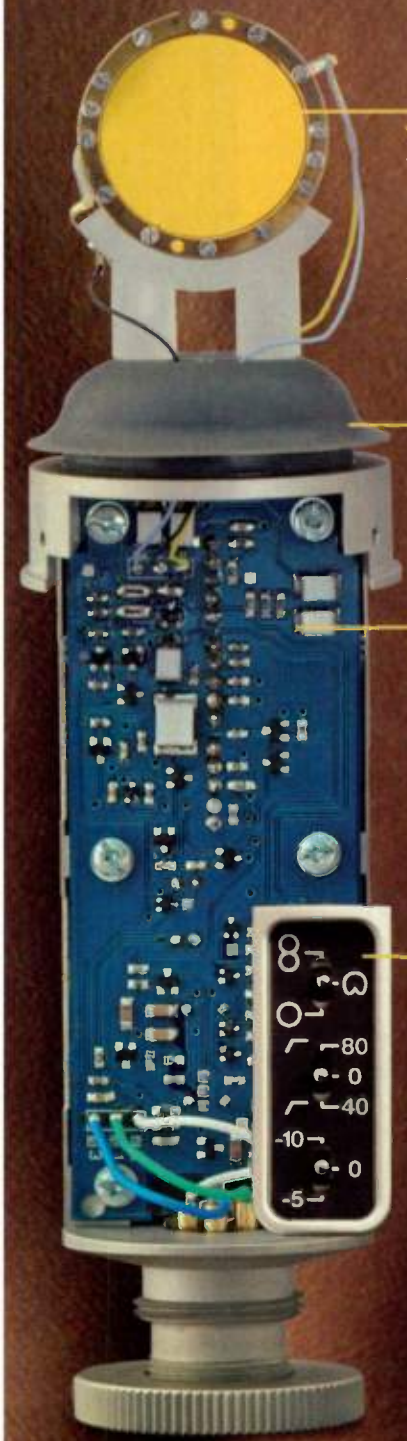

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Vocal Magic

By Mike Levine

It would be natural to assume that as digital audio technology has become more sophisticated, the job of the engineer and producer has gotten easier with regard to editing vocals. After all, today's digital audio sequencers and editors give you precision that even the most skilled analog engineer—armed with a razor blade and splicing block—could only dream about. But inevitably, as the tools get better the bar gets raised, and now producers and engineers are expected to work miracles—correcting problems with pitch, dynamics, and even phrasing. "Fixing it in the mix" has become a complicated task indeed.

To give you an in-the-trenches perspective on some key editing issues and techniques for both lead and background vocals, I spoke with six successful producer-engineers who shared their expertise on such subjects as making a comp track, controlling dynamics, editing out unwanted noises, EQ'ing vocals, and much more.

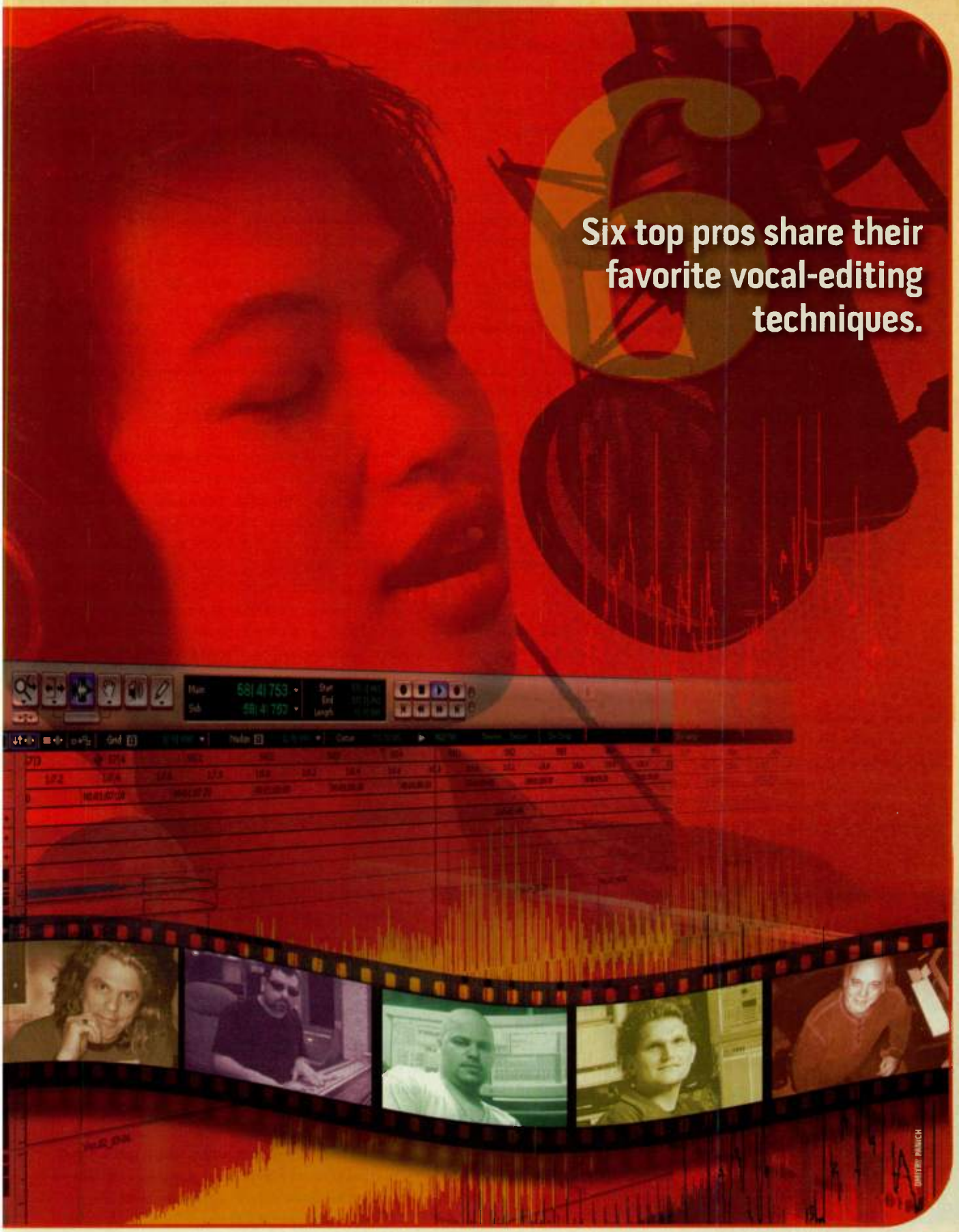
Steve Addabbo is the owner of Shelter Island Sound (www.shelterislandsound.com) in New York City and has produced numerous artists, including Suzanne Vega and Shawn Colvin. He recently finished working on a CD for newcomer Sonya Kitchell that was released on the Velour/Starbucks label.

Bob Power (www.bobpower.com) is a producer, engineer, mixer, and songwriter who has worked for a huge list of artists, including Erykah Badu, D'Angelo, Me'shell N'degeOcello, Citizen Cope, the Roots, and Chaka Khan. He's currently working with singer-songwriter Andrea Wittgens.

Rail Jon Rogut (http://railjonrogut.com) recently engineered Ry Cooder's *Chavez Ravine* CD (Nonesuch, 2005). Other clients have included such artists as Aaron Neville, Mark Lanegan, and the String Cheese Incident. He just completed work on an album with producer Mike Clink and artist Sarah Kelly.

Johnny "Juice" Rosado (www.publicenemy.com) has produced and engineered numerous artists, including Slick Rick, C&C Music Factory, Mandrill, and Mavis Staples. He's the regular producer-engineer for Public Enemy.

Rick Sheppard has been the longtime engineer for R&B megaproducer Dallas Austin. Sheppard has engineered tracks for Michael Jackson, Madonna, Bjork, Macy Gray, Mick Jagger, Gwen Stefani, and Natalie Cole, to name a few.



LEFT TO RIGHT: STEVE ADDABBO (BY LINDSAY MARCUS), BOB POWER, RAIL JON ROGUT (BY ADAM FULLER), JOHNNY "JUICE" ROSADO, RICK SHEPPARD (BY MICHELLE LAPIC), ED STASIUM (BY AMY HARTMAN)

Ed Stasium's (www.edstasium.com) credit list includes Mick Jagger, Talking Heads, the Ramones, Living Colour, the Smithereens, the Reverend Horton Heat, and many more. When I spoke to him he was finishing up a project for an up-and-coming band called Lourds.

Leveling Out

Before getting into the nuts and bolts of vocal editing, it's worth noting that editing is a lot easier if the vocal levels are properly recorded. In addition to the crucial issues of vocal performance, mic and preamp choice, and mic placement—which are beyond the breadth of this story—it's critical that your vocal tracks be recorded hot enough to give you full fidelity, but not so hot that you get distortion.

Several of the interviewees recommended that if you're recording at 24-bit resolution, which has more headroom than 16-bit, it's prudent to shoot for levels that average around –5 dB. They suggested that if the song has wide dynamic swings, you might want to manually ride the mic-preamp gain during recording. There was general agreement that lightly compressing to disk (with compression ratios typically between 2:1 and 4:1) is also a good idea. (For a more in-depth discussion from the six engineers on getting good vocal levels, see **Web Clip 1**.)

Get It Together

In most cases, the end result of a lead-vocal session is an abundance of takes that need to be sorted through, cut up, and reassembled into a final composite or "comp" take. "To me, one of the marks of a really good producer is how they comp," says Power. "Sometimes a good producer can get a really compelling performance out of just an okay singer, if the producer's a good comper."

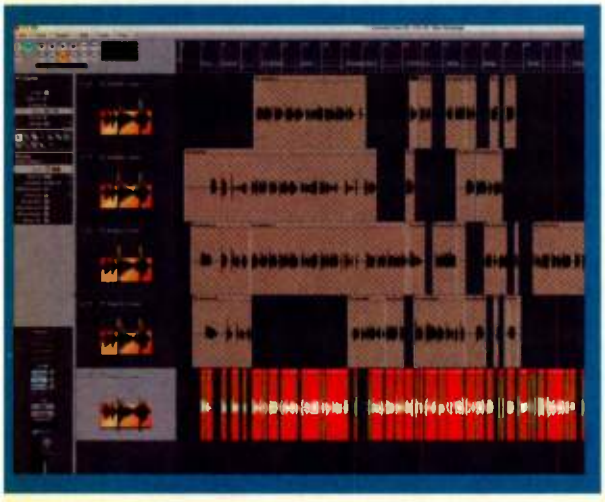


FIG. 1: This screen shot from a Natalie Cole session shows Sheppard's method for vocal comping in Logic Pro. The vocal-take tracks are assigned to the same audio voice, cut up by phrase, and auditioned one line at a time. The keepers are pasted into the comp track at the bottom.

Before you start recording vocals, it's important to create a system to keep track of the quality of the singer's performance on a line-by-line (or sometimes even word-by-word) basis for each take. Some use a *comp sheet*, which is a piece of graph paper with the lyrics written out on it. There are vertical numbered columns after each line where you can check off the "keepers" or grade the performance of each line on each take, or whatever works best for you.

Instead of a comp sheet, some people take notes on a lyric sheet or even on blank paper. "I take physical notes on a legal pad with each Playlist number and what I thought of each performance," says Rogut. (In Digidesign Pro Tools, a track can have unlimited Playlists, which are nested tracks within one track, only one of which can play back at a time.) Rosado, a Cakewalk Sonar user, says, "You can write notes within the program, so I do that."

However you do it, you'll save yourself time in the comping process by taking notes as the singer is recording. Just don't be too obvious about it, because no singer will like the feeling of being "graded" as they work. "I keep notes on the comp sheet," says Power, "but not in a way that will be either demoralizing or obtrusive to the artist."

The How and Why

So what are some of the actual comping methods these engineers use? Their procedures are dictated in part by the architecture of the different sequencers they use.

Rosado uses Sonar's Layers feature for his vocal recording and comping. It allows multiple takes to be recorded into one track; they show up as layers that can be muted, unmated, and freely manipulated. "I just split the tracks where I want them to be, and I mute the clips [regions] that I don't need," he says. "Eventually, after I find the parts I really like, I bounce it down to one clip."

Stasium takes advantage of Pro Tools' Playlist feature. Each vocal take gets recorded to its own Playlist. When it comes time for comping, each Playlist is sequentially numbered (which Pro Tools does automatically) and easy to audition and copy and paste from. "I remember the days of doing vocal composites with multitrack tapes and just rewinding and switching between the tracks," says Stasium. "Sometimes it would take up to two days to make a vocal comp if you had like 10 or 12 tracks of vocals."

Sheppard and Power are both Apple Logic Pro users (both use Logic on top of Digidesign TDM hardware, and Power also uses Pro Tools), and they take advantage of that program's feature for recording multiple takes that are assigned to the same audio voice. When auditioning takes, only one can play back at a time, which streamlines the auditioning process. "First, I cut up the vocals in phrases," says Sheppard, "then I go through each take by unmuting one vocal line at a time. It goes really fast, and it's nice to see everything at once" (see **Fig. 1**).

"I set up a cycle, a loop, which is really easy and handy," explains Power, "and then open up one channel, listen to it, open up take 2, listen to it, take 3, take 4, take 5, take 6. And as that goes on, when I hear something I really like, I actually cut it front and back, on the source track."

If it's a complex comp, Power will often color code a

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region after he separates it. "I have a color coding scheme based on how dynamite the little segments are," he says. "Red means really good and really exciting. Orange means pretty good and pretty exciting. Green means it's acceptable. Blue means maybe if we really need it, etc. But everybody will come up with their own. The advantage to color coding is that you can put together a comp pretty quickly, and then later on, if you're not sure about a line or a word, you can very quickly look up and say, 'Oh wow, there's another orange one up there'" (see Fig. 2).

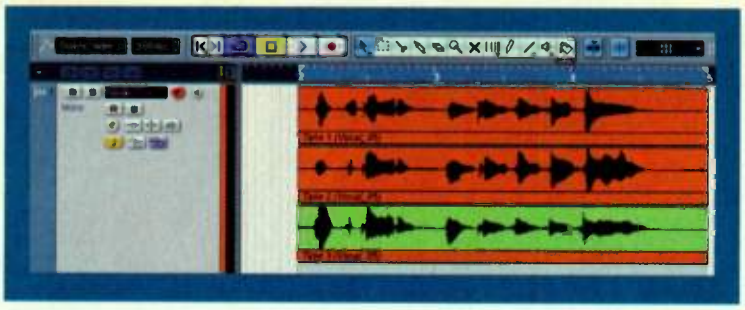

FIG. 3: Many sequencers, including Steinberg Cubase SX (pictured here), allow you to record vocal takes as nested tracks within a single track. Such features facilitate vocal comping by making it easy to manipulate and audition the various takes.

Addabbo takes a different approach. "I've just always been a kind of real-time vocal producer," he explains, "where I'm building a performance with the artist in a live setting." He'll start by having the singer do a few complete takes that are close in feel and execution to what he's looking for. Then, instead of continuing to record a series of complete takes, he'll use those first performances as a starting point and will begin punching in bits and pieces live with the singer. (He keeps all the lines he replaces, in case he changes his mind later.)

Here's an example of his procedure: say the vocalist sings a good first verse and chorus, but then the second verse is not spectacular. Addabbo would duplicate the Playlist, creating a copy of what was already recorded. Next, the singer would resing the second verse, with Addabbo punching into the duplicated Playlist. The singer would continue on until they reached another part Addabbo was not sure of, and then he'd duplicate the Playlist again, and punch from that spot. "It doesn't mean that I'm not going to have a lot of choices later if I decide I don't like something," says Addabbo. "But at least I'm getting a cohesive performance that both the singer and I are vibing to together."

FIG. 2: One way to help keep track of the takes you're comping is to color code them by quality. If your sequencer allows for color coding of individual regions (such as in this example from Logic Pro), you can even color code individual phrases.

Although Addabbo does these cumulative comps in Pro Tools, his method could be used in any digital audio sequencer. It's easiest if your sequencer has a nested track feature such as the Takes in MOTU Digital Performer, the Layers in Sonar, or the Lanes in Steinberg Cubase SX (see Fig. 3). But even if it doesn't, you can duplicate your lead vocal track before each punch, and punch into that new track.

Rogut also likes to use Playlists when recording vocal takes in Pro Tools. But afterward he splits them out onto five separate tracks and also creates a new empty track called Vocal Comp above the source tracks. He'll copy and paste the best sections from the source tracks to the Vocal Comp track, keeping their region names intact to make it easy to see which vocal track they came from.

Rogut then assigns the source tracks and the comp track to the same audio voice, and uses Pro Tools | HD's voice-stealing feature—which gives playback priority to the top voice (the Mute Frees Assigned Voice feature should be enabled for this)—allowing him to easily audition switches between different lines or words. By leaving a "hole" in the comp track for the line he's auditioning (see Fig. 4), whichever of the vocal-take tracks he unmutes below the comp track will play during that cutout section.

Fading Away

When the sections you're comping together have natural pauses between them, then the process is mainly just one of identifying the pieces you want to use and assembling them in a new track. But what if you have to graft the first part of one line with the second of another? Or substitute one word at the end of a line? Or even put together a word using the first syllable or sound from one word and the second from another?

When editing two adjoining sections together, first experiment with moving the transition point slightly. A little more to the left or right can make a big difference in the smoothness of the edit.

If you're having trouble finding a natural-sounding edit point, zoom down to the sample level and make the cut at the zero-crossing (see Fig. 5), which is where the wave crosses the centerline. This will assure that the amplitude will be the same on either side of the edit and will help eliminate clicks and pops. Whenever possible, you should make your edits at zero-crossings.

Once you've found the best possible edit point, you'll want to utilize your sequencer's crossfade function to

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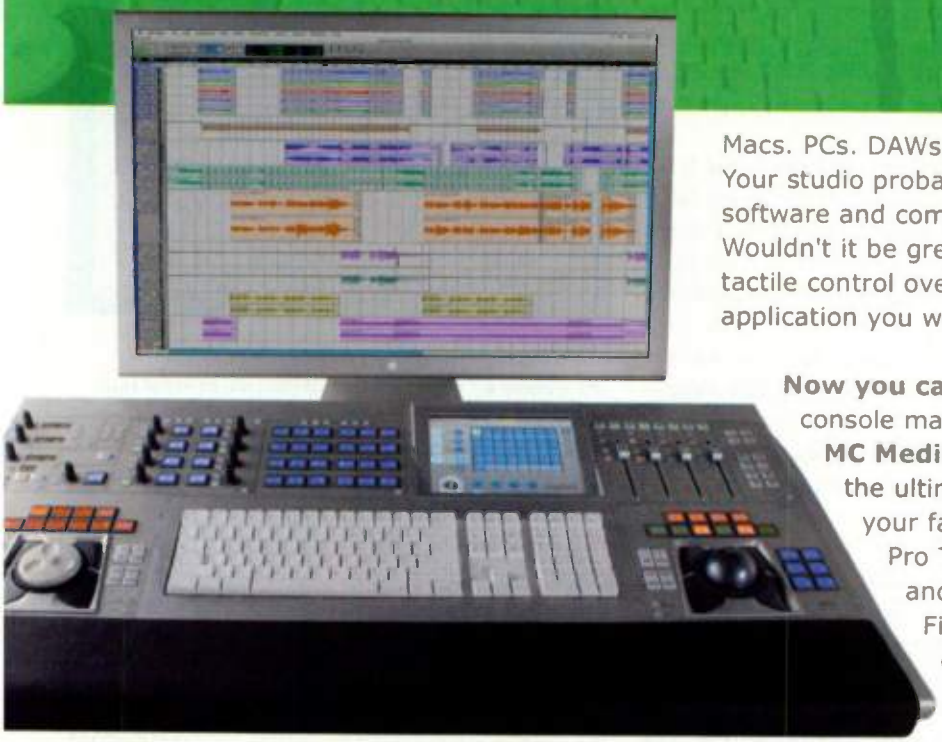

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make the transition even more seamless. A crossfade gives you a smooth transition from one section to another, rather than an abrupt one. If properly set—you have control over the length and shape—a crossfade can make the edit point much less noticeable. Some digital audio sequencers can be set to automatically generate crossfades at every edit point.

The crossfade's length is an important factor. If the fade is too long, you'll hear unnatural overlapping. Short fades (set when you're zoomed in pretty close on the waveform) are often quite effective, although one that's too short might not have enough of a smoothing effect.

"When you're going from take to take, with Pro Tools or any other DAW," says Stasium, "sometimes you're going to get a little click or a pop or some little thing—especially if you're in a word. You've got to find the right crossfade. I crossfade every piece, no matter what. I try to make it sound as natural as I possibly can."

Grafting and Crafting

I asked the experts if they often have to splice words together from syllables, letter sounds, or both. "You can get as crazy as you want," says Addabbo, "and I've certainly been there in terms of chopping up words." However, the overall consensus was that constructing words through editing was not something to be done unless necessary.

"Have I had to do that? Yes," says Sheppard. "Do I want to do that? No. Hopefully, you can get a word out of someone, but there have been times when it was a great performance, but at the end of the first time she said 'this' and the next time she said 'thith' because of whatever. So I can take that *ess* and throw it on the end of the next one and you won't know the difference."

Power offered a couple of additional techniques for when you just can't get your edit to sound natural. "I find a lot of times, if a crossfade doesn't work, try fading out on the left segment and fading in on the right. Many times that will fix what a crossfade can't." That technique is particularly easy to do in Logic, which allows you to type in a fade-in and fade-out value for each audio region, but can be accomplished in other digital audio sequencers or editors, too.

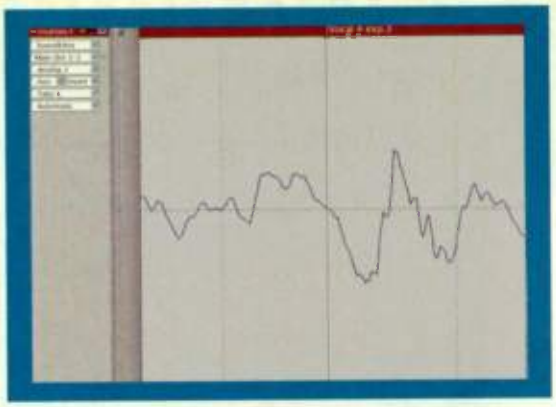

FIG. 5: Zooming to the sample level and making your edit at a zero-crossing will help assure that your edit will be free of clicks and pops.

Power's other edit-finessing trick starts with putting the two sections being edited together on separate tracks. "Then lengthen the left piece a little and then fade out on that, and lengthen the right piece a little and fade in on that, so they actually overlap just a bit," he suggests. "A lot of times you can make it work that way."

Once you've finished comping and crossfading, you're going to have a track with a lot of regions in it. At that point "you should consolidate the track," says Rogut, "so it's just one large audio file. This also allows you to remove the source tracks from the session to reduce its size."

FIG. 4: Rogut's method for auditioning tracks when comping uses Pro Tools|HD's Mute Frees Assigned Voice feature. In this example, when the transport gets to the "hole" in the Vox Comp track, the audio from the track below will play for that section.

Every Breath You Take

Dealing with breath sounds is another important issue in vocal editing. Although it's tempting to just remove them all, doing so can sound unnatural in many musical situations. "Breaths are a really important part of a vocal performance," says Addabbo. "I've had people go out and record breaths sometimes, if I don't have the right one *[laughs]*."

But you have to be careful with them. "I go in and I clean up each line," says Sheppard. "If breaths are needed or give the song a good feeling like it's real personal, then I'll just leave them, but every once in a while you get a singer gasping for air, and you don't want that." He says you should make your editing decisions on a breath-by-breath basis. "Does it enhance the track or does it sound like they're drowning?"

You have to be particularly careful of how compression affects the sounds of breaths. "Sometimes you

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hear stuff on the radio," says Addabbo, "and you can tell how squashed these vocals are, because the breaths are sometimes almost louder than the words. You can just see the meters moving on the compressor."

One way to deal with such problems is to use your sequencer's volume automation to bring the level of the breaths down to where they're not sticking out (see Fig. 6). "You're counterbalancing the tendency of the compressor to open back up when you're below the threshold," says Power.

Another issue with breaths is when two cross in the transitions between edited sections. In that scenario, you'll have to finesse the edit point to try to find the spot where the breath sounds natural. "You'll have to move the file a little bit left, a little bit right, to catch the breath properly," says Stasium.

Rogut finds that it generally sounds more natural to keep the breath from the right-hand (later) side of the edit. "Say there's a breath at the end of line 1 and another at the beginning of line 2. I would normally keep the breath of line 2 and cut the one from line 1."

Rogut says that if he's having a problem getting a breath to sound right around the edit point, he'll sometimes move the breath to its own region, and then time-stretch it to the length he needs to make it sound the most natural.

VOCAL-TRACK IMPROVEMENT TIPS

- After comping, smooth out vocal levels using automation or offline plug-ins. This will make mixing easier and allow vocals to be compressed more evenly, which will keep the tone more uniform.
- Use pitch correction sparingly, only correcting words or notes that are out of tune. Avoid correcting an entire track unless you're going for that "down-the-middle" sound.
- Consider cutting out low frequencies below the vocal's range to reduce muddiness in the mix.
- If the vocal track needs more air, try gently boosting the high frequencies (usually between 18 and 20 kHz).
- Get rid of sibilance using a de-esser plug-in or by lowering the volume of *ess* sounds with volume automation (or both).
- For clicks and plosive pops, try finding a word on another take to replace the offending one. If you can't, try zooming in and correcting with the pencil tool, or actually cutting out the offending sound and possibly time-stretching the file to fill the gap.
- If breaths sound unnaturally loud, reduce their volume with automation or cut them out where appropriate. Cut out breaths on background vocals. Be careful of your compressor's tendency to pull up breath sounds.

FIG. 6: One way to make breath sounds less obtrusive—without cutting them out entirely, which can sound unnatural—is to use volume automation.

Pitching Forward

The six pros generally agreed that pitch correction is a tool to be used only as needed. "I don't like to autotune the entire vocal," Addabbo says. "I'll go in and say, 'Okay, that word's a little flat.' I'll go in and bring up that one line, as opposed to just putting it on stun. It depends. If it's a real poppy, slick record, you may just want to let the thing run through it. But in general, I like to pick and choose what I autotune."

Addabbo uses Antares Auto-Tune for pitch correction. His standard method is to create a new track called the Fixer track. Onto it, he'll copy whatever sections of the vocal need correction. "Generally, what I'll try to do is to fix the stuff that really hurts my ear or bugs me," he says. "That could be just the end. Singers always tend to drop their pitch at the end of a line." He'll then instantiate Auto-Tune as an insert on the Fixer track, capture the track's contents into Auto-Tune's Graphical mode, and correct the problem areas using the various tools provided. After making sure that no clicks or pops have been created by the processing, Addabbo records the Fixer track with Auto-Tune back onto the original vocal track (on a duplicated Playlist).

Rosado has had a lot of success correcting pitch with Celemony Melodyne and lately with the new V-Vocal feature in Sonar 5 (see Fig. 7). He finds V-Vocal both convenient and powerful. "It does its correction with very few artifacts," he observes.

Sheppard often uses Wave Mechanics Pitch Doctor, a TDM plug-in, which has a manual pitch-correction mode that can be controlled with automation. If he only needs to correct a word here or there, however, he will typically use Auto-Tune. "Auto-Tune is great," he says, "as long as you use it only where necessary." Like Addabbo, Sheppard will make a new region out of the offending phrase, word, or syllable, apply correction (generally in Graphical mode), and then record it back onto the original track.

Power will sometimes use Auto-Tune when he needs mild fixes, but he's not comfortable with too much pitch correction. "Unfortunately, autotuning has become a

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part of what people expect to hear when they listen to a record," he says, "and not even in the Cher exaggerated sense. But when I listen to the radio, I can hear it a mile away. This is what modern records sound like. I hate it. I do love things in tune. Everyone who knows me knows that pitch and time are way up on the scale for me. But I really don't like that laserlike, right-in-the-middle intonation. It doesn't sound right to me."

In the Background

Background vocals present some different challenges to edit and mix compared with lead vocals. The ends of lines can be ragged when you have several people singing them, and problems like breaths and pitchiness can be exaggerated. Naturally, the best way to avoid such problems is to get the vocals right in the first place. "When I do group vocals, I make sure that they're very well rehearsed," says Rosado. "I try to keep the problems to a minimum."

In addition to a well-rehearsed performance, there are other techniques for making sure your background vocal performances sound tight when they're recorded. "If there are words with *esses* or *tees*," says Addabbo, "a lot of times I'll tell the background singer, 'Just don't sing it. The lead singer is doing that *ess*.' If you have five people singing *esses* at the end of the line, you're going to hear 'esssss.'"

Likewise, if all the singers have to sing a hard consonant like a *tee* sound at the end of a line, getting them all to sing it at precisely the same time can be very tricky. "I have the background singer not really sing the *tee*," Addabbo says.

Another issue to consider when recording a lot of layered background parts is that any sonic problems with those tracks are going to be multiplied. Your mic choice is important. You don't want to use a mic that will accentuate an unpleasant frequency. Power gives this example: "If you have, say, 24-tracks of background vocals out of a sort of nasal-sounding singer, or that were recorded through a nasal-sounding mic [or both], you cannot possibly cut enough of that bad thing out of there, because you have 24 instances of it."

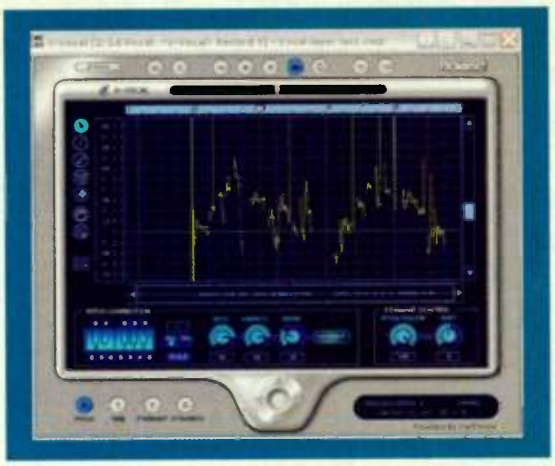

FIG. 7: Cakewalk Sonar 5's V-Vocal section (developed in conjunction with Roland) gives users built-in pitch correction and the ability to manipulate both pitch and time.

Your decisions about editing breaths will also be different in regard to background vocals. In most situations you should just cut the breaths out. "You've got to be really careful," points out Addabbo, "because you can't have 6, 8, or 12 breaths there—however many vocals you stack up. There's no reason to have 12 people breathing on your record. I'll definitely clean them up." (Waves recently released a breath removal and reduction plug-in called DeBreath, which is part of its Vocal Bundle.)

Addabbo adds, "I'll stay in [Pro Tools'] Grid mode in case I want to fly the parts around." No matter what sequencer you're using, it makes sense to edit the boundaries of your background vocals on the grid (assuming you've recorded to a click), because then it's a lot easier to copy and paste the parts to different sections of the song.

In Sync

If you're editing background vocal tracks on which the phrasing of the singers isn't particularly tight, there are plenty of editing solutions. One key for tightening up the background tracks is to make the singers' phrases all end together.

Sheppard's method for doing that is typically to use volume automation and fade down vocals that hold for too long so that they end at the same time as the other singers'. You can use the same technique to make sure that the background vocals start their lines together. "I always do fade-ups on everything," says Rogut. If a harmony vocal is too short at the end of a line, Rogut will often use time-stretching to lengthen it.

Several of the engineers also said that they'll sometimes use Synchro Arts VocAlign (see Fig. 8) to tighten up the harmonies. VocAlign is available as a plug-in and as a standalone application for both Mac and Windows. "It actually captures and time-stretches a vocal to make

EDIT SAVERS

When doing a close edit on a vocal (or any other type of audio track), you have a number of options you can try to make the transition sound as natural as possible.

1. Experiment with lengthening one side of the edit and shortening the other. (Adjust while looping the audio if possible.)
2. Zoom to the sample level and make the edit at a zero-crossing.
3. Crossfade at the edit point.
4. Fade out on the left side and fade in on the right (instead of using a crossfade).
5. Move one side of the edit to a separate track and overlap it slightly with the original track.

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it match the phrasing of the original," says Rogut. "But you have to be very careful because it's so good that it will make it phasey." Sometimes after Rogut uses VocAlign, he'll manipulate the vocal further if the plug-in created phasiness. "You can shift it [the vocal phrase] around a little bit, or slightly time-stretch it afterwards," he says.

If you're going to be time-stretching or doing other destructive edits to your vocal tracks, be sure you're working on a copy, not the original audio file, so you can backtrack if your edits don't work out.

Taming the Beast

Even with their levels compressed on input, your vocals are almost certain to need additional dynamic adjustment in the mix. On this topic, the six engineers offered solutions that included both automated volume rides and compression.

Sheppard recommends evening the levels on a vocal track with volume automation before you mix it, and then writing those changes to disk. "I do that first," he says. "That way, you won't get that heavy compression on something." You generally want to keep the level to the compressor relatively even, because higher input volumes trigger more compression, which can change the character of the sound. "If the level difference is really crazy between the verse and chorus," he says, "I'll put them on separate tracks."

Power takes a different route to the same destination. "When someone sings soft and low, for example, instead of just turning up the volume on the channel, I will often turn up the input to the compressor. This does two things: one, it turns it up, and two, it keeps the sonic character the same."

Rogut likes to do vocal-track volume adjustments at the same time he's comping. "While you're doing all the crossfades between all the comp selections, you check that each line matches EQ-wise and levelwise to the rest of the vocal. It's part of the comping process," he says.

It's hard to generalize what types of compression settings to use on vocal tracks. A lot depends on the song, the singer, and the type of compressor. Typically, however, if you're working on a ballad, you'll set the compression relatively light, to keep the singer sounding natural and dynamic. For an up-tempo song where you want an "in-your-face" vocal sound, you'd compress much more heavily. "For a real driving, dense rock track, you're going to need to squash that vocal pretty hard to keep it powerful and to keep your track powerful," says Addabbo.

FIG. 8: Synchro Arts VocAlign unifies the phrasing between vocal tracks.

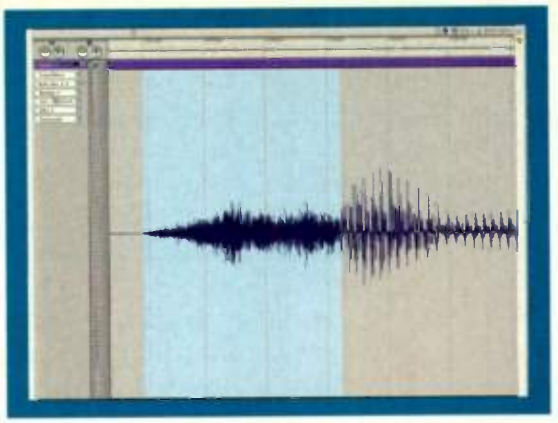

FIG. 9: The blue highlighted section shows a sibilant *ess* sound. Because of their denseness, such sounds are easy to spot when you look at a waveform display.

EQ Me

Deciding how to EQ a vocal track depends a lot on the sound of the track and its context in the mix. That said, there was pretty wide agreement among the interviewees that getting rid of low frequencies below the range of the vocal is usually a good idea.

Rosado has found a lot of problems with low-end rumble and noise on vocal tracks recorded in home studios and sent to him for mixdown. "People don't understand what a low-cut filter is," he says, "or they have mics that don't have low-cut filters. A lot of the very inexpensive condenser mics don't have them. I heard a truck once on a vocal. I said, 'Is that a truck?'"

Even with well-recorded vocals, it's often useful to get rid of unneeded low frequencies. "I would definitely pull out anything that's not used," says Sheppard.

"The vocal sits better when you roll off a lot of that bottom that you don't need," says Stasium. "Depending on the vocalist and the key of the song and 18 other items, I'll roll off somewhere between 40 Hz and sometimes up to 150 Hz. I'll sometimes use a highpass filter."

Boosting in the high frequencies is another typical vocal EQ treatment. "I like air in the vocals," says Stasium, "so frequently, but not all the time, I'll boost it at 18 kHz to get some air into the vocal. I've always kind of done that, even when I was working with SSLs and with Pultecs."

Power will also boost the really high frequencies. "Sometimes I kick up 20 kHz, even though there's not a lot up there," he says. "What happens is that the curve extends down, and you end up gently nudging up the entire top end. Conversely, if you use a shelf at 10 kHz—although it may go all the way out to 20 kHz—it might be too harsh."

Automation can be used not only for volume rides, but also for EQ that changes throughout a song. "I do

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automated EQ rides throughout the range of the song to even out the vocal sound through the different ranges of the person's voice," says Power. "A lot of times, something will sound really good in a certain range of a person's voice, but then when they go down deep, all of a sudden 150 Hz comes up like 6 dB, and all of a sudden it sounds kind of tanky and woolly."

Corrective Action

Sometimes you have to use editing and DSP (or both) to get rid of problems on a vocal track. One of the most common is excessive sibilance, which can be corrected, in most cases, with a de-esser plug-in. A de-esser is a compressor with a sidechain input (see "Silencing Sibilance" in the April 2000 issue of EM, available at www.emusician.com). It allows you to compress a specific sibilant frequency within the vocal.

"The more you compress," says Power, "the more you need de-essers. The closer you mic, the more you need de-essers. The more humid the day (when you're using condenser mics), the more you need de-essers."

The sibilance might not become noticeable until you start EQ'ing and processing the track. "Probably you're brightening your vocal a bit when it's in the track because you're competing with instruments, cymbals, and stuff that's all very powerful," says Addabbo. "And there's this little vocal in the middle of it, and that's the most important part of the record anyway, so you better get that thing right. So you make it bright, you make it loud, and all of a sudden the *esses* start to become overbearing."

Most de-esser plug-ins have "Listen" functions that let you hear only the frequencies you're attenuating. "Listening to the sidechain is very important," Power says. "If you listen to the track for a little while, and you start flipping through the frequencies, you'll really find the areas that—when you put it all back together—sound offensive. In some cases, there's more than one area. Sometimes I'll set a de-esser around 3 kHz and I'll put another one up at 7 kHz. I also ride the threshold on the de-esser. As you know, it sounds okay up to a point, and then it doesn't sound good at all. So I do [automated] rides on the threshold."

Your sequencer's volume automation can also be used for quashing *ess* sounds. "You go in and pick the spot where that *ess* is," Addabbo says, "and you can usually see it because the high-frequency content is easy to spot (see **Fig. 9**), and you can just dip it there. You'd be amazed sometimes—you can dip it 10 dB and it still sounds natural because the *esses* are so prominent. You can get away with a lot."

Besides *esses*, sometimes you get palate sounds from the singer and other anomalies, like clicks or plosive pops (see **Fig. 10**), which can mar an otherwise good take. "That's one reason that I do multiple takes," says Rosado. "I could always crossfade

that part out and put another part in." Addabbo agrees: "Replacing the syllable, word, or phrase from another take would be my first choice."

But if you can't find a suitable replacement for the offending segment, you could zoom down to the sample level of your sequencer or audio editor and try to "draw" out the noise with the pencil tool. "If that doesn't work," Rogut says, "I would try removing the click or pop by cutting it out completely—and then trying to time-stretch the surrounding audio to fill the gap."

"Sometimes I'll cut out the very, very peak of that plosive and crossfade the rest in," Rosado says, "and it will be enough to soften it up. You can't take too much out, because obviously then there will be a gap. But if there's enough happening in the song, you won't miss it."

Stasium has another trick that he uses for pops: "I also will use a highpass filter, and put it in there with the computer just for that word. I'll automate in the highpass filter. Just roll everything off under 100 Hz or 120 Hz or even 150 Hz during the pop. It's so brief that you don't notice the filter coming in and out."

Be sure to make a duplicate copy of the audio file in question before you start drawing in waveforms. In many digital audio sequencers, pencil edits (or their equivalents) are destructive.

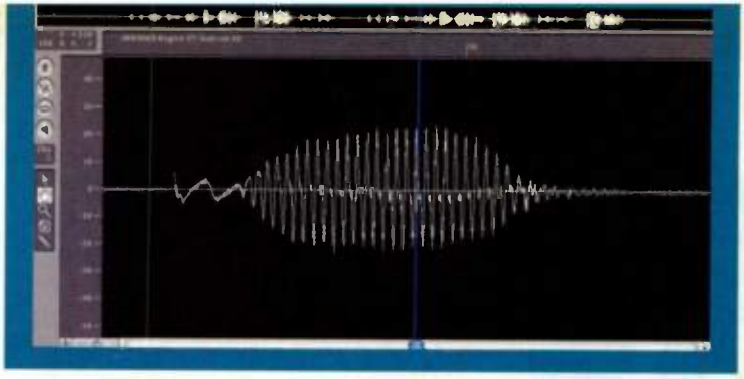


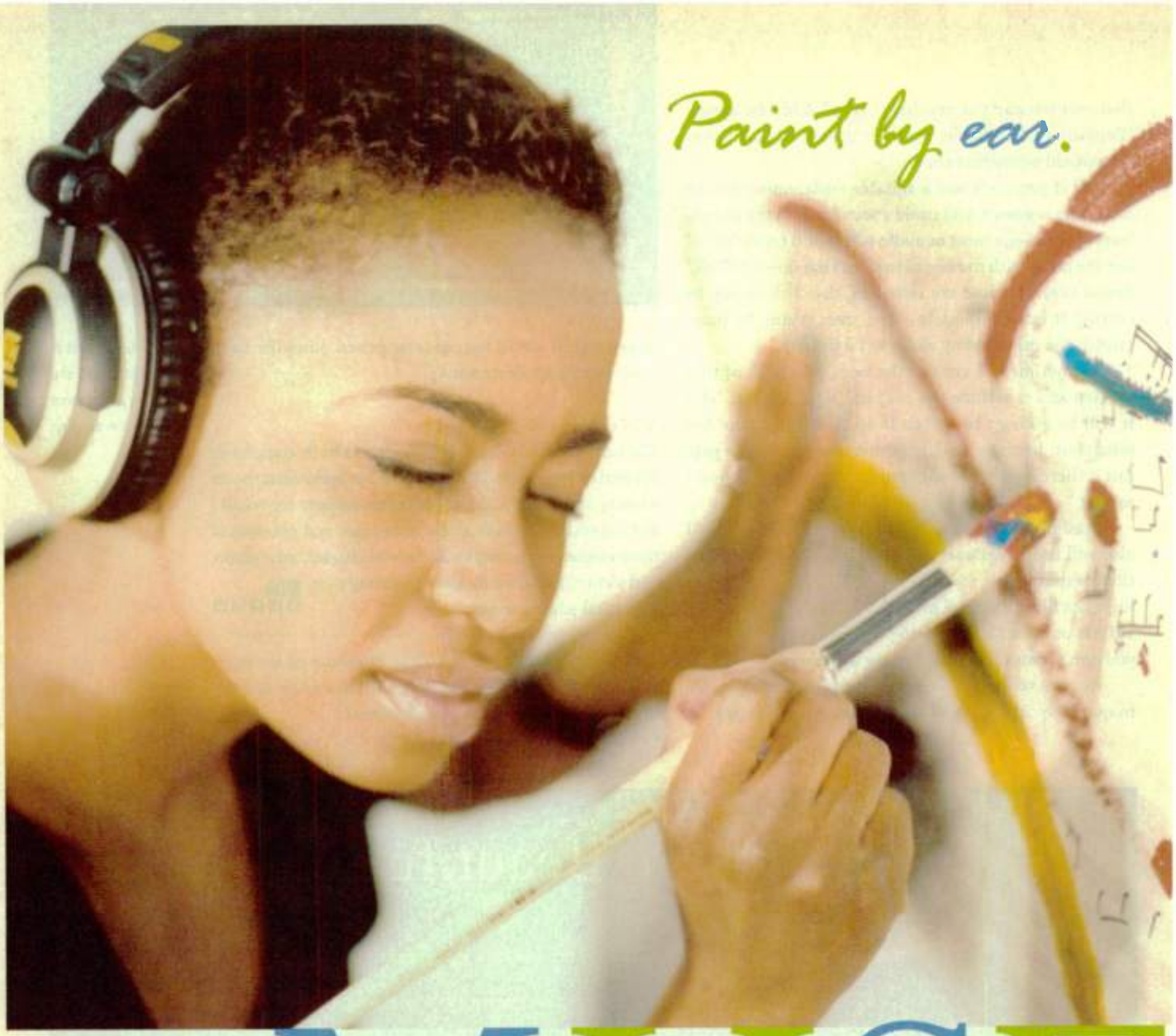
FIG. 10: This is how the sound of a plosive *p* looks in a waveform display.

Vocalization

The subject of vocal editing is too broad to be completely covered in one article. (See **Web Clip 2** to learn what the six experts said about ambient effects treatments for vocals.) But hopefully, this look at the techniques and opinions of these engineer-producers will give you insights, inspiration, and plenty of food for thought regarding your own vocal-editing sessions. EM

EM WEB CLIPS

Mike Levine is an EM senior editor. Thanks to all the interviewees for their time and cooperation, and to Michael Cooper for additional technical assistance.



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Mixing Down and Speaking Up

By Paul Tingen

When it comes to mixing or stirring up controversy, Dave Pensado is front and center.

Master mixer Dave Pensado enjoys causing consternation among his peers by making controversial statements about the craft of recording. He takes great pleasure in contradicting conventional audio wisdom and, in the process, delights many up-and-coming recordists, while simultaneously enraging large groups of the pro-audio establishment.

To give you an idea, here's a typical Pensadoism: "It is better to sound new than to sound good." Adding fuel to the fire, he says, "These old farts that are spending their time trying to sound good have all become irrelevant. The only time we ever see them is when the Grammys give them some BS award for something that doesn't matter anymore. Instead, spend your time trying to sound new, because that is what pop music is about." Take that, anyone who has spent a lifetime pursuing high-fidelity audio.

Or take this statement: "Show me a guy who doesn't like a particular format, and I'll show you a guy who doesn't know how to use it." Ouch. Or, during a time when most people seem to think of the late '60s and early '70s as the golden age of pop music, what about the following: "I'm in an extreme minority, but I think that the music being made today is the best ever." And further: "Trying to protect trade audio secrets makes as much sense as Ernest Hemingway trying to guard his verbs. So I put my exact plug-in settings on the Internet."

Pensado loves saying that many plug-ins sound better and offer more options than old-fashioned outboard gear. He says, "The reason the old-timers are bitching about the digital

stuff is that they have not taken the time to figure out what the various plug-ins are good at. I greatly admire and respect the skills these guys have, but I just don't understand why they don't apply the same vigor to today's technology as they did with analog when they were 18 years old. I wish they would turn loose on today's technology and blow us all away with the records that they are capable of making."

COURTESY DAVE PENSADO

Pensado with Mary J. Blige.

Impeccable Résumé

Pensado's words may rile many, but his track record is inarguable. He is an extremely successful mixer who has worked on numerous hit records. Earlier this year, he enjoyed three No. 1 hit singles in the same month: Mary J. Blige's sumptuous "Be Without You," Beyonce's "Check on It," and Keyshia Cole's "Love." Pensado has also mixed hits for the likes of Christina Aguilera, the Black Eyed Peas, Destiny's Child, Missy Elliott, Ice Cube, Lil' Kim, Brian McKnight, Mya, Pink ("Lady Marmalade"), the Pussycat Dolls, Justin Timberlake, and Warren G. + Christina.

Pensado honed his audio skills during the '70s and '80s—years he spent exploring Atlanta's club and hip-hop scene and engineering in both live and studio settings. In 1990 he moved to Los Angeles hoping to hit the big time, and within three months he had mixed the No. 1 hit "Do Me Baby" for Bell Biv DeVoe. His phone hasn't stopped ringing since.

Most of Pensado's work has been of a hip-hop and pop-oriented nature, which adds a crucial qualifier to his combative statements—that is, he isn't talking about music that requires the most pristine audio possible. "Of course, there are kinds of music that require 8 trillion bits and extensive skills," he admits, "because they are about capturing the ultimate jazz sound or the most beautiful orchestral sound. We could not do what we're doing today had it not been for the guys who're doing that. But that's not the type of music that I do; I do pop, hip-hop, and rock. And 'Stairway to Heaven' could not have been any better, whether it had been made in 8 trillion bits or on a half-inch tape machine.

"That's a great song, and the engineering, production, and mix work together to create something that moves us. That's what contemporary music is supposed to do. We shouldn't be centered on elite snobbism about what has the most bits. We should talk about the types of music we create and present to the public, and the fact that the only limitation is our imagination."

While revealing everything about his technical skills and settings, Pensado simultaneously deemphasizes the importance of technology. "My settings won't work on anything other than what I used them on," he explains. "Publicizing them may give young engineers and producers a starting point and the confidence to be creative and try their own thing. You have to master the technical side so proficiently that you forget about it while working. But that doesn't mean I have a great respect for technical skills.

"You can teach anybody to get a great vocal sound, but you can't teach what a great vocal sound is. Taste is what distinguishes one mixer from another . . . which brings me back to the old-timers, who think that it's their technical know-how that separates them from being irrelevant. But it's not—it's their taste."

> "Taste is what distinguishes one mixer from another."

Fifty-Fifty Proposition

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because there are things he is good at and things he is not so good at. "I want to be in a position where I am challenged, but not so much so that what I'm doing is going to suck. When I decide to do the mix, I'm a big believer in listening and writing down what you like and what you think could be better. For example, the delay in the vocal sound in the second verse may be incredible, but the shaker and the hi-hat aren't moving the song enough."

Pensado currently works almost without interruption at Larrabee Studios in Los Angeles. He doesn't have his own facility, because he likes the feeling of being taken care of and having top-notch stuff around him. "I can have a major technical problem at 3 a.m., and it will be fixed at 3:05 a.m. In a home studio, you don't have that," he says. He also works extreme hours—from noon until 3 a.m.—at least six days a week. His ferocious appetite for work earned him the nickname "Hard Drive." In the past decade, as hard drives have become omnipresent, the name has become doubly apt.

"The sessions arrive on hard drive or DVD disc," Pensado says, "and almost always in Pro Tools format. I haven't seen tape in maybe three years! Very occasionally there's a Logic session, which my assistant immediately transfers to Pro Tools. He also puts the songs in a layout that I like, with the vocals at the top of the session, the drums and percussion next, and then the instruments at the bottom. After that, I go back to my original notes and listen to the rough mix again, because there's usually a more recent one, and take more notes.

"Now all my ideas are starting to produce a direction, a game plan. If the song is percussion-driven, like a hip-hop track, I start with working on the drums. If it's a song that centers around an amazing vocal performance, like Christina Aguilera, I start with the vocals and weave everything around that.

THE MP3 FACTOR

These days, all engineers must deal with having their mixes end up as MP3s. Not surprisingly, Pensado has a few opinions on the subject.

"Whatever works to disseminate our product to the largest number of people for the least amount of money is a wonderful thing. Right now, that's MP3 and the iTunes format. It allows millions to hear our creativity. From an engineering viewpoint, I would prefer it if we could get a little better fidelity. But we've always been obliged to cater to all formats. We have always had to make sure that mixes sounded good on little mono TV speakers, hi-fi speakers, car speakers, the radio, and now computer speakers. And technology always comes along to make that possible. Take the Yamaha NS10 speakers: everybody hates them, but we all know that if your mixes sound exciting and bright and you can hear the bottom end on NS10s, they will probably sound good everywhere.

FIG. B: Pensado uses Waves' L3 Multimaximizer limiter to boost levels without going over when mixing for MP3.

"Today there are more formats than ever that we have to cater to. I mix to 1-inch tape and to 44.1/88.2 kHz, or if it comes in to me on that, I mix to 48/96 kHz. We choose the master from those alternatives. The 44.1/88.2 kHz converts more easily to CD and sounds only slightly better. In any case, the emphasis with different resolutions—44.1, 96, or 192 kHz—is on the word *different*. One is not better than the other. Using 16-bit is better for sledgehammer hip-hop-like tracks, while higher resolutions and sampling rates are better for the airy, spacious stuff.

"The main problem with MP3 occurs with width, and some subtleties such as reverb decays and spaces between instruments can get lost. Usually the two octaves below 100 to 125 Hz are hard to get to come through on MP3. And sometimes the vocal gets sucked into the mix, or the transient response on the snare just isn't there. I make my own MP3 files using the Pro Tools MP3 Codec (highest quality at slowest speed), and if there's something I don't like, I may alter the mix to compensate.

"I always put the Waves L3 Multimaximizer across the mix to cater for MP3. The settings don't vary too much from the screen shot [see Fig. B]. The main point is to keep the signal from going into the dreaded 'above 0' distortion territory. Think of the L3 as a 5-band parametric with a compressor on each band. Or think of it as five compressors, each compressing only a small assigned part of the frequency spectrum.

"The MP3 files I make sound pretty good, and I'm planning to mix specifically for MP3. I'll start mixing Christina's new album next week, and I want to talk with her about that. She has the power to get the record company to release our own MP3 files."

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"So I start with what I judge to be the foundation elements of the mix, get them to a point where I think they're great, and then bring more elements in. With every new element that I bring in, everything starts changing. I take a five-minute break every hour or so, and when I come back into the room, things always sound different and I'm better able to distinguish the good stuff and the flaws. At that stage, mixing is like sculpting. Not that I want to compare myself to him, but Michelangelo would start with a blank rectangular block of marble and chop away until it looked the way he wanted. For me, mixing is a process of taking out what I don't want until I'm left with what I do want.

"At that point, I have something I can evaluate, and that is when it becomes the most fun. I have something that's organic and has a life of its own—that has the creativity of the writer, the performance talent of the artist, and the vision of the producer. Then I can take things to the next level, bring in the elements that are going to move somebody, and create something that's more than the sum of the parts.

"That is when my songs become different from those of other mixers. I often change them quite a bit, perhaps adding the lowest octave, 40 to 80 cycles, to the kick drum—things like that. Also, when a song comes to me, it may already be a year and a half old, and it may be another year before it's released. Part of my job is to anticipate where sounds will be going in one or two years. So I may add some samples to make sure it will sound fresh by the time it reaches the radio."

A Little off the Top

Pensado explains that he can take more chances when artist and producer are present, because he can give them more ideas to choose from. If he's on his own, however, he mixes a lot more conservatively. In his customary colorful language, Pensado draws an analogy between mixing and hairdressing: "Some people want a small trim; others want to change their entire look. So some clients want the rough mix, but a little better, whereas others tell me to do whatever I want. The main problem is finding out what they want."

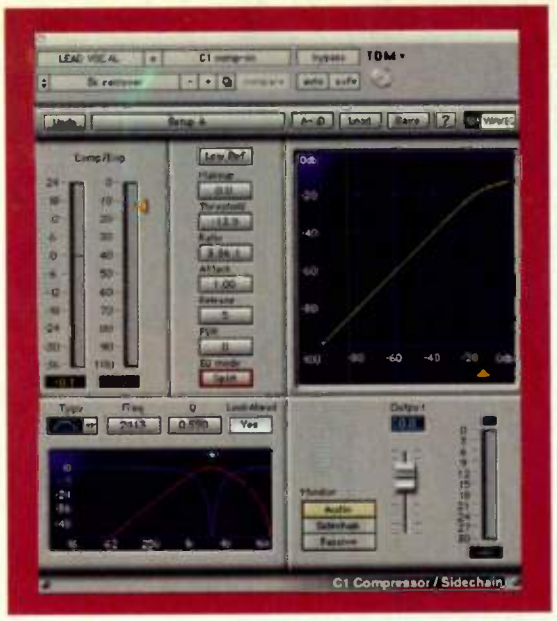

FIG. 1: One of Pensado's favorite dynamics plug-ins is Waves C1 Compressor/Sidechain. This screen shot shows the settings he used for Mary J. Blige's vocals on "Be Without You."

Pensado returns to his analogy of the mixer as sculptor, this time discussing the tools of his trade, notably effects gear. He says, "I use plug-ins about 60 percent of the time now. When you're working with a piece of marble, you have tools to knock off huge chunks, to knock off small pieces, and even to polish the marble. The beauty of plug-ins is that you can get microscopic—you can really polish with them. You can pull out a band as narrow as 998 to 1,002 cycles with the Waves Q-series. In general you can't do that in the analog world.

"But please, especially for up-and-coming people, don't consider that a rule—it's just a guideline, and we're talking EQ here. But in general I like to do the broad carvings in the analog world, with stuff like a Neve, an Avalon 2055, a GML, an API 550 and 560, and a Pultec. Everybody uses those. Then I use my plug-ins to add colors, shades, and nuances. We have such a wide and varied range of plug-ins available; just consider that between companies like McDSP and Waves we have 50 EQ plug-ins to choose from. They are all amazing and all do things that were not available in the analog world. I still use the McDSP E6 [shelving equalizer] a lot. The high end has a 'grain' that I just love on vocals. And the company's F2 is very useful for rolling off low end on reverb returns or making the bass and kick drum fit with each other.

DAVE PENSADO: A SELECTED DISCOGRAPHY

Beyonce, "Check on It" on *#1s* (Sony Urban Music, 2005)
Mary J. Blige, "Be Without You" on *The Breakthrough* (Geffen, 2005)
Keyshia Cole, "Love" on *The Way It Is* (A&M, 2005)
Flipsyde, "Happy Birthday" on *We the People* (Universal International, 2005)
The Pussycat Dolls, "Beep" on *PCD* (A&M, 2005)
Kelly Clarkson, "Miss Independent" on *Thankful* (RCA, 2003)
Christina Aguilera, "Beautiful" on *Stripped* (RCA, 2002)
Kelly Rowland, "Stole" on *Simply Deep* (Sony 2002)
Christina Aguilera, Pink, Mya, Lil' Kim & Missy Elliott, "Lady Marmalade" on the original soundtrack of *Moulin Rouge* (Interscope, 2001)
Pink, "Get the Party Started" on *M!ssundaztood* (Arista, 2001)
The Black Eyed Peas, "Request + Line" (featuring Macy Gray) on *Bridging the Gap* (Interscope, 2000)
Ice Cube, "Until We Rich" on *War and Peace*, vol. 2 (Priority, 2000)
K-Ci & JoJo, "All My Life" on *Love Always* (MCA, 1997)

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"In the compression world, I still like one of the oldest plug-ins ever invented—the C1 Compressor/Sidechain by Waves [see Fig. 1]. When I find an irritating frequency in a vocal, which typically would be somewhere between 1 kHz and 2.5 kHz, I put that on the sidechain and let the compressor pull that out for me, so I don't have to have a separate fader with that notched out.

"In the old days, we would dedicate two to six faders to just the lead vocal. For example, one fader would have the straight vocal; another fader would have a bit of low end rolled off, because most vocalists sing more softly in the verses; and a third fader with the low end put back in and a little notch around 1K for the chorus. Then you would pull up the fader you wanted. But today, because you can automate plug-ins, you don't need multiple plug-ins, and you do anything you want under one fader.

"I also like Waves C4, Renaissance Compressor, and new SSL plug-ins; McDSP C4 and M2000; and Bomb Factory LA3A. The C4 and M2000 are basically 4-band compressors. Think of them as parametric EQs that have four bands and a compressor on each—they are tremendously powerful tools. The SSL EQ plug-in and G-series compressor plug-ins are incredible. They sound exactly like the SSL. The LA3A sounds spectacular on pianos, and I often send the vocals through that before I send them through an analog compressor. Again, it's better to sound new than good, so when the whole world is using Pultecs, don't go out and do it as well! Figure out something else; find a plug-in that gives you something better."

Pensado says that he has no time for the very concept that makes many people reach for vintage gear. "I don't like the word *warm*. For me it's another word for *dull*. I love saying that, because it pisses a lot of people off! I have noticed that different keys and tempos affect the kind of gear I like using. For example, F#, A, and E are bright keys and best with tube gear, while B♭, E♭, and some of the slow tempos sound best with newer gear and plug-ins."

Clearly concerned that people will take his statements the wrong way, Pensado adds, "That isn't a rule; it's just an observation. And I would love to hear some feedback on it." With that, he offers his email address (fdpen@ix.netcom.com) and says, "I get about 20 to 30 emails a day, and I hope that in giving loads of technical details I don't do more damage than good. It's not about the gear—it's what you do with it. So I hope that people think that it's good to know how I do it, so that they can find their own way of doing it. In the end, you don't sell your engineering skills—you sell your creative skills." EM

Paul Tingen is a writer and musician living in France. He is the author of Miles Beyond: The Electric Explorations of Miles Davis, 1967–1991 *(Billboard Books, 2001), a book on early weird funk experimentation. For more information, visit www.tingen.co.uk.*

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Evolving Grooves

By rachMiel

Create killer rhythm tracks with layers and envelopes.

In the early days of electronica production, grooves often repeated without variation, and listeners loved it. These days, beat lovers demand more from their grooves. Unchanging loops have become passé; instead, electronica fans want to hear grooves that evolve.

This evolution can happen in various ways. A groove might start sparsely, riddled with musically compelling holes, and gradually increase in density until all the holes are filled. Or it might begin with all its notes at midvolume and proceed toward a dramatic alternation of loud, accented notes and soft, understated ones. Or it might move from dry to wet (flanged, filtered, delayed, or otherwise processed) and back to dry. A groove might play with slice reordering, slice panning, slice reversal, and so on.

One effective and fun way to build an evolving groove is to first create several variants of the original groove. Next, overlay the original with the variants, keeping all in sync. Finally, use volume envelopes to mix the original and the variants in a way that yields the desired results.

Groove Variants

You can use whichever techniques you are comfortable with to create groove variants: manual cutting and pasting, running the groove through effects boxes, beat slicing and rearranging, desampling, resampling, or some other process. The important thing is that the variants come together in a compelling evolutionary mix.

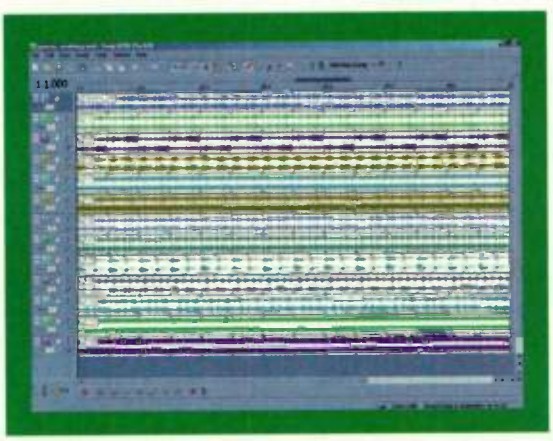

FIG. 1: The original groove loop (on top) and 12 loop variants are loaded into Acid 6. Each loop is present for the entire mix, and tempos are matched at 140 bpm.

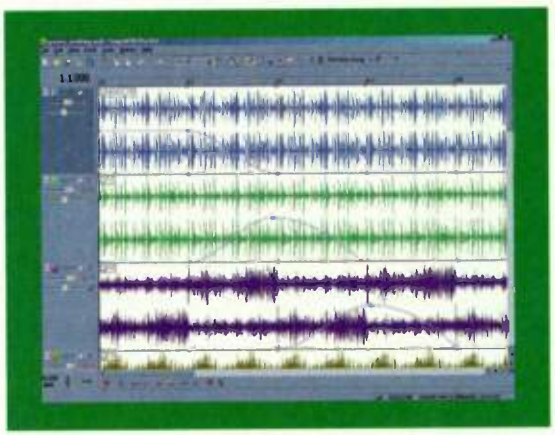

FIG. 2: In this excerpt from the groove mix, volume envelopes create the effect of the original groove crossfading into variant 1, which in turn crossfades into variant 2.

Assuming that you want your evolving groove to remain in the same tempo and meter throughout, the original groove and all variants should match in tempo and meter. For example, to create an evolving groove in 4/4 time at 142 bpm, the original and the variants should all be in 4/4 at 142 bpm. Integer multiples will also work: 2/4, 8/8, or 8/4 time at 71 or 284 bpm, for example.

If you're an experimental groovester, you can dare to create variants with noninteger multiples of the tempo and meter. Start with simple relationships. If the original tempo is 120 bpm, try creating variants at 80 bpm or 160 bpm. If the original meter is 4/4, work with 3/4 variants (3/4 syncs up with 4/4 every 12 beats) or 5/4 (5/4 syncs with 4/4 every 20 beats). If you're feeling really adventurous, foil the grid entirely—create 127.9 and 211 bpm variants, 11/8 and 15/16 variants, and so on. If you keep the original groove present throughout, you can bridge the gap between order and chaos.

Once you've created all your groove variants, you're ready to overlap them with the original. Fire up your sequencer of choice, such as Acid, Live, Cubase, or Sonar, and load in all the grooves, making sure that they are properly synced in tempo and meter (see Fig. 1). All grooves should be fully present for the entire passage;

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you'll use volume envelopes to control the degree to which each is present or absent in the mix. Unless you want to change the base tempo, set the sequencer tempo to that of the original groove so that it plays back exactly as you created it.

Envelopes and Evolution

Now comes the exciting part: creating the evolution. Use volume envelopes to fade groove variants in and out of the mix. Your first step is to get a sense of what evolution you want, and then order the groove layers accordingly from top to bottom visually. That makes the compositional logic clearer when you're ready to draw your volume envelopes.

Next, insert a volume envelope into each groove layer. Doing this in Acid is a snap; simply select all your tracks (Edit→Select All or Ctrl or Command + A) and insert the envelopes (Insert→Envelopes→Volume or Shift + V). Now that your groove layers are overlapped and synced, your evolution is clearly laid out (from top to bottom), and your volume envelopes are inserted, you're ready to start evolving.

You have three types of volume envelope relationships to draw on. The first is simple *crossfading*, in which one groove layer (or set of layers) fades out while another fades in (see Fig. 2). The speed and character of the crossfade is determined by the duration and shape of the fade-in and fade-out: linear, slow, fast, or smooth, for example.

The second relationship is *splicing*, in which one groove layer ends abruptly (for instance, with a very fast fade-out), and the next begins abruptly, like a tape splice (see Fig. 3). A variant of this would be to have the two layers playing together, and then end one suddenly.

The third relationship is *multilayering*, in which one groove layer persists while others fade in and out (see Fig. 4). The persistent layer's volume can remain steady or can change gradually or abruptly. For example, you might want the original groove to start off full bore, and then decrescendo to an audible background presence and remain at that level for the entire passage while other groove layers become more audible, affecting the beat, the meter, accents, and even the tempo.

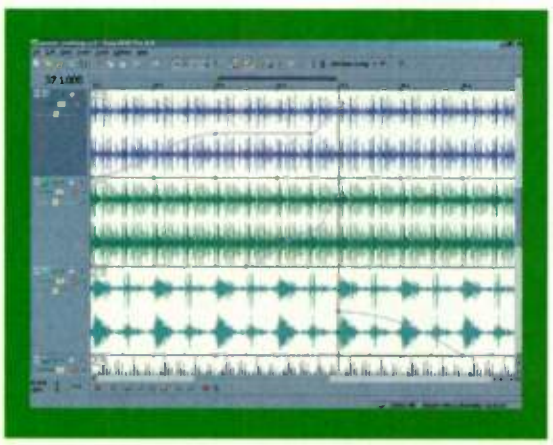

FIG. 3: In this example of splicing, variants 6 and 7 increase in volume, and then end abruptly at the same time that variant 8 begins abruptly.

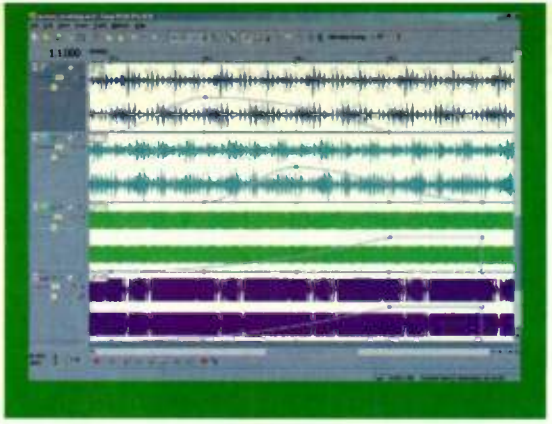

FIG. 4: This screen shot shows an example of variant multilayering. Variants 11 and 12 persist (with a gradual buildup), while 9 and 10 fade in and out over them.

Evolver

I created an evolving-groove example (see **Web Clip 1**), which consists of a 4/4 loop at 140 bpm and 12 variants, all in 4/4 and at 140 bpm to ensure that the groove mix remains synced. I used two Ohm Force plug-ins to create the variants: Quad Frohmage Filterbank and Ohmboyz Delay. Some variants played with the rhythm of the original groove, mostly by using delay to double beats (quarter notes to 8th notes, 8th notes to 16th notes, and so on). Additional variants modified the timbre through filtering, flanging, or some other sound-shaping tool.

The groove evolution took the following form: original groove, variations 1 through 12, and original groove. The mix, which is 1 minute and 39 seconds in length, begins with 4 bars of the original groove. Over the next 4 bars, the original crossfades to variant 1. Over the next 12 bars, a similar succession of crossfades occurs. At variant 6, things get a little different: variant 6 (a pivotal comb-filter pitched version of the original) remains at full volume for an extended period and is joined by variant 7. Variants 6 and 7 rise in volume, and then stop abruptly at the exact time that variant 8 starts (at full volume). Variants 8 through 10 crossfade in and out similarly to 1 through 6, while 11 and 12 build slowly (layering over 8 through 10) to full volume and then stop abruptly, giving way to 4 bars of the original.

The technique outlined here for creating evolving grooves is simple, flexible, and powerful. All you need are a loop, a set of variants, and volume envelopes to make anything from smoothly crossfading organic grooves suitable for downbeat electro-ballads to wildly chaotic and complex grooves. EM

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rachMiel is a composer of experimental electronic and acoustic music.

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Expressive Control

By Jim Aikin

Personalize sounds with multitarget MIDI modulation.

Controlling more than one parameter from a single MIDI Control Change message is an excellent way to add expression to your synth sounds. In "Square One: The Matrix" (see the June 2006 issue of EM), I explained matrix modulation, the primary tool you would use to set up that type of control. In this column, I will discuss some of the musical applications of matrix modulation. I'll assume you have a synth equipped with a modulation matrix and understand how its features work.

The 8-bar filter swell has become a cliché of electronic music: as a rhythmic synth riff loops, the filter gradually opens until the synth dominates the mix, and then closes down again so that the riff recedes. To give this technique a fresh spin, try adding overdrive distortion as the filter opens. Overdrive increases the amplitude of the signal, so you may need to use the same modulation source to slightly reduce the synth's output level.

Controller Scaling

To do that, you need to use controller scaling and controller inversion, both of which you can do using the amount parameter in the modulation matrix. When the amount of modulation is small, the destination parameter changes only a little in response to a big change in the incoming data. If the amount is less than zero, the amount of modulation is inverted. When it's inverted, raising the controller value (by pushing a slider up, for instance) lowers the parameter and vice versa.

Some synths, such as VirSyn Tera 3, which I've used for this column's audio examples, use multiplication rather than addition to control modulation amount (see Fig. 1). In that case, an amount of less than one inverts the modulation.

In addition to opening the filter, you can add a tempo-synced delay line with a wet/dry mix of 40 percent or so. That is especially effective with a ping-pong delay algorithm, which will bounce the synth riff around in the stereo field. You can also blend in a suboctave oscillator (see Web Clip 1).

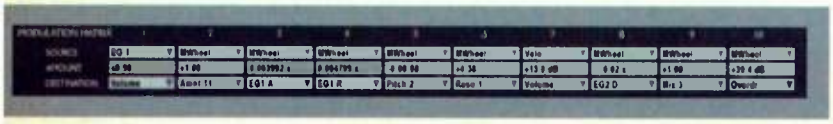

FIG. 1: The modulation matrix in VirSyn Tera 3 has two banks of ten routings each. A MIDI mod wheel can be applied to numerous destinations in different amounts.

Envelope Shaping

Check whether your synth allows envelope segment times (attack, decay, release, and others) to be chosen individually as modulation destinations. If so, you can tighten up a sound by decreasing the attack and release times together. You can start with a gentle sound that has short to medium envelope times (100 ms to 250 ms) and pull both times back to zero by moving your MIDI slider up. Web Clip 1 uses this technique.

Another way to add aggression is to increase the depth of envelope modulation. Try using a second envelope to modulate the frequency of a synced oscillator, as heard in Web Clip 2. Use only a small amount of envelope to start with, and add more while also shifting other parameters. In Web Clip 2, an LFO modulates panning, and its depth and rate both increase as the mod wheel is pushed up.

If you're using the mod wheel to add vibrato, you can also lengthen the amplitude envelope release time using the same MIDI message. That will have no effect on the envelope if the mod wheel is pushed up and pulled back in the middle of the note, but it will allow you to add a long, fading vibrato tail to the end of a phrase. (Try increasing the reverb wet/dry mix at the same time.)

Morphology

Many synths offer numerous ways to change tone color, so look closely at the destinations available in your modulation matrix. Ultimately, this technique is a form of patch morphing, which several hardware and software synths also have. With morphing, one patch transforms smoothly into another; instead of defining several modulation routings separately, you define and store the "start" and "end" patches separately, and then assign a single MIDI modulation source to the morph parameter.

A morph can include any or all of the voicing parameters that have a continuous range of values, but switchable parameters (such as an oscillator's waveform selector) usually aren't morphable. That is because morphing them would cause abrupt and possibly jarring changes in the sound. If, however, the amplitude levels of the oscillators are controllable, you can create the effect of changing waveforms during a morph by crossfading from one oscillator to another. EM

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Jim Aikin writes about music technology and plays electric cello. Visit his Web site at www.musicwords.net.

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The Ins and Outs of LFOs

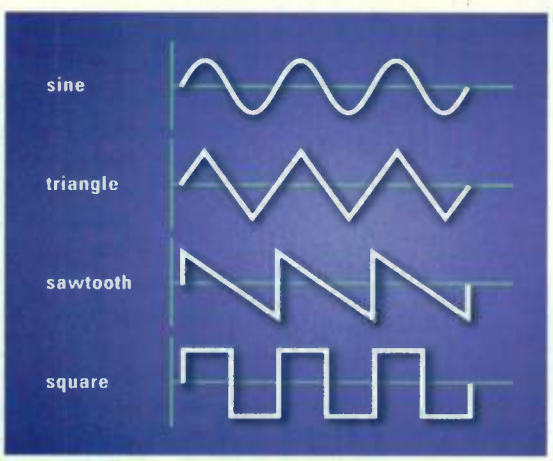
By Jim Aikin

Create a variety of dynamic synth effects with LFOs.

String and woodwind players almost always enrich their long notes by using a technique called *vibrato*. To produce vibrato, string players wiggle the finger that's stopping the string so that the pitch moves up and down slightly during the course of the note. The result is a warmer, richer sound that stands out in a subtle way.

Because synthesizers were initially keyboard based, musicians couldn't add vibrato to synth sounds by wiggling their fingers. But vibrato was too useful a musical technique to go without, so synths were equipped with a mechanical form of vibrato. The pitch of a synth's oscillators could be modulated by a signal from a special *low-frequency oscillator* (LFO).

LFOs quickly proved useful for synthesis techniques in addition to vibrato. Today all synthesizers, most other electronic instruments, and many effects processors have LFOs. High-end instruments may give you four or more LFOs and let you route their outputs to a variety of synthesis parameters so that sounds can be animated in complex ways. In this column, I'll examine the features of the LFO and explain some of the things LFOs are used for.



CHUCK DAHMER

FIG. 1: Standard LFO waveshapes include sine, triangle, sawtooth, and square waves.

Basic LFO Parameters

An LFO is a *modulation source* that produces a repeating signal. Except in a few special situations, the signal rises and falls in a regular, periodic way. Although the LFO is a type of oscillator, we can't hear its output. Instead, the output is used as a control signal for changing some parameter of a synth or an effect (for more on modulation, see "Square One: The Matrix" in the June 2006 issue of EM).

The most important parameters for programming an LFO are waveform, frequency, destination, and amplitude. The term *rate* is often used in place of *frequency*, and *depth* is used in place of *amplitude*.

Most LFOs have a set of basic waveforms, as shown in Fig. 1 (for more on oscillator waveforms, see "Square One: Making Waves" in the October 2005 issue of EM). If you think of an LFO as modulating pitch, it is easy to understand what these shapes do: the sine and triangle wave cause the pitch to rise and fall smoothly, for example, and the square wave causes the pitch to alternate between two values, which is a musical effect called a *trill* (see **Web Clip 1**). Your owner's manual probably has diagrams showing the waveforms that the LFOs on your instrument can produce.

The frequency of a low-frequency oscillator is, by definition, low. Some LFOs can produce signals having a frequency as low as 0.02 Hz (one cycle every 50 seconds). The upper frequency limit for many LFOs is around 20 Hz. Some LFOs can produce frequencies above 20 Hz, but when that type of signal is used for modulation, the result is a buzzy sound that covers up the individual cycles of the LFO waveform.

The LFO's destination is the sound parameter that it's modulating. If the destination is oscillator pitch, the LFO will produce vibrato or possibly a trill. If it's the synthesizer's output level, the LFO will produce tremolo. You'll find other destinations on many synths, including filter cutoff frequency, panning, and the amount of some timbral effect, such as FM. If no destination is chosen for the LFO, you won't hear it, no matter how you set the other parameters. Some LFOs are hardwired to specific destinations (see Fig. 2), but others can be connected to various destinations as needed.

Some synths have a switch that sets the LFO's destination in the LFO itself. You may be able to select only one destination for the LFO, or you may be able to switch on several destinations at once. Other instruments have a switch or a knob on the destination module for activating the LFO as an input.

If the LFO's amplitude is zero, again, no modulation will be heard. As the amplitude is turned up, the depth of the LFO's effect will increase. As with the choice of destination, some instruments let you control LFO

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amplitude in the LFO itself, while others have an LFO amount knob in the receiving module.

The amplitude amount parameter might be *bidirectional*, allowing it to be set to a value that is less than zero. That inverts the LFO waveform. In the case of sine, triangle, and square waves, inversion often makes little audible difference. But being able to invert the sawtooth wave is essential, so that it can produce either a rising slope or a falling slope. If the LFO's output can't be inverted with the amplitude knob or a dedicated invert switch, you'll probably see rising and falling sawtooth waves as separate choices in the waveform selector.

Take It up a Notch

Clever instrument designers have come up with a wealth of variations on the LFO. Here are a few of the features you might see:

LFO sync locks the frequency of the LFO to the tempo of a sequencer. The rate of a synced LFO is set in musically meaningful units, such as half notes or eighth-note triplets, rather than being expressed in abstract units, such as hertz. That is especially useful for creating filter sweeps that repeat every two or four measures, in time with your music.

FIG. 2: The buttons along the top in the LFO section of Spectrasonics Stylus RMX may look like destination selectors, but in fact, RMX has three separate LFOs, each with a single dedicated output routing. One LFO modulates the output amplitude of the voice, another modulates filter cutoff, and another modulates panning. Because RMX is a percussion instrument, it has no pitch LFO.

The **freerun/retrigger switch** controls whether the LFO will restart its waveform each time you strike a key (retrigger) or ignore your keystrokes and continue to cycle (freerun). Retrigger mode is useful if you want a reliable modulation shape on each note, but it can sound a little artificial. Freerun mode sounds more natural.

The **phase knob,** which is active only in retrigger mode, controls the point at which the LFO waveform starts when a new key is struck. That is especially useful when the LFO is producing a trill; you can determine whether each note starts with the lower or the upper of the two pitches.

The **LFO delay knob** controls the amount of time that will elapse after the beginning of a new note before the LFO's amplitude rises from zero to its preset level. This onset delay allows you to play fast notes without using vibrato (which would obscure their pitch). The vibrato kicks in only on longer notes.

Some instruments have both an LFO delay knob and a **slope or ramp knob.** The purpose of this control is to smooth the transition between no LFO effect and the full LFO effect. Here again, the purpose of the parameter is to allow you to mimic the performance of a classical string or woodwind player, who will sometimes begin a long note with no vibrato and then add more vibrato during the course of the note, increasing the perceived intensity of the sound.

FIG. 3: By clicking-and-dragging on the LFO waveform in Steinberg Xphraze, you can change its shape. The smoothing knob controls how smooth or stepped the waveform will be.

If the slope and delay parameters are bidirectional, you may be able to start each note with LFO modulation, and then have the modulation fade out. That can be used with a fast LFO frequency to add an interesting noise burst to the beginning of each note.

Most LFOs include an **amplitude-modulation input** in some form, which is often linked to Mod Wheel messages (CC 1). The purpose of the input is to allow you to control the depth of vibrato or some other type of LFO effect from the mod wheel while playing a keyboard part with your other hand.

LFO rate modulation allows the speed of the LFO to change during the course of a note. Increasing the rate of vibrato slightly as the depth increases makes the sound more expressive. Some LFOs can also change the rate slightly in a random way, which gives a less predictable and therefore more "human" sound.

LFOs on software synthesizers increasingly have more waveforms than the standard sine, triangle, square, and sawtooth types. You may be able to design your own waveform (see Fig. 3), import a complex sampled wave, or choose noise (a random, nonperiodic signal) in place of a repeating waveform.

Another option often found among the waveform selections is **sample-and-hold.** This effect goes back to the early days of synthesis. The LFO module produces a stepped signal in which the level of each step is random rather than a repeating waveform. The speed with which new steps are produced is controlled by the LFO's rate parameter. Sample-and-hold is typically used to modulate pitch or filter cutoff; it's a special effect that sounds very electronic.

Spice of Life

Not all synthesizer patches need the animation provided by LFOs. But if there were no such thing as an LFO, synthesizers would sound so dull and bland that no one would want to listen to them. The LFO is as essential an ingredient in sound design as spices are in cooking. EM

Jim Aikin writes regularly for EM and other publications and Web sites. He is the series editor of Backbeat Books' Power Tools music-technology books.

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MySpace for Musicians

By Fran Vincent

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The Basics

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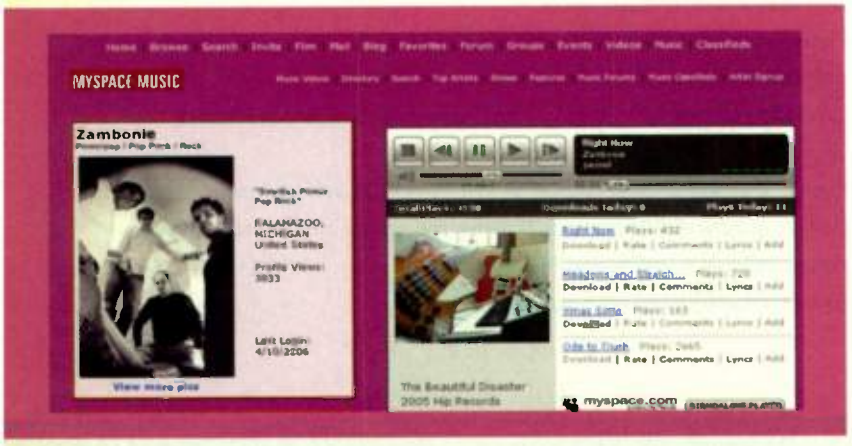

FIG. 1: Your MySpace Profile can be customized using HTML. If you don't know HTML, you can download free MySpace templates from many sites on the Web.

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Customizing Your Profile

Your profile is customizable, much like any HTML page (see **Fig. 1**). If you are HTML savvy, you can compile your own style sheets to customize it. If not, do some research on the Web. Go to your favorite search engine and type in *MySpace layouts* or *MySpace backgrounds*, and you will find plenty of information and many free template layouts. Using these premade layouts is often the best way to compose your page. You simply copy and paste the code into your profile and tweak it a little if necessary. Generally, you will be using Cascading Style Sheets and pasting the code into a field (for example, the Bio field) within your profile.

Keep background colors and foreground font colors compatible. The fastest way to make visitors leave your page is to force them to read dark blue type on a black background. Tie in the look of your profile with your image as an artist and with your Web site. Make it reflect the type of music you play.

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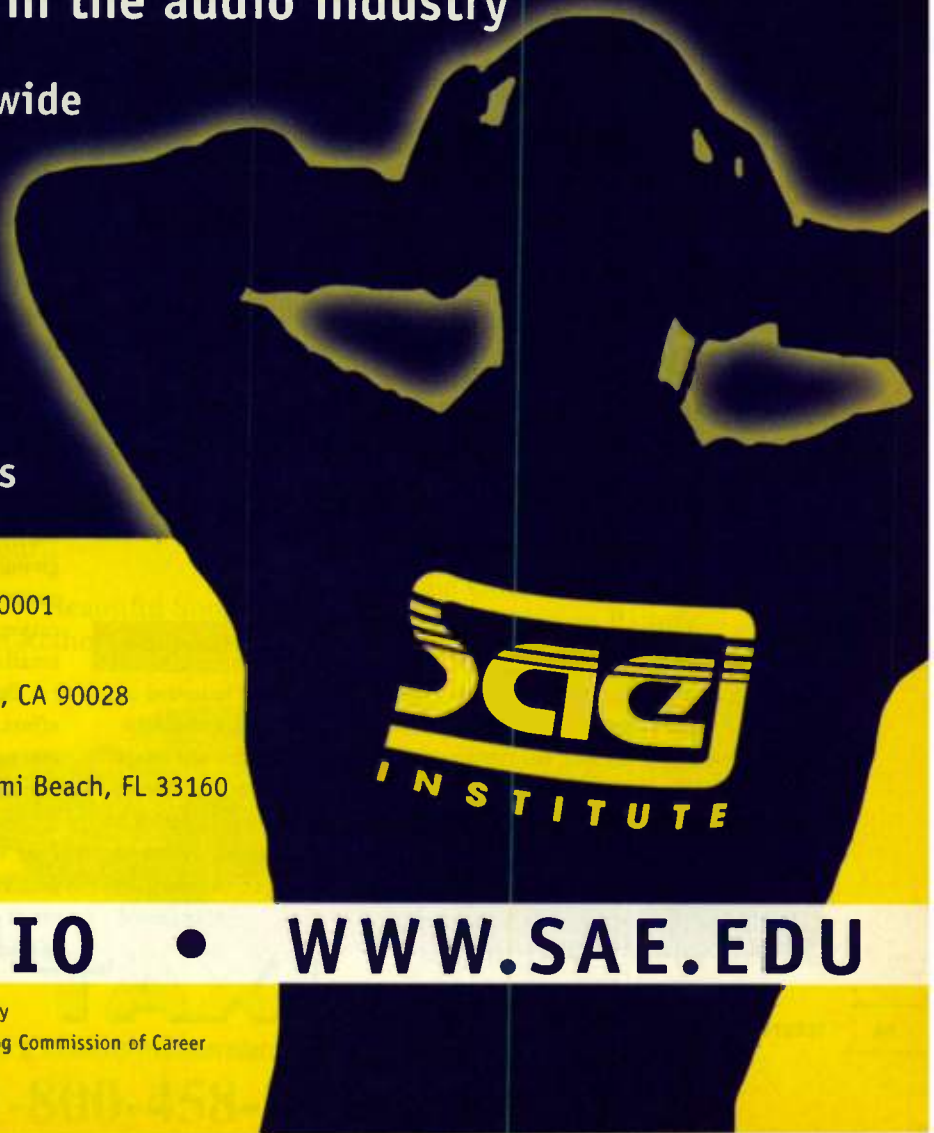

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CAKEWALK Dimension Pro 1.2 and Rapture 1.0 (Mac/Win)

For pure expression, these synths excel.

By Len Sasso

Dimension Pro 1.2 and Rapture 1.0 are the second and third virtual-instrument plug-ins released by Cakewalk since acquiring rgc:audio and the prodigious talents of René Ceballos. (The first synth in this line, Z3ta+, was reviewed in the February 2006 issue of EM, available at www.emusician.com.) Dimension Pro and Rapture have very similar architectures, the primary difference being that Dimension Pro is oriented toward multisample playback, whereas Rapture is designed for wavetable synthesis. Dimension Pro, with its 7 GB sample library, is aimed at the keyboard workstation market. Rapture, with its fat layering and built-in step generators, targets electronica and trance composers.

Dimension Pro and Rapture mark the first Cakewalk releases for the Mac; VST, AU, and RTAS plug-ins are provided. Windows users get VST, DXi, and RTAS plug-ins. Dimension Pro 1.2 for the Mac is in Universal Binary and will run on both PowerPC and Intel Macs. Rapture 1.1, in Universal Binary, will be released in the second quarter of 2006.

FIG. 1: Dimension Pro's Element control panel houses multisample and DSP controls (top), EQ and effects controls (middle), and graphical filter and LFO controls (bottom).

The Expression Engine

Both Dimension Pro and Rapture are based on Cakewalk's Expression Engine, developed by Ceballos, which supports a variety of synthesis techniques. In addition to playing standard audio files in the usual formats, the sound generators can read files in the SFZ format, also developed by Ceballos. SFZ files contain links to audio files together with information about how to play them. For example, an SFZ file can contain the key- and Velocity-zone information for multisample playback.

SFZ files go far beyond simple multisample playback,

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

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however. They can direct the Expression Engine to interpret single-cycle-waveform files as wavetables, they can tell the Expression Engine to treat impulse files as wave guides for physical-modeling synthesis, and they can turn on Expression Engine effects such as the Body/Damper Resonance effect that Dimension Pro uses for grand-piano samples.

Because both Dimension Pro and Rapture use the Expression Engine, each can use the other's SFZ content. For instance, Dimension Pro can play the more than 200 wavetables that come with Rapture. Conversely, Dimension Pro includes a folder of wave guide presets that can be loaded into Rapture. In short, the Expression Engine enables a lot of synergy between these two synths.

Because no dedicated SFZ editor is available at this time, SFZ files can be created and edited only in a plain-text editor. Their format is fairly simple, however, and there's a detailed description of SFZ version 1 at www.rgcaudio.com/sfzformat.htm. Version 2 adds a few bells and whistles, but most can be deduced by examining the SFZ files in the Dimension Pro and Rapture libraries. For example, it took only a few minutes to add the Body/Damper Resonance effect to the SFZ file for one of the factory guitar presets or to convert an impulse response file I had lying around into a wave guide SFZ file.

Elements

Dimension Pro and Rapture have multiple sound generators called Elements; Dimension Pro has four and Rapture has six (see **Fig. 1**). Elements can be layered or

Dimension Pro and Rapture mark the first Cakewalk releases for the Mac.

spread across the first few MIDI channels for multitimbral operation. You can turn Elements off to save CPU resources, and you can copy and paste settings between Elements as well as save individual Element setups to disk. Unfortunately, you cannot freely assign Elements to MIDI channels to, say, create a multitimbral patch in which each voice uses two Elements.

An Element starts with a sample player, implemented by the Expression Engine just described. The sample player feeds a DSP section followed by a 3-band EQ and an effects section. Rapture's DSP and effects sections differ slightly from Dimension Pro's, and Rapture's are a bit more extensive.

Instead of a modulation matrix, each Element in Dimension Pro and Rapture has dedicated envelope generators and LFOs for filter cutoff, filter resonance, pitch, amplitude, and pan position. The envelope generators support an unlimited number of breakpoints, adjustable segment shapes, and looping. The LFOs offer numerous waveshapes and optional host-tempo sync.

As you might expect of an electronica-oriented synth, Rapture also comes loaded with step generators. As with the envelope generators and LFOs, there's a separate step generator for each parameter of each Element (see **Fig. 2**). You can have as many as 128 steps in a generator pattern, and the step generators can be synced to host tempo. Unfortunately, step size is fixed per generator pattern, and changing the number of steps changes the step size automatically, making it impossible to create odd-meter patterns while synced to host tempo.

FIG. 2: Rapture's modulators include a dedicated step generator for pitch, filter cutoff, filter resonance, pan, and amplitude.

Hands On

Envelopes, LFOs, and step generators are great, but for many things, there's nothing quite like direct control. Both synths offer a MIDI modulation matrix as well as MIDI Learn. Using the matrix, you can assign 16 MIDI controllers to virtually any synthesis parameter. MIDI Learn allows you to target any knob or button, and offers both range and polarity settings. For example, with a couple of clicks you can set up a mod wheel to crossfade between two Elements.

Both synths also offer an onscreen X/Y Controller. Rapture's is set up in the modulation matrix, whereas Dimension Pro's is solely for mixing its four Elements. A handy Desaccel numerical introduces hysteresis for smoother control using the mouse.

Cakewalk has made these synths extremely keyboard- and mouse-friendly. Once you focus on a knob or numerical by clicking on it, you can use a mouse's scroll wheel to

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adjust its value. Double-clicking resets knobs and numericals to their default values, and the Arrow keys can be used to adjust the value. When you focus on a button that is part of a group—the Element selector buttons are an example—the top-row number keys or the Right Arrow and Left Arrow keys select among the buttons in the group.

From the Top

As mentioned, each Element's signal path begins with a sample player. Both the Dimension Pro and Rapture players allow you to set the Element's MIDI note, Velocity, and Pitch Bend ranges as well as its transpose in semitones, detuning in cents, and key tracking.

Key tracking is set in cents per key within a range of –200 to 200; negative values invert the keyboard. For example, a setting of –50 makes the keyboard track downward in quarter-tone steps, and a setting of 200 makes it track upward in whole-tone steps. For those who want to go beyond equal-tempered tunings, both synths support the Scala microtuning format, and more than 3,000 Scala tuning files are provided.

The Dimension Pro sample player's Shift numerical is used to change the tuning of each sample in the multisample map without shifting the mapping. You can use that in conjunction with the Transpose numerical, set to an equal but opposite value, to radically alter the timbre of a multisample—think Munchkin effect (see **Web Clip 1**).

Rapture has a few interesting sample-player tricks of its own. You can set the phase at which a waveform starts when a key is pressed. Using different phases in different Elements greatly expands the waveform palette. A Multi button activates multiple oscillators, which are spread across the panorama and detuned in accordance with the Detune setting. The Ring Modulation button activates a second oscillator to ring-modulate the first, and their relative tuning is controlled by the Detune parameter. You can also set any Rapture Element to ring-modulate the outputs of all lower-numbered Elements. As you might imagine, setting Element 6 to ring-modulate Elements 1 through 5 really fattens things up.

FIG. 3: Rapture's global effects chain starts with two multi-effects in series. Those are followed by stereo step generators for amplitude, a 3-band parametric EQ, and a master multi-effect.

DSP, EQ, and Effects

The next stage in an Element's signal path is the DSP section. Dimension Pro has a fixed DSP chain consisting of a LoFi effect followed by a multimode filter, which is followed by a Drive module. Rapture offers the same LoFi effect, two multimode filters, and a slightly different Drive module. Those four modules can be reordered in virtually any serial configuration, or Filter 1 and LoFi can be placed in parallel with Filter 2 and Drive. For example, you can use parallel or series highpass and lowpass filters to create variable-width notch or bandpass filters.

The filters, which are the same for both synths, offer 16 modes or can be bypassed. Lowpass, bandpass, highpass, and band-reject filters come in 1- and 2-pole versions. The 1-pole versions are nonresonant and are perfect for gentle timbral coloring. A 1-pole allpass filter is useful for phase adjustments. A 2-pole peak filter adds a 6 dB boost to the 2-pole bandpass filter's 12 dB rolloff. For more dramatic filtering, you can use the resonant 4- and 6-pole lowpass and highpass filters. Finally, there is a comb filter for phasing-style effects and a pink filter for slightly darkening the timbre.

The Drive modules feature five overdrive types: Tube, Soft, Mid, Hard, and Asymmetric. Shape and Drive knobs control overdrive amount, and Dimension Pro's Tone control adds a touch of lowpass filtering. The LoFi effect consists of separately activated bit-reduction and downsampling stages.

Each synth's signal path ends in a 3-band parametric EQ followed by a multimode insert effect. The 24 modes are divided into three categories: delay, reverb, and distortion. Controls differ according to the selected effect. The reverb and distortion effects were initially available only in Rapture and are a welcome addition to Dimension Pro 1.2.

Global Effects

Running multiple Elements with individual effects processing can be CPU intensive. When you don't need individual effects, you can turn them off and use the global send effects. Dimension Pro has a fairly basic

PRODUCT SUMMARY

CAKEWALK Dimension Pro 1.2

software synthesizer
$359
$99 for Project5 version 2 users

PROS: Large multisample library. Lots of envelope generators and LFOs. Computer-keyboard- and mouse-friendly. Excellent MIDI remote implementation.

CONS: No multisample editor. Multisamples vary in quality. Can be CPU hungry.

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global effects setup consisting of a modulation effect followed by a reverb. Rapture's global effects section is more complex. It begins with two multi-effects processors. Those feed separate step generators for right- and left-channel amplitude. The final stage consists of a 3-band parametric EQ and a master multi-effect. Each of the three global multi-effects is identical to the Element multi-effects (see Fig. 3).

In another CPU-saving gesture, both synths allow you to chain the output of one Element into the signal path of the next at the EQ input. You can use that to process several Elements with the same EQ and insert effect. You can also use it to process a single source with several insert effects. Of course, that doesn't save CPU, but it makes for more flexible effects processing.

From the Factory

On the Element level, Dimension Pro and Rapture are fairly easy to program, and they are similar enough that once you know one, you pretty much know the other. But having multiple Elements and the consequent proliferation of envelope generators, LFOs, and effects still leaves you with quite a bit of programming. Cakewalk has provided a large preset library for Dimension Pro, which is especially welcome in view of the lack of a multisample editor. Rapture's preset library is more modest but still useful.

Presets are selected from a separate Browser window (see Fig. 4). The Libraries are organized in banks by category, and you can use the Arrow keys to step through the presets in a bank and use the Return key to load the selected preset.

Dimension Pro's 25 banks cover the major instrument categories. The collection is rounded out with several banks of ambient sounds (Dimensions, Pads, and Orchestral Scapes), keyboard splits, grooves, and genre-specific banks (Techno & Trance, Electronica, Ethnic, and World).

Grooves and Slices

Groove multisamples contain sliced audio files with the slices mapped to consecutive notes starting with C3 (MIDI Note Number

PRODUCT SUMMARY

CAKEWALK Rapture 1.0

software synthesizer
$259

PROS: Lots of envelope generators, step generators, and LFOs. Computer-keyboard- and mouse-friendly. Robust global effects section. Excellent MIDI remote implementation.

CONS: Step generators have fixed step size. Preset library could be larger.

FEATURES	
EASE OF USE	
QUALITY OF SOUNDS	
VALUE	

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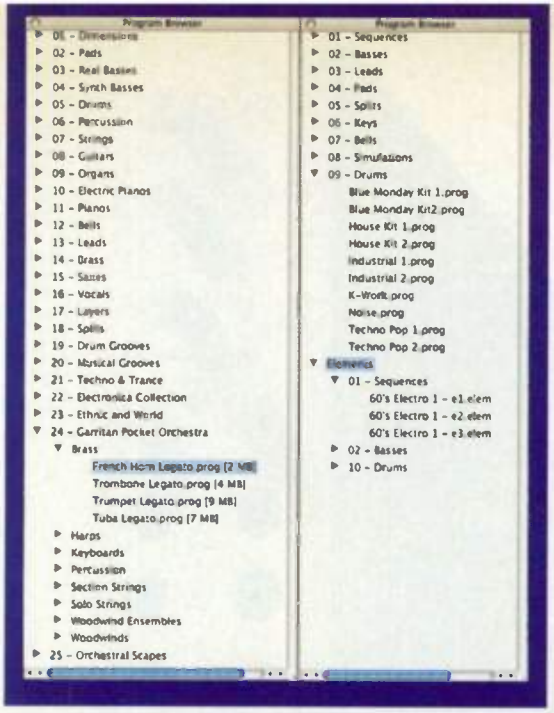

FIG. 4: The handy file browsers allow you to select presets with the computer keyboard's Arrow keys, and then load them by hitting the Return key.

60). The octaves below C3 trigger the entire groove using classic-style pitch-shifting and time-stretching. C1 (Note 36) triggers the groove at its recorded pitch and tempo. In a very nice touch, you can drag-and-drop a MIDI sequence to play the individual slices with the correct rhythm from the Dimension Pro graphical user interface to a MIDI track in the plug-in host. Dimension Pro can also load and automatically map Propellerhead REX2 files.

In addition to the traditional synthesized sounds—basses, leads, pads, keys, and so on—Rapture's preset library contains a few drum kits, several Element patches, and 30 synthesized simulations of acoustic instruments. My favorite bank, Sequences, contains 169 presets illustrating the use of the step generators (see **Web Clip 2**).

Both Dimension Pro and Rapture are good values. Dimension Pro's keyboard workstation–style collection of sounds covers a lot of territory, and you can quickly create new sounds by mixing and matching existing Elements. Rapture is most interesting for its step generators, and again, mixing and matching Elements is the quick road to creativity. Because of the synergy between these two synths, it would be nice if there were a bundled price.

EM WEB CLIPS

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

FIG. 1: The Saffire bundles a 2-in/8-out audio interface with a software control panel and accelerated reverb, compression, EQ, and guitar-amp simulation plug-ins. The compact device handles 24-bit audio at sampling rates as high as 192 kHz.

FOCUSRITE Saffire (Mac/Win)

A desktop audio interface with onboard DSP.

By Doug Eisengrein

The Saffire is a digital audio interface that features two input channels, eight output channels, S/PDIF and MIDI I/O, and a collection of plug-ins that take advantage of its onboard DSP. The hardware is housed in a 24-bit, 192 kHz–capable minitower powered by FireWire. The Saffire is compatible with Mac OS X and Windows XP, and it includes Steinberg Cubase LE software. Two features that make the Saffire stand out are its hardware-accelerated plug-in suite and twin Focusrite mic preamps.

Outside the Box

The lower section of the Saffire's front panel hosts two balanced ¼-inch TRS inputs, two balanced XLR inputs with a global 48V phantom power switch, and two ¼-inch stereo headphone jacks with discrete level knobs (see **Fig. 1**). In the middle are a Monitor Level knob with Mute and Dim buttons (the latter instantly reduces levels by –12 dB), S/PDIF In and MIDI I/O activity indicator LEDs, and a button to enable MIDI Thru. The front panel's top section has two Input Gain knobs, each with three LEDs that indicate input level, and a Line/Instrument button with LEDs to indicate your selection. For each channel, only one of three input options—mic, line, or instrument—is available at a time.

The rear panel sports two 6-pin FireWire ports, eight balanced ¼-inch TRS outputs, MIDI I/O, S/PDIF I/O, and a power-supply terminal (see **Fig. 2**). With eight outputs, the Saffire can handle multichannel or surround sound. Dual FireWire ports allow you to daisy-chain additional FireWire devices.

Hardware-hosted compression, reverb, amp modeling, and EQ, along with VST and AU plug-in duplicates, are included. The Saffire's software component, SaffireControl (Mac/Win), provides configuration and routing controls and serves as the user interface for all the Saffire's hardware-powered effects.

FIG. 2: The Saffire's rear panel has analog audio and S/PDIF jacks, as well as FireWire and MIDI ports. The power terminal attaches to the included DC adapter, but because the Saffire is normally powered by the FireWire bus, the adapter is required only for 4-pin FireWire connections.

I tested the Saffire on a 1.33 GHz Apple iBook G4 running Mac OS X 10.4.3 with Apple Logic Pro 7. I also installed the Saffire's drivers on a 2.4 GHz Intel Celeron–based PC running Windows XP. My Mac recognized the Saffire immediately when I plugged it in. On both Mac and PC, automated wizard programs helped me install the software and authorize the plug-ins.

Adventures in Software

Although SaffireControl is a formidable application, its labeling is a bit arcane. A single window is divided into three sections containing controls for the input stage, for sequencer tracks, and for mixing and processing through all the hardware outputs (see **Fig. 3**). The second section contains level faders for ten sequencer tracks; although they might prove useful for some users, I didn't have any need for them.

The application's input-stage controls apply to any pair of analog inputs (mic, line, or instrument). Both channels supply a welcome addition: a long, tricolored level meter that extends the 3-LED meter on the hardware's face. Two other pairs of level meters display the processed and direct signals side by side. The direct meter doubles as a S/PDIF-input meter, but I wish the user interface were large enough to accommodate discrete meters for everything. Also present are buttons for launching the compressor and your choice of either the EQ or the guitar-amp simulator plug-in, as well as another button for swapping the order of the processors in the signal path—a nice touch. The Stereo Link button, which joins all the channel controls, is useful for stereo sources.

In the hardware-outputs control section, all the controls are configured in five stereo pairs. Each channel pair provides numerous options, such as the ability to custom-mix the input levels, turn on and adjust the reverb, custom-mix the software-output levels, fade between the direct input and the software return, and add or subtract master gain. This section confused me at first, because certain labels such as Output Mix and Input Mix sit right next to each other. Once I'd read the clearly written PDF manual, however, I found the array of controls useful and wide-ranging.

DSP Power

The plug-ins are basic in appearance, and all but one sport minimal controls, yet each sounds surprisingly good. The four hardware-powered plug-ins—straightforwardly named Compressor, EQ, Amp Simulator, and Reverb—are duplicated by unaccelerated plug-ins.

Compressor includes presets for typical scenarios, such as recording vocals, bass, or piano. Though the presets sound reasonably appropriate, the compressor's Advanced mode supplies adjustable input and output levels with corresponding meters; Threshold, Ratio, Attack, Release, and Gain controls; and a Bypass button. Compressor is precise and capable of a broad range of effects. I was able to add just a subtle touch to quiet the vocals and more to really squash the bass.

EQ is a fully parametric 4-band equalizer. In Template mode, four dials control gain for four predetermined frequency bands. Advanced mode switches the bands to parametric, with Frequency, Gain, and Q controls for each. There is ample frequency overlap between bands, and lowpass, highpass, shelving, and bandpass options are featured on each end. EQ offers precise frequency-carving controls, and its sound is very transparent.

The simple Amp Simulator features British, American, Combo, and Bass tones, each adding a distinct color. You can dial in high, mid, and low boost, and a generic Drive control adds distortion. Though it doesn't stack up to a hardware amp, I was able to add some interesting grit to an Ibanez

PRODUCT SUMMARY

FOCUSRITE Saffire

FireWire audio interface
$499

PROS: Clear sound. Low noise and abundant gain. Easily transportable. Solid construction. FireWire bus power. Balanced ins and outs. Nice plug-in suite.

CONS: Cramped software user interface. Low headphone volume. Only two simultaneous inputs. Global phantom power.

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Reverb furnishes three controls: Room Size, Diffusion, and Tone. Despite its sparse interface, I liked the reverb plug-in very much. It might be a bit too pristine and glassy-sounding for some tastes, but I was able to simulate everything from wooden-sounding drum rooms to massive cathedral spaces, and the dry signal was never obscured.

FIG. 3: SaffireControl furnishes a software front end for the hardware. It gives you complete control of input and output levels, signal routing, and plug-ins, allowing the Saffire to function as either a mixer or a standard sound card.

Hardware Trials

I performed my first test using a preexisting multitrack session in Logic Pro 7. I routed all the tracks to outputs 1 and 2 and monitored the Saffire through my mixing console. The sound was initially choppy, but changing the buffer setting from 64 to 256 samples corrected the problem. I connected the Saffire's six remaining analog outputs to the mixer and randomly reassigned the tracks' output through all four stereo pairs. When I soloed a track and played it through each of the four output pairs, the volume and audio quality sounded comparatively consistent. I was impressed with the Saffire's sound; I could hear everything in the mix clearly at low levels, and with the Monitor control set to maximum (with an additional 6 dB from Logic), the Saffire had lots of undistorted headroom despite hot output levels.

Next, I tried out both headphone jacks with a pair of AKG K 240 Studio headphones, and the sound was clear, though not especially loud. By default, headphone output 1 mirrors line outputs 5 and 6, and headphone output 2 mirrors 7 and 8. Using the SaffireControl software, you can assign custom mixes of any output combination for additional control; such flexibility scores points, but I yearned for more headphone volume.

I tested the digital I/O by simultaneously routing all tracks to S/PDIF, connecting the output to the input, and recording a few tracks. When I compared the level and quality of the original signal and the recordings, I heard no differences. I performed a similar test by connecting analog outputs 1 and 2 to the line inputs, and I recorded a few overdubs using a Yamaha P-120 piano, again with positive results. At minimum gain, I could clearly hear low-level audio. Even at maximum gain, just before clipping, the sound was robust, with very low noise. Gain sweeps were smooth and wide, and the Saffire's knobs felt firm to my fingers despite their plastic mold. The Saffire sounded quite transparent; in fact, the P-120 recordings sounded practically indistinguishable from the actual instrument.

SAFFIRE SPECIFICATIONS

Analog Audio Inputs	(2) balanced XLR mic with 48V phantom power (switchable); (2) balanced ¼" TRS line/high-impedance instrument (switchable)
Analog Audio Outputs	(8) balanced ¼" TRS; (2) ¼" stereo headphone
Digital Audio I/O	coaxial S/PDIF, 16/24-bit, 44.1 kHz–192 kHz
Data I/O	(1) MIDI In, (1) MIDI Out/Thru (switchable), (2) S400 FireWire
Frequency Response	20 Hz–20 kHz ±0.1 dB
A/D Dynamic Range	104 dB (A-weighted)
D/A Dynamic Range	110 dB (A-weighted)
Power	6-pin FireWire; DC adapter (with 4-pin FireWire)
Dimensions	6.69" (W) × 6.69" (H) × 2.55" (D)
Weight	2.43 lbs.

Preamplified

I switched an input to Instrument mode, plugged in the Ibanez bass, and recorded an overdub. The Saffire had more than adequate gain and headroom. Across several takes with various pickup settings, the Saffire captured a full tone with plenty of low-end punch. Subtle nuances of finger slides and note sustain came through clearly, and achieving proper gain levels was easy.

To test the mic preamps, I made three 24-bit, 96 kHz recordings with an Audio-Technica

AT2020 large-diaphragm condenser mic. The sources were tenor male vocals, hollow wind chimes, and the Ibanez bass played through a Tube Works solid-state combo amp with a 12-inch speaker. In all three cases, the preamps captured subtle details and exhibited a smooth, wide frequency range. Vocals were lively and up-front. The bass, though not as punchy as in direct takes, sounded thick and round. The chimes were crystalline, with ample high-frequency sparkle and long sustain.

I also recorded a silent track with a stray XLR cable connected and the input gain set at about three-quarters. On playback, I heard very little difference between the recorded clip and the surrounding empty space—a testament to the Saffire's low noise.

Interface-off

I compared the Saffire's preamps to those of three competing interfaces I had on hand: an M-Audio ProjectMix I/O (a FireWire interface and control surface that costs about three times as much), an Echo Layla 3G (a PCI interface for about the same price), and an Edirol UA-25 (a USB interface for about half the price). Though the Edirol had a good frequency range, it was noisier than the Saffire, and in fact, the Saffire was quieter than all the others.

The Echo and the M-Audio waged good battles, though the Saffire had a level of detail that was really noticeable, especially in the midrange. I heard a bit more presence in the M-Audio's bass response, whereas the Echo's was fairly equal, yet the Saffire's clarity still stood out. Each interface displayed positive traits and its own distinct color. I won't say that the Saffire was my absolute favorite, but it certainly held its own against the competition on hand.

Like a Champ

Focusrite makes a strong case against its competitors with the Saffire. The unit sounds great while adding no distinguishable noise, is well built, provides ample outputs, and includes an invaluable set of hardware-accelerated plug-ins. The Focusrite preamps nicely captured everything I threw at them, and the Saffire's signal routing, combined with the software, offers plenty of flexibility.

I did end up with a wish list, including louder headphone outputs, discrete phantom-powered preamps, a separate S/PDIF level meter in SaffireControl, and additional inputs. But considering the Saffire's price, I'm not complaining much. Had I not purchased a new audio interface just prior to writing this review, I would seriously consider the Saffire.

Doug Eisengrein is an electronic-music composer and multimedia developer in the San Francisco Bay Area. He frequently contributes to Remix *and* EM.

FIG. 1: Compared with the previous model, Access Music's Virus TI has twice the processing power, more than twice as many memory locations, and loads of new features.

ACCESS MUSIC Virus TI Desktop

Bridging the gap between hardware and software.

By Geary Yelton

If you know your way around virtual analog synthesizers, you're probably familiar with the Access Virus, if only as an object of desire. Now in its fourth generation, the German-made Virus isn't just one synth; it's a product line that has been earning the respect of electronic musicians since 1997. The Virus TI (which stands for Total Integration) is available in three form factors: the 61-key Virus TI Keyboard ($2,765), the 37-key Virus TI Polar ($2,765), and the instrument I had for review, the rack-mountable Virus TI Desktop ($1,995). All three models share a common hardware user interface, and for the first time, they offer a software user interface as well.

Compared with the Virus C, the Virus TI furnishes many improvements and some exciting new features (for more information on the Virus C, see the August and September 2003 issues of EM, available at www.emusician.com). Most important is the concept of Total Integration, which allows the instrument to serve as an essential element for computer-based recording. The Virus connects to a computer by means of its USB 1.1 port, which carries audio, MIDI, and control data. A software plug-in called VirusControl (Mac/Win) lets you control the Virus from within your audio sequencer. Total Integration gives you a choice of hardware- or software-based control with sample-accurate timing and a connection that automatically compensates for any system delay.

In addition to integrating hardware and software, the Virus TI delivers twice the polyphony of the Virus C, two cool new oscillator modes, expanded modulation capabilities, and a substantial increase in effects-processing power. It stores well over twice as many programs, with 16 rewritable ROM banks and 4 RAM banks, each containing 128 Single patches. Each patch can store its own arpeggiator settings, and you can edit and store Single patches within a Multi patch without affecting the source Singles. The Virus TI can also function as a 16-channel MIDI interface and a 2-in/6-out audio interface for your computer.

First Contact

Physically, the Virus TI Desktop bears a strong resemblance to the Virus C Desktop (see **Fig. 1**). Its 160 × 32-pixel LCD can show four lines of text and graphics, and its black-on-white rather than black-on-red color scheme improves viewing from different angles. The wood end panels are mahogany-stained instead of black. A few of the controls have been rearranged, and there are a few more buttons and LEDs; otherwise, the layout is almost identical. A BPM indicator LED constantly flashes the current tempo, and a Tap button has been added. When you turn most knobs, the LCD very conveniently displays the new value as well as the previous setting.

A small but significant addition is a dedicated Shift button; pressing it in combination with other buttons

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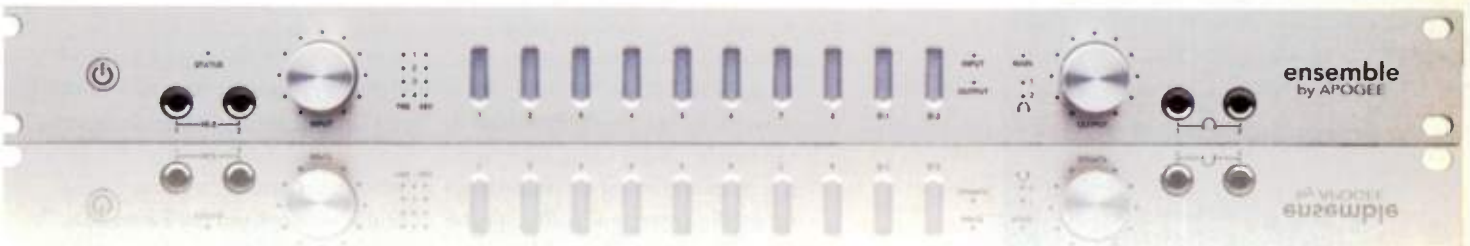

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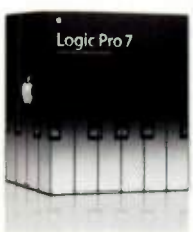

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and knobs lets you access supplementary functions, without extra front-panel controls. The labeling for Shift functions is in dark red ink, however, making it rather difficult to read in sections where the background is dark gray rather than black.

Like previous models, the Virus TI Desktop gets its power from a sizable lump-in-the-line transformer. To turn the power off, you must press both Transpose buttons simultaneously, which isn't very intuitive. Pressing one of several buttons turns the power on. Although the Virus TI is in Standby mode as long as it's plugged in (as indicated by a slowly blinking LED), it still takes about 13 seconds to power up.

Instead of the Virus C's Soft Knob 1 and Soft Knob 2, the Virus TI has three assignable knobs labeled Value 1, 2, and 3. By default, they are assigned to either the three most appropriate performance parameters for the current patch or whatever parameters you're editing in the LCD at the moment. Pressing the Shift button lets you use the three knobs to scroll through lists of Category, Bank, and Program selections.

On the backplate, a USB port and coaxial S/PDIF ports accompany the power connection, headphone jack, three MIDI ports, two unbalanced TS inputs, and six balanced +4 dBu TRS outputs (see Fig. 2). According to Access's specifications, the D/A converters support 24-bit sampling rates as high as 192 kHz, but Mac OS X's Audio MIDI Setup recognized only 16 bits at either 44.1 or 48 kHz.

The module ships with its rack ears unattached. When rackmounting the Virus TI Desktop, you can rotate the rear-panel connections 90 degrees with the aid of a Phillips screwdriver and a hex wrench. The procedure requires a bit of surgery and involves disconnecting a multiport connector. When the operation is complete, the jacks and ports will be on the baseplate, opposite from the front panel, rather than on top when rackmounted. That allows you to easily make connections without needing to leave room in your rack above the Virus.

The Virus TI consumes all the bandwidth on a USB connection. According to the user manual, connecting the synth to a USB hub is not recommended, and in fact, such a connection doesn't work at all. Nor does it work when you connect the Virus to any USB port other than one mounted directly on your computer. Considering that I have 29 USB devices and my Power Mac G5 has only 3 USB ports, connecting the Virus proved a bit of a challenge. Using a powered hub made no difference, so I had no choice but to dedicate one of my computer's ports to the Virus. I ended up disconnecting several devices and reconnecting them individually whenever they were needed.

Enhanced Synthesis

In addition to Classic mode, which generates conventional modeled-analog Virus waveforms, Oscillators 1 and 2 offer two modes that are new to the Virus family: HyperSaw and Wavetable. HyperSaw is a sawtooth generator that can produce as many as nine waveforms at the same time. You can sync or detune those waveforms and change their number (called Density) in real time, giving you loads of control over the timbre's thickness. You haven't heard *thick* until you've layered 16 HyperSaw oscillators in Unison mode, each with 9 sawtooth waves, and added 8 suboscillators for a total of 152 audio sources per note (see **Web Clip 1**).

Wavetable mode works a lot like wavetable synthesis on the Dave Smith Instruments Evolver and a handful of other digital synthesizers. For either oscillator, you can select from 72 preset wavetables with names such as Glass Sweep and Robot Wars. An Index control lets you select one of 127 smoothly cross-faded waves within the wavetable. If

you route one or more modulators to scan the wavetable, the Virus can produce an impressive variety of animated sounds that range from glassy and vocal-like to brittle and abrasive.

Previous Viruses had two main operating modes: Single, which produced only one timbre on a single MIDI channel, and a multitimbral mode called (appropriately enough) Multi. In the Virus TI, what was previously called Multi mode is now called Sequencer mode. Whenever you're using VirusControl, the Virus is in Sequencer mode. Instead of referencing Single patches as Sequencer mode does, a new Multi mode lets you edit individual Parts without affecting the original Single patches and then save 16 Multis without any dependencies. This is a big deal because it's very much like having 16 monotimbral synths that you can edit individually without changing any parameters stored within their source patches.

For the most part, the Virus TI's impressive range of effects is identical to that of the Virus C, though happily, you can now control reverb and delay independently. Previously, the reverb, delay, and envelope follower were shared within a Multi, but now each Single patch in Multi mode has

PRODUCT SUMMARY

ACCESS MUSIC **Virus TI Desktop**

analog modeling/wavetable synthesizer
$1,995

PROS: Spectacular sounds. Capacious timbral storage. Hardware or software control. Well-designed user interface. Easy-to-read display. Doubles as external effects processor and audio interface.

CONS: Requires direct USB connection. Manual lacks depth. Some hard-to-read labeling. Software is unfinished.

FEATURES
EASE OF USE
QUALITY OF SOUNDS
VALUE

MANUFACTURER
Access Music
www.access-music.de

its own settings for those effects, greatly increasing the instrument's flexibility. No matter what mode the Virus is in, all effects types are simultaneously available. And thanks to the Virus's audio inputs, the effects can process external sources, too.

Pulling the Plug-in

Access Music gained several years of experience creating software versions of the Virus for Digidesign's TDM and TC Electronic's PowerCore platforms. Like TDM and PowerCore expansion cards, the Virus TI is based on the Motorola (now Freescale) 56000 series of DSP chips. Consequently, writing software to control Virus hardware must have been a natural next step for its programmers. VirusControl borrows several design features and control capabilities from the soft-synth plug-ins, but its user interface is more attractive and sophisticated.

VirusControl operates as a VST plug-in in Windows 2000 and XP and as a VST or AU plug-in in Mac OS X; an RTAS version is forthcoming. The plug-in requires a minimum 60 MB of free disk space and 512 MB of RAM.

Because the Virus's firmware and VirusControl software are updated frequently, I logged on to Access Music's Web site and downloaded the latest version, 1.0.9. Installation was straightforward and required that I restart the synth and my computer. I was using a dual-processor 2.3 GHz Power Mac G5 with 4 GB of RAM running Mac OS X 10.4.5, and my host software was Apple Logic Pro 7.1. (Digital Performer users take note: MOTU and Access Music are working together on a solution to some latency compensation issues.)

VirusControl does something that soft-synth users usually take for granted: when you save a song on your computer, it remembers all your synthesizer settings. It does require keeping the Virus TI in Sequencer mode, however, and it automatically switches to that mode whenever you run the plug-in.

FIG. 2: The Virus TI is the first generation of Viruses with connections for coaxial S/PDIF and USB, in addition to analog audio and MIDI ports.

Poring Through Pages

VirusControl 1.0.9 has nine edit pages that give you many perspectives on the Virus TI's programming depth. Every time you make a change on the hardware's front panel, that change will occur simultaneously in the plug-in window. Whenever the cursor is over any knob, button, or menu in VirusControl, an Information Bar at the bottom of the graphical user interface displays the parameter name and its value.

To the left of each page is the Part Control Bar, which determines which of the 16 Parts is affected by changes you make in the nine pages. From the Part Control Bar, you can load, save, and mute Single patches and set their volumes and pan positions.

FIG. 3: VirusControl is a software plug-in that integrates the Virus TI with audio sequencers. The Filter page offers control of two filters, two envelope generators, and the amplifier.

The Easy page gives you access to the parameters you're most likely to use, such as filter cutoff and resonance, effects sends, and amplitude attack and decay. On the Oscillator page, you can control the same parameters that you control with real knobs and buttons in the hardware's Oscillators and Mix sections. The Common page presents pull-down menus and fields you can click-and-drag to change soft-knob assignments, categories, bend range, and other patch-specific parameters that don't belong on the other pages.

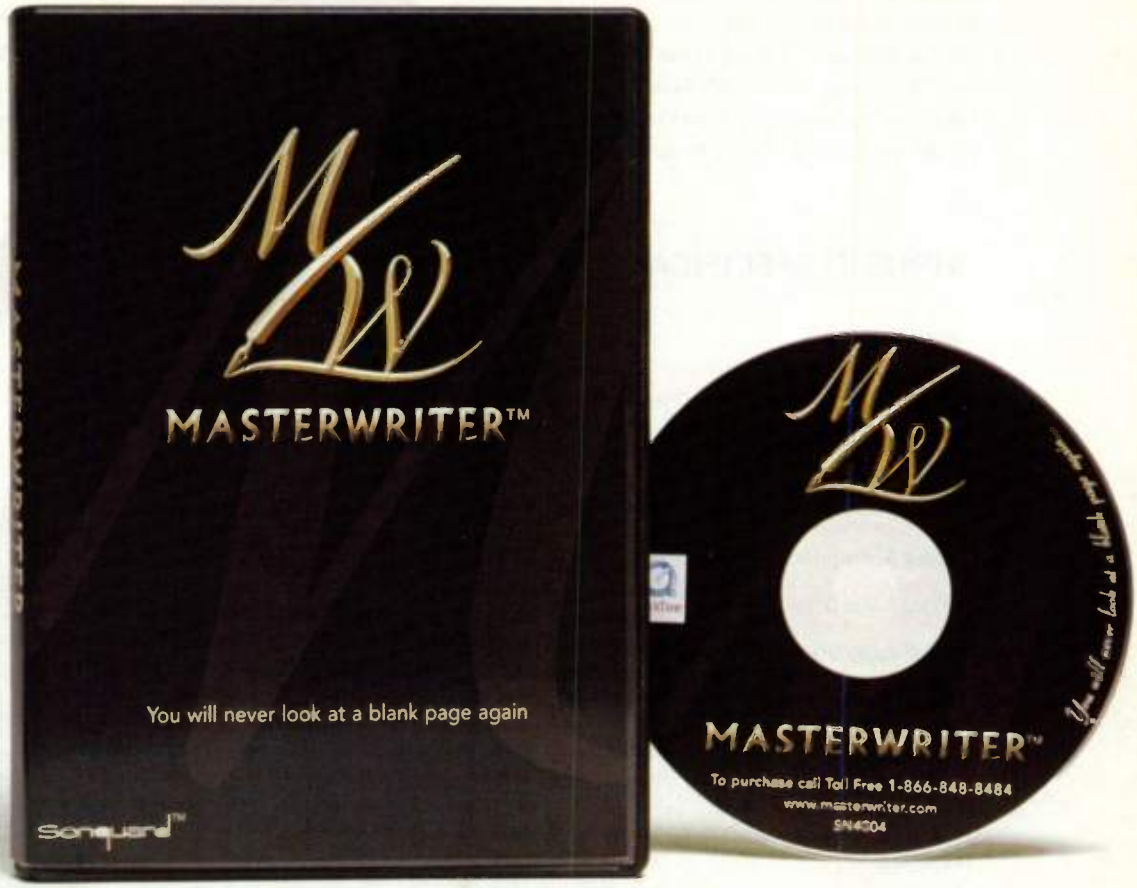
VirusControl's Filter page reproduces all the hardware's controls for the amplifier and both filters (see Fig. 3). Envelopes are graphically displayed with breakpoints you can click-and-drag. When you're editing filter parameters using hardware alone, they're shown as text in the instrument's LCD; that makes it especially gratifying to see them all graphically displayed at the same time in a nice, big window.

Likewise, I much prefer changing LFO parameters on the LFO page to using the hardware's LCD, especially when I'm using real knobs with my eye on the computer display. The LFO page supplies controls for all three LFOs and routing for the most common LFO modulation destinations.

The FX page furnishes knobs, buttons, and pull-down menus for controlling all the effects parameters (see Fig. 4). Each effects type has its own section, giving you instant access to comprehensive controls for the vocoder, ring modulation, analog boost, and seven other effects types.

The Matrix page provides a clear visual interface for routing modulation sources and destinations. It contains six identical slots, each with a pull-down menu for selecting from 27 modulation sources. Additional pull-down menus let you select three destinations for each source. Each destination has a slider to govern the modulation amount, along with a handy bar graph that lets you see that destination's mod depth at a glance.

The Arp page makes creating user arpeggios much quicker and more intuitive than if you were to rely on the hardware alone. Unlike previous generations of Viruses, the Virus TI lets you create user arpeggio patterns, with one user pattern for each Single patch. Using either hardware or software, you can select an arpeggio



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from a list of 63 presets, and one user pattern as well as its direction, range, beat resolution, maximum note length, and swing factor. If you want to set the number of notes in a pattern, add rests, or individually specify the duration and Velocity of each note, you'll need to use the Arp page's Pattern Editor. One surprise is that when you begin creating a user arpeggio from scratch, the default resolution is a very fast 1/128 note.

Access expects Virus TI software version 1.1 to be available by the time you read this. Until then, one page that's missing is the Remote page, which will allow you to define and edit templates for using the Virus TI as a universal control surface. According to Access, it will allow the hardware to store as many as 32 preset scenes, each with up to 32 named knob assignments. Access plans to provide a library of remote templates for third-party software.

VIRUS TI SPECIFICATIONS

Sound Engine	analog modeling, wavetable synthesis
Maximum Polyphony	(80) notes, stereo
Multitimbral Parts	(16) in Multi or Sequencer mode
Memory Locations	(16) flash ROM banks × (128) Singles; (4) RAM banks × (128) Singles; (16) Multis
Analog Audio Inputs	(2) unbalanced ¼" TS, –10 dBV
Analog Audio Outputs	(6) balanced ¼" TRS, +4 dBu; (1) ¼" stereo headphone
Digital Audio I/O	coaxial S/PDIF; 16/24-bit, maximum 48 kHz
Data I/O	(1) USB 1.1, (1) MIDI In, (1) MIDI Out, (1) MIDI Thru
Control Inputs	(2) assignable ¼" TS pedal inputs (Keyboard and Polar only)
A/D/A Conversion	16/24-bit; maximum 48 kHz A/D, 192 kHz D/A
Keyboard	Desktop: none; Keyboard: 61-note semiweighted; Polar: 37-note semiweighted; Velocity, Channel Aftertouch
Oscillators	Oscillators 1 and 2: Classic, HyperSaw, Wavetable waveforms; Oscillator 3: (4) modeled analog, (64) digital waveforms; Suboscillator: square, triangle waveforms; variable-color noise generator
Filters	(2) resonant multimode (lowpass, highpass, bandpass, bandstop); Filter 1 also has (4) analog-modeled resonant lowpass modes
LFOs	(3) with (68) waveshapes
Envelope Generators	(2) ADSSR (attack, decay, sustain, slope, release)
Modulation Matrix	(6) slots, each with (1) source, (3) destinations
Arpeggiator	(64) presets, (1) user pattern per Single
Display	160 × 32-pixel LCD
Effects	analog boost, chorus, delay, distortion, envelope follower, EQ, phaser, reverb, ring modulation, vocoder; (145) simultaneous
Software Component	VirusControl plug-in (VST for Windows; VST, AU for Mac OS X)
Dimensions	Desktop, 18.5" (W) × 3.2" (H) × 7.4" (D); Keyboard, 39.2" (W) × 4.6" (H) × 14.6" (D); Polar, 22.3" (W) × 4.4" (H) × 13.2" (D)
Weight	Desktop, 7.2 lbs.; Keyboard, 27.6 lbs.; Polar, 19.2 lbs.

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Just Browsing

One of VirusControl's most useful features is the Browser page (see **Fig. 5**). The Browser lets you view and rearrange the 2,560 patches stored in the Virus's memory banks and, thanks to the Public Library function, patches stored on your hard disk as well. You can quickly and easily navigate through the patch banks by clicking or by using your computer keyboard's Arrow keys. To replace a non-ROM patch with a new one, just drag it from one location to another—from a Public Library location to a RAM bank location, for example. Because patch locations aren't associated with their names, however, I do wish that selecting a Part would highlight its assigned patch in the Browser.

To try out new Virus patches, you no longer need to use your sequencing program to transfer Standard MIDI Files (SMFs) to your synthesizer. Instead, just copy them to the appropriate folder on your computer, and they'll appear in the Browser's Public Library the next time you run VirusControl. You can store literally thousands of patches on your hard drive in SMF format. When you find a keeper, just copy it over to one of the RAM banks. You can find plenty of Virus patches, including Access's complete libraries from previous models, online.

Version 1.1 will introduce two new Browser functions. Sort by Category will automatically build banks of sounds within the same category. Search by Name will instantly find patches on your hard drive whose names match whatever text you type in, no matter how few letters you type.

FIG. 4: You can specify all the effects parameters on VirusControl's FX page. Nine effects types are independently and simultaneously available to each Part, but only one Part at a time can use the vocoder.

Access the Future

The Virus sound, as always, is absolutely gorgeous. If you love electronic music, you're sure to love the Virus TI's tremendous range (see **Web Clip 2**). Its musical points of reference are all over the map, from Tangerine Dream and Kraftwerk to BT and the Chemical Brothers, from Pink Floyd and Kitaro to Brian Eno and Vidna Obmana. The Virus does well at imitating analog and digital synthesizers from just about any era, and it makes way for the future. Its palette has something for everyone, but its forte is generating electronic timbres—whether ambient, electronica, or

experimental—rather than emulating traditional instruments. Still, some very good acoustic simulations are on hand. However, as with any large patch collection, you'll find plenty of unusable sounds, too.

The instrument's ease of use has taken a big leap forward with the introduction of VirusControl. If you've become as accustomed to using soft-synth plug-ins as I have, you'll probably take to the Virus TI like a duck to plum sauce. You'll really feel as though you're working with a soft synth. The Virus TI Desktop also operates as a normal synth module, of course; for onstage use, you may never need a computer unless you want the convenience of rearranging patches in the Browser page.

I didn't experience many problems during my review. A few times, VirusControl apparently lost touch with the hardware, and I had to quit and restart Logic Pro to continue working. Once, I turned on the Virus and it froze up; it wouldn't play and its LCD remained blank. I had to unplug it from the power supply to shut it down. My only other complaint is that the user manual is too brief and doesn't go into enough detail on some points.

Unlike most hardware instruments, previous generations of the Virus have received ongoing firmware updates for free, so even older models have gained new features. I expect the same will be true of the Virus TI. If history is any indication, the Virus TI will have a hedge against obsolescence—bravo to Access Music for such outstanding service to Virus owners.

FIG. 5: On the Browser page, you can quickly load any Single patch stored in the Virus's memory or on your computer's hard disk. To rearrange them, just click-and-drag.

The Virus TI outperforms its forebears in every respect. It sounds wonderful, it's easy to use, and it offers some very cool new capabilities that other synths do not. If you're in need of a hardware synth—especially if you do a lot of sequencing—you can't go wrong with a Virus TI.

EM WEB CLIPS

EM associate editor Geary Yelton began playing and writing about synthesizers in the mid-1970s. He lives in Charlotte, North Carolina.

FIG. 1: The TonePort UX2 USB interface lets you plug in guitars, microphones, and line-level effects, and send the output to either analog or digital gear.

LINE 6 TonePort UX2 (Mac/Win)

More than just a USB guitar interface.

By Orren Merton

Several years after its release of GuitarPort, Line 6 has released its next generation of USB guitar processors—the TonePorts. Currently offered in two versions, the UX1 and UX2, the TonePorts each have a hardware front end that interfaces with a software processing rack of Line 6 guitar, amplifier, cabinet, and effects models. The TonePorts have significant new features, including one or two XLR microphone inputs with phantom power and, in the case of the UX2, a S/PDIF output.

The design of the TonePort system allows the UX1 and UX2 to straddle the middle ground between outboard hardware processors and software plug-ins. Your guitar signal passes through a hardware input and is then sent to your computer through USB, where it is processed by the GearBox software. Next it is routed to the recording software of your choice, and then back to the TonePort for monitoring.

This setup lets you monitor your signal through GearBox with minimal latency while setting your host software's buffer at a high level. That results in better host performance than you could achieve with a plug-in, which would require a lower buffer setting to avoid excessive latency.

The UX2, which has more I/O options than the UX1, is the focus of this review. The UX1's features are summarized in the sidebar "A Brief Look at the UX1."

Ports of Tone

The front panel of the UX2 (see Fig. 1) has two XLR microphone inputs and a button that globally turns on the phantom power. There are no pad switches. You get two ¼-inch instrument inputs: a high-impedance Normal input for passive electric-guitar pickups and a low-impedance Pad input designed for active pickups.

The UX2's twin VU meters light up when the interface is connected to a USB port and can display the input, send, monitor, or output level, depending on which is selected in the GearBox software. The meters do not have a numerical scale on them, just Min and

TONEPORT UX2 SPECIFICATIONS

Analog Inputs	(2) XLR mic, (1) high impedance, unbalanced ¼" TS instrument; (1) low-impedance, unbalanced ¼" TS instrument; (2) unbalanced ¼" TS line; (1) ¼" TRS stereo monitor; (2) ¼" footswitch
Analog Outputs	(2) balanced ¼" TRS, (1) ¼" TRS headphone
Digital I/O	(1) USB, (1) S/PDIF output
Maximum Mic Gain	50 dB
Signal-to-Noise Ratio	guitar in to USB: 110 dB; mic in to USB: 107 dB; line in to USB: 109 dB; USB to analog out: 107 dB
Dimensions	10" (W) × 3" (H) × 6" (D)
Weight	1.38 lbs.

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{ music production USB Automation/Transport controller }

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Max notations. There is a clipping LED for each meter to indicate peaks. The front panel also has a ¼-inch headphone jack.

Rear Ports

The rear of the UX2 has two unbalanced ¼-inch line inputs for line-level signals such as mixers and keyboards. Next to those are two ¼-inch jacks for standard momentary or toggle footswitches. You also get a ¼-inch Stereo Monitor In jack, which allows you to plug in and monitor external audio sources, and a pair of balanced ¼-inch analog output jacks.

The S/PDIF output operates at either 24-bit, 44.1 kHz or 24-bit, 48 kHz. Line 6 considers the TonePort to be 96 kHz compatible because the audio driver can upsample the audio to 96 kHz, but the S/PDIF output remains at the original 44.1 kHz or 48 kHz sampling rate. I would have preferred real hardware options for 88.2 kHz and 96 kHz.

If you connect the S/PDIF output to another digital device, the UX2 must be the clock master. That's because there is no S/PDIF or word-clock input on the UX2 to establish sync with another digital unit. I was disappointed by that, because I prefer using the more precise clock on my other digital audio gear. Nonetheless, the S/PDIF output is useful for standalone recorders, DATs, and other devices. The S/PDIF jack always outputs the same signal being sent to the analog outputs.

The top of the UX2 has four knobs (see Fig. 2): the Mic 1 Gain and Mic 2 Gain knobs give you 50 dB of adjustable gain. Near the headphone jack is the Phones knob to control headphone volume. Behind the Phones knob is the Output knob, which adjusts the volume at the analog outputs.

PRODUCT SUMMARY

LINE 6 TonePort UX2

USB recording and modeling interface
$269

PROS: High-quality modeled sounds. Intuitive GearBox software. Hum noise-reduction feature. Programmable footswitch jacks. Expandable sound set. ToneDirect monitoring allows high host-software buffer settings.

CONS: Can't process recorded tracks in GearBox. UX2 must be digital clock master. Can't use both instrument inputs simultaneously. No trim knobs for instrument inputs.

FEATURES
EASE OF USE
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Getting in GearBox

The GearBox software acts as the control center for the UX2 hardware (see Fig. 3). All of the TonePort's audio processing happens inside GearBox. Signals input through the UX2 get sent directly to GearBox. After that, the program sends them to the UX2's audio driver and analog outputs. The audio driver has four sends and one stereo return. The recording software that you're using, be it the included Ableton Live 4 Lite (Mac/Win) or any other recording software, then selects the four recording sends of GearBox as its audio input device.

FIG. 2: This photo shows the top and rear panels of the UX2. There are knobs to control mic gain, headphone volume, and output volume.

The UX2's monitoring scheme, called ToneDirect monitoring, lets you hear your guitar signal after GearBox processes it but before it has passed to your recording software. The latency is not as low as you'd get on a device that has direct hardware monitoring, but it's low enough (it was an acceptable 10 ms on my system) to let you set a high buffer in your host software.

GearBox allows you to address the UX2's inputs in a variety of ways. You can plug in two simultaneous mic or line sources, plug in one instrument and one mic, or input a single instrument, mic, or line-level source. You can't use both instrument inputs simultaneously. If you plug instruments into both, only the Normal input will be heard.

When using one of the dual-input configurations, you can choose the Dual Mono option, in which GearBox processes each signal separately. One signal is sent from send 1–2 and the other from send 3–4. I wish that GearBox had software returns so that you could route recorded tracks from your host through GearBox's processing.

GearBox has knobs to adjust the monitor signal to the TonePort, the record signal to the recording software, the pan position of the recording signal, and the master output volume to the UX2. There is also a handy button to boost the signal to your recording software by 18 dB. When the level going into my recording software was too low, I was able to fine-tune it by engaging the +18 button and lowering the record-volume knob. Also useful is the Mute Line Outs button, which shuts off the signal from GearBox to the UX2.

Another nice GearBox feature is its hum-reduction algorithm, which operates using a simple Learn button. Although I didn't have significant hum in my setup, it silenced the little that I did have.

Listening In

If the UX2 is your only audio interface, you can hook it up to your audio monitors and hear the input signal from the UX2 and the output of your audio software. If your speakers are already hooked up to another audio interface, you

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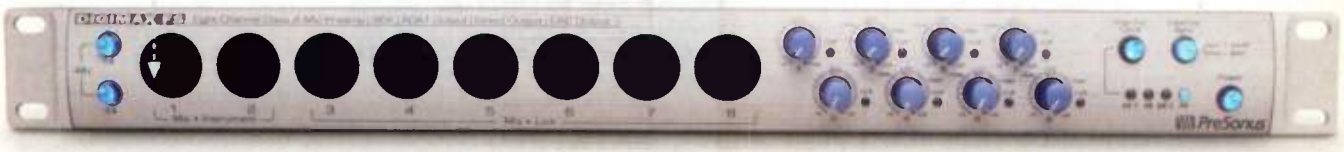

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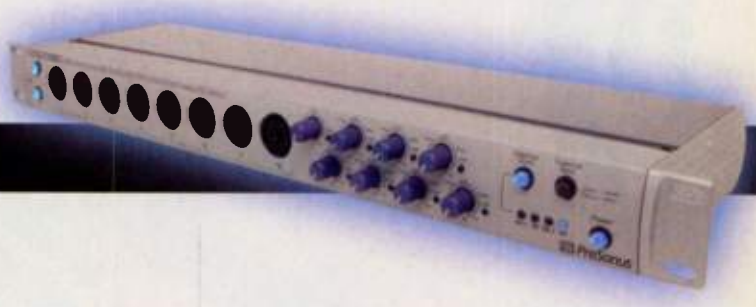

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can connect the UX2's outputs to the inputs of that interface and use its direct-monitoring capabilities. In my case, I have my studio monitors connected to an RME Fireface 800, so I monitored the UX2 by connecting its S/PDIF output to the Fireface's S/PDIF input.

If you run Mac OS X Tiger, you can use the Audio MIDI Setup utility to create an aggregate device made up of the UX2 and another audio interface. That will allow you to record the input signal from GearBox into your host and monitor it through the software of that other interface. I tested an aggregate device in Apple Logic Pro 7.1, recording my guitar through the GearBox software and monitoring the signal through the analog outputs of my Fireface 800, and it worked like a charm.

A BRIEF LOOK AT THE UX1

The TonePort UX1 ($175; see Fig. A) is the little brother of the UX2. It offers a single mic input (with a mic pre) and a single high-impedance instrument input. Like the UX2, it provides stereo balanced TRS outputs, a ¼-inch stereo monitor input, and unbalanced ¼-inch line inputs. However, it doesn't have the S/PDIF output and footswitch jacks.

FIG. A: The TonePort UX1 has less I/O than its more capable sibling, but it offers the same GearBox software.

The GearBox software is the same for both the UX1 and UX2, so both devices can access the same included DSP and (optional) model packs. For the home recordist who doesn't need the additional inputs and outputs of the UX2, the UX1 is worth looking at.

Stomp Your GearBox

GearBox also lets you configure the UX2's two footswitch inputs. You can set both to control either GearBox or your audio software, or you can assign one footswitch to each. When used to control GearBox, a footswitch can be assigned to control the tap tempo function, toggle individual GearBox effects on or off, activate the tuner, advance presets, or mute some or all of the recording sends.

FIG. 3: The UX2 includes Line 6's GearBox software, which offers control over the TonePort hardware and access to the Line 6 amplifier and effects models.

When used to control audio software, each footswitch can be set up to send a variety of MIDI CC messages or to send Mackie Control or MIDI Machine Control transport commands. When used in that way, the TonePort shows up as an available MIDI or control-surface input in your host software, depending on how you've configured the footswitch in GearBox. Simply select it as a controller or MIDI destination, and you can use the footswitch with your host.

Currently, Line 6 doesn't ship the UX2 with printed documentation. But plenty of useful information is offered through GearBox's Help menu and on the Line 6 Web site.

All About Tone

GearBox has dozens of models of classic amplifiers, speakers, stompboxes, and even mic preamps. Many of the models are from the PodXT processor. The effects cannot be freely placed in the chain, but most offer pre and post switches to allow them to be placed either before or after the speaker cabinet. You can also buy extra model packs from Line 6, including the rest of the PodXT sounds or packs of classic models, metal models, and effects models.

Line 6's models range from fair to excellent, and this batch is no exception. Of the classic amps offered with GearBox, I like the Plexi and Brit J-800 models the most. I'm less enthused about the Fender models, which seem a bit duller than real Fenders I've owned. But I really like the Line 6 originals such as Chemical X and Spinal Puppet.

GearBox's mic preamp models are designed for use with incoming mic signals. They offer pleasant warmth and color, but don't expect them to turn the UX2's preamps into Neves or Avalons.

Of the sounds in the optional model packs, the amp models in the Metal Shop collection, such as the Bomber Uber and Diety, are punchy yet retain the sound of the guitar. From the FX Junkie pack, I was particularly inspired by effects such as Phaze Eko and the flangers and choruses (see **Web Clip 1**).

Arriving at TonePort

There's a lot to like about the UX2. It's affordable, it gives you a number of I/O options, the GearBox software is easy to use, and the model collection sounds good and is expandable. Sure, there are some features that I wish Line 6 had included, such as S/PDIF input and trim knobs for the instrument inputs. But at its current price, it's hard to complain.

If you're in the market for a stereo USB interface that offers instrument and microphone processing, the UX2 should be on your short list. EM WEB CLIPS

Orren Merton is the author of Guitar Rig 2 Power! *(2006) and* Logic Pro 7 Power! *(2004) and the coauthor of* Logic 7 Ignite! *(2005), all published by Thomson Course Technology.*

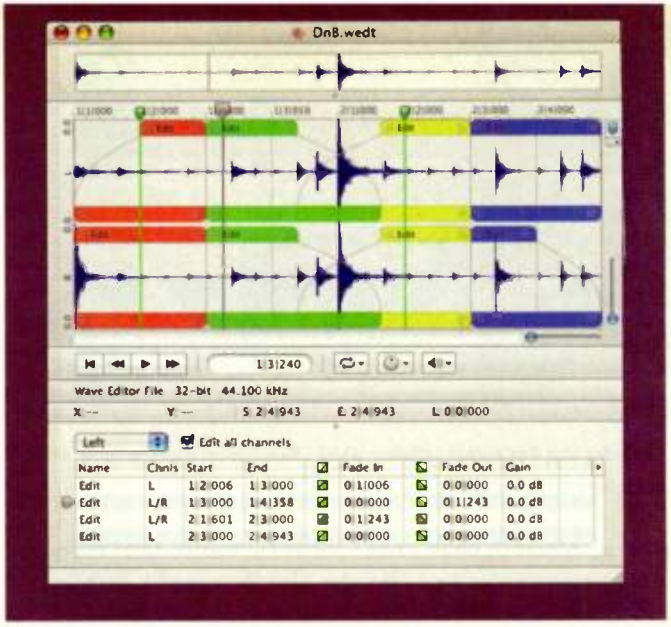

FIG. 1: The Document window is divided into bars that can be repositioned and resized. SmartEdits, shown with color borders in the Waveform bar, are movable audio clips.

AUDIOFILE ENGINEERING Wave Editor 1.2.1 (Mac)

Layered audio editing in an Aqua package.

By Len Sasso

Minneapolis studio owners and self-proclaimed Mac zealots Matthew Foust and Evan Olcott set out to develop the audio editor of their dreams. Their goal was to incorporate the features they wanted in the studio while adhering strictly to Apple's Aqua GUI standard. The result is an application that screams OS X (and requires Tiger). If you're a seasoned Apple user, you'll feel right at home with Wave Editor.

Wave Editor concentrates on audio file analysis, editing, and effects processing. It draws heavily on the paradigm of graphics-editing software such as Adobe Photoshop and QuarkXPress. Instead of having multiple tracks for playing and mixing audio files, it lets you separate an audio file into layers, within which you can move audio around and apply individual processes. Wave Editor doesn't offer playlists, but you can use SmartEdits to accomplish the same thing with some interesting twists (see Fig. 1). It supports AU effects plug-ins but not virtual instruments. In short, Wave Editor won't replace your digital audio sequencer, but it does what it does very well.

Got It Covered

Wave Editor can open, save, and edit audio files in all the standard formats, like WAV, AIFF, and Sound Designer II. It can import any format supported by Core Audio and export the compressed formats MP3 and AAC. And it can edit files with any number of channels in either interleaved or Sound Designer II split format.

For slice-based Acid and Apple Loops files, Wave Editor allows you to edit the slices, and then save the results in the original format. It can also create files in those formats from scratch. The program can read files created by Propellerhead ReCycle (REX, RCY, and REX2), but because of Propellerhead's proprietary compression scheme, it can't save in those formats. You can, however, save ReCycle files in other slice-based formats, complete with their slice markers.

Wave Editor's user interface is completely customizable, and you can have as many window-setup templates as you'd like. You can even apply a template to an already open document. That means you can instantly change layouts to suit what you're doing.

The Document window is arranged in bars, Aqua-style. The Waveform bar, which is where all editing takes place, is always present. The remaining six bars—Information, Overview, Playback, Channels, Edit List, and Location—can be toggled on or off as well as rearranged and resized. Most bars have their own Options dialog boxes that allow you to customize

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their content. Labels and Layers drawers fold out on either side of the Document window.

Processing

Wave Editor has nine built-in processes and, as mentioned, supports AU effects plug-ins. It would be nice if VST plug-ins were also supported, but most commercial effects now come in both formats, so that isn't a big issue. The built-in processes are gain change, normalization, phase inversion, DC-offset removal, time-stretching and pitch-shifting, beat slicing, threshold trimming, reversing audio, and sampling-rate conversion. All processes are layer based, and most can be applied to the entire layer or the current selection.

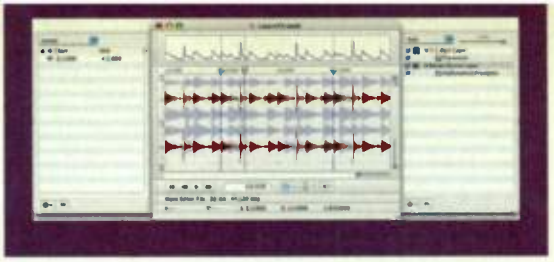

FIG. 2: Layers with different channel counts can coexist within the same Wave Editor document. The drawer on the right shows each layer with its own effects plug-ins.

You can pitch-shift in cents or musical intervals. You can time-stretch by percentage, by tempo, or in ruler increments, which are user selectable. When time-stretching, you can choose to either preserve the file's original pitch or use classic-style speedup, which changes time and pitch.

The beat-slicing process, called Thresholds, uses standard attack and delay decibel thresholds with a convenient Interval parameter that keeps the slices a minimum distance apart. The strategy when using Thresholds is to get as many automatic slices as possible, and then manually adjust them as needed.

Get Smart

You can think of SmartEdits as clips or snippets taken from the audio file. By default,

PRODUCT SUMMARY

AUDIOFILE ENGINEERING — Wave Editor 1.2.1

audio editing software
$250

PROS: Standard Mac OS X Aqua interface. Clever adaptation of layers paradigm from graphics-editing software. SmartEdits for managing audio snippets within layers.

CONS: Documentation is online and not searchable. No provision to automatically move regions or slices to separate layers.

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each channel of the file is a single SmartEdit, but you can snip individual channels into more SmartEdits as desired. You can also use markers and regions as in other audio editors, and Wave Editor can convert those to SmartEdits.

SmartEdits can be moved, resized, faded in and out, and crossfaded with adjacent SmartEdits. They can also be dragged to other audio documents as real audio clips or as links to the audio in the source document. That's Wave Editor's answer to playlists: open any number of source documents, create SmartEdits of the portions you want in your playlist, and drag linked versions of those SmartEdits to a new document, which then functions as your playlist. You manage crossfades, spacing, and relative levels using the SmartEdits in the new document.

The optional Edit List bar gives you control of a variety of SmartEdit parameters. You can numerically set a SmartEdit's start, end, and fade times. You can choose from among five fade types. You can change gain, lock position, and temporarily deactivate a SmartEdit, which silences and hides it. The Edit List displays a single channel, but you can choose to have the edits apply to all channels.

You can also drag SmartEdits to, from, and within the Edit List, and that's often an easier way to move them around. For example, sliding an edit up and down in the list is the simplest way to swap SmartEdit positions, although care must be taken with overlapping SmartEdits. You can set an Edit List option so that SmartEdits dragged to it are linked by default, rather than copied.

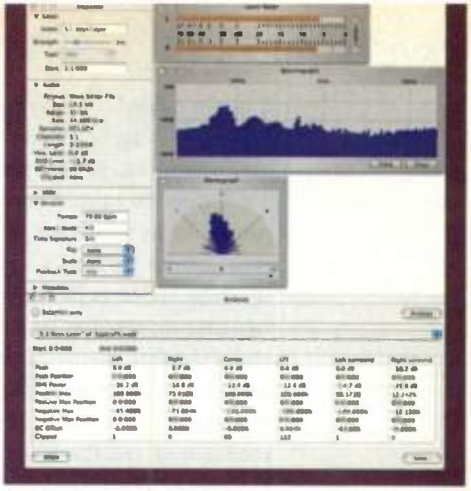


FIG. 3: The Inspector, Analyzer, Level Meter, Spectrograph, and Stereograph windows provide complete details for each audio file layer.

Layers

Layers are Wave Editor's answer to multitracking, although they allow for some additional possibilities. The concept of layering is borrowed from the graphics software paradigm, and as with graphics software, layers are part of Wave Editor's internal document format and must be collapsed to produce standard audio files.

Layers are almost completely independent. They can have different numbers of channels, different sampling and bit rates, and different formats. For example, you could load a 5.1-surround AIFF file into one layer and

a stereo Apple Loop into another (see Fig. 2). Until collapsed, each layer retains its own properties, its own SmartEdits, and its own plug-in effects.

Layers do impact each other, however. They are processed in order from the top down, and there are four options for how a layer affects the ones above it: it can be added as in standard mixing or subtracted as in phase-inverted mixing, duck the layers above it, or ring-modulate them. Ducking is useful for voice-overs and video-censor editing. You can use Wave Editor's waveform generator to produce a waveform for ring modulation, and then pitch-shift and time-stretch that to vary the effect (see **Web Clip 1**).

What You See

Wave Editor has several useful utility modules in addition to the waveform generator. The Recorder can record as many as 32 channels of audio, and you can use an audio-routing utility such as Cycling '74's Soundflower to record audio from other applications running on the same Mac. The Level Meter, Stereograph, and Spectrograph windows give you a visual readout for identifying clipping, stereo imaging, and EQ problems (see **Fig. 3**). The Loop Editor gives you sample-level control over loop tuning.

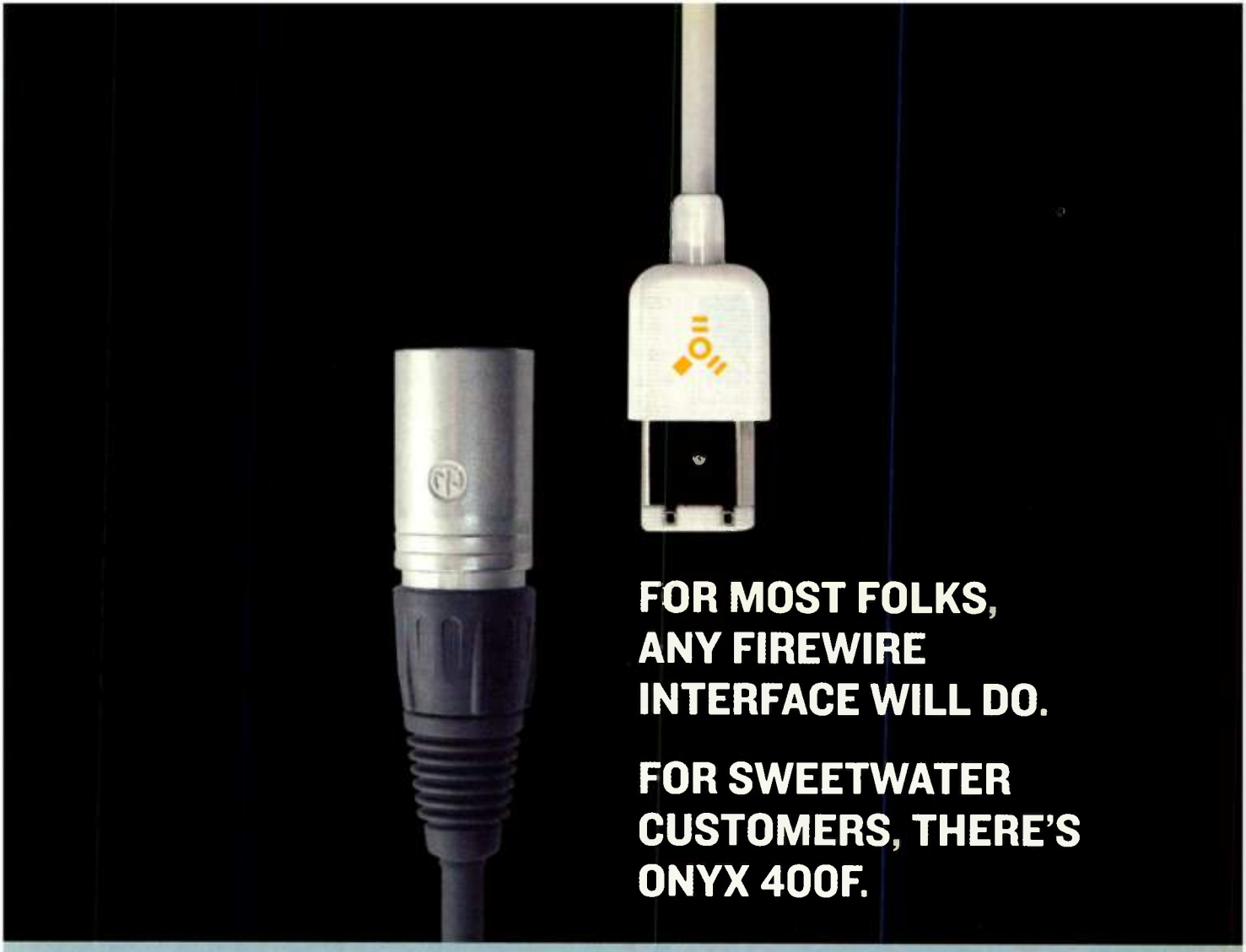
The Analyzer window identifies peaks, clipping, DC offset, and positive and negative maximums. The Inspector window has panes for audio file statistics, MIDI triggering information, tempo settings, and metadata contained in the file header. MIDI triggering allows you to control playback from a MIDI keyboard with MIDI Note Number controlling either playback speed or start position relative to markers in the audio file. You could use that to trigger individual slices in REX files, for example.

User-definable keyboard shortcuts are available for virtually every Wave Editor function, and many scrolling, zooming, and playback functions are available only through keyboard shortcuts. The ability to customize most aspects of its display and operation is one of Wave Editor's greatest assets.

If you're used to another audio editor, it may take you a little while to get used to Wave Editor's adaptation of graphics-editing techniques. But the advantages those techniques offer are well worth the effort, and the pure Aqua interface is a pleasure to work with.

EM WEB CLIPS

Len Sasso is an associate editor of EM. Visit his all-new Web site at www.swiftkick.com.



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QUICK PICKS

NATIVE INSTRUMENTS

B4 II (Mac/Win)

By Doug Eisengrein

EM ★★★★★ hot pick

With the release of B4 II ($229; upgrade, $99), Native Instruments has substantially improved its already stellar Hammond B-3 and Leslie emulation, B4. New features abound, including a greater selection of organ tones and speaker configurations and a new graphical user interface with five views instead of two.

In addition to B4's original sounds, B4 II integrates all the tonewheel sets from the Vintage Collection expansion, giving you simulated B-3s in various states of deterioration, two Farfisas (see **Web Clip 1**), three Vox Continentals (see **Web Clip 2**), and an Indian harmonium. You can choose from 13 speaker setups that range from assorted guitar-amp cabinets to single- or dual-rotor Leslies. Other enhancements include adjustable tonewheel leakage, a more flexible Pedal Bass section, MIDI controller mapping, a MIDI Learn function, and templates for popular MIDI controllers.

B4 the Beguine

I tested B4 II on a 1.33 GHz Apple iBook G4 running Mac OS X 10.4.3, and on a 2.4 GHz Intel Celeron–based PC running Windows XP. Installation and online authorization were simple and trouble-free. A single CD-ROM supplied Mac and Windows installers, a Read Me file, and a PDF that duplicated the included printed manual. B4 II runs standalone and as a DirectX plug-in in Windows XP, as an AU plug-in in Mac OS X, and as a VST or RTAS plug-in on either platform.

Native Instruments B4—already a great tonewheel organ emulation by any standard—just got better with the introduction of B4 II.

At the top of the GUI are buttons to select the five views: Manual, Organ, Expert, Preset, and Setup. When B4 II runs standalone, a sixth button activates a full-screen view, but rather than enlarging the GUI, it simply blacks out everything else onscreen.

Manual view displays the organ from a bird's-eye perspective, as in the original B4, and adds a Brake switch. Brake lets you rapidly stop and start the rotator for some interesting dynamic effects (see **Web Clip 3**). Organ view displays sections for the Organ, Pedal Bass, Tube Amplifier, Cabinets, and Microphones controls; the image of a glowing tube responds to MIDI input. Expert view adds a graphical display of the tonewheel choices, as well as Percussion, Vibrato, Reverb, and Bass and Treble Rotor sections. Organ and Expert views also display larger versions of all the drawbars and tone-shaping controls from Manual view.

Along with a well-designed Preset Manager that lists 48 of the included 120 presets at a time, Preset view has buttons for File and List Operations, a Standard MIDI File player, and an Audition function that tests the current preset. In Setup view, you can specify preferences such as MIDI channel and controller assignments, key splits, and additional drawbar parameters.

Organ Donor

Combined, B4 II's new goodies offer a wealth of possibilities. The new tonewheel sets alone are worth the upgrade. In addition to harmonium, Vox, and Farfisa, they supply ten slightly detuned tonewheels and six B-3s ranging from the factory-fresh B-3 Pure to the obviously abused B-3 Trash. B4 Trash and B4 Filthy were my favorites for their wonderfully raw, gritty tones (see **Web Clip 4**).

In Organ view, the Tube Amp section has been redesigned to accurately emulate a Leslie's amplifier using Guitar Rig's Tube Response technology. The Tube Amp's tonal character is quite distinct from the original's and produces a nice overdrive. Bass and Treble knobs replace the previous Body and Bright controls, allowing you to drive or cut the bottom or top end. The selection of cabinets, in addition to the open and closed Leslies and even a direct box, range from a Fender Tweed to a Roland Jazz Chorus. In the redesigned Microphones section, you can balance the cabinet tone with or without the rotor. An Air control adjusts early reflections and replaces the previous version's Distance knob.

My favorite new feature is the reverb. It has four flexible controls and furnishes Spring and Studio types. Spring works well with vintage timbres, and on attack produces a convincing *boing*. The Studio type sounds natural rather than digital, and with some minor surgery it can reproduce small or large spaces (see **Web Clip 5**).

B4 You Go

The original B4 didn't give me much to complain about. But with B4 II's slicker GUI, comprehensive MIDI control, and additional tonewheel and speaker varieties—not to mention the convincing tone that made the first B4 an award winner—I can only applaud Native Instruments. I grew up with a real B-3 in my parents' house, and B4 II sounds awfully close to the real thing. And in many obvious ways, it's better.

EM WEB CLIPS

Value (1 through 5): 5

Native Instruments

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BARBER ELECTRONICS

Tone Press

By Eli Crews

EM ★★★★★ hot pick

Most stompbox compressors do one of two unflattering things to a guitar signal: they either kill the attack of the instrument or let the sharpness of the attack through in a negative way, overaccentuating the harshness of the pick or pluck. Barber Electronics' Tone Press provides a unique

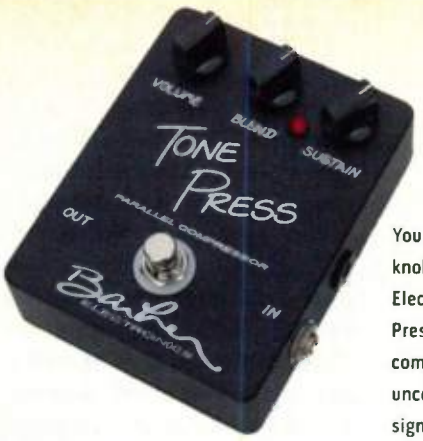
and musical solution to this problem. A feature called Parallel Compression uses a phase-coherent wet/dry control usually not found on compressors. The ability to mix in a direct signal (a capability usually found only on time-based processors) makes this unit stand out in a crowd of currently available effects pedals.

Press Your Luck

The Tone Press is housed in a sturdy but surprisingly light 3.5 × 4.5–inch cast-aluminum enclosure, its silver lettering stark against the matte black finish. You get standard TS input and output jacks, with the input jack doubling as a switch to engage the internal 9V battery. There's also a jack for an optional external DC power supply. The three pots are capped with ridged plastic knobs that are easy to adjust with your tennis shoe—a plus for a stompbox. The Volume knob determines the overall postcompression level of the pedal. The Blend knob sets the percentage of compressed signal, and the Sustain knob functions as an amount control. The higher it is set, the more compression is applied to the input signal. The final control, a tiny trim pot labeled Color located inside the unit, can be accessed only by taking out the four Phillips screws on the bottom of the box. (That is also how you change the battery.) According to the manual, Color sets the tonal attitude of the pedal, ranging from Open Snappy to Round Vintage. The pedal also has an LED that shows when the circuit is engaged and a high-quality footswitch that provides true bypass when off.

Tone for Days

Plugging a guitar into the Tone Press made me an instant convert to its compression method. I was sold on this pedal from the moment I engaged its switch. The box quickly and obviously improved the tone of every guitar or bass signal I ran through it at varying degrees of compression. With the Blend knob fully down, the Tone Press served as a clean boost (about 8 dB) pedal, which allowed me to sweetly overdrive my pretty clean Fender Bassman 135 and Ampeg SVT. With the Blend knob fully up, the Sustain control was highly responsive, and produced useful results throughout its wide range. At the extreme, it was a little too compressed for my taste, but backing it down a bit made for delightfully punchy bass sounds with lots of sustain and shimmering yet thick guitar chords.



You use the Blend knob on the Barber Electronics Tone Press for mixing compressed and uncompressed signals.

As good as the Tone Press sounded on its own, the real fun came when I interfaced it with other effects pedals. With a distortion or an overdrive effect, the pedal

gave me a wide range of tonal variations, from subtle to drastic. Putting the compressor before the other pedals yielded the most interesting results, and the reverse worked well, too. As for the Color adjustment, it sounded best when fully open, which is its default setting. (Any control that's reachable only by first removing four screws is best left alone anyway.)

I like testing stompboxes in the studio as inserts on drums and vocals and in other nontraditional applications. The Tone Press shone when processing a drum-room mic and taming a shouted vocal track. In both cases, the pedal sounded best with the Sustain knob cranked and the Blend knob favoring the dry signal, similar to the technique of heavily compressing one signal from a multed track and blending the result in softly with a dry version.

Stop the Presses

My overall impression of the Tone Press is very favorable, from its indestructible build quality to the sweet sounds it gave to my various tones. I highly recommend this pedal to anyone curious about guitar or bass compression. Those of you who are unsatisfied with your present compression pedal owe yourselves a listen to this one. At an unbelievably low $139.95 list price, what have you got to lose besides lousy compression artifacts?

Value (1 through 5): 5
Barber Electronics
www.barberelectronics.com

By providing distinctive grooves and phrases played by live musicians, Big Fish Audio's *Jazz Quartet* captures the essence of the postbebop style.

BIG FISH AUDIO

Jazz Quartet

By Marty Cutler

For some musicians, creating music from preordained riffs is antithetical to the very concept of jazz. Nevertheless, even if you don't create improvised music (the core signature of jazz) from a sample set, you can certainly capture some of its flavor. That's because jazz has a unique harmonic, rhythmic, and sonic stamp that distinguishes it from most other types of music. Big Fish Audio's *Jazz Quartet* ($99.95) gathers characteristic grooves and phrases in a construction-kit collection that has WAV, REX, and Apple Loops versions on a single DVD-ROM.

The collection's subtitle, *Traditional Jazz Construction Kits*, might suggest the music of King Oliver, Louis Armstrong, and other purveyors of early jazz styles. In reality, the disc focuses on the modern-sounding, postbebop style that emerged during the mid-to-late '60s, with elements of Grant Green, Horace Silver, and others.

Jazz Quartet's musical phrases cover many textures and moods. Although the included PDF document gives you a list of file keys and tempos, having more information would be helpful. Other than the 16-bit Apple Loops, all the tracks are 24-bit, 44.1 kHz files. I tested the Apple Loops in Apple Logic Pro 7.1, and the WAV files in Ableton Live 5.02. I checked REX-format files in Propellerhead Reason 2.5 and converted and loaded them into Spectrasonics Stylus RMX 1.5 to good effect.

On the DVD, folders divide their content by file format. Except for the REX-format folder, files follow an identical organizational layout. Tempo and key subfolders hold variation subfolders (also grouped by tempo and key) containing one file for each of the four instruments: bass, drums, guitar, and piano. Along with the individual instruments, you get a fully mixed stereo file. The REX-format folder groups everything by instrument first, with subfolders gathered by key and tempo.

Phrase Shifters

Tempos range from a laid-back 52 bpm to 148 bpm, but the Apple Loops and REX versions adapt nicely to tempo changes without any noticeable time-stretching artifacts. However, the more freely phrased material inherent in jazz will inhibit how far you can alter tempos, because the looseness can sound exaggerated at tempos over 20 bpm above or below the original.

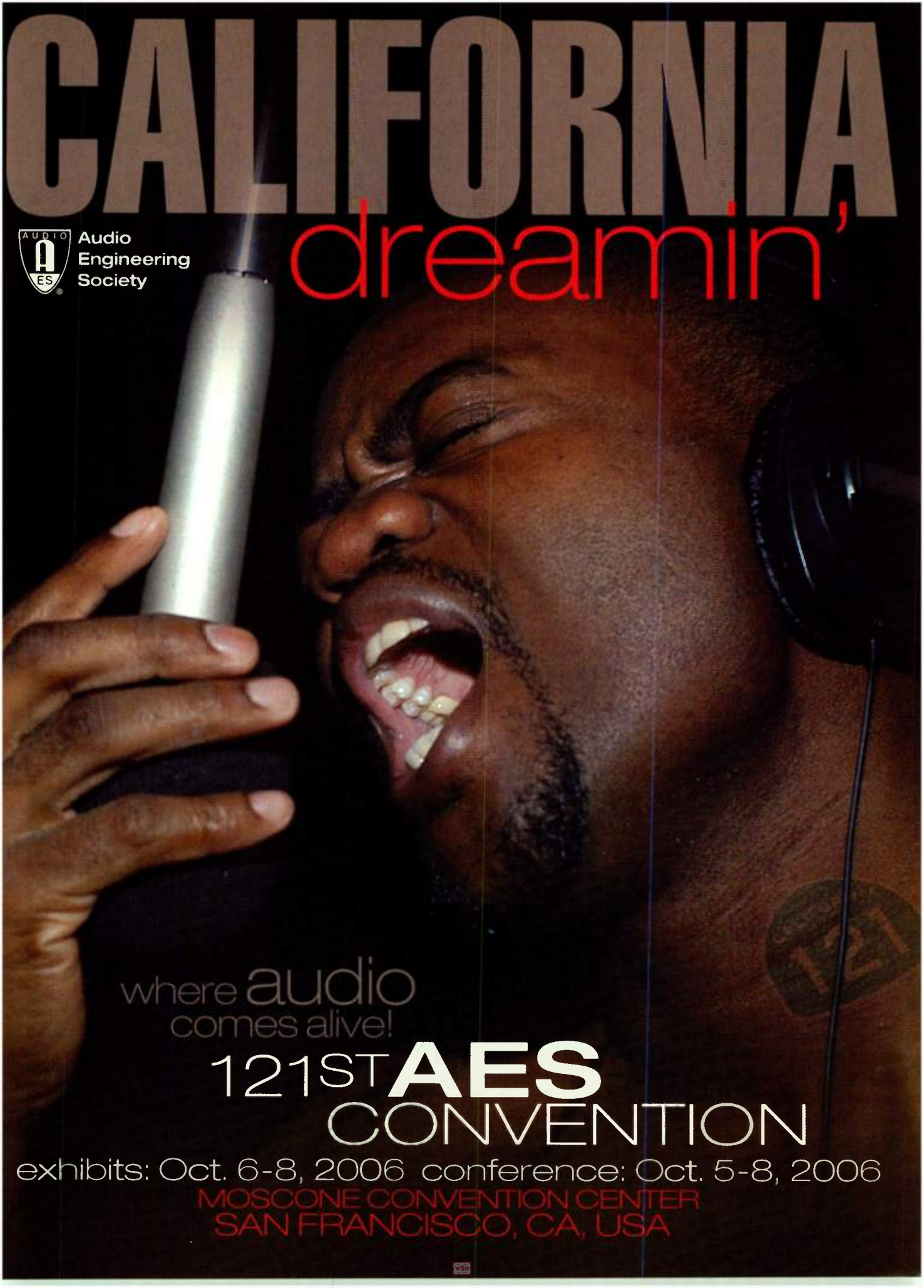
As relaxed as the playing is, the groove between the musicians is strong. Drums combine loose-limbed stick and brush work, and the bass parts are tuneful, swinging, and dynamic. The guitar comping is warm and varied without getting in the way of the groove. Occasionally, the piano parts are a tad too active, as if the musician couldn't decide whether to play lead or accompaniment. For the most part, though, the pianist plays excellent rhythmic accompaniments and peppers the parts with an occasional funky flourish.

After putting the REX files through their paces in Reason 2.5's Dr:rex, I converted them to Stylus RMX format and added a changing, improvisational touch with RMX's Chaos Designer feature. Although Chaos Designer could potentially have wreaked havoc by unnaturally truncating some of the looser feels, most of the files translated nicely. In a few instances, a quick tweak of the amplitude envelopes in the RMX plug-in smoothed out playback. A few tracks suffered from a bit of leakage from other instruments, but adding other tracks easily masked the problem.

Close Enough for Jazz

In *Jazz Quartet*, the parts are greater than their sum. Taken individually, the grooves and comping are terrific. The drums, bass, and guitar parts are easily worth the modest price. As a construction kit, however, the more-active piano parts tend to restrict the music, and presumably you'd want to create your own melodies and solos. Still, there's a lot to appreciate here. Although many recordings of the postbebop era have already been musically strip-mined for sampled funk clichés, *Jazz Quartet* commendably avoids such stereotypical feels and lets you mix and match the instruments in true construction-kit style for more variety. If you're intent on re-creating the feel of classic jazz of the '60s, make sure to check this one out. EM

Value (1 through 5): 3
Big Fish Audio
www.bigfishaudio.com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Universal plug-in control

As a Universal Sound Platform, **Native Instruments KORE** operates not only as a plug-in within Digital Performer but also as an instrument host application. It allows you to integrate all your VST- and Audio Units-based software instruments and effects into a single, unified interface. KORE provides greatly increased control, overview and ease of use in all creative situations. Both Native Instrument's own range of instruments and effects as well as third-party products are supported. The seamless integration with KORE's advanced hardware controller gives hands-on control with unprecedented analog feel, finally turning today's software synthesizers and samplers from applications into true instruments.

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Headphone amplifier

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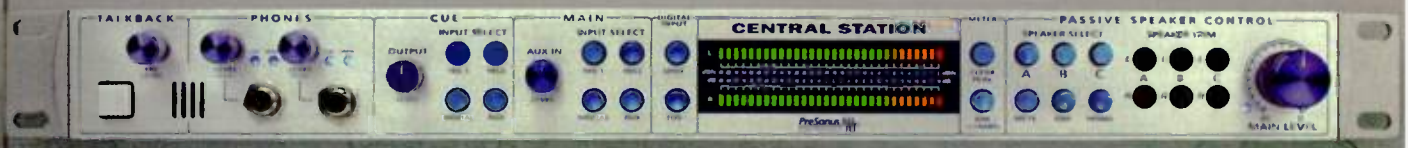

Control room monitoring

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 signal path uses no amplifier stages including op amps, active circuitry 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, talk, 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 in both 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.

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Fragged

By Larry the O

In my recent interview with Geoff Emerick (see "Production Values: The Long and Winding Road" in the May 2006 issue of EM), the former Beatles engineer excoriated modern recording technology for its tendency to grab the spotlight away from performers, absorbing people's attention, and removing spontaneity from sessions. My first reaction was that Emerick's comments seemed curmudgeonly, but on reflection, I find I agree with him in large part.

In my experience, there's a correlation between session flow and the flow of ideas. A smoothly flowing session doesn't guarantee a smooth flow of ideas, but poor session flow inevitably inhibits creative flow. And as I see it, the cumulative effect of several types of fragmentation in the digital studio is poor session flow.

Let's start by discussing how studio time gets fragmented. Although session setup in a multitrack analog studio takes easily as much time as setting up a session in a computer-based studio, once the analog studio is set up, you can work for long periods with only infrequent stops. With DAWs, you must constantly manage computer tasks that take 15 seconds here, a couple of minutes there. The problem is not the lost time; I'm not convinced that sessions in analog studios are faster. Rather, the problem with computer-based sessions is that they are too often composed of short chunks of time, with frequent interruptions.

There's a correlation between session flow and the flow of ideas.

Lack of control is also a problem in most DAW sessions. With a bit of practice, human beings can be remarkably adept, precise, and expressive with controls for grabbing, turning, flipping, pushing, and sliding. Computers, on the other hand, involve very little physical interaction. As quickly as you might learn to navigate menus, that process will always involve an interpretive layer of language skills well beyond that required in a typical analog studio. MIDI control surfaces can help a lot, especially with controlling soft synths, but incomplete or incompatible implementations often degrade the benefits of the control surface.

My point is not that analog gear is functionally superior. Computer-based audio-production systems offer incredible power and flexibility that is unattainable in the analog world. Controlling computer-based audio, however, is less intuitive and physically facile than with traditional analog technology, and that is a significant issue.

When people work in the studio alone or with just one other person, fragmented attention can also be a significant problem. Emerick commented on how difficult it can be to engineer and produce simultaneously, because it is hard to do two things at once and still do both effectively. Yet in many DAW-based studios, one person must often play the multiple roles of producer, engineer, artist, and computer technician. That kind of fragmentation leaves little room for right-brain activity.

Furthermore, a typical personal-studio DAW user has his or her eyes on the screen, hands on the keyboard, and mind on the computer. That puts the emphasis more on the medium than on the message. In fact, having one's mind get sucked into the computer happens so often that one loses awareness of the sense of flow and becomes inured to its absence from the session.

While fragmentation can wait to be addressed in the future, the most pressing question has to do with what can be done to improve things now. Obtaining a control surface, even one with limited functionality, is an important step. But preproduction is at least as important. With well-thought-out templates, macros, and presets, session flow can be substantially improved. But getting all of those things thought through and put together requires a major commitment of time.

I hope that people who have heretofore settled for a keyboard, mouse, and monitor will somehow make it economically desirable for manufacturers to develop better physical interfaces. I would also like to see software houses be forced by powerful processors and a popular uprising to make elegant, intuitive programs. Most of all, I hope we all can remember that the computer is a means to an end and not the reason that we are in the studio. EM

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