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FEBRUARY 2007

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Designing Sounds in
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REVIEWS
**Ableton Live 6.0.1
Moog Little Phatty
Digidesign Strike 1.0
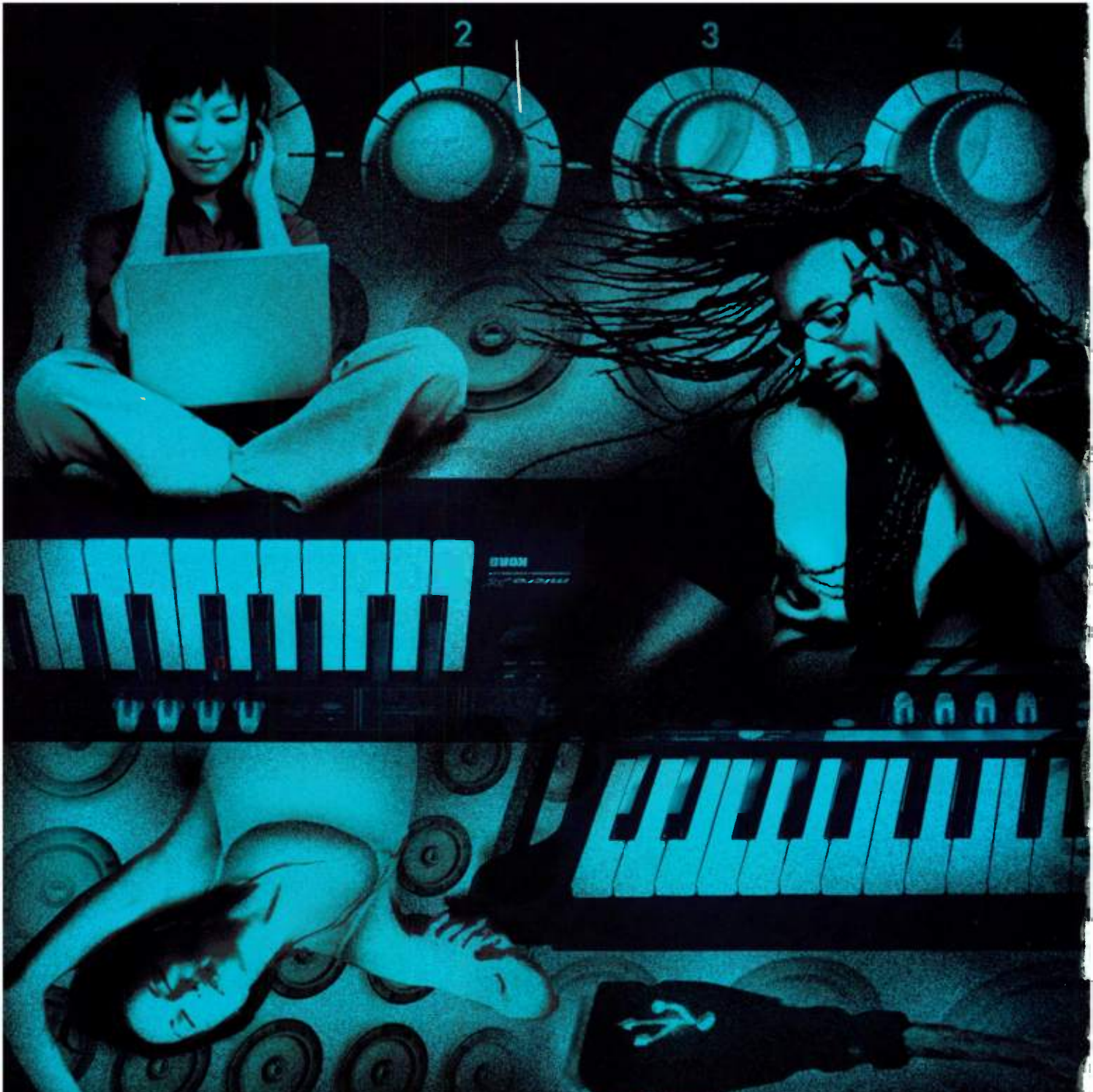Muse Receptor 1.5**
and 9 more

Gil Norton

*Old-School
Contemporary*

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Letters

Reamp

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www.reamp.com

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and 9 more

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Gil Norton

Old-School Contemporary

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- Assimilator Konnekt native fingerprint EQ included
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FABRIK C INTEGRATED

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February 2007

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STUDIO-QUALITY RECORDING IN YOUR HAND.

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The Rest of the Story

Why hasn't EM reviewed Product *X* yet? After all, it has been advertised for months! Some obvious reasons are that it takes time to test a product, write the review, and then produce the issue.

Some less obvious reasons are that before we can start the review process, we have to obtain the product. But unlike some competing magazines, we don't review prereleases, prototypes, or betas; we wait for the finished version.

Getting a review copy of software is usually easy, because providing it is simple and inexpensive; the main expense associated with software is development, not inventory. At most, we're talking about a DVD or CD, some packaging, and perhaps a printed manual. In some cases, we simply download the software and a PDF manual (if there is one).

Hardware is another matter, however, because available inventory is a significant factor. Each review unit costs the manufacturer a chunk of change, and although we return it after the review has been completed, it can't be sold as new. Some companies reserve the first units to fill back orders and provide dealer samples, and the press must wait for a later shipment. We understand, and we try to be patient.

Occasionally, manufacturers delay sending a review unit because they are afraid the product might receive a bad review, but that's rare. More often, reviews are delayed because we have to replace a defective or damaged unit. And occasionally a reviewer misunderstands something about the product, and they have to go back into the studio to retest it.

Sometimes we finish reviewing a product only to find it has been redesigned or updated during the review process or has been discontinued entirely. Murphy's Law apparently has a corollary that states this happens only after the review has been researched, written, and edited. For example, we announced the Open Labs MiKo computer workstation in our July 2006 "What's New" column. The MiKo involves complex technology, though, and it wasn't ready for July. So readers who expected a review by autumn may have been puzzled when none appeared. But in fact, we requested a review unit as soon as we saw the prototype. We got a MiKo shortly before the AES show and sent it to a reviewer a few days after we received it. He proceeded to edit the review and fact-check it with Open Labs. Not until the end of the fact-check process did we learn that the review was moot because the company was redesigning the MiKo and the current model had been discontinued.

The new MiKo will surely be faster and better, and we're still eager to review the product. The upshot, though, is that if you have been waiting for an EM review, you're going to have to wait a while longer. We hate it when this happens, but such is review biz.

As Paul Harvey used to say at the end of his broadcasts, "and now you know the rest of the story."

Steve Oppenheimer
Editor in Chief

JULIE BIRUM

World Radio History

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Sometimes great musical ideas just turn up in your head. And it's these spontaneous moments of creativity and emotion that you want to capture quickly. Cubase 4 makes it easier than ever before to get your ideas onto your hard disk. For example, the new Track Presets system lets you call up exactly the right sound for any instrument or voice almost instantly. Open a track, select the preset – including EQ and effect settings – and that's it. This is just one of the ways in which Cubase 4 keeps your head free for what's most important: your music.
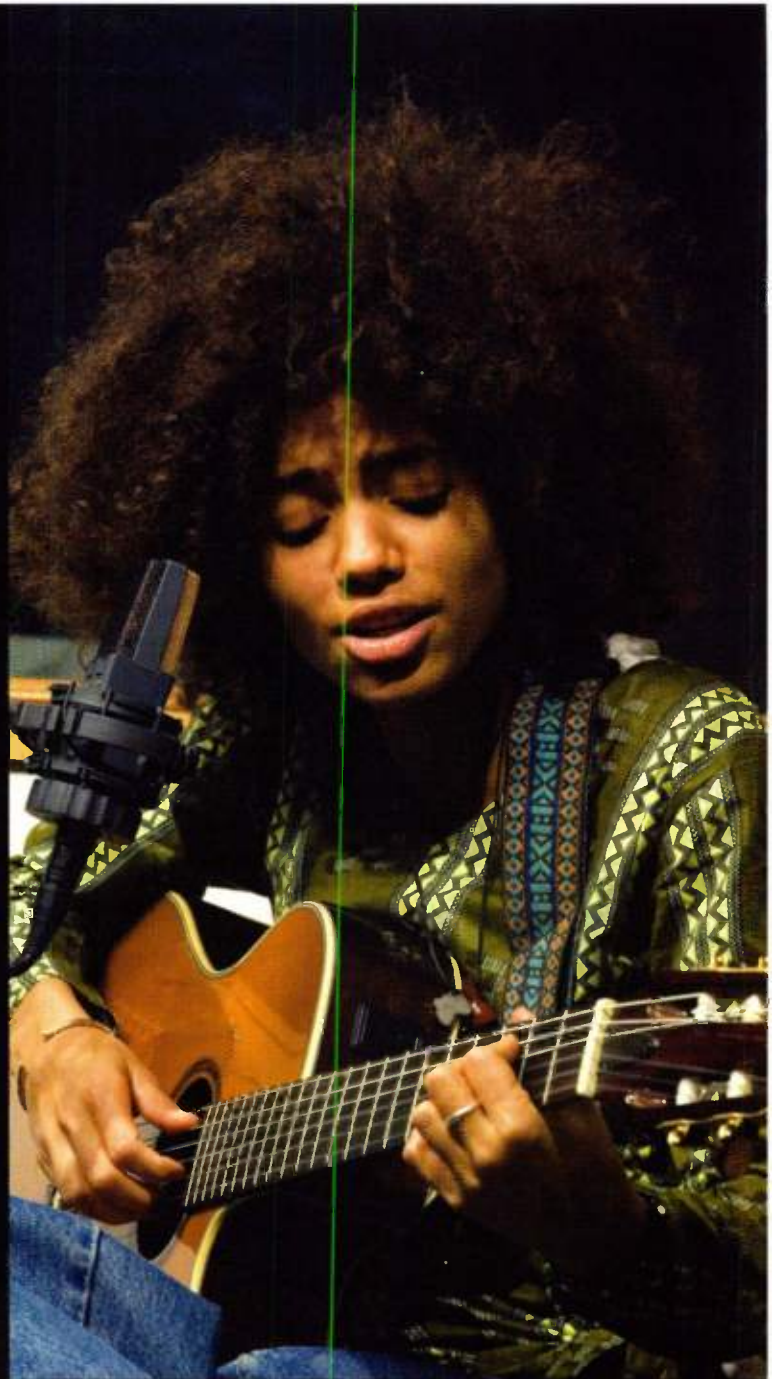
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Breakthrough technology. More creativity. All 4 you.

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Letters

the years ahead. Thanks to EM, and especially to Larry the O, for all the memories.

Joseph R. DeCamp, MD
Fort Wayne, Indiana

Yes on 88

I don't make it a practice to email every publication I peruse; however, I thought your article on virtual pianos was both timely and informative (see "Software Eighty-Eights" in the October 2006 issue of EM). As a pianist, I've been searching for a good software piano for my compositions, and I've found it in Synthogy Ivory thanks to your article. If I had unlimited financial resources, I would purchase the Steinway, Bösendorfer, and Yamaha C7 themselves, but I don't, so the virtual piano is the next best thing.

The fact that Charlie Otwell coauthored the article was another clincher for me, since he is eminently qualified as a pianist. I'm a fan of his from when he played with Poncho Sanchez. I play Latin jazz and salsa piano, so I'm acutely aware of the pianists of such genres. Charlie earned my respect and admiration when I first heard him on piano in "Siempre Me Va Bien," which is still one of my favorite Poncho tunes. Anyway, I greatly enjoyed the article and reread it a few times before making my final decision to buy.

I really enjoy reading *Electronic Musician*. As I stated previously, I find the articles to be both timely and informative. Keep up the good work.

Frank Villafane
via email

High Five

If you go to www.emusician.com/sod, you'll find a great tutorial video on basic synth programming from Korg senior sound designer Jack Hotop. The problem is, it sputtered and froze on me several times, and on deeper reading, the start page said the video is not configured to work on Macs. Yikes!

EM editor in chief Steve Oppenheimer responded to my frustrated query personally and in a timely manner. After juggling his suggestions and trying both Firefox and Safari, still no dice. Apparently, it's just a mysterious computer "thang" that we all encounter at times. He said that there was some work pending with the company that built the video application, so it may snap to by year's end. It worked for him within Mac OS X 10.4 and Firefox and Safari, so I'm personally stumped. But hey, it may work for me by Christmas, and that would be fine.

Despite its refusal to play past the midpoint for me, the video will surely work within Windows and most Mac rigs. More to the point, Steve did his bit to unclog the problem, which is much appreciated. He also removed the erroneous Mac tag from the Web page. EM has always been a good publication, and when the editor in chief himself will step in over such a small matter, it speaks well of the whole operation. I just felt that it deserved public kudos. EM is a good magazine to read and a smart site to add to your bookmarks.

HellPope Huey
via email

HPH—Thanks for the kind words. We hope to have a new video-hosting solution soon, although it's not in place as of this writing. The Hotop video runs fine on my home Mac (running OS X 10.4.7) and on my office Mac (running OS X 10.3.9). But as we all know, strange things sometimes happen with computers and the Web. —Steve O **EM**

emusician.com

EMspotlight

Peter Frampton: the Guitarist

DAVID DOBSON

Despite his superstar status as a singer-songwriter, Peter Frampton is first and foremost a guitarist. He celebrates his six-string passion on his first instrumental album, *Fingerprints*, accompanied by members of Gov't Mule, Pearl Jam, the Rolling Stones, and the Shadows. In this exclusive interview, Frampton talks about recording at home, finding the right guitar and amp combination, and capturing inspiration whenever it strikes. By Gino Robair. emusician.com/em_spotlight

On the Home Page

EM Web Clips

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*.

EM Guides Online

Get detailed specs on thousands of music-production products with our free online Computer Music Product Guide and Personal Studio Buyer's Guide.

EM Show Report

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

EM get on the bus

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

http://blog.emusician.com/the_bus

EM news

A weekly update on new hardware and software releases, manufacturer contests, and pertinent industry news.

emusician.com/news

EM newsletter

Sign up for our free online newsletter, eMusician Xtra, for up-to-the-minute information about new products, software upgrades, and more.

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Think inside the box.

The MX200 Dual Reverb Effects Processor changed everything by combining the rich sound of Lexicon® hardware reverb with the flexibility of full automation and control within your DAW program via VST® and Audio Units plug-ins.

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Griffin Technology iTalk Pro

As long as the Apple iPod has been available, iPod owners have wished for an affordable device that could convert their favorite MP3 player into a portable audio recorder with CD-quality specifications. By shipping the iTalk Pro ($49.99), Griffin Technology (www.griffin-technology.com) has realized that wish. When you plug the iTalk Pro into a compatible iPod's dock connector, a pair of onboard electret microphones can record 16-bit stereo WAV files at 44.1 or 22.05 kHz, without cables.

In addition to internal mics, the thumb-size device has a stereo minijack to connect external audio sources. A multifunctional button serves as a record off/on switch and takes advantage of the iPod's software to toggle the automatic gain control between two levels. The iTalk Pro works with all iPods that support video, as well as with second-generation iPod Nanos.

IK Multimedia StealthPlug

For the mobile-computer-based guitarist, the StealthPlug (Mac/Win, $129) is a guitar-to-USB audio interface from IK Multimedia (www.ikmultimedia.com) that comes with a generous software bundle. The StealthPlug contains a high-impedance instrument preamp and an A/D converter that handles 16-bit audio at 44.1 or 48 kHz. Its 9-foot cable has a ¼-inch plug at one end and a USB plug at the other for connecting any electric guitar or bass directly to your computer. The bus-powered device has volume up and down buttons, an activity LED, and a stereo headphone minijack that doubles as an output for your guitar amp or powered speaker.

AmpliTube 2 Live, which is available only with the StealthPlug, is a standalone application and AU, RTAS, and VST plug-in. It emulates 3 guitar and bass amps, 5 speaker cabinets, 2 microphones, and 9 stompbox effects. The standalone version features SpeedTrainer, an audio player with variable pitch and playback rate that makes it easier to learn songs. The bundle includes Mackie's multitrack digital audio sequencer Tracktion 2.1 and the 16-channel sample player SampleTank 2 SE. You also get the 5-band mastering EQ from IK Multimedia's T-RackS and 500 MB of sampled guitar, bass, and drum loops from Sonic Reality.

PreSonus FireStudio

One of the most exciting of several new products from PreSonus (www.presonus.com) is the FireStudio ($899.95), a FireWire audio interface with 26 inputs and 26 outputs that are all available simultaneously. The FireStudio gives you 8 Class A microphone preamps, as many as 16 channels of ADAT Lightpipe I/O, and support for 24-bit audio at rates as high as 96 kHz. It incorporates JetPLL jitter-reduction technology for fast, stable synchronization over a wide range of clock frequencies. And according to PreSonus, frequency response is virtually flat from 20 Hz to 50 kHz.

The single-rackspace unit has 8 combo XLR and TRS inputs on the front panel, 2 of which accommo-date high-impedance instrument signals. On the rear panel are 4 Toslink ports, coaxial S/PDIF I/O, 16-channel MIDI I/O, word-clock I/O on BNC jacks, stereo RCA inputs, 2 send inserts, 2 return inserts, and 10 TRS outputs. A zero-latency, $36 \times 36 \times 18$ software mixer affords real-time control of all signal routing.

The FireStudio ships with PreSonus ProPak Complete (Mac/Win), a comprehensive software suite that furnishes applications such as Steinberg Cubase LE and Sonoma WireWorks Riffworks Jr. and plug-ins such as Wave Machine Labs Drumagog LE and Wave Arts MasterVerb LE. A selection of virtual instruments includes FXpansion BFD Lite and Applied Acoustics Systems Lounge Lizard Session; you also get drum loops from Keyfax TwiddlyBits and Discrete Drums. In addition, the optional Monitor Station Remote ($229.95) is available. It operates as an outboard speaker manager, input switcher, dual headphone amp, and talkback system.

SUPERSTAR

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Celemony Melodyne Plugin

Celemony (www.celemony.com), maker of Melodyne Studio, Cre8, Uno, and Essential, has introduced Melodyne Plugin (Mac/Win, $299), an AU-, RTAS-, and VST-compatible edition. Like previous versions, Melodyne Plugin lets you intuitively correct and modify the pitch and timing of monophonic audio tracks. Although each instance handles a single audio track, you can assign plug-ins to as many tracks as necessary for shifting the pitch of individual notes or phrases and squeezing or stretching their durations as desired.

Using flexible context-sensitive tools, Melodyne Plugin allows you to manually edit vibrato, change formant structure, and modify other parameters in real time. You can use macros to correct intonation and timing automatically and apply host automation to control pitch, formants, and volume. A multilevel undo function encourages experimentation, and selectable scale grids ensure harmonically correct transposition.

Melodyne Plugin is available for free to owners of Melodyne Studio. It will cost owners of all other versions $149 until the end of February 2007, and $199 after that. Also until the end of February, an upgrade will be available for $199 to owners of Antares Auto-Tune, Serato Pitch 'n Time, and SoundToys PitchDoctor.
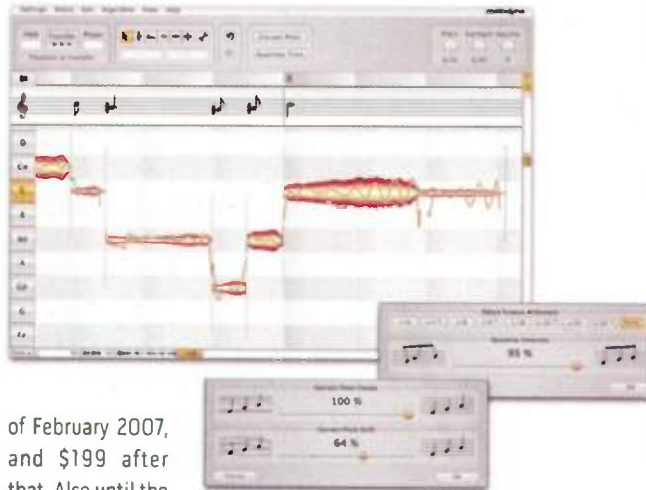
Get Smart

In *Using Reason's Virtual Instruments: Skill Pack* ($29.99), published by **Thomson Course Technology PTR** (www.courseptr.com), author Matt Piper guides you on a journey that explores the synthesis and sampling capabilities of Propellerhead's flagship software, Reason 3. Five comprehensive chapters explain the inner workings of Subtractor, Malström, NN-XT, Redrum, and Combinator. You'll learn how to navigate their control panels, import and edit sounds, program your own timbres, create original patterns, and master their expressive capabilities. The 189-page book continues on the included CD-ROM, which contains PDF files with 72 additional pages investigating NN-19, Dr:rex, and making your own ReFills. The disc also gives you patches, samples, loops, and other materials for use with the book, and a handful of demo files from M-Audio's ProSessions series.

Remote recording presents lots of unique challenges that are best conquered through experience. Audio engineer, microphone designer, and studio owner Bruce Bartlett, along with technical writer Jenny Bartlett, share a wealth of experience and advice in *Recording Music on Location* ($39.95), from **Focal Press** (www.focalpress.com). Subtitled *Capturing the Live Performance*, the 296-page book tells you how to make superior recordings outside of the studio environment, whether you're recording rock or jazz in a nightclub or a symphony in a concert hall. The book is divided into two halves: one about recording popular music, and the other about classical. The first chapter describes the equipment you'll need, including portable recorders, microphones, and mixers. Subsequent chapters discuss session planning, setup and recording techniques, mixing, and editing. You'll learn about mic specifications, stereo and surround miking techniques, and troubleshooting. A 25-page glossary follows extensive appendices detailing stereo imaging and binaural techniques. The book comes with an audio CD that demonstrates various techniques discussed by the authors.

Steinberg WaveLab Studio 6

For PC users who want sophisticated audio-editing tools but don't need surround or DVD-authoring capabilities, Steinberg Media Technologies (www.steinberg.net) is now shipping WaveLab Studio 6 (Win, $399.99). Like WaveLab 6, WaveLab Studio 6 is a multichannel editing, mastering, and CD-burning application that supports VST and DirectX plug-ins, video clips, unlimited undo, and files of unlimited size. It offers most of the same capabilities, including audio looping and disc labeling. It comes with top-notch plug-ins for EQ, resampling, and removing noise and gives you a sonogram-like Spectrum view aligned with the Wave view. WaveLab Studio 6 provides sample-accurate editing and processing at rates as high as 192 kHz with 32-bit floating-point resolution.

So what can WaveLab 6 do that WaveLab Studio 6 can't? The full version supports QuickTime, spectrum editing, and sampling rates as high as 384 kHz. Audio Montage gives you as many as eight independent outputs (as opposed to two in WaveLab Studio 6), with an unlimited number of tracks and more effects per track. You can run multiple instances of WaveLab 6 and render multiple regions. Its CD and DVD support is more extensive, allowing you to import and compare audio CD images and create DVD-Audio discs. Check out Steinberg's Web site for a complete comparison of features.
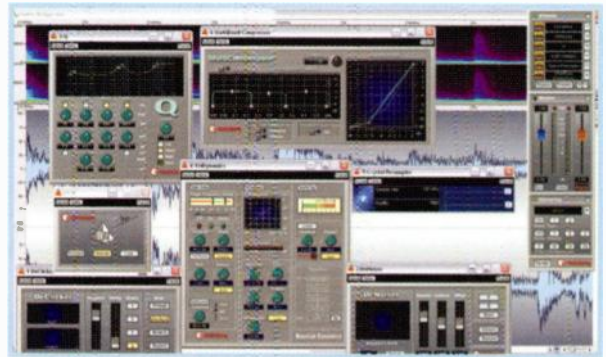
Download of the Month

ANDY WARE ANALOG BOX 2 (WIN)

Analog Box 2 (ABox2) is a free and very powerful modular software synthesizer designed and implemented by Andy Turner. You can download it from his Web site at www.andyware.com. Calling ABox2 a modular synth doesn't do justice to all its capabilities. A quick tour of the many *jambots* (jamming robots) in the AndyWare online gallery reveals an endlessly fascinating sound-design and algorithmic-composition tool (see **Web Clip 1**).
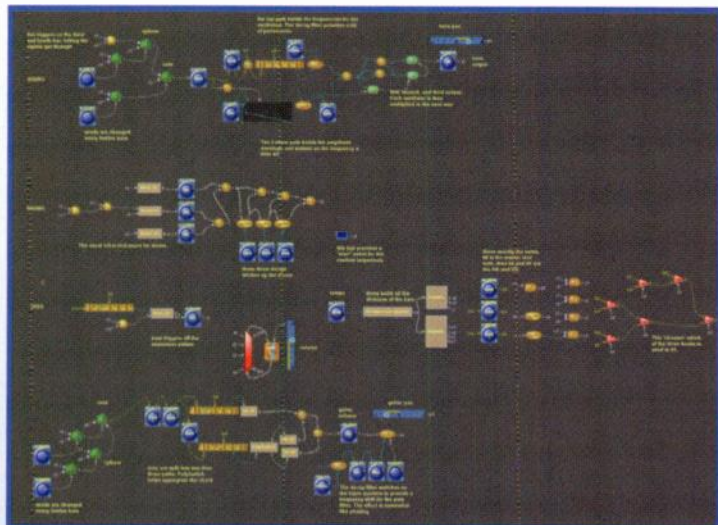
Building devices with ABox2 is not for the faint of heart, often requiring you to do the math from the ground up. Fortunately, it supplies a substantial library of prebuilt modules—sound generators, complex filters, sequencers, and so on—and with a little practice, you can combine them in your own creations. Downloading and tweaking jambots and other user circuits is much easier, less time-consuming, and lots of fun. ABox2 hosts VST instrument and effects plug-ins, so you can build interesting jambots without having to create your own ABox2 synths to play them.

Although you can build virtual instruments in ABox2, it's better suited to building jambots. ABox2 has no provisions for routing audio or MIDI output to other software running on the same PC, although you can use third-party utilities to do so. There is a way to capture audio generated by ABox2 in a WAV file, but it's very utilitarian. Recording is probably best accomplished by networking your PC to another computer.

Regardless of whether you're interested in building your own ABox2 circuits, downloading ABox2 and some of its jambots is well worth your time. The files are compact, and the "How did they do that?" factor is high.

EMWEB CLIPS

—*Len Sasso*

Sound Advice

Samplemeister Eitan Teomi's long-awaited sample library for Kontakt 2, *FlyingHand Percussion* ($259), is now available from SoniVox (www.sonivoxmi.com). The 13 GB, 24-bit collection supplies samples of hand drums and handheld percussion instruments that have been skillfully organized and layered to enhance their playability. Packaged on four DVDs, *FlyingHand Percussion* gives you a selection of articulations, dynamic levels, and microphone positions that will impart nuance and realism to your sampled percussion tracks. Using a proprietary technique called *legato drumming*, the Kontakt 2 Instruments respond to how you play by introducing variations in technique and tone. The huge array of sounds range from frame drums and congas to anklungs and BoomWhackers. If you'd like a taste of *FlyingHand Percussion*, go to SoniVox's Web site and download 600 MB for free.

If you're a fan of '70s-style analog synthesis and a software sampler is your weapon of choice, a downloadable collection from **Soniccouture** (www.soniccouture.com) has your name written all over it. *Synthi AKS for Kontakt 2* ($85.10) furnishes 1 GB of 24-bit samples in Kontakt 2 and EXS24 formats. All sounds are played on an EMS Synthi AKS, a British monosynth made famous by the likes of Pink Floyd and Brian Eno. The collection contains 25 multisampled

instrument sounds ranging from lead solos to filter sweeps, 27 hits and effects kits from insects to dentist drills, and 94 beat machines from caustic loops to Floydish ostinatos produced using a CV-to-MIDI converter. All instruments give you front-panel control of synthesis parameters and a convolution spring reverb.

With the release of *BFD Percussion* ($249), **FXpansion** (www.fxpansion.com) continues to enlarge its line of expansion packs for virtual drummer BFD 1.5. *BFD Percussion* supplies an impressive variety of expressive instruments and ready-made grooves from around the world. The 26 GB collection of stereo 24-bit samples features maracas and timbales from Latin America, djembe and ashiko from Africa, darbuka and doumbek from Arabia, gongs and udu from Asia, bodhran from Ireland, and much more. Each is recorded with a choice of articulations and mic positions and up to 46 Velocity layers. A hodgepodge of found sounds include buckets, a cardboard box, saucepans, trash cans, and even the proverbial kitchen sink. A new feature in the Groove Librarian affords access to nested trees of groove bundles with matching kits.

In 2005 soundware developer Chris Hein proved his sampling prowess with the release of *Chris Hein Horns*. Now he's done

it again with *Chris Hein Guitars* (Mac/Win, $449.95), a Kontakt Player 2–based instrument published by Best Service and distributed by EastWest (www.soundsonline.com). The standalone version and AU, RTAS, and VST plug-ins are paired with 18 GB of 16- and 24-bit content. The collection features nine sampled instruments ranging from electric, jazz, steel-string, and nylon-string guitars to mandolin and banjo, each recorded with different playing positions and up to 13 Velocity layers. *Chris Hein Guitars* takes full advantage of Kontakt Player 2's scripting and programming functionality and provides real-time access to dozens of articulations and variations by means of keyswitching and MIDI Control Changes. Effects include reverb, delay, chorus, phaser, flanger, and compression. **EM**

World Radio History

World Radio History

Gather Round the reacTable
By Scott Wilkinson

Several people can play this instrument simultaneously.

Most musical instruments, electronic or otherwise, are designed to be played by one person. If you bring several individuals and their instruments together, you have an ensemble. But what if you could turn that paradigm around by bringing several people together to play a single instrument, collaborating on the final audio outcome?

That's exactly the idea behind the reacTable, a research project being conducted by the Music Technology Group at the Universitat Pompeu Fabra (http://mtg.upf.edu/reactable) in Barcelona, Spain. The reacTable is a computer-based musical instrument that was designed to be played by several people simultaneously.

As its name suggests, the reacTable's basic form is a table—a structure around which people traditionally gather to converse and collaborate. In this case, the tabletop is round (to allow all participants equal access) and made of translucent plastic. Under the table is a video camera that senses the position, shape, and orientation of specially designed objects placed on the surface. This information is used by a computer to generate visual images on the table with a projector below the surface and audio signals from a software-based synthesizer.

The sound engine was inspired by modular synthesizers, such as those from Moog and Buchla, and object-oriented musical environ-

ments, such as Max/MSP and Pure Data. Different objects on the table represent different synthesis modules, such as sound generators, modulators, controllers, sequencers, and mixers. Connections between the modules are represented graphically on the table by images of audio and modulation waveforms, among other shapes (see Fig. 1). Even more interesting, the connections are made automatically and dynamically, based on the modules' functions and their proximity to each other.

The first reacTables are about one meter in diameter, allowing as many as four people to participate in a performance. As each person places modules on the table and moves them around, the connections between them change, creating an ever-evolving soundscape. In addition, turning a module in place varies an associated parameter, much like turning a knob on a hardware synth.

To involve more people, multiple reacTables can be networked together, with the icons of modules on each surface appearing on all of them. They can also be networked over the Internet; in fact, one of the first performances included one reacTable in Barcelona and another one in Linz, Austria. Participants at one table can't directly affect the modules on another table, but the modules from all tables combine to create the final sound, which all participants hear. Each reacTable has its own audio system, and only control signals are transmitted over the network, minimizing latency.

The video camera scans the tabletop in real time, and its signal is analyzed by a program called reacTIVision. The resulting data goes to a connection manager, whose job is to establish the connections between modules. This information, along with the type, location, and orientation of each module, is sent to a visual synthesizer (which controls the projector) and to the audio synthesizer. Waveforms from the audio synth are also sent to the projection system to be rendered on the table surface.

The reacTable was conceived to serve a wide variety of purposes. For example, it can be part of an interactive exhibit for untrained visitors to play with. It can also be used as a sophisticated instrument by skilled musicians in a concert setting. One particularly fascinating aspect of the device is that it merges the processes of building and playing a musical instrument into a single activity, which opens new creative vistas for all who want to expand their musical horizons. EM

FIG. 1: The reacTable is played by placing and moving specially designed objects that represent synthesis modules on a translucent tabletop. Connections between modules are made dynamically and are represented by waveforms and other shapes.

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Recording the Spoken Word

By Perry Anne Norton

Expert tips
on producing
voice-overs.

pportunities for voice-over (v/o) production have increased dramatically for project studios, mimicking the DIY paradigm shift that continues to rock the music industry. Increasingly, clients needing v/o talent and related audio services are bypassing bigger studios to hire more cost-efficient producers for everything from commercials to interactive voice response (IVR) systems.

There are basic but critical differences between music production and voice-over production. If you learn and implement elements that are unique to v/o clients and talent, however, v/o production can be a good way to augment your existing studio offerings and bring home a bigger piece of the audio-production pie.

Let's Get Physical

Voice-over recording calls for an acoustically neutral space with no reflections. When I walk into a studio, I want to hear an almost oppressive silence. To prepare for v/o work, consider all the reflective areas in your studio. If you don't already have carpeting, try putting a thick rug at the base of your recording area to supplement any other soundproofing. Even the music stand holding the talent's copy should be covered with a carpet remnant, a towel, or perhaps some acoustic foam to eliminate harsh reflections. If you can't achieve such an environment without overhauling your

existing space, you might consider investing in a whisper booth (also called a vocal booth or an isolation booth). A booth's cost will vary depending on its size. As an alternative, try converting an existing closet (see **Fig. 1**). And don't forget that your CPU may be your most significant noise source; be sure to isolate it.

If you can't completely eliminate ambient noise or reflections, invest in noise-reduction software to clean up your files after recording. When necessary, I use the AudioSuite noise-reduction plug-in Digidesign DINR for Pro Tools just before rendering my files to WAV or MP3, but you probably won't need to do that if your space is dead quiet.

Once you're happy with the sound of your recording environment, set the stage with some common-sense props for your voice talent. Having certain items on hand, such as a stopwatch and drinking water, helps eliminate factors that could add to the time you'll spend editing. A stopwatch is handy for actors to use before recording to practice voicing commercial copy running 30 or 60 seconds. In the booth provide plenty of room-temperature water, which is better for the voice than hot or cold liquids. Adding a few drops of lemon to the water will keep lip-smacking and mouth noises to a minimum. Chewing gum between takes also helps keep the mouth moist. Be aware that some medications as well as excessive salt can contribute to dry mouth.

Whether mouth noises occur between syllables or words or throughout a passage will determine whether you'll want to do another take or try to edit them out (see **Fig. 2** and **Web Clip 1**). To discourage mouth noise, your talent should avoid consuming caffeine and dairy products. Contrary to the popular belief that tea soothes the vocal cords, the tannins in tea actually dry out the voice, and caffeine can contribute to readings that sound rushed or nervous. Dairy products foster the production of phlegm, resulting in throat clearings that must be edited out. Keep water and healthy energy foods such as fruit on hand for longer sessions. Make a list of these tips and keep it handy (see **Web Clip 2**).

Usually, you'll want your actors to stand because they can achieve better breath support. For long sessions, though, you should have sturdy, quiet stools or chairs available. A set of headphones for each actor is an obvious necessity. Set up headphones so that actors can hear each other as well as you. Use a filter or windscreen halfway between each actor's mouth and mic to minimize pops on the letters *b* and *p* and other plosive breath noises (a *plosive* is a consonant sound produced when airflow is stopped in the vocal tract).

Mic Check

Which mic you choose for a specific actor will depend on the timbre of his or her voice, just as it would with a singer. A particular mic may sound brittle or harsh and be less forgiving of pops with one voice, yet it may emphasize another person's warmth and resonance. Because voice

FIG. 1: This photo shows voice-over pro Rick Adamson's converted closet, which serves as an isolation booth.

actors are typically closer to the mic than singers (3 to 4 inches rather than 8 or 9) and tend to stay in one place, find a mic that best matches the actor's voice and minimizes audible breathing and pops, but is as transparent as possible to convey the actor's best voice quality. You want the talent's vocal personality to drive mic choice, not the other way around. Many actors know which mics show them off best, so be sure to ask. You should avoid omnidirectional mics, as they pick up too much room sound and can make actors sound off-mic or in the distance.

High-end studios are more likely than personal studios to get requests for expensive mics. If you can afford a Neumann TLM 49, U 87, or U 47, more power to you. Fortunately, cost-conscious project-studio owners can find excellent and affordable alternatives.

For less expensive mics that are still quite effective, Dom Camardella of Santa Barbara Sound Design likes the Audio-Technica AT4033/CL for its crispy, bright sound. Cautioning that the AT4033/CL can sometimes be sibilant and trebly, however, he also suggests the AT4047/SV, which he describes as warm and rich with a strong output that's not overly bright (see **Fig. 3**). It's more expensive than the AT4033/CL, says Camardella, but its sound resembles that of a Neumann U 47. He says the AT4050 is another good option.

Camardella asserts that Australian manufacturer Røde Microphones has better quality control than many of its competitors and makes some superb mics with a great deal of consistency. If you want a dynamic mic that

Contain Yourself

We're not necessarily referring to your enthusiasm for these four exciting new MG-Series mixer models. It's more about the unique single-knob compressors on their mono inputs, designed to keep loud from getting too loud and soft from getting lost in the mix. Combine this with MG's other performance-enhanced features and it's hard to resist a visit to your local Yamaha Live Sound dealer for closer examination.

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performs like older mics, he suggests the Heil PR 40 for its silky top end, openness, presence, and high gain. Blue Microphones has a stable of reasonably priced mics that

roughly 3 to 4 inches—may vary with an actor's natural projection and resonance. Whether an actor is speaking straight ahead into the mic or off to one side, you should place the mic capsule at approximately the height of the speaker's upper lip. Placing the mic 20 to 30 degrees off to one side minimizes pops and gives the actor an unobstructed view of the copy (see **Fig. 4**).

Many actors know which mics show them off best, so be sure to ask.

he recommends. Camardella describes them as silky, with good signal-to-noise ratio. The top of Blue's line is the Blue Bottle with a changeable capsule.

Providing another perspective, v/o talent Rick Adamson gets fantastic results from his Shure KSM32. He likes it because he doesn't want anything with too much coloration, especially because he uses it for straight-ahead corporate work and not pumped-up promos. He says it offers an extended frequency response for an open, natural-sounding reproduction of the original sound source.

As a female, I like the Shure SM7B dynamic mic for its good signal-to-noise ratio and simply because it really complements my voice. It has a large diaphragm, bass roll-off, and presence-boost controls and comes with two different windscreen options. Because the input tends to be a bit on the low side, however, I need to boost the gain.

Placement and Positioning

The number of mics and their placement will depend on the actors and the space available. Although singers need more flexibility to move around and use a mic for dynamic and varied output, actors require fewer fluctuations in volume. Adjust the recording levels to the actor's volume, rather than having him or her move farther from the mic.

Mic distance for voice-overs—

FIG. 2: Mouth noise is a common occurrence when talent is nervous or dehydrated. It would be faster to record another take than to edit this recording.

If you're recording two or more actors in an acoustically treated studio, it's usually preferable to have them standing in the same room and seated only for very long sessions. Because actors are more comfortable working off each other, that arrangement makes for a better performance. The best situation is to have one mic for each actor, but even directional mics can still pick up off-axis sounds. To avoid bleeding or spillover when you're recording actors on two different mics simultaneously, try to position the mics according to their polar pattern to achieve maximum off-axis rejection. Fortunately, actors' lines are rarely spoken simultaneously, which means that if one actor's voice spills into another actor's mic, you can use the mute button or correct the problem with editing.

If two actors must share a single mic, you're better off with a bidirectional figure-8 pattern. As an engineer, you'll need to be forgiving of people with less sophisticated mic technique. You may still have problems if one actor has a booming voice and the other has a soft voice. If you have to, place the mic a little bit above and angled down a bit toward the boomy actor so that plosives will come out under the mic.

Recording and Editing

Voice-over recording is generally less labor intensive than recording a singer. A clean signal path, a good condenser mic (or a dynamic mic with a good preamp that has very little color), and only slight (if any) compression during recording are the three most important factors.

The best strategy is to get a clean sound up front by capturing the voice's natural warmth at a sufficient level. Just let the mic, the actor, and the *proximity effect*—a directional mic's tendency to boost low frequencies at close range—do the work for you (see **Fig. 5**). Dom Camardella calls the proximity effect "the voice of God" or "the Isaac Hayes effect." Dynamic mics are often better for achieving the effect. Voice actors take advantage of proximity to add fullness to their vocal sound. However, working close to the mic can emphasize undesirable sounds such as breathing and plosives that can lead to editing headaches (see **Web Clip 3**).

You can edit out distracting or unnecessary *catch breaths* (quick inhalations) to tighten up a recording if necessary, but experienced v/o talent will mark breathing points in the script in advance so that breathing sounds natural and doesn't draw attention to itself or add length to carefully

timed copy. As a rule, you don't want any reverb or EQ going in, though for certain actors with sibilance issues you may want to incorporate a de-esser or a tiny bit of frequency-dependent compression after the fact.

You might be surprised at how often clients want to edit the raw vocal files themselves. They may add EQ or compression to your recorded files to match other spots they've created—so remember, clean is king. Know your client's expectations up front and use EQ sparingly, if at all. You'll understand the inherent challenges if you're the one editing prerecorded files with a lot of compression or EQ on them. I know an editor who had to improvise and replace every single *f* sound in a long file, because it arrived completely compressed or gated to the point at which that consonant was lost. Because the recording involved a celebrity talent who would have been expensive (and embarrassing) to call back to the studio, it became a tedious project that fell on the editor.

Copy That

While it may seem obvious, it's critical to ensure that you have final, approved copy. Ask about any timing constraints (for TV or radio spots) so that clients don't

FIG. 4: The most effective miking technique for recording voice-overs is to place the mic between 3 and 4 inches from the voice actor's mouth, with the mic capsule aligned with the actor's upper lip and a windscreen positioned halfway between.

waste your time and their money rewriting copy during or after a session. Time the copy yourself before the session. Advise the client if it's likely to run too long, and make editing suggestions if you have a good working relationship. Determine in advance if you will charge extra for that guidance.

Editing copy is the client's job, but as a project studio producer, it often becomes your problem. You don't want to rely on fixing it in the mix for v/o work any more than you do for music. It's amazing how many times the wrong "approved" copy comes into the studio, is recorded, and then must be edited or rerecorded (for more on copy preparation, see **Web Clip 4**).

Under most circumstances, capturing the spoken word is not as demanding as capturing a singer's performance. When you are recording dialog, consider recording multiple tracks, which simplifies tweaking for vastly different voice qualities. Number each take and mark the preferred ones in the script as you record, while slating each new take with an audible marker of where you are in the script. For example, record the slate "Oxford University Press, Lesson 4, take 3," followed by a visual cue for your actor to begin speaking. Number all takes in your hard copy as you go, and circle the best ones; that will speed up your editing later (see **Web Clip 5**). Another task that may come later is recording *pickups*. Pickups are retakes recorded out of sequence that you or the client decide should be redone, usually at the end of a session (sometimes days later).

As a project studio producer, your clients may come from other cities, states, or countries. To gain a client's trust, voice-over pro Rick Adamson suggests taking clients through a dry-run phone rehearsal for style and pacing. After the recording session, he often delivers two full

FIG. 3: Audio-Technica's AT4047/SV is a versatile multipurpose studio mic that works quite effectively for recording voice-overs.

takes of everything, keeping the file clean for the client's engineer to compress or EQ as they see fit (for more on dealing with off-site clients, see **Web Clip 6**).

Talent in the Raw

From time to time, clients may suggest using new, unproven talent because they want "regular-sounding" people. Be aware that raw talent tends to "pop" and go off-mic more often, because their technique will also be raw. Keep in mind that pro voice actors employ the same techniques as other actors. Help your talent find ways to relate to the copy, use timing and phrasing, make the read sound spontaneous, or get into character. As in music, a good sense of cadence and an ear for natural inflection both go a long way (see **Web Clip 7**).

Jon Van Horn, former engineer for Full House Productions in New York and now a project studio producer, stresses the importance of a good working relationship with clients and talent. "I always try to make them laugh and be casual, and above all, to make people really, really comfortable to get the best results, whether they're new talent or old pros. Half of the work is how you relate with the voice-over person you're working with. Be supportive and professional with everyone. You always get better results and repeat business if you're encouraging and positive—so keep the peace, and use diplomacy."

File Delivery

Know what final file format is required by your client before you begin your session. That will be determined in part by how the audio will be used. For example, many of my clients are authors who want short excerpts of their books recorded for use on their Web sites. They may look to you for guidance because they won't care about bit resolution or sampling-rate terminology.

A session recorded using 16-bit resolution and a sampling rate of 44.1 kHz and then rendered to MP3 with a compression rate of 128 Kbps or higher will produce a file that is of decent quality without bogging down a Web site. Because of their small size, finished MP3 files can often be emailed to the client for approval. Other clients, including those who put voice to picture for commercials or other video, will likely require uncompressed WAV files or even 24-bit, 48 kHz recordings. If you're rendering higher-quality (and thus larger) files, you'll need to deliver them by FTP or put them on a CD to send to your client.

O Clients, Where Art Thou?

It's easy to start locally and offer to do public service announcements (PSAs) for non-profits or commercials for local businesses on the cheap. Use your imagination and contact companies you think could benefit from using audio as a marketing tool. Avoid high-maintenance, one-off clients if you can; as the saying goes, "Some money costs too much." Look for clients who respect your time and who pay promptly by PayPal or other secure system.

You might also take a look at producing

FIG. 5: The proximity effect is a low-frequency boost that occurs when someone sings or speaks close to a directional microphone. Voice actors can add fullness to their sound by taking advantage of this effect.

audiobooks. The drawbacks are the labor-intensive nature of the recording process and the need for access to a wide range of vocal talent. Paul Ruben in New York is a widely recognized audiobook producer. Rather than trying to compete, I work with emerging authors creating short promo clips (see **Web Clip 8**). If you want to explore audiobook production, you'll need to cozy up to the big publishing houses.
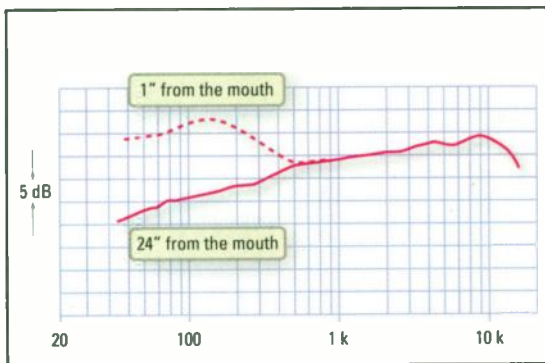
Good Night, and Good Luck

The voice-over industry is a service industry. Ultimately, you will need to please your clients. You will need to put aside your own ego, even if your client or talent is difficult to work with. Never be dishonest, but if someone makes a mistake, take responsibility as the engineer rather than making the client or actor feel inadequate, and you'll get a better product. Use the best equipment you can afford, and keep trying new methods to improve efficiency. Learn from your mistakes, but not at your clients' expense. With a bit of imagination for wooing off-the-radar clients, voice-over production can be a good opportunity for a music producer-engineer with the right studio environment to expand his or her client base and bring in extra cash. **EM**

How to put the
finishing touches on
your recording project.

Do-It-Yourself Mastering

By Michael Cooper

Mastering is essentially the art of applying signal processing to finished mixes in order to enhance them or correct perceived problems, sequencing the processed tracks in the desired order for playback, and exporting the resulting file in the correct format for delivery to the replication plant for mass production. Although mastering is a highly specialized process, so many affordable mastering-oriented products have been introduced to the market over the past several years that almost anyone can now master their own recording project. However, *can* doesn't always mean *should*.

Many engineers look to mastering as a solution for fixing a mix that has problems they can hear but can't quite figure out. For example, they might feel their mix lacks punch or is fatiguing to listen to but don't know how to make it sound otherwise. In this case, indiscriminately piling on additional processing during the mastering stage won't solve anything and could, in fact, make the finished masters sound a lot worse. Here it makes a lot of sense to hire a great mastering engineer to add fresh ears and skill sets to the project.

If, on the other hand, your mix has problems that you can readily identify and know how to fix, it makes more sense to remix than it does to try to correct a bad mix in the mastering stage. Because processing applied during mastering typically affects *all* elements of a mix, correcting one thing might make another thing sound worse. For instance, boosting EQ in the mastering stage to make a dull-sounding snare drum sound brighter might make vocals too sibilant or the cymbals sound harsh. (There are other ways to fix such a problem, which I'll discuss later.) In this case, it would be best to remix the song and apply needed EQ only to the snare drum track before mastering the project.

All that said, mastering can work miracles on poorly wrought material that cannot be remixed for some reason, such as when the multitrack master is lost or damaged or exists only in an obsolete format for which there is no playback mechanism. In that case, the 2-track mix may be all that is available to work with, and mastering is the only recourse to improve the program.

Mastering yields the best results when used to make great mixes sound even better. It can also make

a program flow better from beginning to end by lending more consistent spectral balance and dynamics to all the mixes on a project so that no jarring changes occur (unless intended) from song to song. Also implicit in the mastering process is preparation of the *premaster* (the file or disc used for replication), which involves sequencing of songs (with gaps and track offsets), documentation (including CD-Text), preparation of reference discs (for auditioning and final approval), and delivery of the master in an error-free format that the replication house can handle.

In this article, I'll discuss what's involved in mastering your own stereo project for CD release. I'll begin with a brief overview of requirements for accurate room response and monitoring setup. Then I'll dive into how to optimize your work flow and evaluate what types of processing may be needed for your program material. Along the way, I'll give a small sampling of some of the mastering products currently on the market. We'll finish with a discussion of preparation of the premaster.

The main focus will be on do-it-yourself projects and not on running a commercial mastering studio. (See the sidebar "Do unto Others" for a brief overview of things to consider when mastering other people's projects.) For the sake of simplicity, I'll talk about mastering an album project inside a DAW, but most of what I'll discuss also applies to other mastering scenarios.

FIG. 1: A multichannel mastering setup in MOTU Digital Performer 5. Each track is routed to a different stereo digital output feeding the same downstream monitoring path so the engineer can monitor each one separately in turn as they all play back. Note the markers placed at the beginning and in peak sections (the latter denoted by the word *loud*) of each file.

FIG. 2: In the rightmost column of the List view for BIAS Peak Pro's Playlist, Vbox plug-ins can be turned on and off as a group with one mouse-click to facilitate A/B comparisons between dry and processed mixes.

Go to Your Room

It is absolutely vital that your mastering work be performed in a room that has highly accurate frequency, phase, and reverb responses. Trying to master your project in an inaccurate room makes as much sense as doing color photo touch-ups while wearing tinted sunglasses. Likewise, unless your monitoring setup is flat and can accurately reproduce the entire audible frequency spectrum, how can you confidently decide which frequencies need adjusting on your program? If you can't truly hear what is going on with your source material, you're just shooting in the dark.

While a full-blown examination of room acoustics, control room equalization, and monitoring setups is way beyond the scope of this article, a few points bear mentioning here. (For an in-depth look at tuning your control room, see "Truth or Consequences" in the November 2001 issue of EM, available online at www.emusician .com.) First, learn what *room modes* (those narrow peaks or dips in frequency response) your room exhibits and keep them in mind while mastering. Before boosting or cutting EQ at or near any room-mode frequencies, listen to the same program from a position in the room where those same modes are not being reinforced to determine if corrective EQ is really needed. Mastering can be effectively performed in a room that has only minor imperfections in response if you know what those imperfections are and compensate for them.

Room tone is another matter. You should make sure the RT60 (essentially the reverb decay time) is not skewed in any one frequency band in your room. Otherwise, you may end up cutting bass frequencies, for example, simply because your room reverberates longer in that band than elsewhere throughout the spectrum

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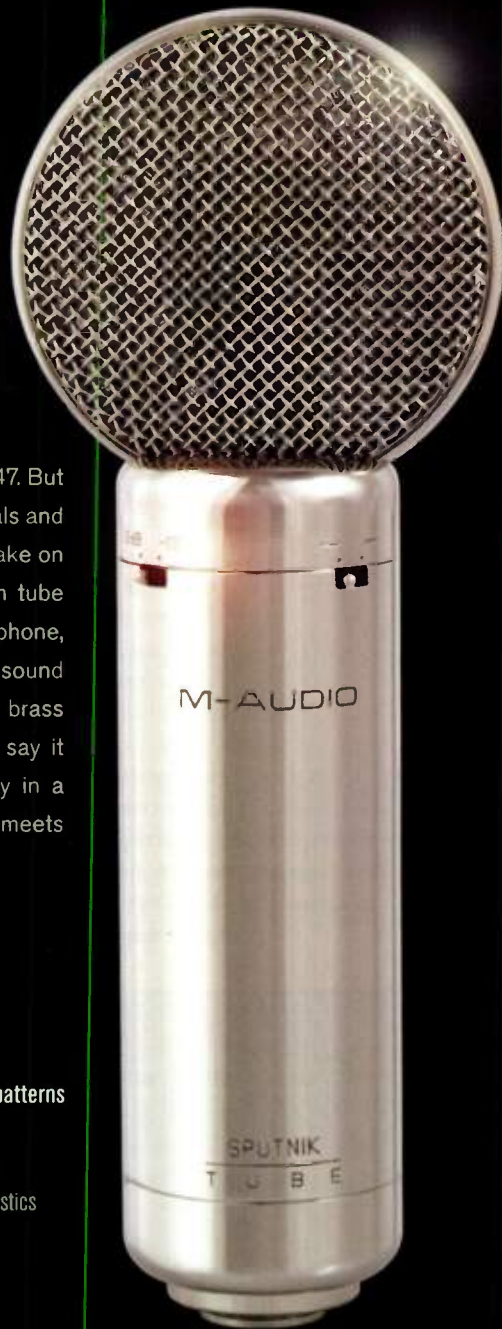
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and not because there is excess bottom end in your mix.

Get with the Program

You should have at least two pairs of monitors on which to evaluate the program. One pair should be full range (extending at least down to 35 Hz or so) or be paired with a subwoofer to extend bass response. The second pair should ideally be band limited (that is, bass deficient) to provide you with an idea of how the finished product will sound when played back on small consumer systems.

Full-range monitors suitable for use in mastering are too numerous to list here. But strangely, there are relatively few high-quality band-limited models on the market, so I can make some recommendations: the Yamaha HS50M and NS-10M Studio (the latter model is discontinued) and the Avant Electronics Avantone MixCubes are the best I've heard for this purpose. Most other tiny close-field monitors I've auditioned attempt to sound like a big speaker in a small box and have way too much bass response to serve as a proxy for small consumer-playback systems (not to mention that their bass response usually becomes highly inaccurate when placed on workstation shelves or a console meter bridge).

Both pairs of monitors (full range and consumer proxy) and the subwoofer should be wired to a switch box (or patched to separate control-room outputs on your mixer) so that you can select playback on either pair—alternately with and without the subwoofer engaged—at the push of one or two buttons. Depending on the capabilities of your gear, you may need to wire up a custom setup to allow simultaneous playback on one pair of satellites and sub. By
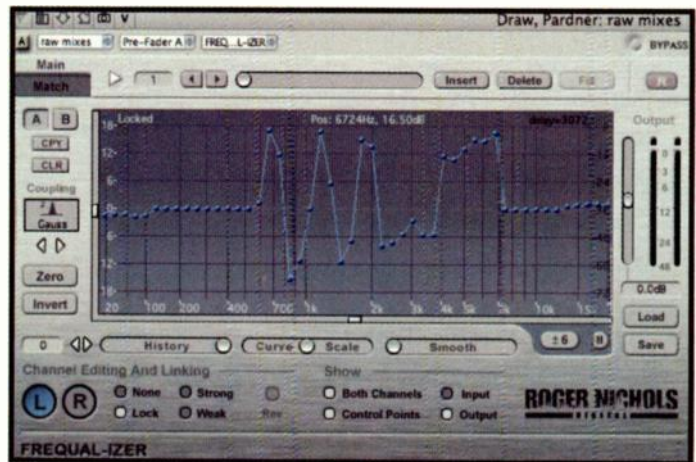
FIG. 4: Almost any equalization curve imaginable can be drawn using a mouse in the RND Frequal-izer plug-in's frequency graph. An overzealous mastering engineer got fired after applying this particular curve to the client's mix.

switching among different playback references while working, you'll be able to tell if your mastering changes will sound good across a variety of audiophile and low-end consumer-playback systems.

Equally important to having high-quality monitors is knowing their inherent frequency limits (bass and high-frequency rolloffs). For example, I know my NS-10M Studio monitors don't reproduce bass frequencies below 60 Hz very well. If I hear boomy bass guitar frequencies on my full-range monitors but the boominess disappears when I switch to my NS-10M Studios, I know the problematic frequencies lie below 60 Hz. And if I listen to playback on my subwoofer only (with satellites muted) and a boomy acoustic guitar track completely disappears, I know the boomy frequencies lie above the 110 Hz high-frequency cutoff of my sub.

During your mastering sessions, you'll occasionally and very briefly want to check your work at a loud volume to make sure extreme low and high frequencies are in proper balance with the rest of the spectrum. (Compared with the midrange band, the human ear is less sensitive to the extreme frequency ranges at low listening levels.) But avoid listening fatigue by resisting the urge to listen at loud levels for more than just a few minutes spread out over each day. And as the day progresses, take more frequent breaks to rest your ears and gain a fresh perspective.

Drag-and-Drop

At the beginning of the mastering session, your first task is to import all your mixes into your digital audio workstation (unless they are on tape, in which case any desired analog processing should be applied before you bring those tracks into your DAW). This is usually accomplished by simply dragging-and-dropping each file into the appropriate folder or window in your DAW

FIG. 3: URS is one of many manufacturers whose parametric EQ plug-ins can adjust boost and cut settings in exacting 0.1 dB steps, suggesting their use in mastering applications. The 7-band URS S Mix EQ plug-in is shown here.

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and, from there, into a blank track. If at all possible, you should be working with 24-bit files for your mixes.

After importing all the audio files for an album project, I will listen through virtually the entire project and make notes for each song as to what types of signal processing may be needed to correct problems or further enhance what's already good. (I'll discuss what some of those treatments might be in a bit.) This gives me a plan of attack for each song while my ears are their very freshest. It also reveals at a glance whether the same problems repeatedly crop up in most or all of the mixes. For example, I often hear unnecessary boosts or cuts in the same bass band for every song when a project was mixed in an inaccurate control room. In such a situation, the engineer compensated for a problem with their mixes that simply didn't exist, and part of the mastering process is to undo that equalization and restore spectral balance.

As I'm listening through and evaluating the project's mastering needs, I'll place markers at the beginning of each song. That will later allow me to quickly jump from songs for which I've already rendered signal processing and level changes to the one I'm currently working on, to check for consistency or artistic compatibility of spectral balance, dynamic range, and so on.

I will also place markers where at least one significant peak for each song occurs. This allows me to make comparisons between songs at those points to ensure that no song is much louder or softer, or more dynamic or compressed, than the others in the finished product.

Avoiding Blind Alleys

Before we dive into signal processing, it's important to set up a good monitoring scheme to facilitate work flow and maintain critical perspective throughout the session. During the course of the session, I may elect to add analog or digital processing or both. I'll want to refer back often to how the audio file sounded before I started
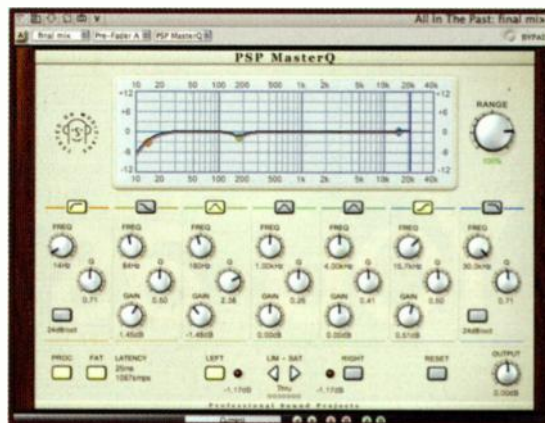

FIG. 6: Here, the 7-band PSP MasterQ linear-phase equalization plug-in is being used on only the left channel of a stereo mix. MasterQ also includes various limiting and saturation algorithms, which may be disabled.

messing with it, just in case I'm making things sound worse instead of better. To facilitate these comparisons, I'll set up multiple monitor-source points for listening to the song I'm currently working on at various stages of processing.

One monitor-source point will be for the original audio files playing back in the track they were first dragged into, with no processing added. A second monitor-source point will be for any of the same audio files that have already been processed in the analog domain and "printed" (recorded with the analog processing rendered to new files) to a new track (see **Fig. 1**). Each of these two tracks gets routed to a separate stereo digital input on my mixer feeding the exact same downstream monitoring path. I also set up a third monitor-source point (feeding another stereo digital input on my mixer) for the audio file currently being processed in the analog domain (monitored post A/D converter so that I'm making an apples-to-apples comparison with already-printed files). With this setup, I can compare—by selecting with a simple button push each monitor-source point in turn—the sound of the audio file for which I'm currently tweaking analog signal processing settings to both the sound of the original, unprocessed file and the sound of the files for other songs that have already had their analog signal processing imparted.

I now have two tracks in my DAW, one containing the original, unprocessed audio files for all songs and another containing the same songs with analog processing added (if they needed it). Typically, if one song in the batch needed some sort of analog processing, all others recorded and mixed in the same studio will need it too, though often to varying degrees. However, if I elect to forgo analog processing on one or more mixes, I can simply copy and paste their original audio files over to the track containing mixes that have had analog processing

FIG. 5: iZotope Ozone 3 incorporates six mastering processors that alternately share the same window. The graphic display for the Paragraphic Equalizer module is shown here.

applied so that all songs at this stage of the game are contained in the same track. I keep notes along the way as to which mixes had analog processing applied and which didn't, in case I need to go back and tweak prior settings. And I either document for each mix the exact settings for any analog gear used or, if possible, save those settings as a custom preset in gear that offers digital storage and recall.

Once analog processing has been rendered for any song files that needed it, I'll trim the head and tail of each processed mix to remove any unnecessary pre- and postroll. Then I'll bounce those files to a third track in my DAW while adding any needed digital processing (which, in some cases, might be limited to only dither).

Digital processing may be provided by plug-ins, outboard devices (such as digital compressors and equalizers), or both. As I add digital processing, I'll once again use multiple monitoring paths so I can compare my work in progress to the sound of mixes already printed with digital processing and also to the same mixes with only their analog processing (if any) applied.

I'll also store the settings used for digital outboard processors and plug-ins as custom presets named after the song they were used on. This makes it easy for me to revisit those settings in case I need to later rework the processing for any particular song. When mastering in MOTU Digital Performer, as I regularly do, I'll save the settings for all the plug-ins used on a particular song as a group in a Clipping; that allows me to later restore the whole kit and caboodle with one simple drag-and-drop operation into Digital Performer's mixer. BIAS Peak Pro users can do essentially the same thing by saving and recalling a Vbox matrix of plug-ins for each Playlist Event; this has the advantage that a Vbox and all the plug-ins it contains can be bypassed or activated with
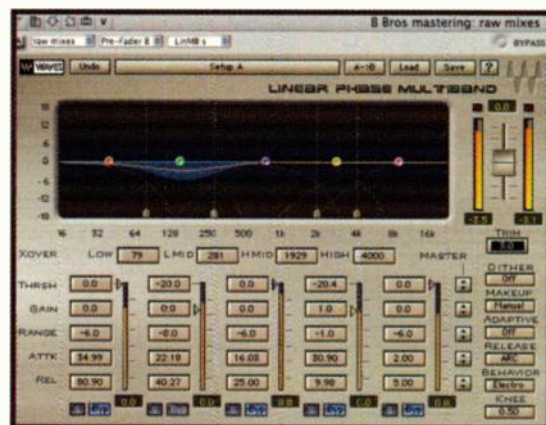
FIG. 8: This screen shot shows the Waves Linear Phase Multiband plug-in being used to reduce boominess in the 79 to 281 Hz band. The other bands are bypassed, leaving their frequencies untreated.

one mouse-click for A/B comparisons between original and processed mixes (see **Fig. 2**).

The above process has been oversimplified for the sake of clarity. In reality, I may sometimes elect to add corrective digital processing (to fix obvious problems with the current mix) before the D/A conversion on the first pass through analog processing if it's obvious I'm going to need it, and if failing to do so would make it harder for me to hear the best settings for the analog gear. Additionally, I prefer to bounce my fully mastered mixes (those rendered with any needed analog or digital processing) to disk and then import the finished file into the third track mentioned earlier, rather than bouncing directly to the track, as the results sound a little more transparent to my ear when working in Digital Performer.

It must be said that the multichannel paradigm I've presented here isn't the only way to master. If you're mastering in a 2-bus application such as Peak, for instance, you can listen over the same output bus to multiple documents in turn (each representing a different audio file, such as processed and unprocessed versions of the same mix) simply by pressing a number key on your computer keyboard. It all comes down to how you want to work. The point is that it's important to set up a monitoring scheme that allows quick A/B/C comparisons.

Put It in Gear

When building your arsenal of mastering gear, it's crucial to know what makes a product specifically tailored to this application. The most distinguishing and vital aspect—beyond superior sound quality, which is a given for mastering applications—is repeatability. Mastering gear must provide a way to recall your exact settings so that you can try different things and always return to prior settings if you take the wrong fork in the road.
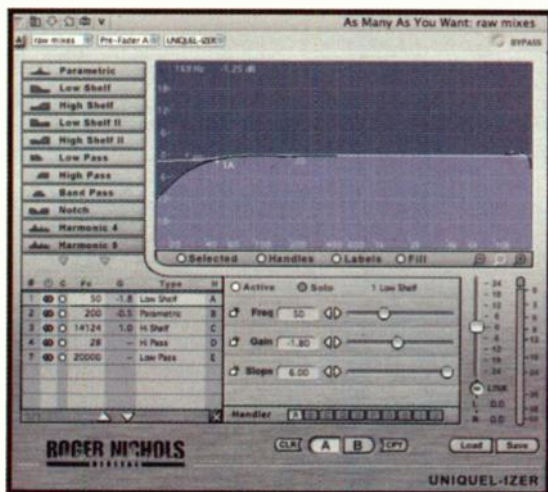

FIG. 7: As long as your CPU can handle it, you can add as many bands of equalization as you want in the RND Uniquel-izer plug-in. Choose from 11 different filter types for each band.

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This repeatability can be attained either through detented or switched control settings (which you must remember in the short term and document for the long term) or by including facilities for digital storage and recall (as do, for example, digitally controlled analog processors). Obviously, DAW plug-ins easily meet this criterion, as all their settings can be stored in user presets. An

Mastering can work miracles on poorly wrought material that cannot be remixed.

increasing number of plug-ins are going one step further by providing two or more work spaces (alternate A and B control setups), allowing the user to try a couple of different tacks and toggle back and forth between them to see which sounds best.

Another requirement for mastering equipment is that all controls have very small incremental steps. For instance, equalizers should ideally have no greater than 0.5 dB steps between adjacent boost or cut settings if they are to be used in mastering. The most important rule of mastering is "First, do no harm," and handling a project with kid gloves requires that the gear used be capable of making very subtle adjustments.

Analog Demagogue

Despite having many suitable plug-ins at my disposal, I often send mixes that have been created entirely inside a DAW out to the analog domain for processing. I've yet to hear plug-ins that can produce the warm, nuanced sound of, for example, a Millennia NSEQ-2 parametric equalizer or Tube Tech SMC 2BM tube multiband compressor. When cold, digital tracks need warming up, some mastering engineers record DAW mixes to an analog half-track machine before they even touch them with EQ or compressors.

That said, digital processing—both hardware- and software-based—has a lot to offer. For instance, if you're looking for a convincing emulation of tube and tape saturation without leaving the digital domain, the Crane Song HEDD 192 (a hardware device) has found favor with mastering engineers. And there are a number of plug-ins

FIG. 9: Universal Audio UAD Precision Limiter is an excellent mastering tool for cranking up the volume and power of your mixes.

and bundles that excel at mastering, not just because they sound great, but also because they have capabilities their analog counterparts can only dream of.

For example, most parametric equalizer plug-ins can adjust gain boost or cut in surgical 0.1 dB steps (see **Fig. 3**). And with the Roger Nichols Digital (RND) Frequal-izer plug-in, you can draw any equalization curve you want with your mouse (see **Fig. 4**). Try doing *that* with an analog equalizer.

There's also your budget to consider—a high-end analog mastering equalizer or compressor might cost you several thousand dollars, whereas a bundle of mastering plug-ins such as the excellent-sounding iZotope Ozone 3 will set you back less than $250 (see **Fig. 5**). In the end, both analog and digital processors can yield excellent results; it's what you do with them that counts. That's what we'll examine next.

In the Trenches

Let's take a look at some examples of what you should listen for when mastering and how to evaluate what types of processing might be helpful to make the changes you want to hear. Of course, an in-depth discussion of signal processing techniques is beyond the scope of this article, but the following mastering tips will give you some useful ideas to pursue further.

First, switching among your different pairs of monitors during playback should give you a good idea of where the spectral balance may be skewed. For instance, if the current mix lacks presence on your full-range monitors but sounds balanced on your band-limited speakers, the problem probably isn't a lack of upper midrange frequencies but too much bottom end.

As you evaluate what equalization may be needed to restore the proper balance, keep in mind that the problem frequencies may be affecting only one side of the mix. For example, a guitar part with excess energy in the upper-bass band may be panned more or less to the left side of the stereo field, in which case cutting in that band on both channels would cause the right channel to sound thin. The PSP MasterQ equalizer plug-in is highly useful here, as it allows you to equalize only one side of a stereo file while leaving the other side untouched (see **Fig. 6**).

The RND Uniqual-izer and Frequal-izer plug-ins both go one better, allowing you to apply different EQ treatments to the left and right sides of a mix simultaneously. With Uniqual-izer, you can add and delete as many bands of EQ as your CPU can handle, choosing from 11 different filter types (see **Fig. 7**).

As you work, listen also to the current mix's dynamics. Does the mix sound full enough? If not, some light stereo-linked compression with moderately fast attack

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and release times and a low ratio should add some pleasing "glue" to the performance. Just be careful not to go overboard, or depth will go out the window and your mix will start to sound two-dimensional, squashed, and lifeless.

Perhaps guitars and keys are blanketing the trap drums too much in your mix. Instead of EQ'ing the traps so they sound brighter, try applying stereo-linked compression with slow attack and fast release times. That should make percussive elements such as kick and snare drums "pop" more.

Boom Town

Say you've got a mix in which the acoustic guitar gets boomy every time it plays the D string, but it sounds fine at all other times. (Make sure this is really the case and isn't just a room mode blossoming at that frequency.) A static equalization cut around 160 to 200 Hz will tame fundamental frequencies for that D string (if fretted below the sixth fret), but it will also thin out all other guitar passages, as well as the entire mix, when that string isn't played. A dynamic approach is called for here.

You can use a split-band compressor such as the Waves Linear Phase Multiband (part of the Waves Masters bundle) or iZotope Ozone 3 Multiband Dynamics plug-ins to put a lid on the 160 to 200 Hz band so that its energy is dynamically cut every time the boominess surpasses the threshold set for that band.

FIG. 11: Track offsets should be set for your premaster to prevent CD players from muting audio at the beginning or end of each CD track. In this example, global track offsets are entered in the Delivery window (Windows→Preferences→Delivery) for Sonic Studio PreMaster CD, a premastering application.

(You might actually need to set the bandwidth wider to also tame boomy formants; see Fig. 8.) Both of these plug-ins can also execute upward expansion (sometimes called bootstrap compression) that will make quiet song passages such as a solo guitar intro louder without compressing the top end of the dynamic range and squashing peaks.

Of course, squashing peaks is where it's at if you want your mixes to be competitively loud. Plug-ins such as Universal Audio UAD Precision Limiter (see Fig. 9), iZotope Ozone 3 Loudness Maximizer, and Waves L3 Multimaximizer and L2 and L1 Ultramaximizer can really pump up the volume of your mixes by clamping down on peaks and bringing up the average level. The danger is in going too far, killing any punchiness and making your project sound like a fatiguing onslaught nobody can stand to listen to for more than five or ten minutes at a time. In fact, for some musical styles, such as classical, using *any* amount of limiting or maximizer processing would be inappropriate. If used, maximizing and dithering (which I'll discuss momentarily) should be the very last processes you apply to your mixes.

Waves L3 Multimaximizer is an especially powerful mastering limiter because it can condition different frequency bands to alter the limiter's response. With L3, you can subtly favor average levels in one frequency band and peak levels in another by changing the plug-in's Gain and Priority settings (see Fig. 10). For example, you can increase the energy in the bass band and enhance the upper-midrange-frequency component of snare drum hits while keeping a firm lid on other bands. L3 includes the Waves IDR word-length-reduction
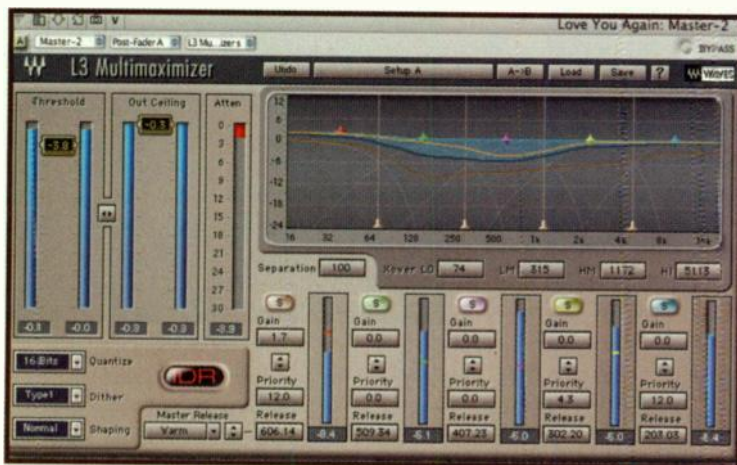
FIG. 10: Users can condition the response of the Waves L3 Multimaximizer's limiter in five different frequency bands. With the settings shown, the average level of a cello section and, to a lesser degree, the crack of snare drum hits are enhanced in a mix while other elements are more tightly controlled.

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algorithm, a highly transparent quantization and dithering process useful for rendering your files to 16-bit format for CD release.

There are many other types of signal processors available for use in mastering, including harmonic exciters, reverb, and stereo imagers. The Waves S1 and iZotope Ozone 3 Multiband Stereo Imaging plug-ins can alter the perceived width of your mix, but not without making it sound more diffuse (which may or may not be appropriate). Mid-side (M-S) processing is an advanced mastering tool that can also be used to adjust your mix's width. By encoding a stereo mix into an M-S matrix, audio common to both channels such as center-panned tracks (the mid component) and audio that's exclusive to both channels (side) can be separated and independently equalized and compressed. For example, you can use M-S processing to beef up the kick and snare drums while raising the level of an uncompressed stereo string pad to widen the mix. (For more information on mid-side processing, see "Front and Center" in the March 2006 issue of EM, available at www.emusician.com.)

As mentioned earlier, before you maximize and dither your files to 16 bits for CD release, you should trim the head and tail of each file to remove unneces-

sary noise at the start and end of each mix. Then create fades at the beginning and end of each trimmed file to avoid any pops or clicks that would otherwise result from potential DC offset or low-level noise instantaneously slewing up or down at butt-splice margins. Finally, after any last-minute fader adjustments or other gain changes are made so that perceived loudness flows smoothly from song to song throughout the entire program, maximize and dither all the files to 16 bits. To preserve the depth of your mixes, make sure you apply no further signal processing—including gain changes—after you dither your files. And avoid dithering more than once if possible, in order to prevent potentially audible artifacts from polluting your files.

Premaster Prep

After all of your mixes have been trimmed, faded, processed as needed, and dithered to 16 bits for CD replication, import them into whatever software application you've chosen to prepare your premaster. (The premaster is the disc or file from which a glass master is made at the replication house in preparation for mass-producing your CD.) If you've been using, for example, BIAS Peak Pro XT to master, your files are already ready for premastering in Peak's Playlist and don't need to be imported. (I'll talk about Peak and other premastering solutions in greater detail in a bit.)

Due to space constraints, I can't completely discuss premastering, but I'll hit on some of the major points. First of all, resist the temptation to make any further gain changes once signal processing has been rendered and your mixes have been dithered; doing so would degrade the quality of the audio portion of your CD.

Your premastering software should automatically assign a different CD-track number to each audio file you import, but you can rearrange the song order and their track numbers if you want. After the song sequence is set the way you like, set the duration of any gaps (silent portions) you want to have between CD tracks. One CD track is typically composed of only one song but may be fashioned to include a medley of several songs or a segue.

CD players take a fraction of a second to fill their play buffers with the current track's audio, during which time the player's output is muted. So if you keep your CD track's

DO UNTO OTHERS

If you're considering opening a commercial mastering studio, one of the first decisions you'll need to make is which formats to support. The best-equipped facilities support a wide variety of analog and digital formats, both for compatibility with clients' masters and, in some cases, for use as signal processors in their own right (as is the case when transferring program material to certain analog tape machines). The commercial operator must weigh the expense of supporting multiple formats against what the market requires and competitors offer.

Mastering other people's projects also demands that extra weight be given to considerations beyond your own creative leanings. First of all, it's important to ask your clients what their likes and dislikes are. That huge bottom end on their mix might be an annoyance to them—something that needs fixing—and not an intentional production value. How do they feel about competitive loudness? Do they want their record to be louder than anyone else's at any cost to sound quality? And if it's a band's project that you're mastering,

which member(s) call the creative shots?

Before you begin working on a project, give a quick listen to short sections of a few tracks and offer your opinion of what needs to be done to fix problems (such as a murky midrange masking vocals and guitars) and also what *can* but doesn't necessarily have to be done in the realm of creative enhancement (such as making the drums slam more or widening the stereo image). Write down your client's responses and direction and use those notes throughout the mastering process as your homing beacon to keep true to the artist's vision for the project.

The need to work fast must be weighed against all potential engineering considerations. Unless you're working for experienced artists or companies, only a very small portion (if any) of the project's overall budget will likely be allocated toward mastering. Whether you're working for an hourly or per-song rate or a flat fee, you'll need to stay aware of the time you're spending, focus on the major issues, and realize when the law of diminishing returns is kicking in and it's time to wrap things up.

default start time, or *track index*, positioned at the onset of the audio program, playback of the very beginning of the track may be cut off. The solution is to program *track start offsets*, which are silent gaps that occur between each track index and the beginning of audio for the respective CD track (see **Fig. 11**). The first CD track typically requires a longer start offset than that for all the other tracks on the CD. It's also important to add *track end offsets* to prevent the *end* of a CD track from getting muted before its audio program fades completely out. (I've been told by a reliable source that this is an issue only when a

CD player is in shuffle or random-play mode.)

Start and end offsets are usually programmed globally so that they are the same duration for every track after track 1 (which we noted gets its own special offset applied). That said, Peak can set different start offsets for each song (in the application's Playlist), which is useful when you want to arrange a specific start point during a crossfade between two tracks. As consumer CD players don't all mute their outputs for the same amount of time while filling their play buffers, play it safe and set the premaster's global offsets to be a tad longer than you think you might need. Safe numbers are 75 *CD frames* for the first track, 25 CD frames for track start offsets, and 15 CD frames for track end offsets. An offset of 75 CD frames is equal to a 1-second duration, so each 25-frame track start offset is ⅓ second in duration and each 15-frame track end offset is ⅕ second long. When calculating how long the silence will last in between each song on your CD, it's important to add the track start and end offsets to the gap time you programmed, to arrive at the real silent-gap length.

If your computer's operating system and drive can write CD-Text data (you'll need Mac OS X 10.4.3 or later if you're using a Mac), you may want to enter this information, which includes album and song titles, the names of the performer or band and songwriter, and ISRC data. (Go to www.ifpi.org/content/section_resources/isrc.html for more information on obtaining ISRC codes for your CD tracks.) Once all text data is entered, use your software to print out your premaster's contents so that you can give a text readout of all PQ codes (which generate the disc's table of contents, or TOC), track offsets, and other data to the replicator.

If your software enables it and your replicator can accept it, export a DDP (Disc Description Protocol) file set and burn it to a CD-ROM. DDP file sets contain error-protected audio data and all the metadata (or "data about the data" on your premaster) needed by the pressing plant. DDP file sets are a superior delivery format compared with Red Book CD-DA discs, which are prone to errors, but not all replicators can accept them.

Learn to Burn

Several premastering programs are currently on the market, some of which are

incorporated into mastering software. Here is a quick look at what's available.

With a $200 list price, Roxio Jam 6 is an inexpensive disc-burning application with a limited premastering feature set and no plug-in support or DDP-export capability. BIAS Peak LE ($99) includes audio file editing, plug-in support, and disc burning, but it can't export DDP file sets either. BIAS Peak Pro 5 ($599) offers plug-in support and advanced editing, mastering, and premastering features, including, with an optional extension costing $399, DDP file set export. BIAS Peak Pro XT 5 ($1,199) adds the company's Master Perfection Suite plug-in bundle and SoundSoap and SoundSoap Pro restoration software to Peak Pro 5's feature set and can also utilize the company's optional DDP extension.

Sonic Studio PreMaster CD ($475 for the download) offers intuitive editing and premastering capabilities but no plug-in support. It is the least expensive solution for DDP file export. However, PreMaster CD also forces you to set the same gap between every song, while even the rock-bottom-priced Peak LE and Jam 6 allow the gaps between songs to be set to different lengths. The 4-track, 4-bus Sonic Studio soundBlade

(native, $1,495; with accelerated DSP, $3,995) does everything PreMaster CD can do but also adds powerful plug-in architecture and advanced editing, mastering, and premastering capabilities. SoundBlade offers expansion options for applications such as restoration as well.

Job Export

There's a lot more to mastering than simply slapping a maximizer on your project and calling it good. (In fact, we've just scratched the surface in this article.) And while many studios offer mastering services these days, there is a wide divergence in know-how and quality between the best and worst of the crop. If you can only afford the mastering services of a bottom-rung studio, you may find that you can obtain better results by doing the work yourself. With an accurate room and monitors, quality mastering gear, good ears, and technical chops, there's no reason not to give it a shot. **EM**

EM contributing editor Michael Cooper is the owner of Michael Cooper Recording in beautiful Sisters, Oregon. Examples of his mastering work are posted at www.myspace .com/michaelcooperrecording.

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Old-School Contemporary

By Paul Tingen

Gil Norton
produces new
acts with classic
methods.

I f you think that most contemporary records are constructed "in the box" by young musicians endlessly peering at screens and applying multitudes of plug-in effects, or that conventional recording studios are for an older generation on its way out, meet Gil Norton. Consider this list of rock musicians, all of whom he recorded and produced using his tried-and-tested old-school ways: Dashboard Confessional, Jimmy Eat World, Feeder, Gomez, Ben Kweller, and Morningwood.

With 26 years of working in recording studios, Norton, an Englishman, definitely qualifies as old guard. At the same time, with credits including Echo and the Bunnymen, Foo Fighters, the Pixies, and Throwing Muses, he's that rare combination of living legend and contemporary, happening producer (see the sidebar "Gil Norton: A Discography"). Norton specializes in producing bands of the post–new wave and grunge varieties, with the occasional oddball thrown in, like the acoustic–alt rock sound of Counting Crows and Gomez or the pop sound of Morningwood.

Norton's productions feature guitar-driven walls of sound, played by real bands in traditional recording studios—in other words, projects that involve strings, pedals, amps, drums, microphones, baffles, mixing desks, and outboard gear. In short, Norton deals in everything that sitting behind a computer screen is not.

Instead, he delivers an invaluable service by carrying old-school studio know-how into the 21st century. The ongoing commercial and artistic success of his approach shows how relevant it remains. The roots of Norton's methods can be found in his beginnings in the late '70s when he studied music and trumpet at Mabel Fletcher Technical College in Liverpool. While there, he played bass in bands and recorded musical acts in the 8-track demo facility of Amazon Studios. Norton recalls, "A lot of bands couldn't tune their drums—some couldn't even tune their guitars *[laughs]*—so I ended up getting heavily involved in the musical side of things and drifting into production."

Speaking from one of his favorite studios, central London's RAK, Norton agrees that his approach is "totally traditional," and adds, "I mainly do bands, so it's not like I'm involved in dance music, which has more to do with sonics, drum machines, loops, and keyboards. When you're working with a real drummer and a real kit, you get a lot of character. So most of the time I work in a traditional studio environment, and it's all about capturing the band as best as possible. It's all about the songs and giving guidance about performances and why you think they're great or could be better. I like the whole psychology of working with a group of people. I like human interactions."

FIG. 1: Adrian Bushby often engineers for Norton, who prefers to concentrate on the production aspects and leave the knob twiddling to others.

Firm Believer

After engineering at Amazon, Norton worked for nearly a decade as a producer-engineer, but the last album he engineered entirely himself was the Pixies' *Doolittle* in 1989. Since that time he prefers to have a good engineer at his side. "When you are worried about the technical aspects of things, it's harder to concentrate on the music and the band," he says. In the past five or so years, he's worked frequently with engineer Adrian Bushby (see **Fig. 1**), who is a producer in his own right. Bushby has engineered for Norton many times, working with Dashboard Confessional, Feeder, Gomez, Natalie Imbruglia, Minogue, Morningwood, My Bloody Valentine New Order, and Kylie Placebo.

Norton and Bushby collaborated recently on Gomez's *How We Operate* (see **Fig. 2**). The CD is the band's fifth, and its first on the independent ATO Records label. *How We Operate* climbed to No. 1 on Billboard's Heatseekers chart and No. 7 on the Billboard Top Independent Albums chart, receiving widespread acclaim (allmusic.com called it "a quiet stunner"). It was also the first Gomez CD that wasn't self-produced. Norton's influence is clear; the songs and arrangements are more downbeat, acoustic, melodic, and integrated than the band's previous offerings.

"When I met the band, the big thing for me was that I wanted them to work as a band," says Norton about recording *How We Operate*. "I didn't want a bitty album of individual writers doing their own thing and the others acting like session musicians. I wanted to put them in a room together and work them through the songs. I am a firm believer in preproduction before going into the studio; to me, it is the most important part of recording. Gomez had never done that, because they tended to do everything in the recording studio. I wanted to make sure that we all understood what each song was about

GIL NORTON: A DISCOGRAPHY

Gomez, *How We Operate* (ATO, 2006)
Ben Kweller, *Ben Kweller* (Red Ink, 2006)
Morningwood, *Morningwood* (Capitol, 2006)
Jimmy Eat World, *Futures* (Interscope, 2004)
Dashboard Confessional, *A Mark, A Mission, A Brand, A Scar* (Vagrant, 2003)
Feeder, *Comfort in Sound* (Universal, 2003)
Patti Smith, *Gung Ho* (Arista, 2000)
Foo Fighters, *Color and Shape* (Capitol, 1997)
Counting Crows, *Recovering the Satellites* (Geffen, 1996)
Belly, *Belly* (Sire, 1993)
Catherine Wheel, *Chrome* (Fontana, 1993)
The Pixies, *Bossanova* (4AD, 1990)
Del Amitri, *Waking Hours* (A&M, 1989)
The Pixies, *Doolittle* (4AD, 1989)
Throwing Muses, *Throwing Muses* (4AD, 1986)
Echo and the Bunnymen, *Ocean Rain* (Sire, 1984)

REALTIME INSTRUMENTS

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"I didn't want a bitty album of individual writers doing their own thing."

and that everyone had their parts worked out before we recorded. They've been touring a lot, especially in the U.S., and I thought it'd be good to have some of that energy on the record."

In addition, the budget for *How We Operate* was relatively limited, which added to the incentive to come to the studio as well prepared as possible. Preproduction took place at the band's rehearsal room and studio in Brighton. "It's a nice environment," says Norton, "and we would try out different arrangements and lots of different drum parts and feels. We didn't demo to a high standard and didn't use any of the demo recordings. It was more a matter of putting ideas down and checking whether they worked. Nearly all the lyrics, arrangements, and rhythms had been completed by the time we began recording in Studio 1 at RAK in October 2005. 'See the World' was the only song that wasn't quite worked out, so we had to spend time in the studio on the rhythm section for it."

Shock and DAW

Norton's main reasons for recording in RAK Studio 1 were its API desk, enormous live room, and extensive mic collection. "We tried to do as much as we could live, and then added overdubs," he says. "The room at RAK is big enough to be split in half, so we had the drums in one section and the amplifiers, keyboards, and vocal mics in the other. The band members were all playing at the same time; there was a little bit of spill, but not a lot. We had lots of screens and blankets and stuff. We made sure the tempos were sorted out and recorded the tracks live as much as we could. We were in the big room for 15 days—each day recording the basic tracks for one of 14 songs, even though only 12 made it to the album."

Norton confesses to having a predilection for older Neve and API boards when recording. "That is where you get the sound," he explains. "So most of what we recorded in Studio 1 went through the API. It's nice and clean; it sounds great. After the two weeks there, we went to RAK Studio 4, which is just

a back room with only an SSL and a small overdub booth. We did a little bit of fiddling with the acoustics there to turn it into a great-sounding mix room. We were there until Christmas, overdubbing mainly guitars, keyboards, vocals, and finally mixing."

For the longest time, Norton considered himself a purist—a man who preferred tube and analog gear. But even though plenty of analog compressors, mic preamps, and EQs were used on *How We Operate*, the CD was tracked using Norton's two Digidesign Pro Tools systems, both filled to the brim with plug-ins.

Norton became a digital convert about two years ago. "Until recently, digital wasn't very good," he explains. "I used the Mitsubishi X850 digital 32-track recorder with Echo and the Bunnymen, and that sounded okay in the beginning. But the more I added, the more I could hear the digital processing, and it sounded harsh. I was still using analog when I did *Recovering the Satellites* with Counting Crows, and ended up filling 100 reels of 24-track tape because we

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NORTON'S OLD-SCHOOL ADVICE

Norton divulges his five top tips for successful production:

1. Be prepared. "As a matter of principle, I do at least two weeks of preproduction with almost every band I work with. You can't get the best results when you write and arrange in the studio, because there's too much pressure. You need time to sit with things and make sure you have the right ideas. The more work you have done before you go into a studio, the faster the whole recording process becomes."

2. Don't settle for less than best. "A producer's job is about bringing the best out of people and their music and making sure that they enjoy being in the studio. One important thing is to be sure that they don't undersell themselves. Sometimes people will say, 'Oh, that will do,' but in reality it may not be good enough. It's important that you challenge them to do better."

3. Always explain yourself. "You have to be very articulate as a producer. If you confuse a musician in the studio, it's the end. You can't just say, 'I don't like that.' You have to give people a good reason why you say what you do and warn them about the things to look out for. It's important for musicians to feel confident and to know what they're doing."

4. A click track is essential. "Lots of drummers these days don't like playing to a click track, but clicks prevent bad takes. That's because click tracks stop bands from pushing or slowing down when the musicians get tired. Some songs need to move in tempo, and you can move the tempo up and down a bit using a tempo map. Having a tempo map also makes editing a lot easier."

5. Record as a band. "I always try to get the whole band to play the songs together live in the studio, and normally I get them to do about three or four guide vocal takes at that time. When you work all day on a song and there's a vocal on tape, it refreshes your head and the band gets to know the song better. You also get very fresh vocals that way. It can take a while for vocalists to get into the mood of a song, and so I normally get a lot of the vocals during these backing-track takes."

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had recorded the preproduction as well. Then, when producing Jimmy Eat World's *Futures* album in 2004, we did a big A/B shoot-out at Cello Studios in L.A. and compared a 2-inch, 16-track Ampex machine with Pro Tools HD 96/24. We all felt that the Pro Tools sounded better, especially for rock music. That was a bit of a shock.

"The big thing is the HD sampling rates. That is where digital has become as good as analog tape—you start getting the air around the sound. It's more punchy than what's coming off the analog tape, so everything that we record now goes straight to Pro Tools via the 192s. I still use analog because it can make things sound softer and warmer, so I mix to ½-inch analog and master off that. Occasionally, I'll move something over to analog tape and back to Pro Tools again just to get the tape compression, and I still enjoy doing tape edits with razor blade. But digital is great for the convenience of editing and recording lots of takes. Tape is so expensive, and you can only record three takes on one piece of 24-track tape. Now you can do eight or nine takes in Pro Tools and see what you have."

And so, to his own surprise, Norton has found himself recording exclusively in the box since 2004, and his working methods are now a blend of old and new. "I have a little [Digidesign] control surface," he says, "but I still prefer to use conventional desks. I'm running one Pro Tools system on my old [Apple] Dual 1 GHz G4, which I use for recording, and the other, which runs on a G5 Dual 2 GHz, is for editing. I try to have the editing done in a different room than the control room, so it doesn't get in the way of recording. I don't think bands like sitting around looking at a screen all day; they prefer to be creative and record stuff. They don't want to sit there while someone is editing, so I tend to keep that away from the band."

Plugged Up

Norton has also become increasingly enamored with plug-ins. "Most plug-ins sound good, but it's very difficult to compare them with outboard gear. You can't say that plug-ins sound better than outboard gear; an 1176 plug-in doesn't sound better than a real 1176. They just each do a different thing. And you can overuse plug-ins, just like you can overuse outboard gear. If you have good instruments and good players and good microphones, you don't need to do a lot of EQing. You shouldn't be using plug-ins to rescue something; they're really for adding a little bit of enhancement here and there."

Norton rarely processed the tracks on the Gomez CD before sending them to Pro Tools. When he did do some processing, he used outboard compressors such as the LA3A, 1176, and Empirical Labs Distressor, as well as the Thermonic Culture Vulture (tube distorter). Once recorded, the material was edited and subjected to all sorts of treatments. (For more technical details about the recording process for *How We Operate*, see **Web Clip 1**.)

Some of the CD's tracks were either recorded or processed remotely, such as the drum programming and vocal mangling on "All Too Much," performed by Ben Frost, an Icelander living in Australia. Frost, a friend of Gomez singer-guitarist Ian Ball, added effects to audio files sent to him over the Internet. MIDI string arrangements were done by another friend of Ball's, Fil Krohnengold, who sent MIDI files back to London, where Norton and company fleshed them out with the aid of a MOTU Mach 5 sampler and samples from one of the Miroslav string libraries and from Norton's own collection.

Pushing the Boundaries

Norton's G5 Pro Tools HD3XL rig has 24 outputs, but the system he runs on the G4 has only 16.

Because he and Bushby wanted to be able to move freely between the two systems, the duo premixed a lot to 16-tracks in Pro Tools as the project went along. This made the final mix, performed by Norton and Bushby together, straightforward. "With all our recent projects, we mixed as we went along," says Bushby. "If we got balances that we liked, we saved them. When it came to mixing the Gomez album, we added a few more plug-ins and used the SSL desk a bit more—especially its stereo compressor, which gives it a sound."

"I like the idea of mixing in the box," adds Norton, "but for now I still prefer mixing on SSLs. I like their automation; it's very simple, and I'm very familiar with it. My HD3XL system is quite powerful and has a lot of plug-ins. I like the Sony Oxford plug-ins. Their EQ sounds great—very sweet; their compressor is good; and I really like their Inflator—we used that a lot. I have the Waves bundle and use the C1 compressor a lot. Reverb is the only area in which I always prefer using outboard, often the Lexicon 224 and 480. I enjoy plates on vocals, with a little bit of a Pro Tools delay."

When asked about the ratio of plug-ins to outboard gear, Bushby estimated that more than 50

percent of the effects used on *How We Operate* were plug-ins. Understandably, his list of favorites is similar to that of Norton's, though he adds a few of his own. "The Waves TrueVerb is probably the best-sounding computer-generated reverb, but springs and plates sound more natural. I love my old spring reverb. I enjoy the [Sony] Oxford Dynamics, which has a little setting for warmth that blends in well and gives it a really nice sound. Waves has a great de-esser that does the job very nicely without being heard too much. Focusrite and Bomb Factory also make good plug-ins; I particularly like the Bomb Factory 1176. As for outboard gear, I used my Chiswick Reach and ADL stereo compressors and my pair of Neve 33114 EQs. I also managed to get a Chandler TG1 compressor and an API EQ into the mix room."

Asked whether he foresees an entirely in-the-box future, Bushby shakes his head. "I don't think I'll ever use just plug-ins. There's something about having gear that you can get your hands on; it reacts in ways that plug-ins can't emulate. You can already do a mix without using any analog gear, but things done entirely in the box can sound a little flat, and there's something that analog brings to everything. If you are doing everything in the box, however, I'd advise at least putting a good outboard stereo compressor and EQ across your mix. It's important for things to go through that stage at least once."

FIG. 2: Norton's belief in preproduction was in evidence when he produced Gomez's *How We Operate*. The band refined the arrangements by recording all the songs first in their home studio.

Norton says, "I'm not a purist anymore. It doesn't matter to me where things come from." As ever, his focus is less on the technology and more on music and performances. This is exemplified on Gomez's *How We Operate*. The sound of the album is first-class, but the atmospherics and performances are what come across most strongly. "My job is about making sure that I get the right performances and that people enjoy being in the studio," concludes Norton. "I'd like them to experience it as a creative place where they can work and feel that they can push their boundaries." **EM**

EMWEB CLIPS

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). For more information, visit www.tingen.co.uk.*

Controlled Chaos By Len Sasso

Get great results from Stylus RMX Chaos Designer.

Chaos Designer in Spectrasonics Stylus RMX 1.5 is a great tool for remixing loops and generating new ideas, but as with any random process, some results are better than others. Setting up Chaos Designer intelligently and capturing multiple passes from which to choose will improve your results immeasurably. Here are some tips and tricks for doing both.

I've used Ableton Live 6 to host Stylus for my examples because Live is particularly well suited to recording and manipulating clips from multitrack virtual instrument plug-ins. But you can use most of the tips given here with any full-featured digital audio sequencer.

Chaos Designer is intended primarily for manipulating sliced audio files in Spectrasonics Advanced Groove Engine (SAGE) format. With the aid of the included SAGE converter, however, you can also import Propellerhead's REX2 files and Akai and Roland Groove Control files for use in Stylus. The Stylus factory library is excellent and large, but it's weighted heavily toward percussion, so keep the imported formats in mind as a path to nonpercussive chaos (see **Web Clips 1 and 2**).

Wreaking Havoc

Chaos Designer independently manipulates seven slice-playback parameters: Pattern, Repeat, Reverse, Timing, Pitch, Dynamics, and Buzz (see **Fig. 1**). Those manipulations are probabilistic; a slider sets the likelihood of each action. The last four effects have additional controls for parameters such as amount and direction.

FIG. 1: Stylus RMX Chaos Designer modulates various playback parameters in real time.

Pattern shuffles the slices. Repeat repeats individual slices. Reverse plays slices backwards. Timing introduces timing variations, which you can tilt toward leading or lagging the beat. Pitch applies a random pitch-shift. Dynamics changes the playback level. Buzz retriggers a slice at very short intervals, with controls for the rate of retriggering and the amount the retriggering rate slows down or speeds up over the duration of a note.

Stylus can play eight audio clips simultaneously (each clip has its own Chaos Designer), and you can route the clips to separate stereo outputs. Host permitting, you can capture multiple Stylus outputs in one pass, but if your host makes that difficult or impossible, you can always record one output at a time. If you're using only the Pattern, Repeat, and Dynamics effects, you can drag a MIDI file from Stylus into a host track and use it to trigger the slices remotely. The other Chaos Designer effects use DSP and can be captured only as audio.

You can get a great deal more out of Chaos Designer by using Stylus Edit Groups. They allow you to group slices by a variety of criteria. Each Edit Group has its own Chaos Designer, so you could, for example, reverse slices on upbeats, pitch-shift offbeat slices, and repeat a manually selected group of slices.

Getting Down

Once you've designed your chaos, you'll want to capture a number of passes of each loop. In Live, Stylus will be hosted on a MIDI track, so route that track's audio output to the input of an audio track; that's how you capture Stylus output A. Next create a separate audio track for each of the other active Stylus outputs. On the MIDI track hosting Stylus, create a MIDI clip with a Clip Envelope to trigger the Stylus AllPlay button. That's necessary to do because Stylus does not sync to the host's transport.

To find the length of each Stylus loop, drag the loop's MIDI trigger sequence from the Stylus file browser to the Stylus MIDI track in Live. You can then look at it in Live's Clip editor to determine the span of the slices in bars and beats.

Start by disabling all the Chaos Designers, setting the Live audio tracks receiving Stylus input to Record, and recording for the length of the longest loop. That gives you unprocessed loops of the same length for all tracks. Next turn on the Chaos Designers and record multiple passes

STEP-BY-STEP INSTRUCTIONS

1

Use Clip Envelopes to automate Stylus RMX's AllPlay button, placing the On event just before bar 2.

2

Assign each used Stylus track to a separate output and set levels and pans to their defaults.

3

With all Chaos Designers turned off, record one full pass of each loop. You can also drag in the slice-triggering MIDI files from Stylus RMX if desired.

4

With all Chaos Designers turned on, record multiple passes of each loop, thereby getting several different Chaos Designer patterns to work with. Duplicate the recorded clips and set each copy's loop to a different pass.

5

Audition each pass and delete the undesirable ones. Create an arrangement by dragging or recording clips and scenes to the Arrangement view.

6
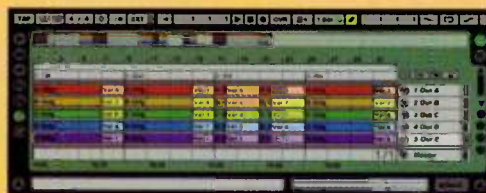
Here, Follow Actions were used to create fills at the end of each chorus. The third chorus was then overdubbed with different Follow Action settings to create an alternate take.

of each loop. For example, if the longest loop is 2 bars, you might record 32 bars, giving you 16 Chaos Designer alterations of each loop. Finally, use Live's Clip editor's Loop settings to single out each alteration, copying clips as necessary. You can now audition individual loops, toss the bad ones, and arrange the others to your liking in Live's Arrangement view. EM

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

Here's That Grainy Day
By Jim Aikin

Using Ableton Sampler for granular synthesis.

Granular synthesis is often used in academic and experimental music but isn't found in many commercial synthesizers. The sounds it produces are somewhat unsettling rather than smooth, but if you're looking for fresh sound effects, granular techniques can be a potent resource.

With granular synthesis, a source sample is sliced into tiny grains, usually between 1 and 50 ms long. The sound is then created by mixing the grains back together. A number of parameters can be modulated. For example, the length of the grain, its pitch, and its starting point (the spot in the original sample from which each grain is drawn) can be changed while the sound is playing.

Grain States

You may want to consider using Csound (www.csounds .com) to explore granular synthesis. The site is free, it's powerful, and it includes sophisticated granular synthesis tools. On the downside, Csound is not well suited for live performance and can be difficult to learn.

An easier way to try out granular synthesis is to use the Sampler plug-in available as an optional add-on for Ableton Live 6. Sampler doesn't provide full-on granular synthesis, but it can produce a surprising range of tones that use granular concepts. You can develop rich, sustained tones that evolve smoothly and gradually, burbling nonsense vocal textures, or thick crackling masses.

The Source

Any sample can be used as a source, but the best sounds are a few seconds in length, have a continuous high-amplitude level, and have lots of variation in tone color. I created the audio examples for this column by recording my own voice speaking a phrase, and then compressing it substantially to get a more uniform level. The audio examples and the Live project used to generate them are in Web Clips 1 and 2 on the EM Web site.

FIG. 1: The Sample tab in Sampler contains the sample start and loop parameters.

In Live 6, I created an Instrument Rack, loaded an instance of Sampler into the rack, and dragged my source sample into Sampler. Using an Instrument Rack makes it easier to create layers, each of which can have its own effects, and to assign modulation routings that can be saved with the rack preset.

Gran-diose

On the Sample tab in Sampler, set a short loop that starts somewhere in the middle of your sample, as shown in Fig. 1. Give it a healthy crossfade value, so as to smooth the edges of the grains, and set the Sustain Mode to either forward or back and forth. If you play Sampler from the keyboard at this point, you'll hear a static tone.

The parameters under the Sample tab can't be modulated directly, but they're available as destinations under the Modulation tab. There, two LFOs and an auxiliary envelope can modulate loop start and loop length. I like setting the LFOs to slow rates to achieve gradual changes.

To animate the tone of your short loop, activate LFO 2, choose Loop Start as destination A, and set the modulation amount for destination A to about 10 percent. This value is a percentage of the sample length, so changing the Sample End parameter in the Sample page will affect how much modulation you get.

The key to developing good granular sounds is experimentation: keep fiddling with parameters until you get a sound you like. By assigning a macro knob in the Instrument Rack to LFO rate, you can move from a smooth, short loop to longer stutters while a note is sounding.

Modulation to the Max

Sampler gives you three ways to modulate the loop start or length from an external MIDI source. You can assign your mod wheel to a macro knob in the Instrument Rack that contains Sampler, assign the mod wheel to a Sampler parameter directly, or switch to Sampler's MIDI tab and choose a parameter such as Loop Start or Loop Length as a destination. (Some of these MIDI routings override others.)

For more-complex sounds, try adding Live's Grain Delay effect to Sampler's output, then modulate a couple of its parameters from a macro knob. You'll be amazed at the richness of the sound textures you can create. **EM**

Jim Aikin writes regularly for EM, Mix, *and other music-technology magazines.*

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Mic Specs Demystified By Brian Smithers

What the numbers and tech jargon really mean.

A microphone is a straightforward device—sound goes in, voltage comes out. So shouldn't reading microphone specifications be a straightforward process? Mic specs are intended to tell you at a glance what to expect from a given microphone—specifically, how analogous the voltage is to the sound source.

Unfortunately, however, by the time you've researched a microphone and waded through the various acronyms, such as *EIN, mV/Pa,* and *dB SPL,* your head hurts too much to make a rational buying decision. The truth is that no amount of research into microphone specifications can tell you what a mic will really sound like. Specs may help you narrow the field, but they are no substitute for your ears.

Once you know the terminology, though, mic specs can be helpful. In this article, I'll explain the concepts and terms you need to know to get the most from spec charts.

Rejection

Aside from transducer type—dynamic, condenser, or ribbon (see "Square One: A Change Is Gonna Come" in the November 2006 issue of EM, available at www .emusician.com)—the first thing to look for in a microphone is its *polar pattern,* or directional response (see **Fig. 1**). A circular graph is used to illustrate the microphone's polar pattern, showing where in a 360-degree radius the mic is most sensitive. Some microphones offer multiple polar patterns, enhancing their usefulness.

A microphone with uniform sensitivity in every direction and plotted as a circle on the polar chart is said to be *omnidirectional.* Mics that are most sensitive to sounds arriving from the front, filling half (180 degrees) of the circular polar chart, are called *directional. Bidirectional* mics, on the other hand, are sensitive to sounds coming from the front and rear, but not the sides. The term *on-axis* is used to describe sounds that arrive at the sensitive parts of the pickup pattern. *Off-axis* refers to sounds hitting the areas of a mic's pickup pattern that are least sensitive.

Cardioid, so named because it resembles a heart shape, is the most common directional mic pattern, while *figure-8* is the classic bidirectional pattern. There are a number of variations, such as *wide cardioid* (which is a compromise between cardioid and omni), that fall in between the basic pattern types. Between cardioid and bidirectional, the pattern gets narrower toward the front of the mic, rejecting more sounds from the sides, while a bubble of sensitivity opens at 180 degrees to the rear. As you move from cardioid to *supercardioid* to *hypercardioid,* the pattern narrows and extends, until it eventually becomes bidirectional.

Freq'ing Out

Microphones respond to some frequencies better than others. The broken lines in Fig. 1 demonstrate how polar patterns vary depending on the frequency range. Manufacturers also create a frequency-response graph that indicates the mic's response to a tone swept from 20 Hz to 20 kHz. In **Fig. 2,** the tested microphone is about 3 dB more sensitive between 3 and 7 kHz than at 1 kHz—the frequency to which the curve is normalized.

Fig. 2 also shows two broken lines below 500 Hz, reflecting a property of directional mics known as the *proximity effect.* When a sound source is very close to a directional mic, the mic's low-frequency response is enhanced. This is the trade secret behind that rich, booming "announcer's voice" sound. The more directional the mic, the more pronounced the boost. The two broken lines reflect the mic's low-end response at a distance of 6 inches and 2 feet from the sound source, while the solid line indicates its response at 3 feet.

When you see a mic's frequency response listed as a numeric range, look for the degree of deviation— "±3 dB," for example. Without that qualifier, the given frequency range is not helpful. Look carefully, and you'll see mics occasionally listed with a "20 Hz to 20 kHz"
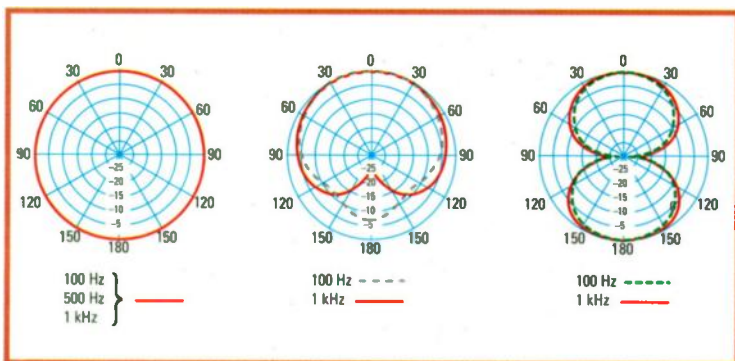
FIG. 1: This figure shows three common microphone polar patterns. An omnidirectional mic (left) responds uniformly regardless of angle of incidence, a cardioid mic (middle) rejects sound from behind, and a bidirectional mic (right) rejects sound from the sides.
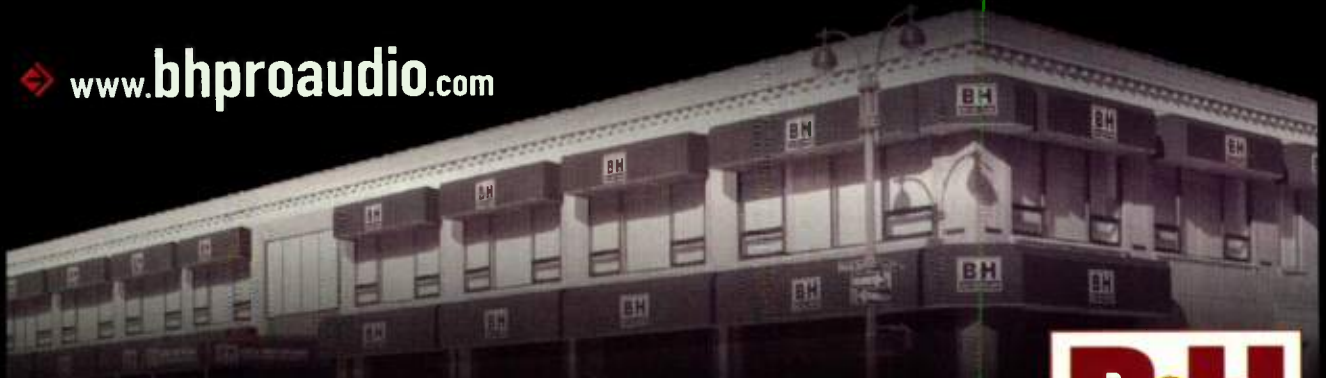
response that turns out to be 20 Hz to 15 kHz ±3 dB, with a significant rolloff toward 20 kHz.

Quiet, Please

If you will be working with very quiet sources, such as those in classical music or field recording, you will need

The first thing to look for in a microphone is its polar pattern.

a microphone that doesn't get in the way sonically. All mics have some degree of inherent noise, or *self-noise*. Self-noise is determined by comparing the inherent noise level with a hypothetical source sound whose level produces the same response from the mic (the *equivalent input noise*, or EIN). This is ordinarily measured in dB SPL, sometimes listed as "dB re 20 µPa," and is often A-weighted. A-weighting approximates the nonlinear frequency response of our ears but also results in a better-looking (lower) number. If the figure is not A-weighted, it will refer to a specification known as CCIR 468-3, and the number will be about 10 dB higher. A mic with an EIN of less than 15 dB SPL A-weighted is very quiet.

You should also know about a microphone's *sensitivity*, the voltage it produces in response to an acoustic stimulus. A sensitivity of 70 mV/Pa means the mic produces an output of 70 mV when presented with an input of 1 pascal (94 dB SPL). Sensitivity may also be expressed in dBV/Pa. For example, a sensitivity of –37 dBV/Pa indicates a microphone whose output is 37 dB lower than that of a hypothetical microphone

FIG. 2: This figure illustrates a microphone's typical frequency-response curve. The microphone enhances frequencies between 3 and 7 kHz. The boost below 500 Hz is due to the proximity effect.

that produces 1V in response to 1 pascal. Converting from one scale to the other requires a bit of math, but in general a range of 10 to 100 mV/Pa translates to an approximate range of –40 to –20 dBV/Pa.

Proper Load

To achieve optimum performance from most modern microphones, the microphone preamplifier's input impedance should be much higher than the output impedance of the microphone itself. Some mic manufacturers recommend a ratio of 5:1, while others recommend 10:1. This type of high-to-low ratio is called *impedance bridging*, and it results in maximum transfer of voltage from mic to preamp in order to minimize signal degradation.

A mic's spec sheet will list the mic's output impedance. If it doesn't also list a recommended *load impedance* for the preamp, you can simply multiply the output impedance by 5 or 10, depending on which of the above ratios you want to work with.

Transformation

With a *transformer*, a signal passes through a coil of wire wrapped around a magnet and induces signal in an adjacent coil/magnet assembly, allowing signal to pass without physical contact between the two assemblies. Changing the number of turns of wire in the output coil, or *winding*, changes its impedance relative to the input winding, thereby changing the output signal level.

In general, a mic with an output transformer allows you to use a longer cable without fear of unacceptable signal loss, and it also increases resistance to noise. Intense low-frequency sounds, however, can overload a transformer, causing distortion. Some engineers will therefore prefer a transformerless design if they can get by with relatively short cable runs, as in a studio (as opposed to a live setting, in which long cable runs are common).
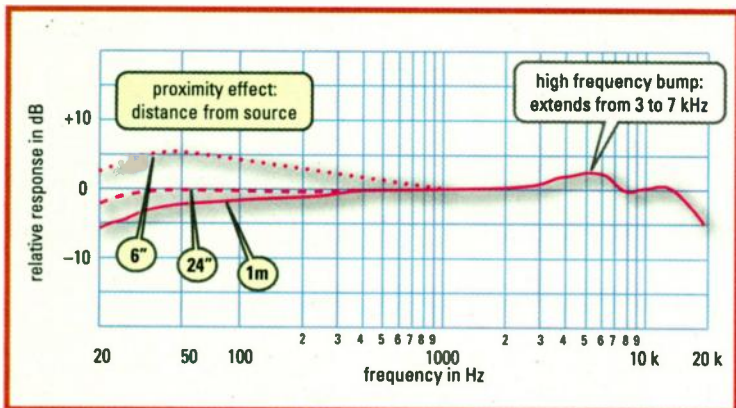
Caveat Emptor

There are some international standards that attempt to codify the way in which microphones are tested and their specifications reported, but few manufacturers follow them faithfully. A wise mic shopper therefore needs to come armed with a broad and flexible vocabulary, a probing mind, and a healthy dose of caution, if not outright skepticism. So armed, one can glean much useful information from Web sites, spec sheets, and glossy sales literature.

Of course, all of that research only serves to narrow the field down to which mics are likely to please your ears. After that, you must judge which mic sounds best on which source. **EM**

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Q&A: Jack Rudy By Mike Levine

Why music supervisors are looking for your songs.

In the music-for-picture world, composers get most of the glory. But there's another group of professionals who have a big impact on which music gets into a film or TV show: music supervisors. While film composers write custom instrumental music to set moods and fit with the action in various scenes, music supervisors choose their music from songs and compositions that have already been written. Music supervisors are therefore a conduit for songwriters and independent composers who are trying to get their music on the screen.

For a better understanding of what music supervisors do, I recently spoke with Jack Rudy (see Fig. 1), who has supervised the music for feature film, sitcom, *Comedy Central*, and more. Rudy brings plenty of musical skill and experience to his job, having worked as a professional harmonica player for years, playing with artists such as Dave Alvin, the Blasters, and John Lee Hooker.

FIG. 1: Jack Rudy frequently places songs from independent bands and artists on TV shows and in films.

COURTESY JACK RUDY

What is the role of the music supervisor in a production?
I complement the work of the composer by providing cues that come from outside of the composer's brain—so band cues, rock music songs, preexisting songs. That's why my specialty as a music supervisor is to stay on top of the music of all current bands: bands that have albums, bands that don't have albums, bands that are playing in clubs.

In the rock world, mostly?
Not necessarily just the rock world—the whole world of music. Anybody who is playing music live for people is at the top of my list. And not just because those songs work well in movies. It's also because those musicians and I have a similar self-interest in a project. What will work for me and my film will also work for those artists in their careers. Obtaining a film placement or a television placement is a very valuable and positive step in a live musician's career. It's kind of like what getting on the radio meant in 1955.

So with a film or TV show, does the audience focus on your music less than it would if it were on the radio?
Yes, but a lot of shows are music driven. A huge percentage of the people who are of record-buying age buy their music based on what they've seen on television. The people who are selecting music for television shows are trusted; their taste is trusted by the watchers of those shows. So in a weird twist of fate, I am able to function in much the same way that, say, Wolfman Jack did back in 1970—by finding songs and helping people discover new music.

What types of scenes require songs rather than music written by the film composer?
There are certain obvious places where we put songs from bands. Source music is one of them—every time a character in a film or show turns on a radio, walks into a bar, sits in a waiting room, or turns on a jukebox. Then there are the cues in which the composer and the music supervisor must work together to decide whether a composed piece or a song from a band would work best. Things such as a musical montage segment sometimes work great with a band and sometimes it works great with a composer's own work.

Can you generalize about what particular attributes of a song make it useful for a film or TV show?

What you're looking for first in a particular scene is credibility (as opposed to musical innovation), and everything follows after.

By "credibility" do you mean that the song has authenticity from a stylistic standpoint?

You're getting there. But I actually mean it more literally than metaphorically. What I'm saying is that you as a listener need to believe it. Forget that it's music for a second. If it were a person talking to you and telling you something, would you believe them, or would you think that they were not credible, not believable?

Based on the lyrics?

Just ask yourself if the music captures an emotional component of a scene that needs to be there. Because the component is emotional, it can't ring false. An example of a song that lacks credibility would be one in which the composer tries too hard to be literal with the lyrics and what's going on in a particular scene and ends up with something that's too matchy-matchy.

What happens when a noncredible song is used?

If there's one insincere note in the song, the actors will look like bad actors. One song in the background that's baloney can wipe out the hard work of an actor, a lighting designer, or a set director.

Do you ever go to music libraries for material?

Yes, I absolutely do. Mostly for instrumental tracks, and mostly if a particular scene requires a style of music that my composer isn't familiar with.

For a song to be considered for a movie, I assume that it has to have a certain level of quality, productionwise.

Yeah, absolutely. These days, everybody has to have high-quality production. Nobody is walking around with a 4-track cassette tape anymore.

There's a big variation in quality on independently released CDs.

That's true. But practically speaking, if the mix is okay, the music is fine by the time it gets compressed and pushed into the background of a TV show. In fact, we've run with MP3s on TV shows and haven't had a problem. What makes the interests of the band musicians align with the music supervisor's is that the guy who's playing in a band has rehearsed and performed these songs in front of people repeatedly. And he has had the benefit of what—to an ad agency, for example—would be a very expensive focus group. He's playing the songs over and over again, and that's why he's getting a crowd to come and see him. That's the reason a well-placed band track will leap off the screen, compared to a custom-composed song for that particular scene. Now that is my own personal taste; that's subjective. And a band benefits from having its song on a TV show or in a film in ways that a composer sitting at home doesn't. Even though they both get performance-rights royalties and some sort of a license fee.

What's a typical licensing fee that an artist would get for use of his song in a movie or TV show?

Some TV shows pay less than $500 for an up-front license fee.

How can artists maximize their exposure to music supervisors?

If your band is successful locally and you are playing a great gig that gathers a large crowd, and you look out into the audience and ask yourself if there might be someone in that audience who might help your career, it's more likely that a music supervisor is out there scouting than a record company executive.

Is it only in L.A. and New York that this happens, or can it happen in, say, Lubbock, Texas?

It could happen to somebody in Lubbock. I'm not going to fly to Lubbock to see a country band perform, but I am going to pick up the phone and I'm going to call someone who knows country music and ask them to tell me who is out there right now playing, and who is good.

Do you require people to get permission from you to submit something?

That's an interesting question. Oftentimes with record companies you can't even find their address. There's all that crazy, weird secrecy. That's not true at all for me—not one bit. If I solicit for a song through a network, I give an address. Whoever answers is submitting based on my solicitation.

So music supervisors are generally more accessible than record company A&R people?

We're on the ground. We're where the rubber hits the road, where music meets film. We are actually interested in finding new music, and we aren't motivated in the way that so many record companies are: by profit and sale of the album. We're seeking out and searching for songs that fit scenes. So when you submit your work to us, we'll truly listen to it. We don't have prejudice between a known band and an unknown band. **EM**

Mike Levine is an EM senior editor.

World Radio History

REVIEWS

Ableton once again delivers a stellar upgrade.

By Len Sasso

L
ive 6 is the latest in a steady stream of upgrades to Ableton's flagship sequencer, and it doesn't disappoint. Live 6 adds many new convenience features as well as some biggies such as video support, the new Sampler instrument, and plug-in Racks capable of housing

parallel chains of instruments and effects. You can purchase Live as a download or in a box, which includes a printed manual and a large collection of presets for Simpler, Live's free sampler. The Simpler presets make up the Essential Instruments Collection (EIC) from SoniVox (formerly Sonic Implants).

Live 6 runs under Mac OS X and Windows. It can function as a ReWire mixer or client application, and it hosts VST and AU instrument and effects plug-ins. It includes a collection of Ableton plug-ins optimized for Live. For this review, I installed Live 6.0.1 on a 2 GHz dual-G5 Power Mac and an 800 MHz G4 PowerBook, both running Mac OS X 10.4.8. I also installed it on a 3.2 GHz Pentium 4 laptop running Windows XP. Performance was outstanding in all cases.

Because Live has been well covered in previous EM reviews, I'll stick to the new features. You can read reviews of earlier versions online in the June 2002,

FIG. 1: Assigning an audio clip in Arrangement view as the tempo master automatically generates tempo changes matching the clip's Warp markers.

GUIDE TO EM METERS

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

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

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June 2003, April 2004, December 2004, and January 2006 issues.

A Steady Clip

Live is primarily about triggering, arranging, and synchronizing audio and MIDI clips. Live 6 introduces three important improvements in audio clip management: you can slave

FIG. 2: Frozen tracks render all clips as audio, but you can still move and resize clips and add mixer automation.

Live's tempo to a time-warped audio clip, simultaneously time-warp multiple audio clips, and crop audio clips directly in Live's Clip view.

It was a nice step forward when Ableton introduced tempo automation in Live 2, but conforming Live's tempo to a long, freely played audio clip was still a tedious job. First you had to time-warp the whole clip, and then you had to go back and manually insert tempo-automation events to match each of the clip's Warp markers. Now, you can designate any audio clip in Arrangement view as the tempo master, and Live will automatically create tempo automation to match.

You can also use the tempo-master feature for a kind of groove quantizing. For example, designate a short drum loop as the tempo master, and then set Warp markers to align all hits to grid lines. The drum loop will sound as it originally did, but other quantized material—MIDI files, for instance—will conform to the drum loop's groove (see **Fig. 1** and **Web Clip 1**).

Live 6 has a few more tricks to simplify working with Warp markers. If you select multiple audio clips of the same length, you can add and adjust Warp markers simultaneously. That's convenient when you've recorded multiple tracks and you want to adjust the timing of all of them. You can also copy and paste Warp markers between clips, and the clips don't need to match in length or in the location of those markers.

On with the Show

Although Live 6 does not offer sophisticated video editing and imports video in only Apple QuickTime format (.mov), it makes it easy to quickly create soundtracks synced to video. You can drag video clips to Arrangement view audio tracks or to clip slots in Session view, but only their audio is imported in Session view.

Video clips on Arrangement

World Radio History

RE

view tracks work the same way that audio clips do—in particular, with regard to creating and managing Warp markers. The simplest way is to designate the video clip as the tempo master and align Warp markers with the desired video hit points. Imported and recorded audio tracks can then be time-warped to sync with the video hit points.

You can drag multiple video clips to Arrangement view tracks and even place them on different tracks, but only one video clip can play at a time, and that's always the one on the lowest track. You can also slice video clips and rearrange the slices, but you can't recombine them to create a single video or add audio directly to a video. You need separate video-editing software for those functions.

Frozen Solid

With new multiprocessor and multicore support, Live 6 delivers a significant improvement in performance. For example, a project on my 2 GHz dual-G5 Power Mac that peaked Live 5's CPU meter at 89 percent and broke up during playback played without problems in Live 6, peaking the CPU meter at 60 percent. In spite of that improvement, a few power-hungry virtual-instrument tracks can still top out Live 6. When that happens, it's time for the deep freeze.

Most sequencers have a freeze option that temporarily renders an audio or a virtual-instrument track, along with any effects processing. The sequencer then plays the rendered audio in place of the real track. Live's freeze function (called Deep Freeze) does that and more (see **Fig. 2**).

When you freeze a Live track, all clips in the Session and Arrangement views are rendered. You can move clips around on the track and between views, split them and shorten them from either end, and, if clip looping is turned on, extend the loop from either end. When a clip is frozen, its Clip Envelopes are also frozen. But mixer-automation envelopes such as volume and panning are not frozen, and you can edit them as on any audio track.

You can, of course, unfreeze a track when changes are needed. Conveniently, if you unfreeze a track and then freeze it again without making changes, the old freeze file is used, and you don't have to wait for another rendering. If you decide no further changes are necessary, you can flatten a frozen track.

That converts the frozen track to an audio track playing the rendered audio and automatically removes all plug-ins and rendered automation envelopes. As an alternative to using Live's Flatten command, you can simply drag or copy a frozen clip to another audio track to get a flattened version of that clip. You can thereby flatten some frozen clips while leaving others for subsequent unfreezing.

What and Where

Live's file browser sports several enhancements. A new Hot Swap mode facilitates swapping samples used on Live tracks or in Ableton sample-based instruments (Impulse, Simpler, and Sampler). Clicking on a sample's Swap button in the File Manager or an instrument's Sample window reveals the file's location in the browser's Hot Swap tab. You can use that as a starting point for auditioning or permanently swapping the target sample with the sample in the browser.

Each browser tab now has columns showing the type, size, date modified, and file path for displayed items. You can hide, reorder, and resize the columns. Bookmarks have been added to the browser, making it much easier to get around. You can add or remove the currently displayed disk location to the bookmark list, but oddly you can't change the order of the list or alphabetize it. Still, the permanent bookmarks (Library, Current Project, Desktop, All Volumes, and Home), together with a few of your own, come in very handy.

Live's project management has been improved in several ways. Live sets are now automatically saved in a Project folder where all ancillary files used in the set are collected using Live's new Collect All And Save command (an improved version of Consolidate Project). If you save a new set in an existing project folder, it becomes part of that project; otherwise, a new project folder is created with the same name as the set.

You use the new File Manager, which shares the Lessons window at the right of the user interface, to manage the files in a set, a project, or the entire Live Library. With it, you can quickly locate all relevant files, display them in the browser, and collect them into the project (see **Fig. 3**). Furthermore, you can collect projects into Live Packs for archiving or sharing. Live Packs are monoliths containing all files used in a project and are created with lossless compression to minimize their size.

Grabbing Hold

The Browser window is also used for Live's upgraded MIDI and Keyboard mapping functions. When you invoke either function, the browser displays all current mapping assignments, allowing you to quickly see, modify, and delete them.

You can now assign the same key or MIDI command to multiple functions where needed. For example, a single MIDI controller can control several volume faders, and

FIG. 3: Live's File Manager window details all file usage in a set or project and provides utilities for finding and replacing files.

REVIEW

a single key can solo multiple tracks. For knobs and sliders, each assignment has its own range, which can be positive or negative. You can use that to assign the same MIDI controller to increase one channel's volume while decreasing another's.

A global preference called Take Over mode determines how MIDI continuous controllers are handled. The target control can jump to the incoming value (None), stay put until a matching MIDI value is received (Pickup), or have its value scaled by the incoming MIDI value (Value Scaling). The last option is so clever that it's hard to imagine a use for the other two—on one hand, there are no jumps, and on the other, the targeted control reacts immediately.

Live's new Instant Mappings support a variety of popular control surfaces, and you can activate as many as six Instant Mappings at a time. When Instant Mapping for a control surface is invoked in Live's Preferences, the control surface's knobs, sliders, and buttons are automatically assigned to Live's mixer and plug-in device parameters, and that includes third-party plug-ins. Manual assignments (made using Live's MIDI Map mode) override Instant Mapping assignments. Although it's not officially supported, you can create your own Instant Mappings by following the instructions in the User Remote Scripts folder on your hard drive. I had no problem creating an Instant Mapping configuration for my discontinued Native Instruments 4Control.

Rack 'Em Up

Ableton introduced device groups in Live 5. They allowed you to collect linear chains of plug-ins in any combination of formats (VST, AU, and Live) into a single device, and then save the configurations of all the devices as a single preset. Device groups could start with a virtual instrument plug-in or be pure effects chains, and all automation and MIDI control was on a per-device basis.

Live 6's new instrument and effects Racks expand on device groups in two important ways: Racks can hold parallel chains of instruments and effects plug-ins, and each Rack has eight macro controls that can be mapped to multiple device parameters and scaled independently (see **Fig. 4**). Device
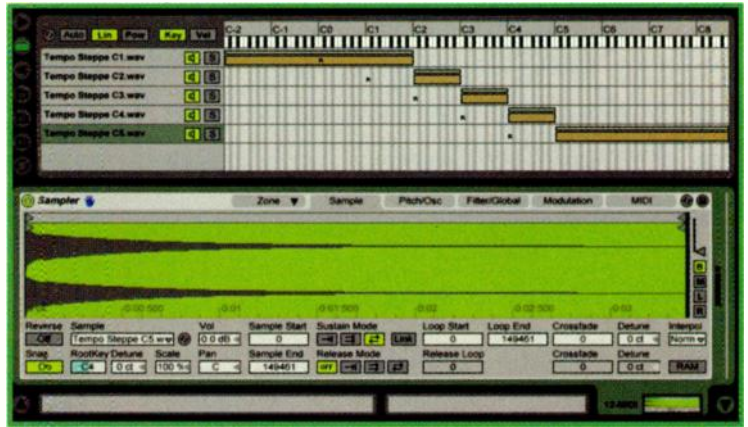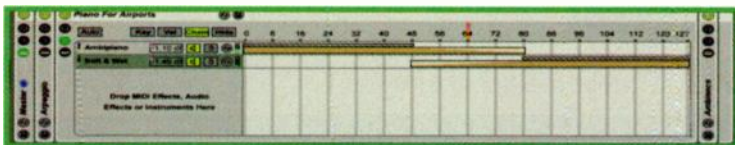
FIG. 5: The new Sampler instrument offers full multisample editing and a host of filtering and modulation capabilities.

groups saved from Live 5 are automatically converted to Racks when opened in Live 6.

You can use parallel chains within a Rack to layer and split virtual instruments and to layer effects. There are three ways to control the mix of virtual instrument chains: by key zone, by Velocity zone, and manually. Effects chains can be mixed only manually, which is done by setting the Chain slider with the mouse or through MIDI. You set the active zone and crossfade region of each chain, and then position the Chain slider to determine the mix of the chains.

Although the devices in a single chain work in series and separate chains work in parallel, you can create more-complex signal flows by nesting Racks and placing Racks in series. For example, you might place a MIDI arpeggiator effect before an instrument Rack containing parallel piano and strings chains and follow that with a Rack containing chorus and reverb. Incoming MIDI will be processed by the arpeggiator, and then passed to the piano and strings Rack, whose mixed output will be processed by the chorus and reverb Rack (see **Web Clip 2**). You can then nest those three Racks in a new Rack to save the whole setup as a single Rack.

By Any Means

Racks are only as good as the instruments and effects you stuff in them, and Ableton has added to and improved its complement of built-in devices. The new Note Length MIDI effect has two modes: Note On and Note Off. In Note On mode, incoming notes are all given a fixed duration that you set in milliseconds or tempo-synced note divisions. In Note Off mode, notes are triggered when MIDI Note Off messages are received.

The Dynamic Tube effect is a tube saturator and envelope follower combined. EQ Eight upgrades the older EQ Four to eight channels and adds left-right and mid-side modes. In those modes you get separate control of each channel. The Saturator and Utility effects have also been upgraded.

On the instrument side, Operator has been beefed

FIG. 4: Instrument and effects Racks manage parallel chains of instruments and effects.

up with a new set of 24 dB filters and a couple of new FM algorithms. Both add significantly to its range, but the big news is the new Sampler instrument. Operator ($149) and Sampler ($199) are paid add-ons to Live.

Sample This

Sampler is a full-featured multi-sampling companion to Live's free sampler, Simpler. Sampler comes with a small library of multisampled instruments, which is greatly expanded by the optional EIC ($119). The EIC instruments are built for Simpler, using a hidden multisample-playback capability, but they are easily converted to Sampler format, making them much more tweakable. The EIC is heavily weighted toward acoustic instruments and is a solid collection, but the electronica crowd won't find a lot here.

In addition to creating your own multisampled instruments, you can import instruments in a variety of common sampler formats, including Tascam GigaStudio, Native Instruments Kontakt 2 (nonencrypted), Creative/Emu SoundFont, and Apple EXS24/GarageBand. Importing multisampled instruments in other formats is a notoriously flaky enterprise, and the less complex the format being imported, the better. I successfully imported SoundFont, Kontakt 2, and EXS24 instruments, though I had to guide Sampler to the EXS24 samples. The results were mixed; for example, all the SoundFonts I imported had their sustain loops disabled.

Sampler has a multitab interface along with a Zone editor that is expandable to the full size of the clip area if needed (see **Fig. 5**). Key and Velocity zones are sized and crossfaded using the same expandable-bars technique used for zoning instruments in Racks. The remaining tabs—Sample, Pitch/Osc, Filter/Global, Modulation, and MIDI—toggle displays for setting the corresponding playback properties. All settings, except those on the Sample tab, apply to all samples; you can't work with sample groups the way you can with most samplers. Ableton's alternative, which is quite effective once you get used to it, is to use multiple instances of Sampler collected in a Rack to create sample groups. Live will do that for you when importing foreign formats. The advantage of this approach, typical of Ableton, is that you don't have to fuss with sample groups unless you need them.

The remaining four tabs are devoted to filtering and modulation. An oscillator and envelope generator provide FM, AM, and pitch-bend effects. A morphable multimode filter and a waveshaper are there for timbral contour. An ADSR envelope generator and three LFOs can modulate two sources each. A modulation matrix routes six different MIDI messages to two destinations each.

Final Mix

Live's mixer is improved in several ways. You can vertically resize the Session view fader section to reveal digital peak and level readouts along with dB-labeled tick marks. Inputs from other audio tracks can now be pre- or post-FX as well as postmixer (after the fader and pan controls). Finally, you can choose between seven crossfade curves for the Crossfader.

One of the best things about Live 6 is that it looks and works almost exactly the way Live 5 (and 4, 3, 2, and 1) does. A cumulative what's-new list of Live's six generations reveals a huge change in functionality, but Ableton has been careful to preserve the basic look and feel. The company has also avoided adding features just to make the list look good. There's a lot to learn in Live 6, but it's mostly easy going. This upgrade **EMWEB CLIPS** is a must-have.

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

FIG. 1: The Moog Music Little Phatty is an old-school analog monosynth with 21st-century advantages, including 100 rewritable presets and comprehensive MIDI control. Although the instrument is monophonic, when it's used as a MIDI controller, the keyboard transmits polyphonic data to any instrument that can receive it.

MOOG MUSIC Little Phatty

The Minimoog spawns yet another worthy descendant.

By Geary Yelton

Although history credits Bob Moog with inventing the keyboard synthesizer, his most successful designs were the results of collaboration. Moog shared responsibility for engineering the Minimoog, for example, with Bill Hemsath, Jim Scott, and Chad Hunt. In early 2005, shortly before learning of his fatal brain tumor, Moog and his engineering staff in Asheville, North Carolina, began designing

Moog Music has cut corners by designing an ingenious user interface.

an analog synth that would be more affordable than the company's flagship Voyager (reviewed in the October 2003 issue of EM, available online at www.emusician.com). The team included longtime Moog employee Steve Dunnington and relative newcomer Cyril Lance, among others, who have effectively taken up Moog's engineering duties since his diagnosis and subsequent passing.

Much to the chagrin of many observers (including me), the new synth was eventually named the Little Phatty. If you knew Bob Moog, however, you'd understand that the moniker is a fitting tribute to a great man who enjoyed a skewed sense of humor.

The Little Phatty (LP) is a monophonic analog synthesizer with a 37-note, Velocity-sensitive keyboard. Its synthesis engine is not as comprehensive in scope as the Voyager's, but its fat sound is unmistakably analog and indisputably Moog. The first and current LP is the Bob Moog Tribute Edition, a limited production run of 1,200 instruments. An even more economical Stage Edition is said to be in the works, though as of this writing, Moog Music has made no official announcement.

Outside Possibilities

With solid-wood end panels and retro-style knobs, the LP looks simultaneously modern and traditional (see Fig. 1). Its control panel is permanently fixed at an angle

for easy access, and its brushed-aluminum rear panel has a unique S shape and sports an enlargement of Bob Moog's signature. On the LP's left side panel are all its connections to the outside world, including mono audio and MIDI I/O, control voltage (CV) and gate inputs, and an IEC power receptacle (see **Fig. 2**).

When you power up the LP, it greets you with a 5-second light show as it cycles through all its blue and magenta illuminated buttons and LEDs. The translucent pitch-bend and mod wheels have a continuous blue glow, and luminous LEDs and buttons indicate which controls are active. In each of four control-panel sections—Modulation, Oscillators, Filter, and Envelope Generators—is a single rotary knob surrounded by a ring of 15 blue LEDs that indicate the selected parameter's current value. Whereas most recent hardware synthesizers supply a knob for each function, Moog Music has cut corners by designing an ingenious user interface that replaces individual parameter knobs with multipurpose knobs and buttons that select their functions.

On the far left is an unlabeled user-interface section containing a 2 × 16–character alphanumeric LCD, unadorned knobs for fine-tuning and changing displayed values, and buttons that let you change modes, enable glide, and perform utility functions such as saving presets and moving the LCD cursor. The Octave Up and Down buttons in this section have a thoughtful feature I've never seen before: they glow blue when you press either of them to transpose by one octave, and they glow magenta when you press again to shift by two octaves. Thus, you can always see your keyboard's transposition at a glance.

In the Output section on the control panel's far right, a simple Volume knob controls the levels of the ¼-inch headphone output mounted just below and the main output on the side panel. An Output On/Off button disables the main out-

FIG. 2: The Little Phatty's left side panel contains an unbalanced audio output, an unbalanced audio input, a gate input, and control voltage inputs for pitch, volume, and filter cutoff—all on ¼-inch jacks—as well as MIDI In and Out ports and an AC power connection.

put but doesn't affect the headphones, making it handy for onstage tuning and cueing up presets.

Because the LP's electronics are analog, the instrument needs to warm up so that its voltage-controlled circuits can stabilize. After 15 minutes with the power on, temperature-regulated chips ensure rock-solid oscillator tuning. Because the LP doesn't furnish an internal tuning reference or automatic tuning, however, you'll want to tune it by ear to an external pitch source. Considering that guitar tuners are so inexpensive, this is a minor, though perplexing, annoyance. On the plus side, the Fine Tune knob makes a dandy alternate pitch-bender and has a range of a minor third up or down.

On my review unit, the keyboard's feel varied subtly from one white note to another. One white key had a lighter touch than the others, and several of them banged against the key bed when I played more than moderately hard. Fortunately, neither problem affected the LP's output, nor were any mechanical noises audible with the audio level turned up. The black keys were much more even and consistent, but they also felt much stiffer. I'm not the first reviewer to receive this particular LP, and I suspect that previous reviewers have played the white keys enough to loosen up the action.

It's a Lean Machine

Two buttons in the user-interface section access Preset mode, which lets you load presets and manipulate their parameters, and Master mode, which lets you alter global parameters and utility functions. To change presets, you can scroll through them sequentially by turning or pressing the Value knob in Preset mode or switch to them directly by sending MIDI Program Changes. You can also scroll through them ten at a time by pressing the Value knob as you turn it.

Although there's no way to switch directly to a nonsequential preset from the front panel, the LP offers a clever work-around using four Performance Sets, each containing any eight presets you specify in a sequential

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PRODUCT SUMMARY

MOOG MUSIC Little Phatty

analog synthesizer
$1,475

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

RATING PRODUCTS FROM 1 TO 5

PROS: Fat analog sound. Intuitive user interface. Extensive MIDI implementation. Excellent factory presets. No wall wart.

CONS: No tuning reference. Inconsistent keyboard. Global filter cutoff slope and release defeat. No noise generator.

MANUFACTURER
Moog Music
www.moogmusic.com

order. After you've set up a Performance Set, you can quickly scroll between its eight presets by pressing the Value knob when the LP is in Master mode. Turning the Value knob in Master mode changes LCD pages, which allow you to set up Performance Sets and specify MIDI mode and channel, keyboard priority, pitch-bend range, and other systemwide parameters.

The LP has two identical voltage-controlled oscillators labeled Osc 1 and Osc 2. Their most outstanding aspect is that the waveforms are continuously variable—you can smoothly transition from triangle to sawtooth to square to narrow pulse by turning a knob or applying

modulation. The Oscillators section has buttons to sync oscillator 2 to oscillator 1 and to independently adjust their octave transposition. By pressing the corresponding buttons, you can use the Oscillators knob to change the glide rate, to tune the second oscillator relative to the first, and to independently adjust each oscillator's waveform and level.

The LP's voltage-controlled filter is lowpass only, which is probably all you'll need 99 percent of the time. It can resonate to the point of self-oscillation, and you can use MIDI to modulate the resonance amount. The filter's Overload parameter lets you dial in a very warm and pleasant distortion without going over the top. You can choose from four filter cutoff slopes—from one to four poles—but surprisingly, slope is a global parameter. The selected filter slope affects all presets; you can't specify the slope for individual presets.

Two identical ADSR generators are hardwired to the amplifier and filter. There's no switch for defeating the release stage as there is on a Minimoog, but you can disable release in the global parameters—an odd compromise, because like filter slope, it affects all the presets. The Envelope Generators section is the only place where I missed having individual knobs, because it takes so long to view or change attack, decay, sustain, and release settings one at a time.

In the Modulation section, you can route the LFO, filter envelope, or oscillator 2 to modulate either the filter cutoff, oscillator 2's pitch, or the pitch or waveform of both oscillators (routing oscillator 2 to itself has no effect). You can choose from four LFO waveforms and use the section's knob to control the LFO's modulation depth and rate.

Dedicated ¼-inch inputs let you route CV sources to directly control pitch, filter cutoff, and volume level. You can also insert a footswitch to serve as a gate that triggers both envelopes—a useful function if you're processing external sounds. The CV sources might be expression pedals or programmable devices such as analog sequencers. You'll achieve a much greater breadth of external control, though, using MIDI Control Change (CC) messages. Virtually every parameter you can change from the front panel responds to MIDI. However, I'm confounded that the LP doesn't respond to MIDI Volume (CC 7) changes, and I would be amazed
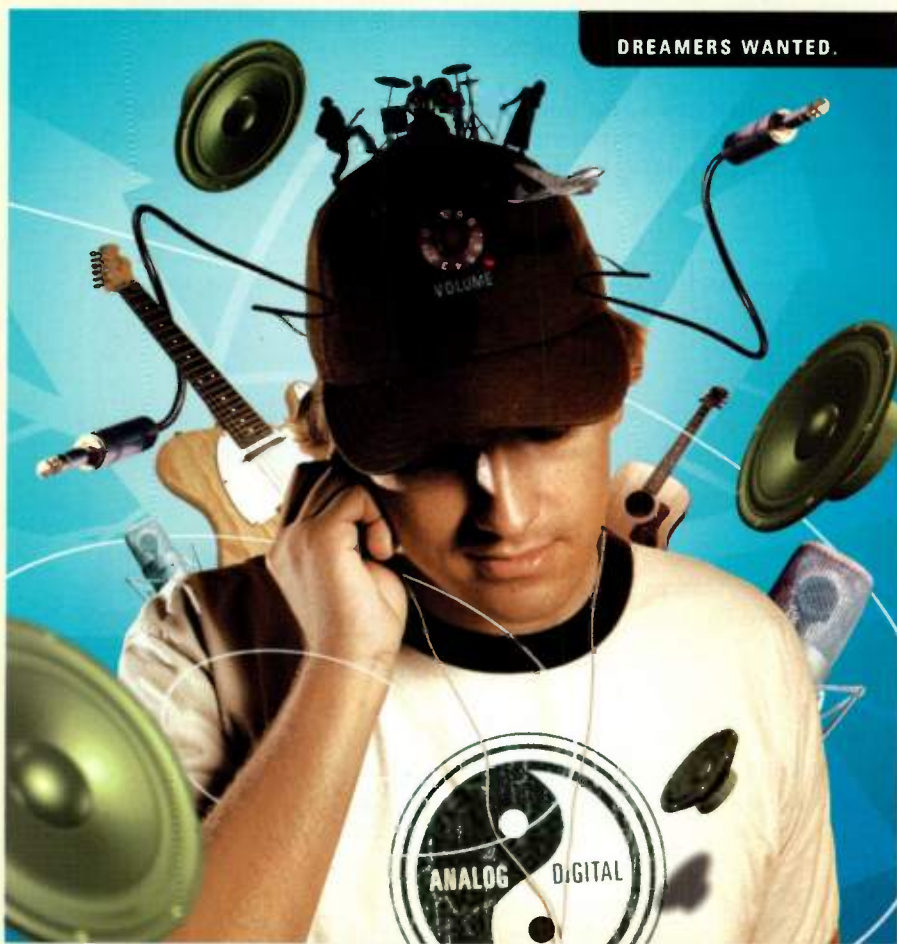
if Moog Music doesn't address this shortcoming in a future software update.

Wired for Sound

Have I mentioned that this thing sounds fat? If you close your eyes, the LP's tonal character is virtually indistinguishable from the Voyager's or the Minimoog's, largely because its lowpass filter is based on the Minimoog's filter. I don't think anyone could mistake the LP's sound for a digital emulation.

The LP stores 100 presets and comes fully stocked with a fine variety of expertly programmed patches. Deeply resonant growls, metallic bells, staccato blips, funky clavs, clangorous tones, arcade sound effects, and Simmons-style drums are all within easy reach, but the emphasis is on classic leads and basses, just as you'd expect (see **Web Clip 1**). I didn't always agree with the modulation assignments, but whenever I wanted the mod wheel to control vibrato rather than filter cutoff, for example, I could reassign it in seconds.

Fat and Happy

The Little Phatty is a true analog monosynth in every sense: it's a monophonic, monotimbral, mon-aural subtractive synthesizer with analog circuitry throughout. Ideal for beginners and pros alike, the LP puts every parameter right at your fingertips with a minimum of front-panel clutter, and the well-written manual makes it easy to learn your way around. Any synthesist who relies entirely on software or all-digital instruments is missing an essential color from his or her timbral palette, and the LP fills that gap admirably.

Although it is obvious that Moog Music economized wherever possible in the LP's design and construction, it is nonetheless a completely professional musical instrument that sounds fantastic and feels built to last. Because of shortcuts and work-arounds such as 1-knob envelope generators, global filter slope, and the lack of a noise generator or effects processor, the LP doesn't offer quite as much flexibility or immediacy as instruments like the Voyager. Nonetheless, the vast majority of users won't notice such minute details, because the LP delivers just what it promises: dynamite sound at an attractive price.

EM associate editor Geary Yelton has been writing about synthesizers for almost as long as Jan Hammer has been playing a Minimoog.

FIG. 1: The omnidirectional mics on the Edirol R-09 are conveniently positioned on either side of the top of the unit.

EDIROL R-09

A pocket-size WAV and MP3 recorder that's ready for action.

By Rudy Trubitt with Bruce Koball

dirol, a subsidiary of Roland, has several field recorders on the market, including the R-1, a stereo deck that's a bit smaller than a videocassette, and the 4-track R-4, which is perhaps the least expensive portable multitrack available. The company's latest entry in the world of portable recorders is the R-09.

A bit chunkier than the original Apple iPod, the R-09 records 2-channel audio at resolutions from 64 to 320 Kbps in MP3 format to linear-PCM WAV files at 44.1 or 48 kHz at 16 or 24 bits (see Fig. 1). The recorder has a pair of built-in electret condenser mics, as well as 3.5 mm jacks on the top for an external mic and a line input. (Plug-in power is provided for mics that can run from this consumer counterpart to phantom power.) The R-09 also has a jack that triples as a headphone output, a line-level output, and an optical digital output.

The R-09, which runs off of a pair of standard AA batteries, is lightweight but doesn't feel cheaply built. The small, high-contrast OLED screen is easily visible indoors but virtually unreadable in direct sunlight. The recorder uses SD memory cards up to 2 MB in size for storage (a 64 MB card is included), and with firmware version 1.10 (announced as this review was going to press), SD High Capacity cards are supported for greater storage amounts and increased data transfer speeds.

The transport controls are grouped around a single rectangular button, which rocks up, down, left, and right to control play/pause, stop, rewind, and fast-forward, respectively. A separate Record button, easily located by touch, sits in the center of the transport area. There are three additional buttons on the front panel: Finder/Menu, Reverb, and Repeat.

Other controls are positioned on the sides of the device: a recessed power button and a pair of up/down input-volume buttons are on one side; a hold switch, up/down monitor

The internal mics were convenient and up to the task at hand.

level buttons, and the headphone/line/optical output are on the other (see Fig. 2). On the back of the R-09 are low-profile switches for mono/stereo (for the external mic), low cut, mic gain (low/high), and automatic gain control (AGC). Although the AGC's operation is certainly audible, I appreciate the inclusion of this feature. It's handy for situations where you don't want to constantly watch the recording levels. Note that Mono mode routes the left input to both channels of a stereo file but does not record a smaller mono file.

Up and at 'Em

The R-09 is extremely quick and easy to deploy; power-up is as fast as we've seen in a portable recorder—about six seconds. Pressing Record puts you into Record/Pause mode, and a red LED hidden behind the Record button blinks brightly. Press Play/Pause or press Record a second time, and the red LED glows steadily as recording begins.

The device has responsive metering, which ranges from −45 to 0 dBfs. A large part of the display is given over to an elapsed-recording-time counter. That, as well as the record LED, makes it easy to tell when you're rolling. A single peak LED lets you know if your levels are clipping the input, in which case it's time to switch the back-panel Mic Gain switch to Low. (This switch attenuates the input about 25 dB.) The bar-graph record-level
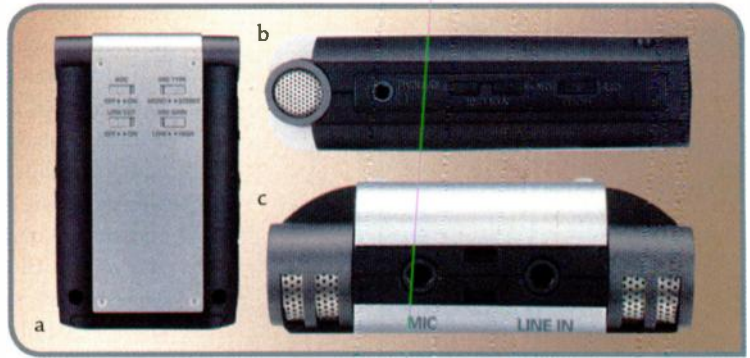
FIGS. 2a–c: Additional controls are located on the back (a) and sides (b) of the unit, while I/O is placed on the top (c).

meters also have a peak indicator, which clears itself after a 1-second hold. The digital up/down input-level buttons provide 29 1 dB steps of record-level adjustment in addition to the 25 dB gain switch.

The built-in mics have a pleasant tonality. Although the capsules are omnidirectional, their mounting yields an acceptable stereo image. Self-noise is just audible with medium-loud sources such as an acoustic guitar, but it becomes objectionable when you crank the input level to record quiet ambiences. However, for many common applications (songwriting, rehearsing, or performance

recordings), the internal mics will provide suitable results.

Alternatively, you can connect external microphones to the R-09 through the 3.5 mm mic input, which provides menu-selectable plug-in power. You can also connect an external mic preamp and high-end mic to the deck's line input. Note that left and right input levels are linked and cannot be adjusted individually.

In our measurements, we found the internal mic preamps (not including any noise contributions from the internal mics) to be significantly quieter than the M-Audio MicroTrack 24/96's, but not as quiet as those of the Fostex FR-2 or the Sony PCM-D1, which is the lowest-noise performer we've tested. There are 13 ms of latency between mic input and headphone output, which could be disconcerting if you're hearing your own voice live (for example, if you're conducting an interview and simultaneously monitoring over headphones), but obviously this won't affect the resulting recording.

When using it as a practice tool, you'll appreciate the R-09's various playback modes, including Repeat Play, which loops any portion within a single recording. Repeat One Track, Repeat All Tracks, and a random-playback Shuffle mode are also provided.

Playback formats include a couple of varieties that the R-09 can't actually record itself: 32 kHz sampling-rate files and variable-bit-rate MP3s. These extra playback formats could make the R-09 more capable as a general-purpose MP3 music player, depending on your library's format.

Battery Life

With standard alkaline AA cells, you can expect up to 6 hours of 24-bit, 48 kHz recording, and perhaps 7 hours or more using NiMH cells. Recording MP3s instead of WAV files boosts the record time a bit, presumably because the amount of data written to the SD card is reduced.

In firmware version 1.02, which shipped with our review unit, the battery status indicator seemed somewhat erratic, especially with low batteries, showing a full charge on power-up but then dropping quickly a few moments later. It turns out that the battery status wasn't being checked immediately. This problem was corrected by firmware version 1.03, which we easily downloaded and installed. You'll get a Battery Low warning when your time is running out; we let the battery die during a lecture recording, and the R-09 closed the recorded file gracefully before turning out the lights. A battery change causes the record format to revert to the default WAV-file setting.

Menus and Files

Press and hold the Finder/Menu button on the front panel, and you'll find the various options that control the deck's behavior. The top-level menu has eight submenus, each of which includes a couple of options. Here's where you'll set sampling rate, file format, display contrast, clock time and date, and all the basic settings you would expect in a device like this.

A quick click of the front panel's Finder/Menu button brings up a scrolling list of the sound files on the SD card. The R-09 offers more computer-like file-management functionality than other portable recorders we've tested: files can be renamed using the R-09's multifunction transport controls; new folders can be created and named; new recordings are automatically placed in the current folder; and existing recordings can be moved between folders. There are also Delete and Copy commands, although the latter would be more useful if the R-09 also had some minimal editing facility, such as a track-split function. My only significant complaint is that the unit's file-naming convention follows the rather uninformative "R09_0001.wav" scheme. When you empty the card of data, the numbering restarts at 0001—I'd prefer it kept counting up to reduce the creation of duplicate sound-file names.

Although Edirol does not publish a list of supported SD cards, we were advised to stay away from "ultraburst" types of media, which are good for digital photography but are a poor choice for audio recording, where sustained throughput is critical.

In the Field
We used the R-09 in numerous recording situations—as a shirt-pocket logging recorder during a full day of field recording in the New York City subway system, as well as for rehearsals, lectures, nature sounds, and concerts. As a log recorder, the R-09's AGC mode allowed my colleague Bruce Koball to focus his attention on running the primary deck (a Fostex FR-2), manipulating his boom mic, and interacting with the subway operators. Besides providing a continuous document of the full-day session, we snagged some usable sounds when the primary deck wasn't running.

As a rehearsal recorder, the R-09's internal mics were convenient and up to the task at hand. In a marsh on the shores of Lake Tahoe, the internal mics were noisy—the midrange rush of 50 ducks taking flight from the still waters of the lake at dawn were certainly audible, but the high-frequency self-noise of the deck and internal mics was obtrusive in the final recording. Plugging my trusty Shure VP-88 into the minijack input yielded excellent results at a live rock 'n' roll show.

Unusual Features
The R-09 does have a few quirks. I'm amused by the inclusion of reverb on this device. Edirol's literature describes reverb as being "the world's most-requested/desired effect." On a multitrack, sure, but on a portable stereo recorder, where it will be applied to the entire track? I don't get it. Fortunately, it's a playback-only effect, so there's no danger of inadvertently engaging it and having it applied destructively to a recording.

Then there's the rather flimsy 2-position sliding door at the bottom of the unit. Half open, it reveals the USB connector and the SD card; releasing a tiny latch opens it further for battery access. The R-09 doesn't draw power through USB when connected, so if you're on internal batteries during file transfers, be sure your cells have enough juice to complete the process.

Also, the R-09's power management will turn the recorder off even when it's mounted on your desktop, leading to a Device Improperly Removed warning message from your operating system. Our wish-list items include an optical digital input, the ability to drop markers while recording, and a prerecord buffer.

Stand and Deliver
Nevertheless, it's hard to find much fault with a compact, moderately priced audio recorder that delivers as reliably as the Edirol R-09. We recommend it.

Rudy Trubitt and Bruce Koball record music and sound in the San Francisco Bay Area. Reach them at rudy@trubitt .com and bkoball@motionwest.com.
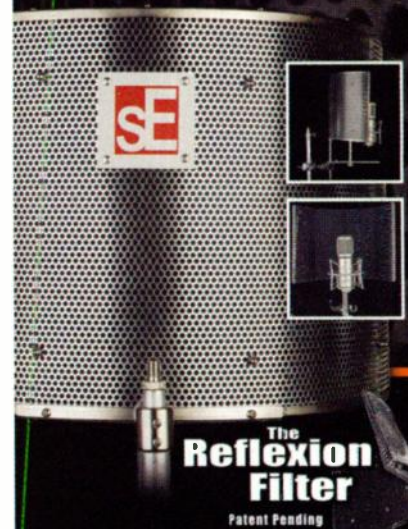
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FIG. 1: Classik Studio Reverb's front panels are color coded. Shown clockwise from top left: Room, Hall, Inverse, and Plate.

IK MULTIMEDIA

Classik Studio Reverb 1.1
(Mac/Win)

Classic reverb makes a comeback.

By Larry the O

I K Multimedia's latest offering, Classik Studio Reverb (CSR), adds reverb plug-ins to the company's sampling, synthesis, and mastering repertoire. Available for Mac OS X (version 10.3 or later, in AU, VST, and RTAS formats) or Windows 2000 and XP (in VST and RTAS formats), CSR runs standalone or as part of a bundle of IK plug-ins.

CSR consists of four plug-ins composing the "holy quartet" of digital-reverberation algorithms—Hall, Plate, Room, and Inverse—and runs at sampling rates of up to 192 kHz. In fact, the word "Classik" in the product's name indicates IK's intention to bring users the sound and features of classic, but unnamed, outboard digital reverbs of the '80s and '90s. With a strong feature set at a price lower than those of nearly all other

reverb plug-ins, CSR presents a real value to those who are on a budget or who like to have several reverb flavors on hand.

Basik Reverb Ingredients

Installation of CSR was simple and problem free. The review copy of CSR used the Syncrosoft hardware dongle for copy protection, but IK reports that it is changing its copy protection. (And good riddance—CSR occasionally lost communication with the dongle and crashed MOTU Digital Performer 4.61, the host in which I did nearly all of my evaluation.)

The four plug-ins use the same basic interface but are color coded to make it easy to see at a glance which one you're working with. The interface is a virtual-hardware front panel designed to look like an early '80s device that used only a 7-segment LED display (see Fig. 1).

Some users like the familiar look of physical hardware, which is the primary reason for virtual front panels. In fact, CSR's graphical user interface is easy to understand. I am of a different

Some users like the familiar look of physical hardware.

REV W

mind-set, however, and have groused about them before: virtual front panels may look cute, but their functionality is inferior to that of both the hardware they emulate and software interfaces designed to suit the control paradigm of a personal computer. I'll consider exceptions for software that models specific well-known hardware devices, but for everything else, the novelty wore off long ago. Manufacturers are so afraid of intimidating their users that they sacrifice real usability to give a product the appearance of usability.

Clicking on the front panel's Load legend calls up the preset menu. Each plug-in comes with around 20 presets, available in both Insert and Send configurations. (The only difference between the two configurations I could see is the Mix setting.) The presets cover the basic applications for each algorithm. For instance, the Plate plug-in has three presets for drums, three for vocals, two emulating gold-foil plates, and three "vintage" or "classic" presets, and the rest are mostly general purpose.

Save, Save As, and Delete buttons right on the panel make it easy to manage presets quickly. You'd better be sure you know what you want to do, though, because there are no confirmation messages for Save and Delete. The front panel also features two temporary storage buffers for A/B comparisons and pairs of LED-ladder meters for input and output levels.

Sonik Seasonings

When you want to edit the presets, you can operate CSR in Basic mode, which provides access to several key parameters, or Advanced mode, which offers five pages of parameters for serious tweakage. The parameters are I/O (mix levels and image width), Time (decay time, predelay, HF damping), Reverb (secondary settings such as diffusion), Color (high- and low-frequency EQ), and Reflections (two discrete early reflections). The Plate program adds an

FIG. 2: Classik Studio Reverb allows control through automation and the Modulation (top) and Macro (bottom) pages.

Echo page that has left and right recirculating delays.

Advanced mode also provides access to some of CSR's most potent functions. The Mod button calls up CSR's modulation matrix, which allows any of the advanced parameters to be controlled by one of the four onboard sources: two 5-waveform LFOs and two simple envelope generators (see Fig. 2). Up to eight parameters can be modulated. I got a wonderfully animated sound by applying a low-rate noise signal to the output image width, for instance.

The other powerful performance-modification feature is the Macro page. This page has eight "slots," each of which allows you to specify one of the four sliders on the right side of the panel as a source, any parameter as a destination, minimum and maximum values, and a curve shape. As an example, I tied Decay Time, Hi Cut EQ Frequency, and Hi Frequency damping frequency to one slider, each scaled differently. Each Macro slider can bear a 5-character name—not a lot of characters, but many more and they wouldn't fit in the small allotted space. As a crowning touch, any parameter, including any of the Macro sliders, can be automated by your digital audio sequencing software.

The only parameters I found myself wishing for were ones that would let me set separate high-frequency (HF) and low-frequency (LF) decay times with an adjustable crossover. HF damping can fill some of these needs, but it's not the same thing, and there are times when it doesn't really work (nor do the low- and high-cut filters work as a substitute in this case).

Puttink CSR to Use

CSR's parameters are reminiscent of '80s and '90s reverb units rather than what you'd find in the newest generation of convolution reverbs. IK also touts CSR as having more controllability with less CPU hit than some

convolution reverbs, and that is unquestionably the case. This makes it viable to use multiple instances in a session.

CSR was probably the last new thing I would be testing on my dual 800 MHz Mac G4, since my Mac Pro finally arrived. (The Mac Pro was not configured in time for this review.) It was impressive to me that I had no problem running several instances of CSR on the old machine while playing 12 to 16 tracks, many with other plug-ins on them.

I tried CSR on a number of sources, including drums, percussion, vocals, vibraphone, and guitar. After becoming acquainted with each of the algorithms, I decided to put together a quick comparison test. I started with a little drum jam centered around dumbek and djembe. I called up a Room drum preset and tweaked it until it sounded good. (The raw preset was too spitty and seemed to lack enough diffusion for the hand drums.) I then tried to create similar presets in Waves Renaissance Reverb, Audio Ease Altiverb 5, Universal Audio DreamVerb, a Lexicon PCM80, and a Kurzweil KSP-8. All of these except Renaissance Reverb cost considerably more than CSR, so it was certainly a trial by fire. (You can hear the results in **Web Clips 1 through 6.**)

Compared with the others, CSR's Room algorithm sounded almost like there was dry sound mixed in, even though I had the mix set to 100 percent wet. The reverb tail seemed somewhat low level compared with the early reflections. However, the reverb tail did have good density and an open timbre, whereas some of the other reverbs had darker sounds. The smaller room and ambience sounds were nice and tight, and I got some great sounds putting a few of those on a vocal and then adding a touch of a second, longer reverb on top of that.

I enjoyed having many parameters available when tailoring the sound. I think IK has struck a good balance between offering sufficient control to give flexibility and main-taining a set of parameters familiar enough to make the plug-ins easy to understand and work with.

When it comes to the Plate program, I must admit I've never been impressed with anyone's digital reverb plates; they generally strike me as not as useful as other algorithms. In fact, the only one that really knocks me out is Universal Audio's Plate 140. CSR sounded fine but sat with the rest of the pack well behind the UA plug-in.

Hall algorithms are the strong suit of reverbs like Altiverb and the PCM80. CSR sounded good but needed some EQ to clean out the low midrange, which was somewhat tubbier than on the other reverbs. Long decay tails on CSR were not smooth unless the high frequencies were fairly heavily rolled off. I noticed that quite a few presets on all the algorithms used the highest setting on the HF damping amount, HF Cut.

In all cases, CSR exhibited good density and provided a lot of variety (available through tweaking). In general, however, I was not so happy when I created settings that allowed a significant amount of high-frequency content in the reverb.

Conklusions

Many reverb solutions are available for project studios today, at a wide range of prices. CSR is one of the most economical choices and provides a solid sound with a rich set of features that make it very versatile. The Macro and Modulation features add even more creative options. The user interface is easy to understand and familiar enough that most people will be able to jump into tweaking parameters easily.

In short, CSR makes a great bread-and-butter reverb and delivers handsomely on an investment for that use. Give it a listen. **EM WEB CLIPS**

Larry the O has stolen his own identity, leaving him with no User account on his new Mac Pro. Then again, some would say he's always been a no-account.
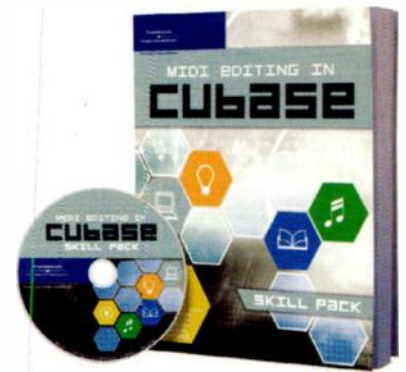
REVI
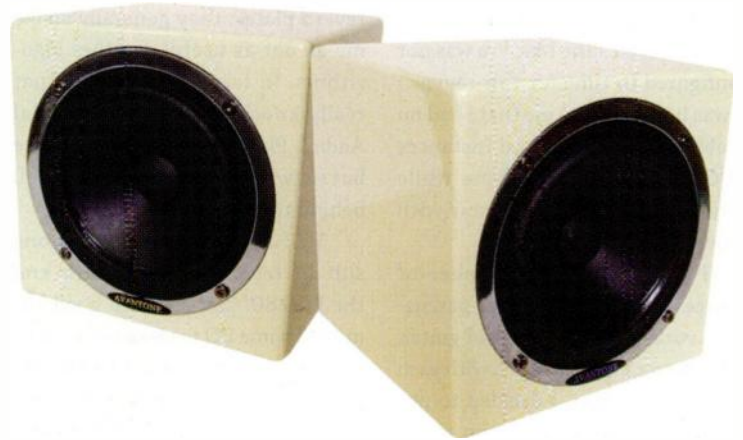
FIG. 1: The MixCubes feature one 5-inch driver per cabinet. All cabinet edges are radiused.

Close-field monitoring just got a whole lot smaller.

By Michael Cooper

I wouldn't dream of mixing without listening part of the time to small, bass-challenged close-field monitors. Listening back to a mix on such a system gives me a better idea of what it will sound like on consumer systems such as tiny boomboxes, cheap car stereos, computer speakers, and clock radios. With bass frequencies no longer masking the overall sound, I can more readily

The MixCubes should not be an engineer's sole mix monitor.

hear whether or not critical midrange elements such as lead and background vocals and electric guitars are at the right volumes relative to each other. Also, the closer the monitor approximates a point source (by virtue of

its diminished size), the more I get the overall picture instead of being lost in a sea of details thrown at me by larger monitors with widely offset drivers. Those details are critically important to hear, but I want the bigger picture too.

Equally important, I can put bass-challenged monitors where they belong—on a console meter bridge or workstation shelf—without upper-bass frequencies sounding too boomy due to furniture resonances and speaker-boundary effects. (It should be a critical design goal for all close-field monitors to have this clarity and evenness in response when placed thus, yet the overwhelming majority of manufacturers overlook it in their quest to build a small monitor with impressive bass response.)

The Avant Electronics Avantone MixCubes, a stereo set of close-field monitors from Avant Electronics, addresses all of these performance issues with amazing

REVIEW

grace. Additionally, because the MixCubes each employ only one wide-range driver, they do not need to use a crossover between high and low frequencies. This avoids the midrange hole and phase anomalies that plague many 2-way systems.

The MixCubes present a refreshing approach, one that's different from that taken by most speaker manufacturers. In a market flooded with close-field monitors, they stand out for providing an eminently useful and unique reference.

FIG. 2: Three-way binding posts on the rear of each cabinet afford connections to a power amplifier.

Popularity Contest

The MixCubes were inspired by, and designed to improve upon, the performance and construction of the Auratone 5c SoundCube, the seminal close-field monitor that enjoyed widespread popularity in the early '80s. Like Auratones, each MixCube features a single 5-inch driver in a sealed cabinet (that is, it's not ported) and employs a passive design (see **Fig. 1**). The speaker's cone uses a blend of paper pulp and mica fibers for rigidity and longevity. The MixCubes' frequency response is stated to be 90 Hz to 17 kHz, with no plus-or-minus tolerances given.

The MixCubes are striking in appearance, sporting a high-gloss buttercream polyurethane finish over their nonlayered, high-rigidity MDF (medium-density fiberboard) cabinets. These are positively tiny monitors, each measuring 6.5 inches in all three dimensions (width, height, and depth).

All cabinet edges are radiused to minimize diffractive effects and comb filtering, as well as to improve imaging. A 7-mm-thick neoprene pad affixed to the base of each cabinet serves as a skid-resistant acoustic decoupler, an outstanding solution for the boomy resonances mentioned earlier that generally plague shelf-mounted monitors. (Incidentally, such decoupling should also improve imaging.) To accommodate unusual speaker setups, each cabinet also features

a mount recessed into its base so you can place the monitor on a mic stand. The cabinets are magnetically shielded and can be put near a CRT monitor without distorting the picture.

On the rear of each cabinet, 3-way metal binding posts accommodate bare wire, spade lugs, and single or dual banana plugs (see **Fig. 2**). Avant recommends using the MixCubes with a power amp having 50 to 200W of solid-state power or 10 to 100W of tube power.

Speak Up

I placed a pair of MixCubes on the monitor shelves of my Omnirax MixStation/02R, behind which I have installed in my control room an Acoustic Sciences Corporation Attack Wall (a modular arrangement of tube traps that tighten up the room's impulse response). Normally I place my shelf-mounted close-fields (Yamaha NS-10M Studio monitors) on Auralex MoPads to decouple them from the furniture. Because the MixCubes have neoprene decouplers, I decided to forgo using the MoPads.

For my tests, I listened to a number of pop and country productions, including my own mixes. The first thing I noticed was that electric guitars and especially lead vocals sounded very forward. Good—now I can really hear those money tracks. I could easily tell whether the lead vocal was suitably louder than the guitars, fiddle, and pedal steel or whether it was being stepped on by those instruments. I could also confidently assess the relative loudness of background vocals compared with that of the lead vocal.

The bottom end of bass guitar tracks was audible but very understated. On the other hand, bass instruments such as bass guitar and kick drum also sounded extremely tight; there was no hint of flabbiness whatsoever.

Rapping my knuckles sharply on the tops of the cabinets, I heard no resonances. These monitors are built solid, and a five-year manufacturer's warranty supports that impression.

PRODUCT SUMMARY

AVANT ELECTRONICS   Avantone MixCubes

passive monitors
$199 per pair

AUDIO QUALITY	5
VALUE	5

RATING PRODUCTS FROM 1 TO 5

PROS: Outstanding proxy for bass-deficient consumer playback systems. Excellent transient response and stereo imaging. Provides useful window into critical midrange band of the mix. Tiny footprint makes placement a snap. Built-in neoprene decoupler. Rock-bottom price.

CONS: None.

MANUFACTURER
Avant Electronics
www.avantelectronics.com

The upper portion of the MixCubes' high-frequency response is slightly soft, which is a good thing. A sizzly high end, in combination with weak bass response, would've made these monitors fatiguing to listen to. Instead, I found their overall sound to be very pleasing and smooth at low to moderately loud listening levels. And despite their subtle high-end and dramatic low-end rolloffs, the MixCubes' transient response and stereo imaging sounded excellent. Percussive mixes were really popping.

Even when I briefly cranked them up to deliver 97 dB SPL (A weighted) at my mix position (3 feet from the speakers), the MixCubes didn't distort at all. I didn't want to damage my hearing by going any louder, but suffice it to say that these babies are no wimps. And with a rated sensitivity of 93 dB at 1W per meter, they are also quite efficient (which means they are relatively loud for a given input).

A Clear Winner

For all their great attributes, it is important to remember the MixCubes' niche purpose and limitations. I would not use the MixCubes as my only, or even my primary, reference monitor while mixing. Doing so would almost guarantee bass-heavy mixes with dog-whistle highs and lead vocals tucked too far back in the mix. But as a proxy for bass-challenged consumer playback systems, used alternately with full-bandwidth monitors, the MixCubes provide unique and invaluable insight.

Avant dutifully states in its literature that the MixCubes should not be an engineer's sole mix monitor. But make no mistake: despite their Lilliputian size and band-limited response, these are not toys or entry-level monitors. The MixCubes are professional tools that provide a highly useful alternate reference when in pursuit of the perfect mix that will translate well to consumer playback systems. Simply put, these are outstanding monitors. What's more, the price is ridiculously affordable. The Avantone MixCubes get my highest recommendation.

Michael Cooper is a contributing editor for EM. You can hear some of his mixes online at www.myspace.com/michaelcooperrecording.

FIG. 1: Vienna Instruments' sample player runs standalone or as a plug-in. Its well-designed interface offers plenty of real-time control over the library's 24-bit, 44.1 kHz sample content.

VSL

Vienna Instruments
Symphonic Cube 1.1 (Mac/Win)

Take sampling (and your budget) to its limits.

By Geary Yelton

A t the 2002 AES Convention, a new company called Vienna Symphonic Library (VSL) announced that it would soon release the largest, most versatile, and most realistic orchestral sample library ever attempted. With private funding and a seemingly unlimited budget, VSL had custom-built a state-of-the-art studio and recorded thousands of hours of samples played by world-class musicians from ensembles such as the Vienna Philharmonic and Vienna Radio Symphony Orchestras. Led by visionary Herb Tucmandl, VSL had begun editing, processing, programming, and assembling those samples into a massive collection of unprecedented proportions.

A few months later, VSL shipped the First Edition of the Complete Orchestral Package, followed by the Pro Edition in 2003 and many volumes in the Horizon Series in 2004. All of them supplied detailed 16-bit sample libraries for popular software samplers, and all but the First Edition are still available.

In 2006, VSL began shipping all ten volumes in its gargantuan 24-bit Vienna Instruments library, known collectively as Symphonic Cube. Vienna Instruments is more than a sample library, however; it's also the

name of the software instrument that serves as its graphical user interface.

Setting the Stage

The ten boxed collections, which you may purchase separately, are Solo Strings, Chamber Strings, Orchestral Strings I and II, Harps, Woodwinds I and II, Brass I and II, and Percussion. They vary in size, price, and number of installation DVDs. Each includes a CD-ROM for installing the Vienna Instruments software instrument, which runs standalone and as a plug-in for AU and VST hosts.

To accommodate so much content, I bought a 500 GB Seagate hard drive and installed it in an external USB 2.0 enclosure. My computer was a dual-processor 2.3 GHz Apple Power Mac G5 with Mac OS X 10.4.8, 4 GB of RAM, and a 16× DVD drive. Although I ran Vienna Instruments standalone and as a VST plug-in in Steinberg Cubase SX3 and as an AU plug-in in Apple Logic Pro 7.1, I spent most of my time working in MOTU Digital Performer 4.61.

Before installation, you must download and install the latest version of Syncrosoft's License Control Center (LCC). Then you go to VSL's Web site to authorize your ViennaKey ($23), a Syncrosoft-compatible USB dongle.

You begin installing each volume by running the Vienna Instruments Library Installer, inserting the first DVD, and waiting while the files are copied and decompressed; on my computer, it took about 50 minutes per disc. The installer will visually prompt you to insert the next DVD. In all, the 10 volumes furnish 29 installation DVDs, most of them double density. You cannot install only certain instruments; you must install the entire volume containing the instrument you need. Nor can you delete from disk any instruments or articulations you don't plan to use.

After installation, the total content added up to about 375 GB—much less than the 550 GB I had expected from VSL's specifications. According to the company, Vienna Instruments decompresses its 24-bit data every time you load an instrument into memory. It uses a proprietary technique with a 3:2 compression ratio, making 24-bit samples on disk the size of 16-bit samples.

Because most users don't have a single hard drive large enough to hold the entire collection, you can distribute the sample libraries on several drives. In any case, you must initially run the included Directory Manager application to tell Vienna Instruments where to find its content.

To authorize Vienna Instruments, you must register each volume on VSL's Web site and then paste the activation code you receive into Syncrosoft's LCC application before you can download a license to your ViennaKey. Installing and authorizing the entire Symphonic Cube took more than two days for me to complete.

Big, Bigger, Biggest

Vienna Instruments is available in three editions: Standard, Extended, and Full. The Standard Library provides the articulations that are most useful for every instrument. The Extended Library, an add-on to the Standard Library, gives you articulations that are more detailed and of greater variety, but few additional instruments. Together, those two editions make up the Full Library, the version that VSL authorized for my review.

When you install any volume, you install its entire content, Standard and Extended. After authorizing the Standard Library, you can use any volume's Extended Library for 30 days from the first time you open its Extended content. If you don't spring for the Extended Library at the end of 30 days, you can't delete the unusable data from your hard drive. I suspect that given the choice, most users would rather reinstall the entire Standard Library than keep hundreds of gigabytes of unusable data on their hard drives.

Virtual Vienna

In the center of Vienna Instruments' GUI is the Selector Ring; clicking on one of five buttons on its perimeter activates a function page (see **Fig. 1**). The Right Workspace contains either the browser or controls for the current

function page, and the Left Workspace displays detailed information about the current page. Near the bottom of the window is a keyboard flanked by Pitch and Volume knobs.

Vienna Instruments arranges sample data in a 3-tier hierarchy of orchestral instruments and variations. Presets contain Matrices, and Matrices contain Patches. A Patch, the most basic level in the hierarchy, is a multisample of an instrument playing a single articulation or event. It could be a solo violin played staccato, a bass clarinet trill, or a tubular bell hit. When you click on Vienna Instruments' Patch Assign button, the browser reveals an extensive list of Patches within folders (see **Fig. 2**).

The next level up in the hierarchy is a Matrix. A Matrix contains one or more Cells, and each Cell contains a Patch (see **Fig. 3**). A Matrix loads a number of related articulations—such as legato, portato, and staccato—and allows you to quickly switch between them during real-time performance. Patches may contain other variations as well; a glockenspiel Matrix, for example, may offer single hits, rolls, or glissandos with a choice of wood or metal mallets. Each instrument comes with a variety of preconfigured Matrices, and you can easily create and save your own.

A Matrix also stores controller assignments for navigating between Cells in real time. You can instantly switch articulations by moving your mod wheel, changing Velocity, or playing faster or slower. Cells are arranged horizontally and vertically, and you can assign one controller to move horizontally and another to move vertically through the Matrix to instantly activate any Cell. Switching between Cells is how you change articulations, and thus control musical expression, in real time.

At the top of the hierarchy are Presets. A Preset contains as many as 12 Matrices arranged in a useful combination, which are selected using keyswitches. Most instruments have a Level 1 Preset and a Level 2 Preset. Generally, Level 1 Presets contain fewer variations; for instance, whereas Level 1 Presets for harp offer only major glissandos, Level 2 Presets also give you minor and diminished glissandos (see **Web Clip 1**). Level 1 Presets contain only data from the Standard Library; Level 2 Presets also contain data from the Extended Library. Consequently, Level 2 Presets take longer to load, but they offer much more flexibility.
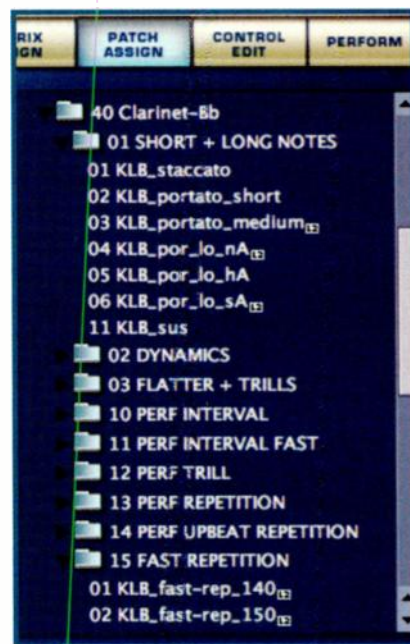
FIG. 2: Vienna Instruments' browser lets you select from an extensive list of Patches. Each instrument-specific Patch offers a single multisampled articulation or some other kind of variation.

VIENNA INSTRUMENTS SYMPHONIC CUBE

Assume Control

Clicking on the Control Edit button lets you assign controllers for switching between Cells in two directions (see **Fig. 4**). You can also switch between two opposing articulations in the same Patch, such as crescendo and decrescendo or up and down arpeggios. Controllers can be keyswitches, the pitch wheel, Note On Velocity, playing speed, or any MIDI Control Change (CC) message.

The Perform Control page provides sliders to control parameters such as attack and release, lowpass filter cutoff, and the depth of crossfading between Patches (see **Fig. 5**). In addition, you can tweak the master tuning, enable or disable release samples, or instantly switch to the slowest Patch in a speed-controlled Matrix.

A MIDI Learn function lets you easily assign any physical controller to an onscreen knob, button, or slider. Clicking on the Map Control button will display a page for assigning and editing controller assignments and defining their response curves. You can also right-click on an onscreen control and then move a hardware control to assign it.

By default, Pitch Bend has no effect. You'll probably want to assign your pitch bender to control the onscreen Pitch knob, but no matter what controller you choose, bends will be limited to a maximum whole step up or down. That's the range I usually assign for Pitch Bend, but if you're accustomed to a larger interval, you'll be disappointed.

Also on the Perform Control page is the Sample Management section. Vienna Instruments' Sample Management scheme is a brilliantly conceived and elegantly implemented technique for conserving your computer's resources. When you click on the Sample Management section's Learn button and then play a track in your sequencer, Vienna Instruments will remember which samples were used in the sequence. If you then click on the Optimize button, the plug-in will delete any unused samples from memory. This procedure very effectively frees up memory that would otherwise be unnecessarily wasted. To reload the deleted samples, just click on the Reset button.

Real Enough?

Frank Zappa complained that orchestral players were uncooperative and obstinate (admittedly, he
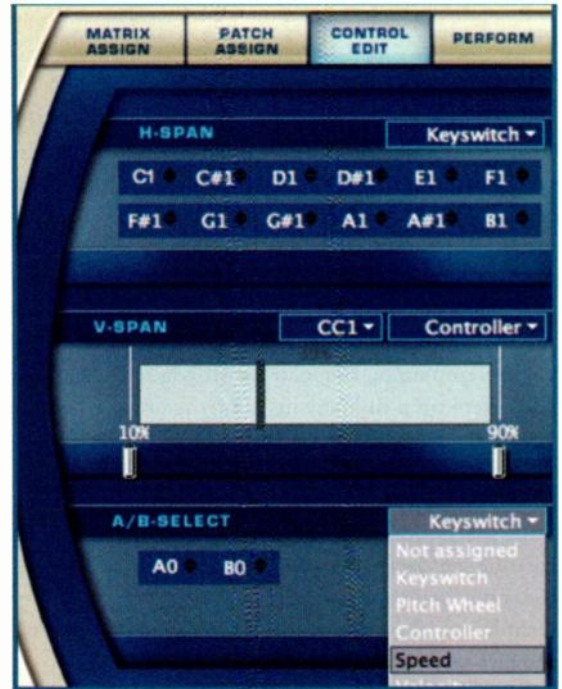
FIG. 4: You can switch between Cells using keyswitches, Pitch Bend, or any MIDI Control Change message, or by changing Velocity or even playing speed. In this example, keyswitching moves through the Matrix horizontally, and Modulation (CC 1) moves through the matrix vertically.

FIG. 3: A Matrix arranges Cells in two dimensions, horizontally and vertically, and you use controllers to change Patches in either direction. The Matrix shown here lets you instantly switch between 12 Muted Trumpet articulations.
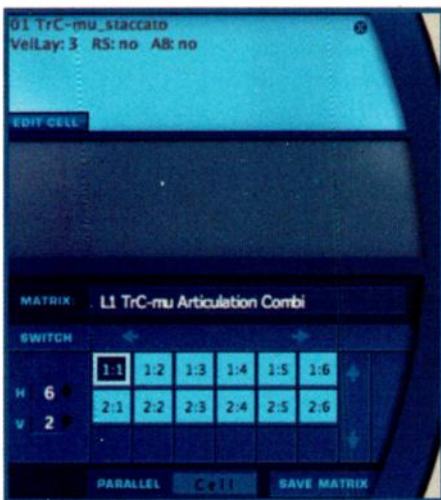
gave them grounds to be that way). He would have loved Symphonic Cube. It does what you want it to without objection, no matter how outrageous your demands. It also excels at just about everything you'd normally want an orchestra to do, and although it will probably take you quite a while to master it completely, at least it can eventually be mastered—you can't say the same of a real orchestra, as Zappa would no doubt have attested.

Limited space prevents me from describing the sounds in much detail; suffice it to say that without exception, they are extremely good. The strings are impressively versatile, even providing Gypsy articulations and scale runs in the Extended Library (see **Web Clip 2**). The woodwinds are stunningly lifelike, with solo versions of 13 instruments and bassoon, clarinet, flute, and oboe trios (see **Web Clip 3**). Brass instruments feature practically every articulation imaginable, as well as offbeat horns such as the cimbasso, Vienna horn, and Wagner tuba (see **Web Clip 4**). The percussion section delivers every instrument you'd expect and some you might not, from taiko drums and finger cymbals to lithophone and Peking opera gong (see **Web Clip 5**). Check out the complete contents on VSL's Web site (http://vsl.co.at).

If you listen to enough symphonic music that you can distinguish one orchestra or conductor from another, Vienna Instruments won't fool you. Even if you're only a casual classical-music buff, careful listening will reveal the sampled forgery if you suspect one. But for most audiences, under most circumstances, and in the hands of a skilled musician, Vienna Instruments will probably fool most of the people most of the time

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Understanding Meters
Building MIDI Loops

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(see **Web Clip 6**). Without a doubt, it is possible to make wonderful music with Symphonic Cube, as proven by many fine recordings on VSL's Web site. The sampling quality is uniformly excellent, and Vienna Instruments sounds just as convincing as the state of the art allows.

The software instrument has some brilliant touches—such as the tremendous variety of techniques for switching articulations and the ability to automatically delete unused samples—and a few annoyances. Installation and authorization were tedious (see **Web Clip 7**), and whenever I loaded a Preset or Matrix, Vienna Instruments demanded to be the foremost window.

Sequencing with Vienna Instruments has room for improvement. The software instrument is not multitimbral, and it receives MIDI data in Omni mode only. That means you'll need to instantiate a new plug-in for every instrument or instrument group you want to record. As long as you remember to delete the unused samples as soon as you finish every track, you can probably load dozens of instances without any resource-related problems (depending on your computer, of course). Nonetheless, I still wish it were multitimbral, because I'd rather not have to keep track of so many plug-ins.

Using Vienna Instruments was frustrating at times, but ultimately the results were always worth the effort. The samples sound generally amazing, and once I had learned my way around, it was fairly easy to find just the articulation I needed at any given moment. However, you can't edit or remap the samples as you can with sampler-format orchestral libraries; that may or may not be important to you.

As with any virtual orchestra, there is plenty to learn before you can get the best performance from Vienna Instruments. Fortunately, VSL's documentation is

PRODUCT SUMMARY

VIENNA SYMPHONIC LIBRARY
Vienna Instruments Symphonic Cube 1.1

virtual orchestra
Standard Library, $4,620
Extended Library, $6,370
Full Library, $10,990

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

RATING PRODUCTS FROM 1 TO 5

PROS: Breathtaking sound. Complete selection of instruments and articulations. Well-designed expressive capabilities. Superb documentation.

CONS: Seriously pricey. Not multitimbral. Problematic copy protection. Can't delete Extended samples from hard disk.

MANUFACTURER
Vienna Symphonic Library/
Ilio (distributor)
www.ilio.com
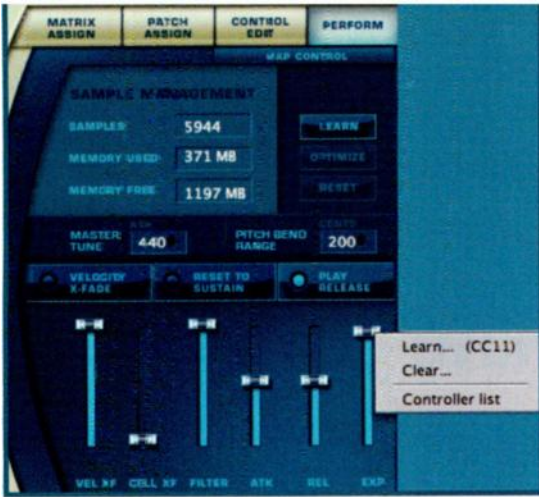
FIG. 5: On the Perform Control page, you can specify parameters such as filter cutoff and pitch-bend range, and you can delete any unused samples from RAM.

top-notch. A 40-page printed manual comes with each volume, as well as a PDF version of the same manual and another PDF that details every Patch, Matrix, and Preset for every instrument in that volume—more than 1,300 PDF pages for the entire Symphonic Cube. Considering Vienna Instruments' cost, though, I'd have expected full paper manuals too. You'll find a ton of knowledge that anyone can access on VSL's Web site, including very detailed information about individual orchestral instruments and orchestration. In addition, you can download 45 minutes' worth of helpful video clips.

As Good as It Gets

Vienna Instruments is groundbreaking, that's for sure. The full-tilt Symphonic Cube has twice the content of any competing virtual orchestra, and it costs more than a Korg OASYS. Whether the cost is a problem depends on whether your alternative is to buy a less expensive sample library or to hire a real orchestra. If you can afford it, you won't regret your purchase. And if you own any previous VSL sample libraries, you may be entitled to a discount for any Extended or Full Library.

Symphonic Cube is remarkably complete, furnishing practically every articulation and orchestral instrument you could ever want—hats off to Vienna Symphonic Library for such a monumental achievement. If you (or, better yet, your employer) can afford to join the exclusive club of Symphonic Cube users, you will come as close as possible to putting a living, breathing orchestra on your desktop.

EM associate editor Geary Yelton lives in Charlotte, North Carolina.

FIG. 1: Strike's handsome main screen has a browser that supports multiple file types (left); macro controls for the style, kit, and mix (center); and a mousable MIDI keyboard for triggering patterns and muting individual drums (bottom).

DIGIDESIGN Strike 1.0 (Mac/Win)

Finally, a drum plug-in for Pro Tools pros.

By Jim Aikin

The acquisition of German software developer Wizoo by Digidesign's parent company, Avid, has been great news for both Digidesign and the legions of loyal Pro Tools users. First came Xpand, a free multisample player that grew out of the technology first used in Wizoo Hypersonic (a Steinberg product); next up was Hybrid, a stunning modeled analog synth; and now there's Strike, an amazingly versatile virtual drummer, which uses some of the groundbreaking concepts first seen in Wizoo Darbuka and Latigo.

Using Strike is not unlike sitting behind the glass in a studio and recording a real trap-set drummer—but with some extras that session producers can only dream of. You can tell Strike what style or beat to play in the verse, chorus, and bridge; request a more complex part or a simpler one; ask for stronger accents or tone them down; change the amount of swing; make the track looser or tighter; indicate the places where you want the verse, chorus, and bridge; add fills; swap in different drum kits without changing the underlying beat; and retune individual drums (without getting up from your chair).

You can adjust the mix of the close, overhead, and room mics; add effects to single drums or the whole kit; teach the drummer an entirely new beat; or, as a last resort, trigger individual drum hits yourself wherever they're needed—all for the price of about three good loop libraries.

As with Xpand and Hybrid, non–Pro Tools users will have to find another solution, because Strike is available only as an RTAS plug-in. (Steinberg Groove Agent 3, which is compatible with VST, DXi, AU, and ReWire, is similar to Strike in some ways.)

Strike One: Styles

Right out of the box, Strike provides a healthy number of musical styles; among the 50 are bebop, boogaloo, Brit rock, Texas boogie, West Coast funk, and a lot more. Each style consists of about 30 patterns—intros, verse grooves, chorus grooves, fills, and so on. The styles are extraordinarily authentic and energetic. They're obviously based on analyses of real drum performances.

When a style is loaded, its patterns are laid out on a MIDI keyboard with a separate pattern on each key, suitable for real-time auditioning and easy sequencing.
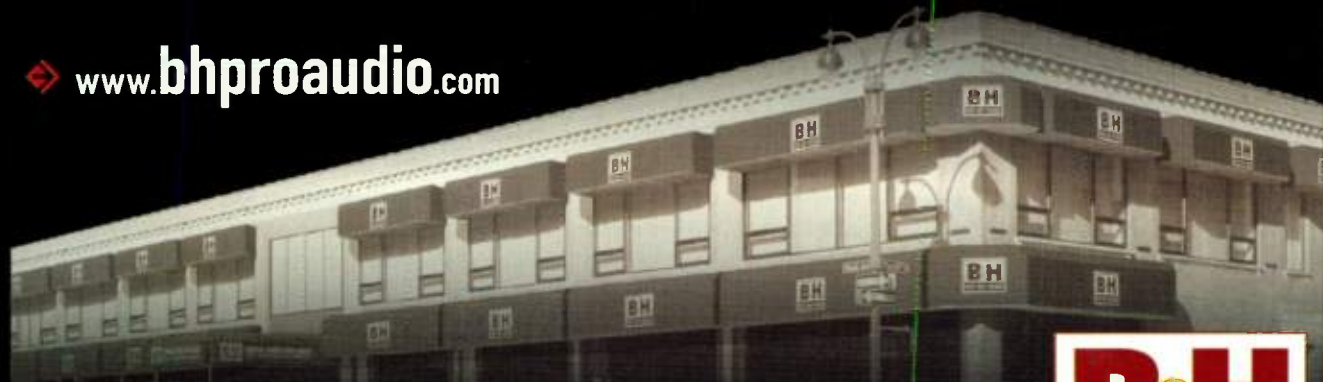
The keyboard can operate in either Latch or Momentary mode. In Latch mode, you can tap a key and the beat will keep going until you stop it; in Momentary mode, it will play only for as long as you hold the key down.

Even before you start tweaking knobs, the mix-and-match possibilities are awesome. You can load a style as a Settings preset, a file type that contains not only the musical data but also the drum kit and the mix. Then you can drag individual patterns from other styles onto various MIDI keys, thereby playing, for example, jazz samba and house patterns with a grunge kit.

The Style knobs (see Fig. 1) give you macro control over the virtual drummer's overall feel. You can adjust the playing dynamics, thus adding more-pronounced accents or smoothing out the track. Hit Variation can be applied to all beats, or to specific beats within the bar. Adding too much Hit Variation tends to undermine the cohesiveness of the groove—but a little helps the beat sound more like it was played by a real drummer. Timing can be adjusted from "natural" to "tight." The Feel knob adjusts from Ahead to Fat. The Groove knob lets you apply a variable amount of 16th-, eighth-, or quarter-note swing. Your own groove templates can be imported. The Intensity and Complexity sliders, which do about what their names suggest, are also worth playing with.

Then there's the Jam button: when switched on, it will add small variations to the beat. And naturally, all of the knobs and buttons can respond to MIDI Control Change messages of your choice, or to Pro Tools automation.

Clicking on the Style button in the lower-left corner of the panel opens up a set of controls with which you can change the intensity, complexity, and dynamics of individual drums. Each drum can also be slid forward or backward in time to alter the feel of the groove. Here again, the parameters are MIDI controllable in real time.

Strike Two: Mixing

Strike provides a conventional console view (see Fig. 2), in which each drum has its own channel. Depending on the drum, you'll find a choice of up to five mic positions that can be blended to create the finished sound. With the kick channel, for instance, you get center, on-axis, front, room, and overhead mics. The snare has top, bottom, room, and overhead mics. I heard no phase cancellation

FIG. 2: Strike's mixer allows you to blend mic positions, add insert effects, and control the levels of individual drums through MIDI.

problems when combining mics—the samples seem to have been phase aligned.

Each channel strip has two insert effects and its own 3-band EQ, which can be switched pre- or postinsert. The inserts are normally applied only to the direct mics, not to the room, overhead, and talkback channels, but the over-heads can be switched into the inserts if desired. The over-head, room, and talkback channels have their own submix strips in the mixer, with their own effects. The master bus has two insert effects of its own in addition to knobs that control the Mic Leakage and Snare Buzz parameters.

The effects algorithms are just what's needed for drums: three different compressors, a brickwall limiter, a noise gate, an envelope shaper, dynamic EQ, three types of distortion, a mic modeler, a filter, a ring modu-lator, and the usual delay, reverb, chorus, and a few other items. The effects are not cheapies: most have four or five parameters, and the mic modeler has more than a dozen choices. The internal mixer has no aux sends, so if you want to apply one reverb to several drums to save CPU power, you'll need to route those drums to one of Strike's eight individual outputs.

Strike Three: Editing

Strike includes five distinct drum kits with about a dozen drums and cymbals apiece. For each instrument, you can adjust the tuning, start point, attack and decay time, and timbre shift. In addition, you can choose to load the economy, midsize, or XXL version of an instru-ment. Some XXL versions load up to 300 samples, so having a computer with a lot of RAM is a good idea. My 3 GHz Pentium 4 machine has 1 GB of RAM, and loading Strike slowed down other applications noticeably. You can give Strike some guidance about how much RAM to use on each drum channel, thus making efficient use of your system resources.

PRODUCT SUMMARY

DIGIDESIGN Strike 1.0

Pro Tools drum plug-in
$299

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

RATING PRODUCTS FROM 1 TO 5

PROS: Versatile editing. Great sound. Eight individual outputs. Wide range of onboard patterns. Choice of mic positioning for samples. Five editable drum kits.

CONS: Works only with Pro Tools. Won't let you import your own samples. Style Editor has no Undo command.

MANUFACTURER
Digidesign
www.digidesign.com

VST-format plug-ins for the Windows platform. You can buy them as well as download time-limited, automation-disabled demos from the Tone2 Web site. ·

Table Talk

FireBird's signal path starts with a pair of identical HCM oscillators, and they are the key to the sound of this synth. You can think of a wavetable as a series of spectral snapshots of a sound evolving over time. Each snapshot is a single-cycle waveform reflecting the harmonic content of the sound, and each wavetable contains 256 snapshots. FireBird ships with 51 wavetables along with 7 single-cycle waveforms for creating standard synth sounds. The wavetables cover a range of electronic and acoustic instruments as well as ten vocoded words and phrases. It would be nice to be able to create your own wavetables, but you can use FireBird's other assets to get a lot out of those provided.

FireBird's wavetable oscillators give you three types of control over how the wavetable is played. You can manipulate the wavetable harmonically by spectrum shifting, mixing several octave-transposed waveforms, or changing the starting position in the table. You can choose the direction and the repeat mode of the table—for example, making the waveform repeat forward and backward, play once and stop, or play from a random position in the table on each pass. Finally, you can choose the speed that playback traverses the wavetable, and that can be set in hertz or synced to tempo. A unison effect distributes multiple detuned voices across the

FireBird packs all modules along with an interactive keyboard onto this compact control panel.

stereo field, and an analog control introduces a random pitch-drift.

The oscillators are mixed and fed to a multimode filter whose modes include the usual filter types along with unusual choices such as amplitude and frequency modulation, downsampling, and formant filtering. The filter has its own ADSR envelope and is followed by an overdrive-distortion module. That is followed by a multi-effect with a variety of time-based processes: reverb, delay, flange, chorus, and so on. For modulation you get an LFO with some interesting, gating-style rhythmic waveshapes and a rhythm-programmable arpeggiator. When you use both the LFO and the arpeggiator, things can get amusingly weird.

Bank on It

FilterBank2 takes the filtering options in FireBird to the next level. It starts with a pair of multimode filters, and you get 53 modes compared with FireBird's 20. The filters can run in series or parallel, and when in parallel, their outputs are mixed, subtracted, or panned full left and right. You can restrict yourself to one filter or bypass both and use only the distortion, delay, and gating effects.

The filter types emphasize the big four—lowpass, highpass, bandpass, and notch—with variations in slope, filter response, and resonance characteristics. Beyond that, you'll find several 2-band EQs, comb filters, a phaser, and a smattering of formant filters. AM, FM, and six flavors of reverb round out the selection. The filters feed a distortion unit with effects ranging from tube-amp saturation to bit crushing to a Warble effect that adds nonlinear harmonics. A basic, tempo-synced feedback-delay effect follows the distortion module. Beyond simple echoes, the delay has spring-reverb and resonator modes as well as a variety of autopanning options.

For modulation you get a pair of multiwaveform LFOs similar to FireBird's, a 16-step gating sequencer, and an envelope follower. You can apply each of the modulators to most of FilterBank2's parameters. For example, in addition to typical volume gating, you can use the sequencer to control the delay time or the LFO frequency. You can use the

envelope follower with negative modulation polarity as a kind of compressor to lower a lowpass filter's cutoff as the input volume rises (see **Web Clip 1**).

The frequency range of the FilterBank2 LFOs runs well into the audio spectrum, and you can route them directly to the audio inputs. That allows FilterBank2 to be used as a subtractive synth, and a virtual instrument version is included for just that purpose. The FilterBank2 effects and instrument plug-ins each come with a bank full of useful presets; FireBird comes with three banks. For the price and range of sounds, these plug-ins are worth a listen.

Value (1 through 5): 3
Tone2
www.tone2.com

ROB PAPEN

Albino 3 (Mac/Win)
By Len Sasso

If you own an earlier version of Albino, there's no decision to make; browse over to the Rob Papen or LinPlug Web site and order the upgrade ($49) to Albino 3. You get four layers, which amounts to four Albinos in one, a slew of new and enhanced effects, an expanded modulation matrix, a sophisticated arpeggiator, and external audio input. If you haven't tried Albino, grab the demo; it's fully functional but belches noise every minute or so.

Albino 3 ($199) is a collaboration between Dutch sound designer Rob Papen of Emu Orbit-3 and Access Virus fame and German software developer Peter Linsener. It runs in any AU- or VSTi-compatible host under Mac OS X or Windows. You can purchase the downloadable version of Albino 3 from the LinPlug Web site. You can purchase the physical package (CD and printed manual) from dealers worldwide or from the Rob Papen Web site.

Analog Style

Each of Albino's four layers has four oscillators, and each oscillator can function as an analog or digital waveform generator, a noise source, or an external input. The digital oscillators mix spectral waveforms

and sound decidedly different from the analog oscillators, which generate their waveforms in real time and then mix in a suboscillator an octave below the fundamental. A Spread control activates as many as five panned and detuned voices per oscillator. You can send each oscillator to either or both of Albino's filters, and you get a variety of frequency- and amplitude-modulation routings among the oscillators and filters.

You can arrange Albino's two filters in series or parallel, and each filter is preceded by an overdrive-saturation effect. The three multimode-filter variants—Silk, Cream, and Scream—differ in both the modes they offer and their characteristic sound. Scream is new to Albino 3, and as you might guess, it is nasty. Also new in Albino 3, you can configure either filter as a comb filter.

The signal path ends in four identical stereo multi-effects units that you can arrange in series or parallel. All the usual suspects are accounted for: cho-

rus, phaser, flanger, filter, reverb, gate, compressor, and distortion.

The Swirl Factor
Modulation has always been a LinPlug strength, and Albino 3 doesn't disappoint. A 16-row modulation matrix routes 4 LFOs, 8 envelope generators, a 32-step arpeggiator, and various MIDI messages to practically any Albino parameter. Furthermore, Albino's robust MIDI Learn implementation lets you assign any MIDI continuous controller to any parameter, and you can save MIDI setups to disk.

The multiwaveform LFOs have their own delay, attack, and decay controls and optionally sync to tempo. The envelopes are switchable between ADSFR and a 5-stage format with independent level and time settings for each stage. All but the Mod envelope have a fixed routing in addition to any routings set up in the modulation matrix. That means you can use the matrix for arcane routings and still

Albino gets everything on a single control panel by toggling the oscillator, filter, modulator, and effects sections between different elements.

have filter and amplitude envelopes.

Albino's arpeggiator is really a cross between a step sequencer and an arpeggiator. Like any arpeggiator, it loops through the currently held chord with a variety of direction options. What's more, each step has its own Velocity, and steps can be silent or tied

to the note from the previous step (see **Web Clip 1**). You can copy and paste arpeggiator patterns between presets as well as save them to disk.

Albino's chord memorizer remembers the intervallic form of a chord. When it is active, single notes trigger chords with that form. The chord memorizer is particularly handy when used with the arpeggiator.

The 2,100 Club

When you match a software developer with a talented sound designer, you get a library of truly usable sounds, and by the third generation that library can get huge. Albino comes with more than 2,100 presets organized by category, and because the sound browser pulls presets directly from disk, you're not stuck with endless bank shuffling. The presets make full use of the arpeggiator as well as Albino's ability to produce really fat multilayered sounds and its multifarious modulation options. If you exhaust the factory library, you can click on the Gen button to randomize Albino's settings. For still more sounds, you can purchase additional libraries of 300 to 500 presets by Bigtone and Ian Boddy on the LinPlug Web site.

Albino 3 is an impressive upgrade to an out-of-the-ordinary synth. If you stuff all four layers with multioscillator presets, turn Spread on, and use lots of effects, you'll probably bring your CPU to its knees, as would any 80-oscillator synth. But with all the variation on offer, you can get great sounds with low CPU usage. Drive it; you'll like it.

EMWEB CLIPS

Value (1 through 5): 4
Rob Papen
www.robpapen.com
LinPlug Virtual Instruments
www.linplug.com

SUBMERSIBLE MUSIC

DrumCore 2 (Mac/Win)
By Mike Levine

Since its initial release, Submersible Music's DrumCore has offered users an innovative software platform for drum-

With additional loops, improved import and export capabilities, a more flexible MIDI drum module, ReWire sync, and more, DrumCore 2 has made drum-track creation even easier.

track creation. The application, which runs standalone and syncs to your sequencer through ReWire, provides a large library of easy-to-audition audio loops, one-shots, and MIDI loops, most of which were performed by one of a dozen well-known drummers including Zoro, Alan White, Matt Sorum, and Sly Dunbar. The loops and other content can be exported through DrumCore's interface into a range of sequencers.

You also get numerous editable MIDI drum kits, an inventive interface design, the ability to import and organize third-party samples, a randomizing function (the Gabrielizer) that works on audio and MIDI loops, and more. So impressive is DrumCore 2 ($249) that it won the EM Editors' Choice Award this year for Drum Machine/Module (Software).

For a look at DrumCore's basic features, see my review of DrumCore 1.0 in the March 2005 issue of EM (available at www.emusician.com). Before focusing on DrumCore 2, let me briefly summarize the major additions to version 1.5, which shipped in early 2005. Most notable in that release was the debut of Windows XP support (the program was initially Mac only) and the Drummer Pack optional expansion packs, which feature extended collections of loops from some of the DrumCore drummers.

Version 2 (or 2.1 for Intel Mac users) adds a number of useful improvements. You get additional content, including 5/4 and shuffle grooves from Terry Bozzio, more jazz-brush loops from Jon Bishop, additional pop grooves from

Ben Smith, odd-time-signature loops from Alan White, and country loops from Nashville session drummer Lonnie Wilson. (An expanded Lonnie Pack, offering a wider range of beats, is available as a Drummer Pack, as are packs from Terry Bozzio, Matt Sorum, Sly Dunbar, Zoro, and, by the time you read this, percussionist Luis Conti.)

MIDI Up

The MIDI drum kit feature has been upgraded significantly. It now gives you multiple ReWire outputs, and the individual drums can be edited for pitch, pan, and volume. With these changes, the kits (multiple kits are included for each of the 12 DrumCore drummers) have become a very powerful feature, offering a wide range of excellent sounds for triggering over MIDI.

Also new to version 2 is support for REX2 and Acid-format loops. You can now import such content into DrumCore, and it will show up similarly to the factory-included content. (Since version 1.0 you've been able to import AIFF, WAV, and SD2 files, as well as MIDI files.) Submersible encourages users to import all their drum loop data into DrumCore and use it as a centralized, searchable database. The importation process requires several steps, but once you get the hang of it, it's pretty easy.

On the export side, the program has always let you send loops directly into a range of sequencing programs, including Digidesign Pro Tools, Steinberg Cubase and Nuendo, MOTU Digital Performer, Apple Logic and GarageBand, and Mackie Tracktion. In many cases, you can just

drag-and-drop from the DrumCore interface.

Version 2 adds another option: you can now export REX2 files from within DrumCore, including all the DrumCore content. This is really handy for working with a sequencer that supports REX2 files, like Pro Tools, because you can export your drum loops and then later change tempos if you want (see **Web Clip 1**).

DrumCore has long supported ReWire for audio routing, but now it supports ReWire sync, which makes it possible to audition content from the DrumCore application along with your sequence, before actually importing the loops. It also lets you sync up to any host tempo, even if it's not a round number. For instance, DrumCore will sync at 124.32 bpm and allow you to export loops at that tempo. Cool.

At the Core
I auditioned DrumCore 2 mainly with Digital Performer 5.1 and Pro Tools 7.1, and I found it to be a solid performer and a great drum-track creation tool. Version 2 is more flexible, more content rich, more reliable, and simpler to use than earlier releases. The improvements to the MIDI drum kits have made it into an awesome MIDI drum module as well. If I had a wish list for the next version of DrumCore, it would include more fills for each individual groove, and support for multitrack drum parts. The latter would offer users the flexibility of mixing and separately processing the looped drums, which would be very useful.

DrumCore 2.5 is due out this February. Improvements will include streamlined drum selection in the MIDI kit, the ability to run as a standalone MIDI **EMWEB CLIPS** instrument, and more.

Value (1 through 5): 4
Submersible Music
www.drumcore.com

ROUTLEDGE

Handmade Electronic Music: The Art of Hardware Hacking
By Tom Duff

Nicolas Collins is an instrument-building animal, and he wants you to be one too.

He has spent the last three or four years touring the world, doing workshops aimed at musicians who want to delve into the ever-deepening world of DIY electronics. His new book, *Handmade Electronic Music* ($24.95), combines his workshop notes with historical sidebars and an eclectic and inspiring CD featuring a panoply of first-class builders and players.

Handmade Electronic Music is not the first book aimed at musical home brewers (the ground was broken by Craig Anderton's *Electronic Projects for Musicians*, in 1975), but it is the first to encourage design and experimentation rather than just a craftsmanlike reproduction of given circuits. It reads more like a science book than a cookbook. In fact, its chapters reminded me fondly of Mr. Puddicombe's eighth-grade science class, with its reassuringly uniform experiment reports, every one with sections ritualistically labeled Purpose, Apparatus and Materials, Method, Observations, and Conclusions.

After an introduction that is by turns inspirational ("Fear not!"), practical ("Try to avoid short circuits"), and poetic-philosophical ("Many circuit hacks are like butterflies, beautiful but short-lived"), Collins's musical-scientific journey begins by observing the vibratory world around us. Almost everything vibrates, either electromagnetically or mechanically. We can capture those vibrations with a guitar pickup, a tape head, or a simple contact mic (made from a Radio Shack piezo buzzer) and turn them toward our own nefarious musical purposes. (Collins's droll humor tends toward the mad-scientist end of the spectrum.)

The World Is My Oscillator
The surprising sounds of highly amplified kitchen equipment are an avant-garde staple: witness Paul Lansky's classic "Table's Clear." We can turn the nefariousness up a notch with a second piezo buzzer configured to inject vibration into our innocent mechanical subjects, turning flowerpots and bicycle frames into effects processors. Having gone this far, it's only a matter of time before the inspiration comes
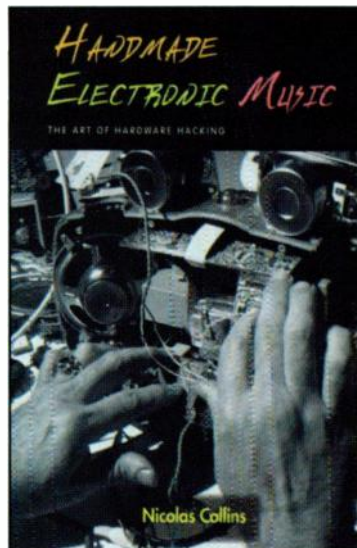
In *Handmade Electronic Music*, Nicolas Collins takes you step-by-step into the exciting world of DIY electronic instruments.

to feed the output back into the input, making infernal pie-tin and toilet-float oscillators.

From there the book introduces the reader to circuit bending, the art of rewiring surplus electronics to do unexpected things. There are good chapters on identifying and mod'ing clock circuits, sussing out useful berds (he recommends a literal hands-on approach), and adding nonstandard controls using photoresistors, pressure sensors made from antistatic foam, and even pencil drawings (graphite pencil lead makes a good resistor). In some ways Collins's brief treatment of the subject goes beyond Q. Reed Ghazala's indispensable, pioneering guide *Circuit Bending: Build Your Own Alien Instruments* (Wiley, 2005), suggesting the use of fruits and vegetables as pressure-sensitive resistors.

Next up is a detailed discussion of breadboarding that starts out with instructions to build simple oscillators from widely available CMOS inverter chips. (Digital inverters are really just high-gain inverting amplifiers. They make great distortion/fuzz boxes, and feeding their output to their input makes them oscillate. A simple resistor-capacitor in the feedback path lets you control the oscillation frequency.) All of the resistive controls mentioned in the

circuit-bending section can be applied here as well, as though they were little resistive modules. Circuit design is a lot like patching a modular synthesizer but at a subatomic level.

Having mastered the inverter-oscillator, we're ready to attack the ubiquitous CMOS NAND gate, which you can think of as an inverter with an extra control input. Wiring it up like an inverter-oscillator and plugging a signal into the extra input gives us all sorts of complex ring-modulation action. Of course the modulation input can be anything at all—another NAND oscillator, a circuit-bent Speak and Spell, a guitar plugged through an inverter-fuzz, whatever.

A series of chapters on video hacking (TVs are full of oscillators!) leads into the book's final section, which combines practical advice about building mixers and amplifiers and dealing with AC power with an inspiring discussion of controller hacking. Video game controllers are really just digital interfaces to a load of switches and variable resistors. Crack one open, and you can easily repurpose it to sense temperature, humidity, motion, weight, air pressure, compass direction, acceleration, gas mixture—the list goes on and on.

Watch This

My son Tim and I have enjoyed many hours following Collins's lead, using a telephone pickup to explore the radiation our laptops emit and wiring up and tweaking some of the book's examples. We've even used our home-brew contact mics in live performances.

Twenty years ago, *Popular Electronics* ceased publication, inaugurating a long lull on the do-it-yourself front. However, things are picking up in the DIY world—notably with the advent of O'Reilly's *Make* magazine—and Collins's book is just in time to catch a newly receptive audience. Despite some quirks (the chapter labeled "Ohm's Law" talks a lot about resistors but doesn't actually explain Ohm's law), I recommend *Handmade Electronic Music* wholeheartedly to anyone interested in exploring sound. **EM**

Value (1 through 5): 4
Routledge
www.routledge-ny.com

World Radio History

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World Radio History

Hit the road with your Dual Core MOTU Studio

Take Digital Performer 5 and the rest of your MacBook-powered MOTU studio on the road for unprecedented portable studio power and performance.

Performance squared

The Apple MacBook Pro is built on revolutionary Intel Core 2 Duo, which packs the power of two processor cores (up to 2.33GHz) inside a single chip. This means you now get mind-boggling dual-processor performance that you can stuff in your backpack and sling over your shoulder. Now add support for internal hard drives up to 200GB, 3GB of RAM, FireWire 800 and double-layer burning SuperDrive, and your MacBook Pro portable MOTU Studio delivers unprecedented speed and performance - to go.

FireWire I/O with 8 mic inputs

The new MOTU 8pre delivers eight mic inputs in one rack space, complete with a five-segment level meter, phantom power switch, 20 dB pad switch and trim knob right on the front panel for each input. Now add two banks of ADAT optical digital I/O for eight more channels — even at 88.2 or 96 kHz. Top it off with main outputs and MIDI I/O, and you've got a FireWire audio interface that turns your Mac into a complete studio that can record your entire band. Or, if you already own an optical-equipped MOTU FireWire, PCI or UltraFast USB2 audio interface, the 8pre is the perfect way to add 8 mic inputs directly to your MOTU interface CueMix DSP on-board mixing via 8-channel optical.

NEW

Liquid Technology

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

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

Waves native processing

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

Ultimate expandable synth

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

Legendary drummers

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

Komplete control

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

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Hit the road with your Dual Core MOTU Studio

Compact MIDI controller

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

Professional pad controller

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

Advanced waveform editing

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

Control room monitoring

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

Texas in a Suitcase By Diane Gershuny

Recording the rootsy, retro C&W of Lonesome Spurs.

Lynda Kay Parker and Danny B. Harvey are the component parts of the quirky country duo Lonesome Spurs. Though both are native Texans, they're based in the countryside north of Los Angeles. Self-described as the "White Stripes of country," the pair's debut CD, *Lonesome Spurs* (Cleopatra Records, 2006), shows off the retro sound they call "honky-tonk garage." Their music is centered around the vocal stylings of Parker—who doubles on rhythm guitar and a bass drum made from a suitcase—and the twangy Tele playing of Harvey.

Harvey produced the disc in his basement studio. He had previously produced two psychobilly bands—13 Cats and the Head Cat (both featuring Motorhead's Lemmy Kilmister and the Stray Cats' Slim Jim Phantom)—and rockabilly godmother Wanda Jackson.

"We do all the recording, mastering, and artwork ourselves," says Harvey. "The recording process, which took three weeks, was done the way most rock records have been recorded since the '60s: we do the basic tracks live, and then replace the instruments that have to be fixed."

The Spurs record through a MOTU 2408mkII and a Digidesign Mbox into a 2 GHz dual-processor Apple Power Mac G5 into MOTU Digital Performer software, and monitor with KRK 6000 close-fields. "We never use more than eight inputs at a time," Harvey says, "and we monitor the output in

stereo, so the recording process is pretty streamlined."

To get his twangy tone, Harvey plugs his James Trussart Steelcaster (or a Fender Telecaster Custom or a Gibson ES-335) into either a Fender Hot Rod Deville 2 × 12 combo, a Music Man HD-130 head with a 2 × 12 open-backed cabinet, or a Vox ToneLab modeling processor. The amps are placed in the bathroom off of the garage and miked with a Shure SM57 or SM58.

Parker's suitcase bass drum is a vintage Samsonite, rigged up with a Duallist kick drum pedal that operates alternating beaters. It's miked with an AKG CB3000 condenser. "It's the only drum you'll hear on the recording," Harvey says. "Lynda uses the Duallist pedal, most often used by metal drummers for superfast bass drum parts, to create a rolling train-beat sound. The main beater is made of wood with a shaker attached, and the second beater is made of felt so it creates a kind of 'ghost' beat."

Harvey and Parker like to cut their basic tracks together in the same room (he keeps about 90 percent of what he's done on the first take). "We record the track as many times as we need to until we're happy with the feel of her suitcase and rhythm guitar and of our interaction," explains Harvey. The close proximity and visual contact helps them keep their recorded performances tight.

An AKG C3000 going through one of the Mbox's mic preamps is the recording chain of choice when Parker overdubs her lead vocals or her 1935 Gibson tenor guitar. "I love those Focusrite Class A microphone preamps," says Harvey. "They give a lot of bang for the buck."

Harvey mixes the tracks down to Digidesign Pro Tools LE 7.0. Once the final tracks have been compiled, he masters the disc using BIAS Peak 4, and then loads that into Roxio Jam to author the CD-R to Redbook specs.

"The freedom that DIY studios provide artists with—to express themselves creatively at any time of the night or day without time limitations, financial constraints, or outside interference," says Harvey, "is priceless!" EM

Lonesome Spurs/Parker and Harvey in the studio.

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