em®
ELECTRONIC MUSICIAN

MARCH 2011

How to do high-quality rough mixes—fast

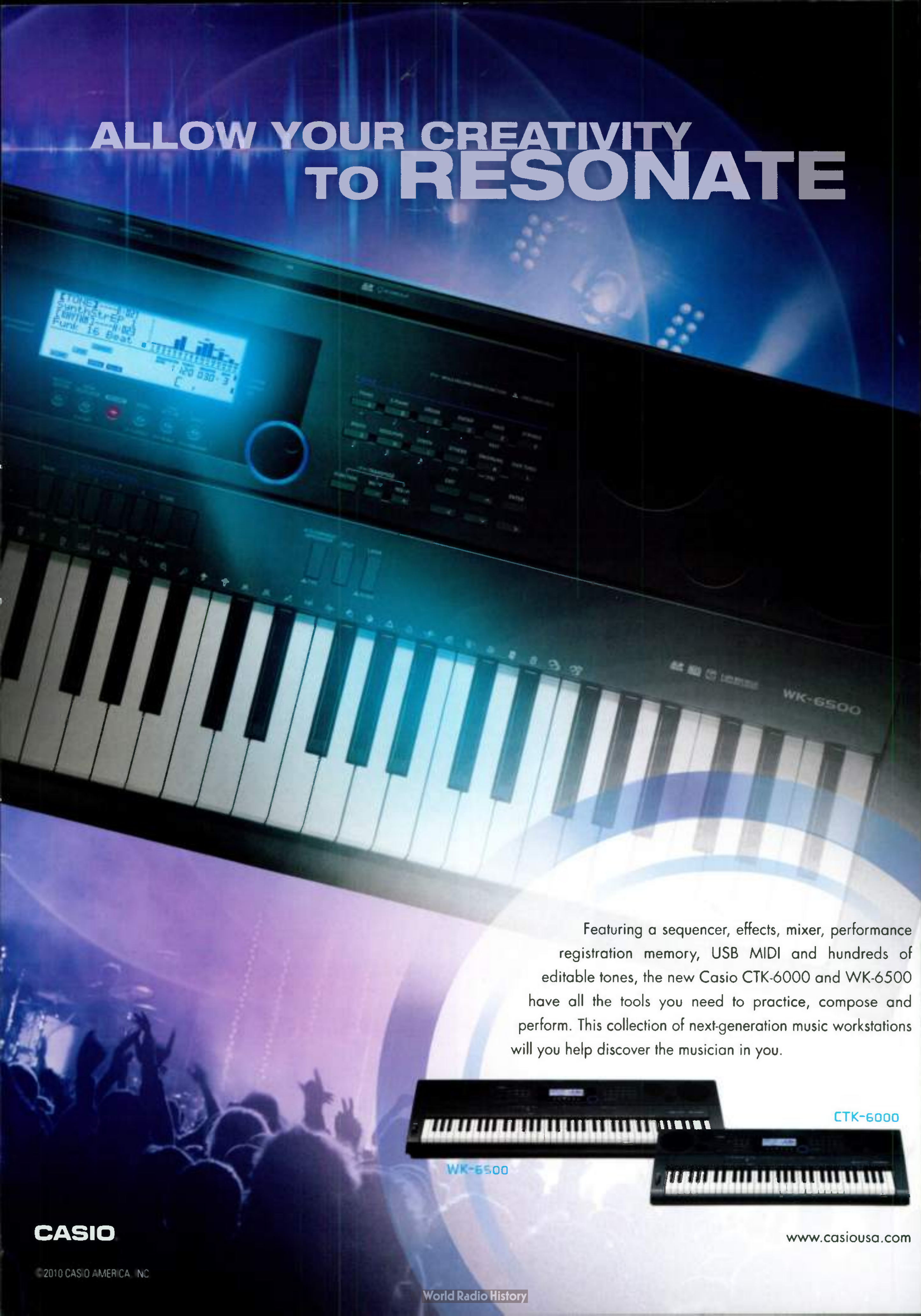
Advanced techniques for step sequencing

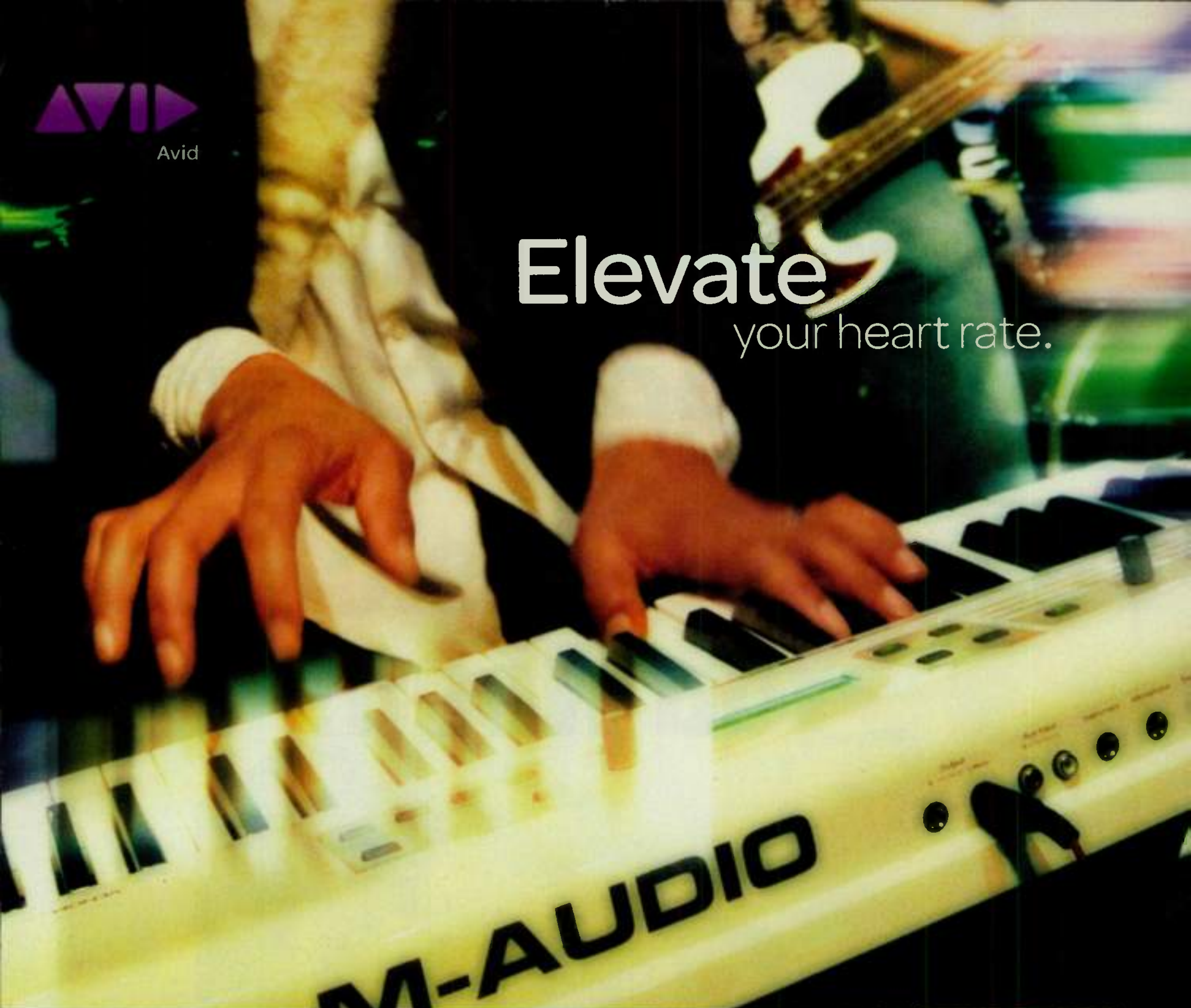
Inside the world of indie game music

WARM UP YOUR TRACKS

8 PLUG-INS THAT GIVE ANALOG FLAVOR TO DIGITAL RECORDINGS

HOW CAKE RECORDED A HIT ALBUM IN THEIR NO-FRILLS STUDIO


ALLOW YOUR CREATIVITY TO RESONATE
WK-6500
Featuring a sequencer, effects, mixer, performance registration memory, USB MIDI and hundreds of editable tones, the new Casio CTK-6000 and WK-6500 have all the tools you need to practice, compose and perform. This collection of next-generation music workstations will you help discover the musician in you.
CTK-6000
WK-6500
CASIO
www.casiousa.com
©2010 CASIO AMERICA, INC
World Radio History


Avid
Elevate
your heart rate.
M-AUDIO

From Instruments to Music Computers,

Sweetwater®

Has the Music Gear You Need!

Sweetwater Is the Best Place to Buy a Computer!

The best way to put your mix together fast is to start with a rock-solid system. With a custom music production computer from Sweetwater, you'll be ready to hit the ground running. Choose one of our custom Creation Station PCs or a Mac from Sweetwater's Apple Store, and have our expert staff set up a complete system, ready to meet your needs. Your computer will arrive fully loaded, with all software installed and all hardware tested. And thanks to our FREE tech support, your entire system will stay up and running smoothly — no hassles no headaches.

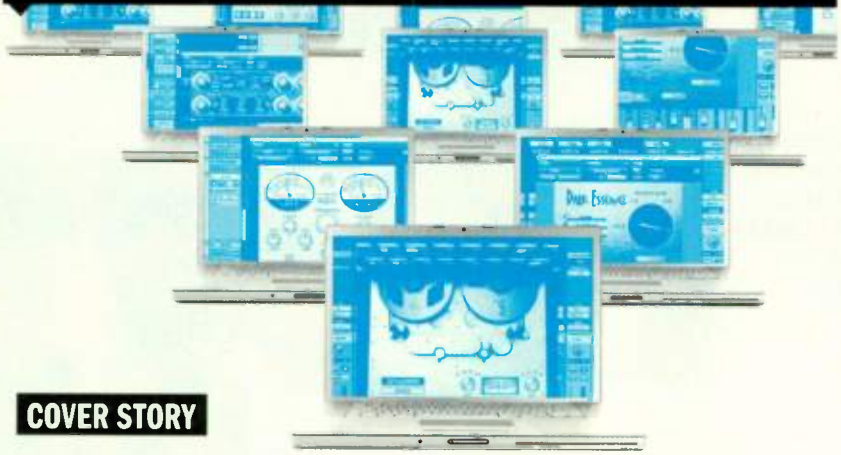
Follow this Tag to check out iZotope Nectar, an all-in-one vocal plug-in that lets you get professional results fast!

(800) 222–4700 • www.sweetwater.com

ELECTRONIC MUSICIAN

MARCH 2011 VOL. 27, NO. 3

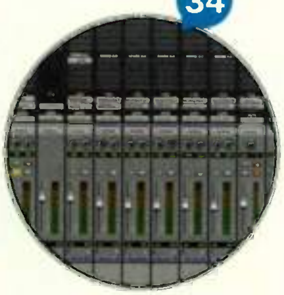
FEATURES



COVER STORY

30

34



REVIEWS

52

57

57

22

42

20

66

COLUMNS

DEPARTMENTS

EM (ISSN 0884-4720) is published monthly by Penton Media, Inc., 9800 Metcalf Ave., Overland Park, KS 66212 (www.penton.com). This is Volume 27, Issue 3, March 2011. One-year (12 issues) subscription is $24. Canada is $30. All other international is $50. Prices subject to change. Periodicals postage paid at Shawnee Mission, KS, and additional mailing offices. Canadian GST #129597951. Canadian Post International Publications Mail Product (Canadian Distribution) Sales Agreement No. 40612608. Canadian return address: Bleuchip International, P.O. Box 25542, London, ON N6C 6B2. POSTMASTER: Send address changes to EM, P.O. Box 15605, North Hollywood, CA 91615.

FIRST TAKE

A Good Forecast

This is my first chance to write this column since the NAMM show, so I thought I'd share with you some impressions. My biggest takeaway was the sense of optimism that was present at this year's show, in marked contrast from the gloomy outlook—due to the recession—of the past couple of NAMMs. Attendance this year was record-setting. More than 90,000 people set foot in the Anaheim Convention Center during the four-day event.

In the music production/recording segment of the show (most of which is clustered in Hall A, especially in a section referred to by some as "software alley"), there were new products galore. We've got a dozen of them profiled in the What's New section of this issue on page 12 and a lot more covered in our NAMM section at emusician.com. As mentioned, there was a general feeling among the manufacturers that 2011 is going to be a good year. I hope that's the case, because a healthy industry will mean that more innovation takes place and more new products are developed and brought to market. Not only is that good for the manufacturers and the press that covers them, but also for the gear-buying public.

On the subject of new products, when you look at the product review section in this issue, you might notice something a little different about the ratings. We are now allowing half points. Instead of being restricted to whole numbers between 1 and 5, *EM* reviewers now have the freedom use the half point (for instance a 3.5 or a 4.5), if they feel it's warranted.

In essence, this modification changes our ratings from a 5-point to a 10-point system, which allows reviewers more subtlety and nuance in how they rate products. This change has been instituted not only for full reviews, which have four different ratings categories, but for "Quick Picks," which have a single overall rating.

Those of you who've been reading *EM* for a while might remember that we have tinkered with our rating system on occasion in the past. For many years, we did use the half point. Then, in 2006, after what I recall was about a three-hour meeting about it (at one period in its history, the *EM* editorial staff had famously long meetings), we decided to go to whole numbers only, thinking it would be more direct and would encourage reviewers to "commit" to a particular rating. But after several years of that, we've decided that a more flexible system would work better. We want to be as fair and accurate as possible with how we rate products.

Photo by Maria Cohen

ELECTRONIC MUSICIAN

EMUSICIAN.COM
A PENTON MEDIA PUBLICATION

EDITOR/SENIOR MEDIA PRODUCER Mike Levine, mlevine@emusician.com
EDITORIAL DIRECTOR Tom Kenny, Tom.Kenny@penton.com
GROUP MANAGING EDITOR Sarah Benzuly, Sarah.Benzuly@penton.com
SENIOR ASSOCIATE EDITOR Andrew Ward, Andrew.Ward@penton.com
EXECUTIVE EDITOR George Petersen, George.Petersen@penton.com
TECHNICAL EDITOR Kevin Becka, emeditorial@emusician.com
CONTRIBUTING EDITORS Michael Cooper, Marty Cutler, Len Sasso, Dennis Miller, Larry the O, Gino Robair, Scott Wilkinson, Geary Yelton
ART DIRECTOR Isabelle Pantazis, Isabelle.Pantazis@penton.com
INFORMATIONAL GRAPHICS Chuck Dahmer, chuckd@chuckdahmer.com
ONLINE AUDIENCE DEVELOPMENT MANAGER Brad Erpelding, Brad.Erpelding@penton.com

SENIOR VICE PRESIDENT Kim Paulsen, Kim.Paulsen@penton.com
PUBLISHER Shahla Hebets, Shahla.Hebets@penton.com
ONLINE SALES DEVELOPMENT DIRECTOR Angie Gates, Angie.Gates@penton.com
SENIOR ACCOUNT EXECUTIVE, WESTERN REGION Janis Crowley, Janis.Crowley@penton.com
EASTERN SALES DIRECTOR Paul Leifer, pleifer@media-sales.net
EUROPEAN/INTERNATIONAL SALES Richard Woolley, richardwoolley@btclick.com
LIST RENTAL Marie Briganti, (877) 796-6947, mbriganti@meritdirect.com
MARKETING MANAGER Tyler Reed, Tyler.Reed@penton.com
CLASSIFIEDS SALES MANAGER Corey Shaffer, Corey.Shaffer@penton.com
PRODUCTION MANAGER Liz Turner, Liz.Turner@penton.com

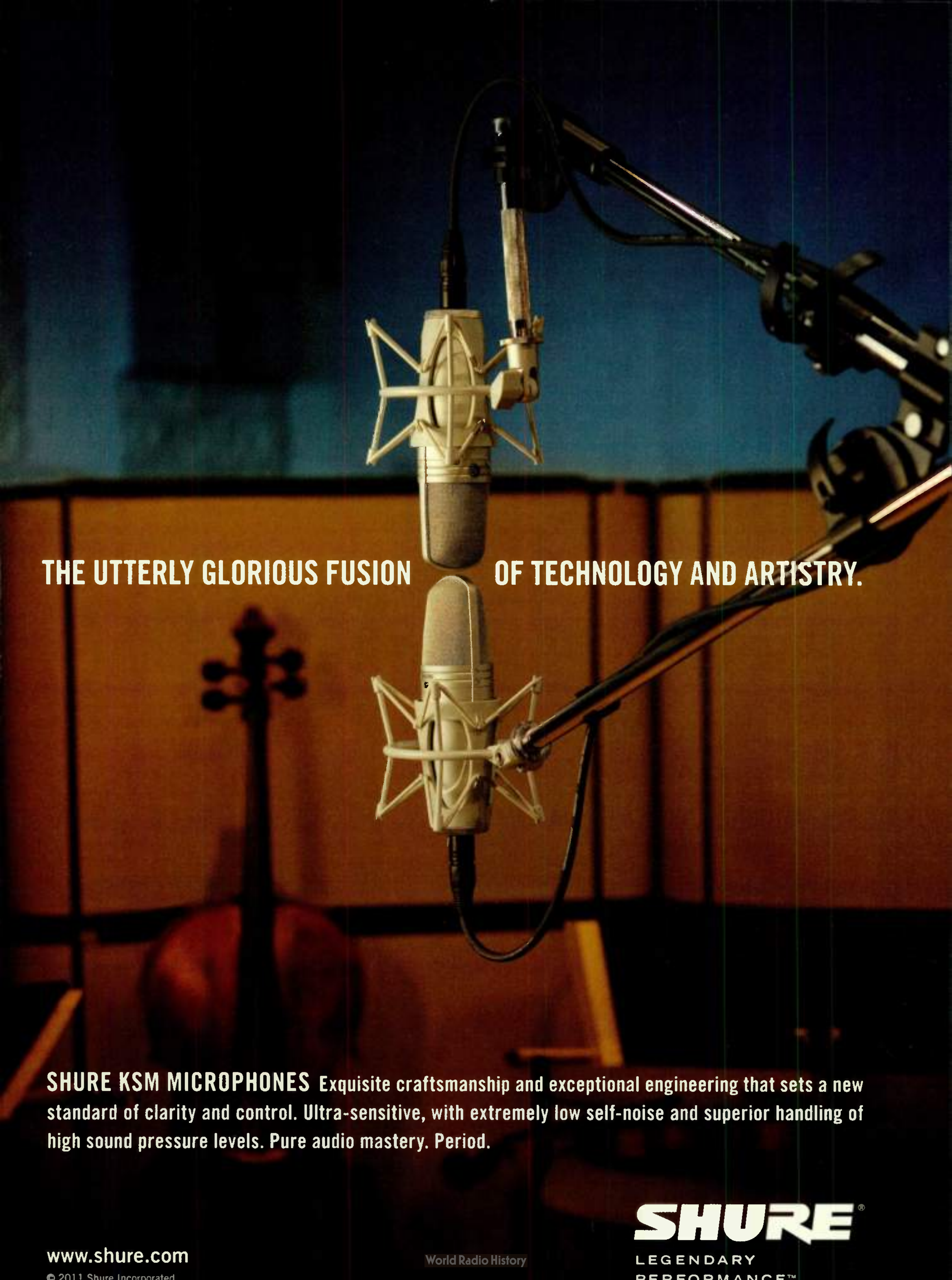
PENTON

CHIEF EXECUTIVE OFFICER Sharon Rowlands, Sharon.Rowlands@penton.com
CHIEF FINANCIAL OFFICER/EXECUTIVE VICE PRESIDENT Jean Clifton, Jean.Clifton@penton.com
EDITORIAL, ADVERTISING, AND BUSINESS OFFICES 6400 Hollis St., Suite 12, Emeryville, CA 94608, USA, (510) 653-3307
SUBSCRIBER CUSTOMER SERVICE To subscribe, change your address, or check on your current account status, go to www.emusician.com and click on Customer Service for fastest service. Or email electronicmusician@pubservice.com, call toll-free (866) 860-7087 (U.S.) or (818) 487-2020 (outside the U.S.), or write to PO Box 16886, North Hollywood, CA 91606.
REPRINTS Reprints are available through Penton Media Reprints, (888) 858-8851, or reprints@pentonreprints.com. Instant reprints and permissions may be purchased directly from our Web site; look for the iCopyright tag appended to the end of each article.
BACK ISSUES Back Issues are available for $10 each by calling (866) 860-7087 or (818) 487-2020.
PHOTOCOPIES Authorization to photocopy articles for internal corporate, personal, or instructional use may be obtained from the Copyright Clearance Center (CCC) at (978) 750-8400. Obtain further information at www.copyright.com.
ARCHIVES AND MICROFORM This magazine is available for research and retrieval of selected archived articles from leading electronic databases and online search services, including Factiva, Lexis-Nexis, and ProQuest. For microform availability, contact National Archive Publishing Company at (800) 521-0600 or (734) 761-4700, or search the Serials in Microform listings at www.napubco.com.
PRIVACY POLICY Your privacy is a priority to us. For a full policy statement about privacy and information dissemination practices related to Penton Media products, please visit our Web site at www.penton.com.

CORPORATE OFFICE
Penton Media, Inc. · 249 West 17th Street · New York, NY 10011

ABM

PRINTED IN THE USA. ALSO PUBLISHERS OF MIX®


THE UTTERLY GLORIOUS FUSION OF TECHNOLOGY AND ARTISTRY.
SHURE KSM MICROPHONES Exquisite craftsmanship and exceptional engineering that sets a new standard of clarity and control. Ultra-sensitive, with extremely low self-noise and superior handling of high sound pressure levels. Pure audio mastery. Period.
SHURE®
LEGENDARY
www.shure.com

FRONT PANEL

DOWNLOAD OF THE MONTH *

U&I SOFTWARE XX (MAC)

By Len Sasso

When you want to create MIDI patterns other than by recording or drawing them in, you typically have two options: use the arpeggiators, randomizers, and step sequencers found in most DAWs, or build your own pattern generators in applications such as Cycling '74 Max/MSP. U&I Software Xx 5.2 (Mac, $99; download from uisoftware.com) offers a powerful alternative by providing a host of graphical tools to build complex note patterns. Some tools are completely rule-based and others include random elements. Many of the tools can generate both monophonic (all notes on one track) and polyphonic parts—you have 24 tracks to play with—and you can limit the notes on any track to a selected key and mode. You can play your creations using the Mac's generic GM synth, Xx's complement of very creative and fully programmable synths from U&I's flagship software MetaSynth, or your favorite AU plug-ins (with some limitations). Using Xx is not exactly a no-brainer, but it bears a strong resemblance to a standard piano-roll MIDI editor. Once you've mastered a few nonstandard techniques, you'll find it easy to use. Best of all, you can export your creations as either audio or MIDI files for use in your DAW.

You can enter and add notes to an Xx project in three ways: play them from a MIDI or computer keyboard, draw them in using one of 11 tools, or use Xx's Multi Generator, a 3-track step sequencer with a twist. The tools let you enter single notes, chords, random note clusters, two- and three-part canons, predefined patterns (you can add your own), and multimeasure/multitrack patterns derived from statistical parameters you set. That last tool, the Random Generator, is context sensitive—its action depends on where you click and the notes already present. The Multi Generator is especially useful for creating drum parts (see Web Clip 1). You get separate bar charts for pitch and velocity for each of its three tracks. You can generate bars randomly or draw them in, and you can set each track's number of steps and scale its tempo. Using different numbers of steps lets you generate long, non-repeating patterns—for example, it takes 42 measures for 3, 7, and 8-beat sequences to repeat. *

OPTION-CLICK

Fig. 1: Peter Cushing's career moved backward between *Star Wars* and *Top Secret*, but reversing sound is a creative goldmine.

BORK! BORK! BORK! YOUR MUSIC

Add Some Foreign Intrigue to Your Tracks

There's a great audio moment in the '80s comedy *Top Secret* when Val Kilmer chats with the big-eyed Swedish bookstore owner (Fig. 1): You quickly realize the filmmakers are simulating Swedish speech by playing the scene backward. I enjoy layering background voices in my productions, so I thought I'd try that tech-

HEAR AN EXAMPLE OF XX IN ACTION * EMUSICIAN.COM/BONUS_MATERIAL

THIS MONTH'S SOUNDTRACK

By Mike Levine

This group of albums takes us on a mainly instrumental journey through deserts, electronic soundscapes, the psychedelic West of the future, and the days of the work week.

STEVE ROACH/BRIAN PARNHAM: *The Desert Inbetween* (PROJEKT)
Roach teams up with Parnham to offer an homage to the desert featuring eerie soundscapes that mix electronic textures with organic instruments like didgeridoo and Waterphone.

Photo by Adam Fleishman

KEN ELKINSON: *Music for Commuting* (AUGUST SON PRODUCTIONS)
Elkinson used eight different synths when recording this three-double-CD box set of ambient-electronic music. Each of the CDs represents a different part of the week, with the music changing from drone-like to more upbeat to spacey as the "week" progresses.

YELLOWBIRDS: *The Color* (THE ROYAL POTATO FAMILY)
The lone album in this month's batch that contains vocals, *The Color* is the brainchild of Sam Cohen from the psychedelic collective Apollo Sunshine. Folk-rock song structures mix with spacey instrumental textures to create what Cohen calls the "4th Dimensional Future West."

THINGS THAT OCCUR IN NATURE: *Snow Flower* (THINGS THAT OCCUR IN NATURE)
This ambient post-rock project features flowing, often arpeggiated keyboard lines, with guitars, bass, and drums woven in around them. Check out the video that goes with the song "Sky (Fast Moving Clouds)" at emusician.com.

ROBERT MILES: *TH1RT3EN* (S:ALT)
Producer Miles cooks up a blend of prog rock and electronic music on this instrumental album that offers synths; cool, textural guitar playing; and much more. Robert Fripp is one of several guest artists.

Photo by Jed DeMoss

THIS MONTH ON EMUSICIAN.COM

ALL NEW// EMBOOKS
The all-new EMBooks give you in-depth tutorial tips and techniques for any recording project. Now available: Synth Programming and Laptop Production at mixbooks.com.

BLOG// ROBAIR REPORT
Our intrepid former editor blogs about all issues relating to music technology and offers periodic updates as he rebuilds his studio from the ground up.

LISTEN// WARM IT UP

Hear the featured saturation plug-ins in use.

LISTEN// IZOTOPE NECTAR REVIEW

Listen to reviewer Steve Skinner put this plug-in through its paces.

nique. First, I pasted an English phrase into Speak It!, a terrific text-to-speech app for iOS ($1.99, future-apps.net). Then I exported the result as an AIFF file and reversed it in Ableton Live.

Even with music, the effect sounded like reversed speech so I took it further. I translated the text into Swedish at **translate.google.com**, pasted that back into Speak It!, and re-rendered it with a Swedish voice. (You can preview the voices at **acapela-group.com**.) The foreign words added the perfect exotic spice.
—DAVID BATTINO, BATMOSPHERE.COM

WHAT'S NEW

*** HOT NEW PRODUCTS FROM NAMM ***

By George Petersen

DAVE SMITH INSTRUMENTS TEMPEST

ANALOG DRUM MACHINE

When master synth designer Dave Smith gets together with legendary drum machine pioneer Roger Linn, a firestorm is likely to be brewing. The result from Dave Smith Instruments (davesmithinstruments.com) is Tempest ($1,999), a hardware analog drum machine with six analog voices, each with two analog and two digital oscillators. Retro and new school in one monster box, Tempest also features 16 pressure- and velocity-sensitive pads, two slide controllers, and onboard effects such as compression, distortion, and beat-synched delay.

MOOG MUSIC MOOG LAP STEEL

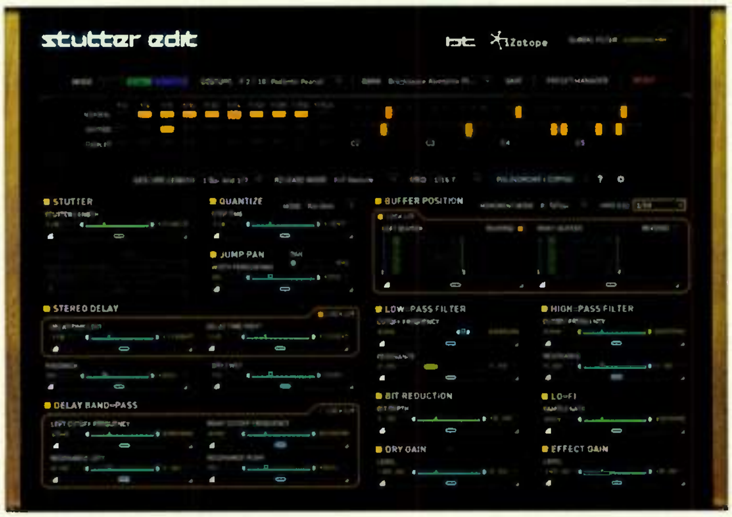
SLIDING INTO THE GROOVE

The award-winning technology from Moog Guitar (moogmusic.com) takes a new bend in the form of the Moog Lap Steel, a 6-string version featuring a shaped body style and raised action that's perfectly suited for the steel player. Offered on a custom basis, the Moog Lap Steel delivers infinite sustain, controlled sustain, and mute modes—greatly expanding the expressiveness of the original instrument—and features an onboard Moog ladder filter that's controllable with a foot pedal. Prices start at $2,895; various wood/finishing options are available.

MISA DIGITAL INSTRUMENTS KITARA DIGITAL GUITAR

NO STRINGS ATTACHED

It's shaped like a guitar, but there are no strings. The Kitara Digital Guitar from Misa Digital (misadigital.com) is played via six switches on each fret of its 24-fret neck, providing all the notes you'd get on a guitar, while the player's other hand "strums" a touchscreen using various articulations to trigger the notes. An onboard synth has 128 preset sounds and a MIDI output. Two models will be offered: a high-density, injection-molded ABS polymer body ($849) and a solid-aluminum style ($2,899); both have 8-inch multitouch screens.



IZOTOPE **STUTTER EDIT**

C-C-C-OOL S-S-SOUNDS

iZotope's (izotope.com) Stutter Edit ($249, Mac/Win, VST/AU/RTAS) is a plug-in developed with BT (Brian Transeau) to emulate his signature stuttering audio effects. It requires a MIDI input from your host to achieve its effects, so it only works with certain platforms, which include Apple Logic, Ableton Live, Pro Tools (7.4+, RTAS only), Cakewalk SONAR, Steinberg Cubase/Nuendo, Image Line FL Studio, Cockos REAPER, and MOTU Digital Performer.

FOCUSRITE **VRM BOX**

STUDIO MONITORING GOES VIRTUAL

Designed for headphone listening, the VRM (Virtual Reference Monitoring) Box from Focusrite (focusrite.com; $99 street) lets users select from 10 pairs of industry-standard near-field and main monitors in an acoustically treated control room to provide various references when listening to their mixes. It's all virtual—of course—and for more variation, convolution-modeled room environments include a living room, bedroom studio, and a pro studio. The VRM Box functions as a high-quality, 24-bit/48kHz USB audio playback interface, and also has a S/PDIF input supporting sample rates up to 192kHz.

ROLAND **GR-55**

GUITAR SYNTHESIZER

Combining PCM synthesis with COSM instrument/amp modeling derived from the VG-99 V-Guitar System, Roland's (rolandus.com) GR-55 offers improved playability, features, and sound quality, including onboard virtual guitars, basses, amps, and synth voices. It can play up to four sound sources at a time: two PCM synth tones, COSM guitar modeling, and normal guitar input. Also featured are more than 900 fully editable PCM sounds, two multi-effects engines, global reverb/chorus/delay effects, an onboard looper, and a built-in USB audio player with foot control. Price with a GK-3 hex pickup is $799 or $699 without the GK-3.

KORG **KRONOS WORKSTATION**

PRODUCTION POWERHOUSE

One of the most talked-about products at NAMM, Kronos (korg.com/kronos) features nine synth engines of various types and can switch sounds seamlessly (or instantaneously) while being played, even from huge 4GB acoustic piano samples, thanks to its Virtual Memory Technology and a fast solid-state disk. Onboard effects, a 16-track sequencer, and 16-track recorder create an all-in-one system. From ethereal textures to complex layered tones, Kronos covers a huge range of sonic territory, as well as classic keyboards, pianos, tonewheel organs, electric pianos, and orchestral, percussion, and pop sounds. The 61-key Kronos-61 is $2,999; the Kronos-73 is $3,499; and the 88-key version is $3,799.

ZOOM R8

MULTITRACK RECORDER

Zoom (zoom.co.jp) took the design of the R24 and reduced the footprint for a portable, powerful music production solution. In addition to 8-track playback and 2-track simultaneous recording (44.1/48kHz at 16/24-bit) using SD memory cards, the R8 ($499) is a digital multitrack recorder, USB audio interface, control surface, and a pad sampler. The R8's built-in drum sounds can be triggered using eight pads and three bank keys to assign sounds to each track and create loops. You can also use the unit's drum machine to create original backing beats, or simply output a metronome for tempo control. The R8 includes a 2GB SD card and supports up to 32GB SDHC cards for a maximum of 100 track hours.

STEINBERG CUBASE 6

CLEARLY ON BASE

The latest version of Steinberg's (steinberg.net) popular cross-platform DAW software, Cubase 6 ($499) and Cubase Artist 6 ($249) offers a bevy of new features, including enhanced workflow features within the Project window and the Track Edit Groups option allowing related events on multiple tracks to be grouped and edited simultaneously. Other features include Lane Track for convenient multitake comping, a redesigned transient and automatic tempo detection, phase-accurate audio quantization, and drum-replacement functions to smooth out any glitches in live recorded drum tracks.

UNIVERSAL AUDIO UAD-2 SATELLITE

FIREWIRE DSP ACCELERATOR

The Universal Audio (uaudio.com) UAD-2 Duo and Quad Satellite family (priced from $899) of DSP accelerator packages (with two or four SHARC processors) put the entire UAD Powered Plug-Ins library within easy reach of any FireWire 800- or 400-equipped computers, with no PCIe card installation required. Compatible with a wide range of modern Intel-based iMacs and MacBook Pros, these let users run larger mixes in Pro Tools, Logic, Cubase, Nuendo, Performer, and more—without taxing the host computer's CPU.

ALESIS STUDIODOCK

I/O FOR IPAD

Apple's iPad has proven itself ideal in all sorts of pro audio/music applications, with a near endless supply of useful apps available. Its main drawback had been a lack of pro connectivity, but the Alesis (alesis.com) StudioDock ($199), an audio interface for your iPad, features two XLR/¼-inch jacks (each with its own gain control and switchable phantom power) and MIDI I/O ports. StudioDock can also connect to a Mac or PC using the USB port to send MIDI back and forth for creative, new applications of the iPad and computer used in tandem.

ARTURIA SPARK

DRUMS 'N CONTROL

Arturia's (arturia.com) latest creation is Spark, a hybrid software drum machine with a hardware controller. Offering analog synthesis, as well as sampled and physical-modeled sounds, the product is designed for creative beat production. Slated to ship in April at $599, Spark also features eight velocity/pressure-sensitive pads, 480 instruments/30 kits (including acoustic drums, modern electronica, and vintage drum machine re-creations), a 16-key/64-step sequencer, looping mode, real-time automation of all parameters, a 16-track mixer, nine onboard effects, and more.

SOUND ADVICE

By Len Sasso

SAMPLERBANKS ELECTROLYZIUM

Electrolyzium ($46.82, download) from sample-library distributor Samplerbanks (samplerbanks.com) is a 1.1GB collection of loops and one-shots from trance-label Anjunabeats producer Márton Levente (aka, Sunny Lax). The material is culled from his work in electro, progressive-house, and trance genres. You'll find bass (76), drum (240), and synth (59) loops in both Acid WAV and REX2 formats. Tempos range from 128 to 134 bpm, and the loops are labeled with key information where appropriate. The one-shots include hundreds of drum and percussion hits with a smattering of sound effects. Beyond that, sampler instruments and their associated multisamples are provided in Native Instruments Kontakt, Apple EXS24, Steinberg HALion, Propellerhead NN-XT, and SFX formats. These include basses, leads, and multisampled percussion. Many of the drum loops are sparse and all are no-kick style, leaving you with ample room to fill them in with the individual hits (see **Web Clip 1**). Electrolyzium is also available in Propellerhead Reason ReFill, Ableton Live Pack, and Apple Loops versions.

TWISTED TOOLS RICHARD DEVINE'S ANALOGUE MICROCOSM

If descriptions like "nonlinear sequential tuned voltage map," and "uses the Cartesian coordinate system to unlock the analog step sequencer from the shackles of linearity" resonate with you, that's all you need to know about Twisted Tools Analogue Microcosm ($39, download). If they leave you a little mystified, check out the video and audio clips at twistedtools.com for a closer look.

Sound designer Richard Devine has unleashed the full force of his extensive analog modular-synthesis studio to create this library of 570 WAV files; no software or DSP plug-ins were used. The samples are spread across five categories: percussion, short effects, long effects, crackles and noise, drones and textures, and bass (bone-rattling sub-bass sounds). The percussion section is perfect for fashioning analog drum kits, whereas the other sections emphasize effects and atmospheres. Some of the longer files (the longest is 43 seconds) have rhythmic content, but most are pulses, whooshes, bursts, or ambiences (see **Web Clip 2**).

Twisted Tools has fashioned 40 kits from these files and formatted them for Native Instruments Battery, Maschine, and Reaktor (for the included MP16 custom drum machine); Apple EXS24; Ableton Live Simpler; and Propellerhead NN-XT. The WAV files are included so you can easily build kits for other sample players. You may start out thinking of this as a sound effects library, but you'll find it useful for much more.

LOOPMASTERS THE FUNKY UNDERGROUND

Producer Darren Payton, creator of Loopmasters' (loopmasters.com) Fidget House titles, branches out to deliver The Funky Underground ($37, download). The 630MB, 24-bit, 44.1kHz collection includes 500 WAV and REX2 loops at 124, 126, and 128 bpm, along with 600 drum and sound effects samples provided as WAV files and in kits formatted for popular samplers such as Apple EXS24, Steinberg HALion, Native Instruments Kontakt, and Propellerhead NN-XT. The material is suitable for all variants of house music but has a decidedly funk flavor. For added flexibility, the 214 drum loops include kick-and-snare, percussion, and tops variations. The 111 bass loops make up the next largest collection. Beyond that, you'll find a good assortment of synth, vocal, and electric piano loops, along with combi loops and sound effects (see **Web Clip 3**).

NATIVE INSTRUMENTS ABBEY ROAD MODERN DRUMS

Native Instruments (native-instruments.com) expands its collection of drum libraries of the decades with Abbey Road Modern Drums ($119, DVD or download). The first three releases in the series feature classic sounds of the '60s, '70s, and '80s. The new collection brings you up to the minute. It comprises a mid-'90s Collector Series kit from Drum Workshop and the mid-'00s Pearl Reference Sparkle Kit, recorded in Abbey Road Studio Two and Studio Three, respectively.

The collection was recorded through the Redd 51 and EMI TG12428 mixing desks, using SSL 95 mic preamps, and modern and vintage mics from Abbey Road's unmatched collection. The analog recordings were then converted to the digital domain with Prism ADA-8 converters. The 17.4GB library (7.4GB compressed) has more than 40,000 samples, with as many as 27 velocity layers for some sounds. In addition to the full kits, you'll find a variety of specialty drums and cymbals. Abbey Road Modern Drums comes with the free Kontakt 4 Player and also runs in the full Kontakt 4. *

By Geary Yelton

From the mid-'80s until at least the early '90s, Yamaha's DX7 was the synth that launched a thousand hits.

Yamaha DX7

In 1983, one synth changed everything

In the 1980s, the synthesizer with the greatest impact was Yamaha's best-selling DX7. It radically altered popular music, heralding an age when electronic instruments saturated the airwaves and digital polysynths became the norm. Yamaha could barely keep up with demand at first, and analog synthesizers were soon considered passé. Demand for older synths dipped so low that you could snap them up for a fraction of their previous prices.

Launched the same year as the MIDI specification, the DX7 was one of the first synths that let you control one instrument by playing another. The DX7 relied on frequency modulation (FM) synthesis, a technique that Yamaha began licensing from Stanford University professor John Chowning in 1974. FM applies simple waveforms (modulators) to modulate the frequency of other waveforms (carriers), creating complex waveforms by generating sidebands. Instead of oscillators, the DX7 had six operators—sine-wave generators, each paired with its own envelope generator—arranged in 32 configurations called algorithms. The resulting sounds varied from pure tones to rich timbres that simulated acoustic instruments more realistically than previous synths. Keyboard velocity and aftertouch had enormous effect on dynamic spectra, making acoustic simulations more expressive than had ever been possible.

CHALLENGE AND RESPONSE

Although you could program the DX7 entirely from its front panel, the process was far from intuitive. The panel furnished an unlit LCD, a pair of sliders, and 42 membrane switches that made 168 parameters available. None of the familiar building blocks of subtractive synthesis were evident, not even a filter. Only a fraction of owners understood exactly how FM synthesis worked, and hardly any understood how to program their instruments beyond the basics. But the DX7's challenging user interface didn't hurt its popularity, and it became one of the first synths you could program on your computer using editor/librarian software.

> Analog synths were soon considered passé.

With room for 32 internal patches and data cartridges each containing 64 more, the DX7 supplied a wealth of new and different sounds. It so authentically replicated the Fender Rhodes piano, and with greater flexibility, that sales figures for real Rhodes pianos plummeted. Synthesized mallet percussion instruments such as marimbas and vibes were suddenly in ample supply, and electronic orchestral strings gained a rosin-in-the-bow realism that wasn't possible with analog synthesis. The DX7 was also one of the first synths that worked with an optional breath controller, making detailed brass and woodwind sounds all the more realistic.

The DX7 had limitations—it was monotimbral and had a monaural audio output—but future Yamaha synths such as the DX7II and TX802 improved on the original. Today, FM synthesizers have all but disappeared, replaced by the much more capable Native Instruments FM8 software and by even more realistic forms of digital synthesis. *

Former EM senior editor Geary Yelton resides in Asheville, N.C.

By Sarah Benzuly

Filippo Beck Peccoz got a bit of sound effects help from his dog, Link.

Crashing Into the Indie Game Scene

Filippo Beck Peccoz creates his own blips and bloops

It wasn't an amazing film soundtrack that got him interested in composing. Nor was it a hit TV theme song or commercial ditty. But at the age of 6, Filippo Beck Peccoz made up variations on the *Mega Man 2* Nintendo game theme by humming them into a tape recorder while he was playing the game. His love for music for games continued at Berklee College of Music (Class of Summer 2009), where he helped create the "Video Game Music Club @ Berklee" where he promoted a dedicated game audio curriculum that was later adopted by the school. Today, in his native Germany, he continues to focus on game audio. "I just love the unique challenges, both compositional and technical, that this discipline is known for," he explains. "You have to think about the unpredictability of a player's actions and how you, as a composer, can construct the best possible system to make the music 'flow' despite of this. Game music has its own voice, and we definitively should let people hear it!"

You've also done composing for art installations. What is your attack for this type of media?

It depends on the artist. For now, I've mainly done sound designy things—like setting up speakers in weird places and jostling waves of sound around the room—to convert the artist's vision into a physical experience. Or, connecting objects the visitor can interact with to sound events.

One installation revolved around the concept of dust: how these tiny elements settle everywhere and change the objects around us in time. The artist brought in an old (and, of course, dusty!) detuned piano, and we layered her live playing with old recordings of her childhood piano recitals, swirling around the darkened room, creating an interesting division between past and present. It was definitively a very conceptual and less-technical project. When live sound interaction is asked for, I tend to use [Cycling '74] Max/MSP as it's fantastically flexible and powerful.

Where are you working out of?

I just moved into a new space, sharing a bigger room with a young developer team from Munich, the Bit Barons. I use Digital Performer mostly; I just love its flexibility in terms of tempo changes, rubato playing, creating tempo maps and such. Also, Version 7 is very stable and reliable on my computer, which is a blessing working on any project. I use a MOTU 828 mk3 interface and a couple of Genelec 8030s, a great investment.

I'm a bit of a Korg fetishist. I have a Kaoss pad, a Kaossilator, and an Electribe hooked up and ready to go at all times and in reach. I love the idea of having a bit of outboard gear but in a compact setting: playing a note on the MIDI keyboard with the right hand and mangling the sound on the Kaoss pad with the left, and then triggering a guitar sample I made with the ribbon controller on the Electribe—the possibilities are endless.

I have two Shure SM81s, which I love for guitars and percussion recording/sampling. Then the ever-present SM57 and SM58. Plug-ins I use are the Vienna Special Edition, the amazing LA Scoring Strings, a couple of Waves plug-ins I still have from my Berklee time, EastWest Silk, and Native Instruments Komplete. IZotope's Ozone has been a favorite of mine for some time, too.

Some say that you're part of the "indie game scene"—those titles produced without a videogame publisher. What exactly does that mean to you?

I've been fortunate to find a pretty large local indie game community in Munich, which also wants to reach out to the bigger groups of people in the U.S. and the rest of Europe. There's lots of opportunities and drive to create great games. The Game Jam is a monthly jam that takes place in Munich. There's also interest in moving the art of game audio forward. Recently, Bit Barons' Alex Zacherl and myself organized a Game Audio Forum, an event geared toward bringing audio people and game developers together, which was a success attendance- and feedback-wise. My personal goal is to make Munich and Bavaria another center for cool indie games.

You recently wrapped up sound for the *Astroslugs* game. What was your process?

First, it was important to determine the general mood of the game. We wanted a bit of an orchestral touch, coupled with a lighthearted beat and catchy melody. I also wanted to make the really pretty artwork in the World selection scene of the game stand out: You are presented a picture of the entire game world, and there are different regions (the classic ice, woods, desert stages) that the player can zoom in by clicking on them. I created variations of the main theme for each zone so that by entering, the player is hearing an appropriate music for each zone.

We did this by starting six music loops at exactly the same time in the game engine, but turning up the volume just on one of those loops. The engine then crossfades between the tracks as the player enters the different zones. The result is a super-smooth, musically coherent transition.

Because the core gameplay revolves around pretty difficult puzzle mechanics, we needed an in-game music that doesn't get annoying even after a long time and doesn't sacrifice the *Astroslugs* vibe to be appropriate for this gameplay. We came up with a little music system in which groups of instruments play semi-randomly over a fixed, kind of "rubato" pad loop. This way, the music never repeats exactly the same way, lending itself to longer sessions inside the same level.

I also created the sound effects for the game and want to thank my dog, Link, for the great chomping/barking sounds that made it into the game after some modifications! It was a lot of fun coming up with what was needed, and I really focused on not using library sounds if possible. That's another reason why I almost always run around with a little portable recorder.

As an added bonus, we decided to release a soundtrack album of a different kind for *Astroslugs*. I gathered a group of musician friends and colleagues, gave them the main melody and bounced the single tracks out of [Digital Performer], and off they went to some arrangement/remix fun. What came out of this project is a nice concoction of different genres and interpretations of the original theme, from progressive trance to solo piano, all the way to metal.

So you do a lot of sampling.

I try to as much as I can. I've always been fascinated by how one can capture sound, contain it onto a tape, CD, or whatever medium. It's been extremely beneficial to me to learn how to use Kontakt—making my own scripts, my own loops (acoustic and electronic), just messing around making tons of little instruments out of everything. When I noodle around on a guitar, I keep a recording. Not just for song-creation/idea-keeping purposes, but also for placing certain sounds as the basis of new Kontakt instruments. Another favorite of mine is the gritty Yamaha FM Soundchip found in the Sega Genesis console. Call me a bit nostalgic, but I think there's nothing like it around!

What's next for you?

This year started out with a series of really cool indie game projects that will keep me busy up until around May. I'm also looking into the sample libraries market to see what has not been covered yet. Also, I'm preparing a game music-related live show, but that's still in the brainstorming phase. Stay tuned and keep gaming! ✱

By Jason Feehan and Randy Chertkow


Feehan


Chertkow

The Right Image

A musician's world is not just about audio, it's also visual. The Web is full of images, all of which shape the way you appear to potential fans. But images are copyrighted and musicians need to know the rules, especially when using them on their albums. Services that manufacture discs or provide digital distribution require you to sign a form that states you have the rights for all of the intellectual property you're using—not just the music, but also the art.

It's at this stage that a lot of musicians realize they didn't secure all the rights for their images. The time to sort this out is before you hire a photographer or someone to create your graphics. It also helps to know a few legal alternatives to acquiring images royalty-free.

Just like music is copyrighted the moment you record it, photos are instantly copyrighted the second a photographer snaps the shutter. The same rule applies with graphic artists and the artwork they create. What may not necessarily be clear is that even if you're paying the photographer or graphic artist, he or she owns the copyright to the images unless there's an agreement that states otherwise. This means you have to get permission each time you use a copyrighted image you don't control, usually requiring a fee for each use. If you end up printing 1,000 copies of your album, that can get costly.

To avoid this, get it in writing that the engagement is a "work-for-hire" when employing a photographer or graphic artist. A work-for-hire means that you will own the copyright to those images. Some photographers might balk at doing a work-for-hire (and may charge a higher rate to give you this option), but the time to work this out is beforehand, not when you're about to make an album and learn the perfect cover shot is controlled by the photographer. Even though it's a work-for-hire, you should still give attribution to the photographer for the images wherever you use them. This is a good bargaining chip to offer when negotiating the agreement.

If you still need to search the Web for that perfect image, there are options to explore. Creative Commons has a set of licenses that many graphic artists and photographers use for their work that allow other people the right to use it for free, as long as you give attribution to the owner, which is easy to do and something the owners deserve. To find images licensed in this way, check out search.creativecommons.org and make sure the license search allows for "commercial use" as it's likely you'll be selling your album. Even if you find images that are licensed only for noncommercial use, you can still contact the owner and work out a deal. Some owners are happy to have their work used on an album and may let you use it for attribution or a nominal fee.

A second option is to use royalty-free images from services such as ClipArt.com or iStockPhoto.com. Some of these services require an annual membership fee for access to their entire catalog, while others allow you to download any image for a one-time fee. Either way, once paid, you can use the royalty-free image again and again and keep whatever you make from it.

Lastly, you can use images in the public domain (anything created by the U.S. government, such as any of NASA's photos). Check out Wikipedia's public domain image resource (wikipedia.org/wiki/Wikipedia:Public_domain_image_resources) page for additional resources.

No matter what you do with your images, knowing these factors ahead of time will help you better engage graphic artists and photographers. If you work an arrangement ahead of time, the rights—and the profits—for both your music and all of your photos will be all yours. *

Randy Chertkow and Jason Feehan are the authors of The Indie Band Survival Guide *(IndieGuide.com).*

Hear your mix, through 15 different speakers, in different environments

Professional Studio

Living Room

Bedroom

. . . .just using headphones.

VRM Box™ is a compact mix monitoring / interface solution worthy of your studio headphones. With patent-pending VRM technology from Focusrite, now you can mix anywhere, anytime.

Mix in the middle of the night without waking anyone. Mix as you travel from the gig to the studio. Mix from the sweet-spot in acoustically perfect environments. Jump from room to room, switching from speaker to speaker, without ever burning a CD or leaving your seat.

Now you really can mix on headphones.

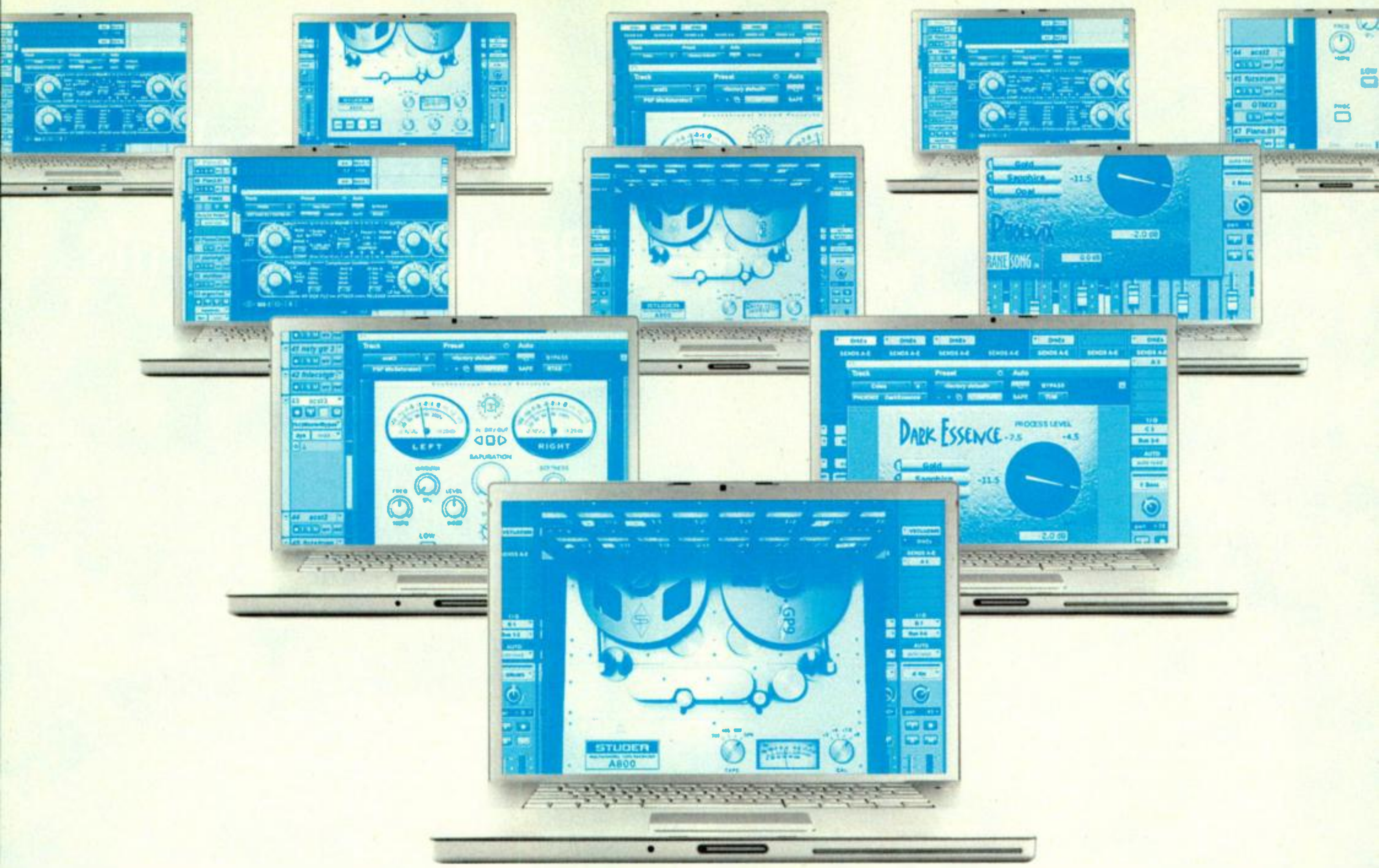
Headphone output quality exceeding interfaces five times its price

Need a recording interface? Check out the Saffire PRO 24 DSP, featuring VRM.

www.focusrite.com/VRM Box

Focusrite®

SOUND IS EVERYTHING



Warm It Up

Eight saturation plug-ins that bring an analog taste to your digital plate

By Eli Crews

Whatever they call it—warmth, dirt, grit—what people long to hear in digital recordings is nonlinear harmonic distortion, also called *saturation*. In the analog world, the main sources of saturation are the three "T"s—tape, tubes, and transformers—and there are a growing number of DAW plug-ins that offer quite convincing simulations of these. I extensively tested eight such plug-ins for this story. Space and logistics limited my full attention to those, but I've included capsule descriptions of seven others in the sidebar, "More Saturation Choices."

Three-T simulations fall into two categories: those that subtly alter the sonic character of your mixing environment and those that cross the line and deliver actual distortion. Although there is certainly some overlap, of the eight plug-ins covered here, HEAT, Phoenix, and the Studer A800 fall into the former category, and the rest alter your sounds more heavily.

AVID HEAT

Avid (avid.com) HEAT ($495) is available only for Pro Tools HD and comes bundled with all-new HD systems. It is more of an add-on than a plug-in, and works quite differently from all the other processing tools in this article. For one thing, you don't add it as a track insert. When you activate HEAT for a given session, it becomes immediately active on all audio tracks. To view and alter HEAT's bypass status for a given track, the HEAT controls must be visible in the Mix window. You can bypass HEAT for each track individually, or you can activate Pre to place the HEAT effect before other plug-in processing on a track.

A meter for each track glows orange, becoming brighter (based on the signal's intensity) as more HEAT processing is being applied. You'll find a master meter, global Drive and Tone controls, and a master bypass button in the master HEAT pane. Drive determines the amount of HEAT processing applied to all non-bypassed audio tracks; turning it clockwise adds more tube-like even harmonics, whereas turning it counterclockwise adds those along with more tape-like odd harmonics. The Tone knob is different from a normal EQ control because it interacts dynamically with the Drive setting and with signal energy. In general, counterclockwise is brighter and clockwise is darker.

If you create a new stereo audio track to print your final mix (as opposed to using the Bounce to Disk option), remember to bypass HEAT on this track. Otherwise, you will be adding HEAT to your mixdown and to each of your individual tracks—in effect, doubling the HEAT! The extra effect won't be printed to your actual mix, so when you export the mix to a new stereo file, it will sound different from the one in your session. This is the only pitfall I've found; overall, HEAT is an extraordinarily user-friendly way to dial in effective, subtle yet powerful shifts in a mix's harmonic content (see **Web Clip 1**).

AVID REEL TAPE SATURATION

Reel Tape Saturation ($295) is also available only for Pro Tools, but it does come in TDM and RTAS/AudioSuite versions so non-HD Pro Tools users can take advantage of what it and the other two plug-ins, Delay and Flanger, in the Reel Tape Suite ($495) offer.

The controls of Reel Tape Saturation are labeled Drive, Tape Speed, Noise, Bias, Cal Adjust, and Output (see **Fig. 1**). There are also three machine types (U.S., Swiss, and Lo-Fi) and two formulas (Classic and Modern). Having all of these options gives you a wide palette of tape saturation, from subtle harmonic additions to downright distortion. This plug-in is adept at adding midrange grit to vocals and guitars, as well as giving a bass guitar a little extra sizzle to poke through the mix (see **Web Clip 2**). I found the noise control handy for adding a slight amount of hiss to a vocal track that had been recorded on tape. That let me edit out in-between noises without leaving a too-silent gap.

CRANE SONG PHOENIX

The Phoenix plug-in ($450, TDM only) was developed by renowned gear designer and Crane Song (cranesong.com) founder Dave Hill, who has a strong background in analog electronics and tape-machine design and service. Phoenix is a suite of five plug-ins esoterically named Luminescent, Iridescent, Radiant, Luster, and Dark Essence (see **Fig. 2**). Each has the same simple set of controls: Input Trim and Process Level knobs, along with three color changes creatively titled Gold (neutral), Sapphire (brighter), and Opal (warmer).

It seems odd to provide five separate components rather than integrating them within one interface, but the good news is that your settings automatically carry over if you swap in a different component. Each has its own harmonic distortion profile. The profiles vary from transparent to somewhat aggressive, although none of them deliver heavy distortion unless your signal is really loud to begin with or you put an additional gain stage before Phoenix.

FIG. 1: Avid Reel Tape Saturation offers most of the control you'd find on a real tape machine.

FIG. 2: Crane Song Phoenix's interface may need a facelift, but the plug-in sounds like gold.

FIG. 3: PSP MixSaturator2 is user-friendly and euphonically rewarding.

FIG. 4: Sound Toys Decapitator captures the analog feel, even in its interface, making it as fun to operate as it is to listen to.

Phoenix does a great job of giving your entire mix a lift, especially if you put an instance on every channel. It's like removing a thin sheet of gauze from your audio (see **Web Clip 3**).

MCDSP ANALOG CHANNEL

Version 5 of McDSP's (mcdsp.com) Analog Channel ($449, TDM; $249, AU/RTAS) has two variations: AC101 and AC202. The AC101 was designed to emulate the saturation effect of analog consoles' electronics. Although it is functionally reminiscent of a standard compressor—having attack, release, and curve controls, as well as a gain-reduction meter—it differs from a standard compressor in that it is not threshold-based; it is always affecting the audio.

The AC202 is a tape machine emulator. It has controls for changing the frequency of the LF roll-off, the amount of head bump, the head type, bias, playback speed, IEC EQ setting, and tape formulation. I found many useful settings on the AC101 and AC202, often landing on that magic mojo that analog electronics and tape provide (see **Web Clip 4**).

Using Analog Channel on spiky instruments—such as a somewhat unruly Wurlitzer electric piano—could give me a distortion I didn't like. But when set right, it delivers a creamy overdrive that tames peaks while still allowing dynamic performance. The Analog Channel's two variations were the only plug-ins on which I had to carefully watch my output level. There is an Auto function for the output level, but I generally prefer manual control. In short, Analog Channel allows much versatility so extra care must be taken to achieve proper gain-staging when setting your parameters.

PSP AUDIOWARE PSP MIXSATURATOR2

PSP Audioware (pspaudioware.com) is a Polish company that is best known for its excellent VintageWarmer2 compression/saturation plug-in (see sidebar). PSP MixSaturator2 takes harmonic manipulation to a different level. It is available only as part of the shockingly affordable PSP MixPack2 bundle ($199, AU/RTAS/VST), which includes other handy mixing tools.

PSP MixSaturator2 has three sections: one for altering LF harmonic content (similar to a tape machine's head bump in the low end), one for emulating the high-end compression that occurs at high levels on analog tape, and one for setting the amount and characteristics of nonlinear saturation (see **Fig. 3**). You get three tube (or Valve) shapes, three tape shapes, and one digital shape, which simulates hard clipping. Changing the shape significantly alters the signal level, which can make comparing settings a bit difficult but each shape has a distinctive sound, especially at higher saturation levels.

If you're driving PSP MixSaturator2 hard, you can engage the OutSat button, which puts a brickwall limiter at the end of the signal path to prevent digital overs. You'll find a Mix knob to dial in as much of the effect as you want in a phase-coherent parallel fashion. I have to say that each of the plug-ins in MixPack2 sometimes acted a bit erratically in Pro Tools, but when it was functioning properly, the PSP MixSaturator2 sounded fantastic, especially considering the price (see **Web Clip 5**). PSP says it is aware of the issues and is working to resolve them in an update.

FIG. 5: The tape reels of Universal Audio Studer A800 actually spin. Although it's a real client-pleaser, you can click the IPS knob to turn it off to keep from getting dizzy.

SOUNDTOYS DECAPITATOR

SoundToys (soundtoys.com) has built analog-type saturation into the lion's share of its effects, such as PhaseMistress and EchoBoy (see my review in the November 2007 issue of *EM*, available at emusician.com), but Decapitator ($349, TDM; $179, AU/RTAS/VST, see **Fig. 4**) is the company's first plug-in with saturation as the star. The interface is straightforward and streamlined, with only six knobs: Drive, Low Cut, Tone, High Cut, Mix, and Output. Of its three switches, Thump adds a boost right at the corner frequency of the low-cut filter; Steep alters the shape of the high-cut filter; and Auto automatically adjusts the output control when you set the drive level.

You get five choices of saturation style, modeled on hardware by Ampex, Chandler Limited/EMI, Neve, and Thermionic Culture. The Punish button adds a whopping 20dB of gain to the plug-in's input stage. It goes without saying that you can produce sounds with this effect that will give you a buzz cut—if you duck in time. Decapitator excels at producing extreme (but also extremely pleasant) distortion. It's also useful at lower drive levels to add a touch of edge to the midrange. That brings out elements that need to assert themselves more in the mix, such as electric guitar, snare drum, vocals, and piano (see **Web Clip 6**).

UNIVERSAL AUDIO FATSO JR., SR.

Emprical Labs EL7 Fatso Jr. was one of the first hardware boxes devoted to emulating the effects of analog tape. Universal Audio (uaudio.com) offers two variations on the EL7: Fatso Jr. and Fatso Sr. ($299, AU/RTAS/VST; requires UAD-2 hardware). The Jr. and Sr. variations share the same basic controls: input, Compression Mode selector, warmth amount, Transformer (Tranny) enable, and output (see **Fig. 4**). Fatso Sr. adds four controls for the compression circuit (threshold, attack, release, and a sidechain highpass filter) and a level control for the Tranny.

Whether or not the compressor is engaged, Fatso adds second- and third-order harmonics that increase as your input level goes up. This emulates how tape and tubes react to dynamics. The Warmth circuit simulates the softening effect that tape has on high frequencies, and the Tranny control adds extra harmonics to low frequencies to make them apparently louder without boosting the headroom-hungry fundamentals. The four Fatso components are interrelated, so a change to the compressor settings affects the sound of the transformer, and so on. The Fatso is highly effective at sculpting the harmonic content of just about

!

ALTERNATIVE SATURATION PLUG-INS

Here are short descriptions of seven saturation plug-ins not covered in the main article but worthy of your consideration.

Antares Warm

Antares (antares.com) retooled its popular Tube plug-in for its Evo suite, which is designed for vocal processing. Warm (part of AVOX Evo bundle; $399, AU/RTAS/VST) has a very simple control set, emulates two different tubes, and has a handy mode that isn't transient-dependent.

Massey TapeHead

Straightforward controls and an extremely low price set TapeHead ($69, TDM/RTAS)—and Massey's (masseyplugins.com) other stellar plug-ins—apart. With only a Drive knob, a 3-position Tone switch, and an Output control, it's extremely easy to dial in thick, rich tones.

Metric Halo Character

Because the effect runs on the DSP of the interface itself, Metric Halo's (mhlabs.com) Character (included with all current company interfaces) can be applied during either tracking or mixing, and it places zero strain on your host CPU. There are nearly 20 different effects, including various tubes, transformers, and mic preamps, giving a tremendously wide variety of color.

Nomad Factory Magnetic

This plug-in appeared during the writing of this article, and it is an affordable way to get into the tape-machine-emulation game. Magnetic ($129, AU/RTAS/VST) from Nomad Factory (nomadfactory.com) has many features and provides great versatility. I found that it delivered some of the most convincingly vintage sounds of the entire lot.

PSP Audioware VintageWarmer2

The PSP VintageWarmer 2 ($149, AU/RTAS/VST) compression-saturation plug-in has been around for almost a decade. VintageWarmer was the first plug-in I ever heard that made me think computers could be good tools for mixing records, and it still gets daily use in my studio.

Slate Digital Virtual Console Collection

This is a simple but extremely effective duo of plug-ins. Slate Digital's (slatedigital.com) Virtual Console Collection ($250, AU/RTAS/VST; iLok2 required) is in beta-testing as of this writing, but should be available by the time you read this. One plug-in models a channel strip and the other models the master bus section of four different legendary consoles. These emulations sound extremely good and can be grouped across multiple channels.

Virsyn VTAPE Suite

I found the sound of the Virsyn (virsyn.de) VTAPE Suite ($219, AU/RTAS/VST; Syncrosoft dongle required) plug-ins—Saturator, Delay, and Flanger—very convincing. Although they gave me a number of problems on my Pro Tools system, they worked well in Apple Logic and BIAS Peak. The interface leaves something to be desired, but their excellent sound is more important. When you push Saturator hard, the distortion remains truly analog-sounding.

—Eli Crews

anything you throw at it (see **Web Clip 7**).

UNIVERSAL AUDIO STUDER A800

The newest addition to the UAD-2 DSP card family, the Studer A800 ($349, AU/RTAS/VST; requires UAD-2 hardware) painstakingly recreates the sound and workings of a single tape machine: the Studer A800 (see **Fig. 5**). You get a selection of four tape types and have control over many other virtual tape-deck functions, including inches-per-second, reference level, hum/hiss amount, emphasis EQ curve, and input and output levels. Notably, you can monitor either directly through the virtual tape machine's electronics or from either of the heads (Sync or Repro), each of which has its own bias and EQ settings.

Perhaps the most unusual feature of the A800 is its ability to gang the controls. The Gang button, when enabled on any instance of the A800 in a given session, completely links the controls for all instances of the plug-in in that session. (Regrettably, the parameters become unlinked when you scroll through presets, but this will be addressed in a future release.) This can be extremely handy for setting up the A800 to work the same way as HEAT, where global changes affect all of your tracks. With the A800, once your global settings are made, you can switch Gang off to fine-tune the saturation type and amount for each channel (see **Web Clip 8**). Keep in mind that using an A800 on each channel comes at a processing cost. For example, 18 tracks of A800 processing used 60 percent of my UAD Quad card running at 96kHz.

GETTING WARMER

I enjoyed using each of these plug-ins. My mixes during the past few months have benefited from the selective use of each, even though a lot of the material was tracked to tape in the first place. In terms of ease of use for the mix's overall sound, HEAT is clearly the standout. The A800, with its ability to gang controls, comes in a close second and is more flexible. Phoenix has the most stripped-down controls, but with five variations and three tone settings, you actually get 15 ways to liven your sounds.

Analog Channel and Fatso add some standard compressor functions, giving you more ways to tame the dynamics while adding oodles of analog character. Reel Tape Saturation excels when driven hard and is straightforward for anyone familiar with tape machines. Priced at less than $200, Decapitator and PSP MixSaturator2 share honors for the most bang for your buck. They are the only two available in all three major formats and don't require special hardware.

Owing to their diversity, a simple comparison of these plug-ins is not easy. However, I did run a section of a song through a few presets on each of the eight plug-ins and you can listen to the results online (see **Web Clips 9 through 16**). *

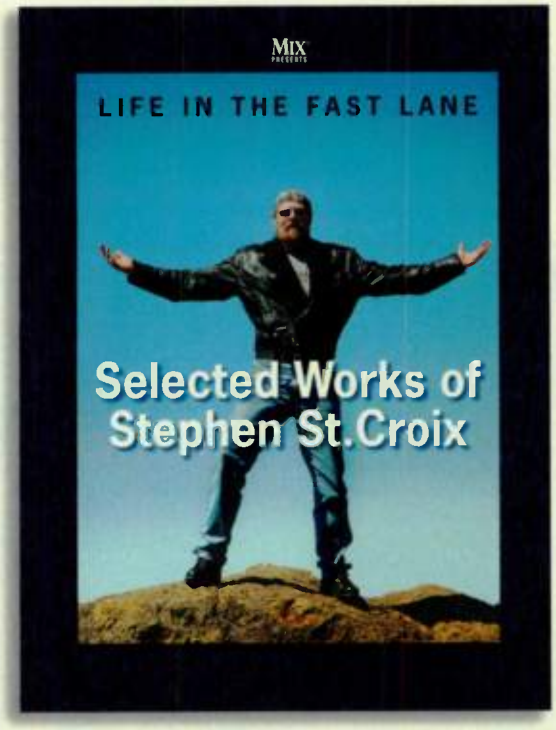
Eli Crews operates New, Improved Recording, a studio in Oakland, Calif.

NEW FROM MIXBOOKS

Life in the Fast Lane

Selected Works of Stephen St.Croix

Stephen St.Croix inspired, provoked and educated *Mix* magazine's readers for 18 years in his one-of-a-kind column, "The Fast Lane." As an inventor, musician and engineer, St.Croix offered his audience a wealth of



knowledge and vision, as well as a Harley-riding rock-star attitude. Now, two years after his death, the editors of *Mix* have selected the best of St.Croix's columns, presented with never-before-seen photos, notes and drawings from his personal files. This book takes "The Fast Lane" beyond the pages of *Mix* and lends new insight into the life and mind of Stephen St.Croix.

To order your copy, go to www.mixbooks.com

MIXBOOKS

Nothin' Fancy

How Cake recorded their new hit album in a solar-powered home studio with some surprisingly modest gear

By Mike Levine

Cake has one of the more distinctive and original sounds in the rock world. Typically categorized as "alternative" by the music press, the band's music is easily recognizable for its catchy melodies, solo trumpet lines, sporadic but memorable synth parts, intelligent and entertaining lyrics, melodic guitar riffs, and occasional vibroslap hits. The recently released *Showroom of Compassion* is Cake's first new studio album since *Pressure Chief* (2004). This is Cake's sixth album of new material (the band also released a B-sides album in 2007), most of which were recorded in commercial facilities. But starting with *Pressure Chief*, Cake has eschewed the commercial studio for a Pro Tools LE-based home setup in Sacramento, Calif.

In the years between the release of *Pressure Chief* and the new album, a lot more has changed for the band. They left their label, switched to solar power for their studio, and stepped up their game as recordists. Despite a very modest gear setup, *Showroom of Compassion* is sonically more akin to their older albums than to the more electronic-sounding *Pressure Chief*.

Cake has always had a very loyal following, but the new album has gained traction surprisingly quickly. It debuted at Number 1 on the *Billboard 200* chart and has received critical acclaim. The band currently comprises lead singer/ guitarist John McCrea, lead guitarist Xan McCurdy, trumpeter/keyboardist Vince DiFiore, bassist Gabriel Nelson, and drummer Paulo Baldi. I spoke to McCrea recently about the band and the production of the album.

Cake is from left: Vincent Di Fiore, Paulo Baldi, John McCrea, Xan McCurdy, and Gabriel Nelson
Photo by: Robert McKnight

We did a story on you guys back when *Pressure Chief* came out in 2004. I can't believe that was your last studio album, that's crazy.

We had to take a little time to re-evaluate what we do and how we wanted to do it, and we rebuilt our studio and made it solar. We also reconfigured our business model. We got off of a major-label deal, and we just thought it was more realistic for today's climate, where increasingly the thing we're selling, recorded music, is, for a lot of people, free. So it's weird to have so much overhead, so many expensive dinners, and the fanciness of a record label. I'm not sure if it's sustainable in today's scenario.

So you're independently releasing your music now?

Through a distributor, ILG. It's our own label, and we're pretty much doing it ourselves, and it's all on our schedule, which is great. One of the things that frustrated us and a lot of other bands is having to work so hard on recording your album and then handing it over to somebody and being really at their mercy, being at the mercy of things that we're not related to at all, to their business model, or their hiring and firing of employees—just random corporate hi-jinks. The stupidity of corporate culture sometimes affects something that you've worked on your whole life. You put everything you have into an album, and then as you're releasing the album, everyone is getting fired because there's a corporate takeover going on.

That must be really frustrating. So now that you're doing it yourself, you guys get to keep everything.

Well, we get to keep most of it.

Minus overhead, I guess.

Yeah, absolutely. We have to pay for everything. We have to write a check for everything that happens. I think people are more careful with their own resources. I've certainly seen bigger record companies being not very careful.

Right, and then they charge it to you. Having your own studio is obviously a way to control costs.

It was definitely another step in the right direction. We had decided that we needed to have our own studio back then when we were still on the label deal. And then, I think the next step was for us to get off the label. So who knows what's next.

Showroom of Compassion is the band's second home-recorded album.

Talk about the changes in your studio since *Pressure Chief*.

We really haven't done a lot. We're still using Pro Tools. We brought in an old acoustic piano. I decided that something that I had resisted for years was piano because I thought it sounded too classy, and I didn't want us to sound classy. But I found this old junker up in Portland and I brought it down. Actually, I feel pretty good about it—about the piano on the album—and probably will have more of it in the future.

So on the album it is a miked acoustic piano?

Yeah. I just sort of stuck the mic in there and found the right place, and kind of a little percussive sound.

What kind of a mic did you use on it?

It's a 58.

Really?

I know, it's crazy. I wanted kind of a crappy sound. I didn't need all of the richness of the piano. That's really why I had avoided piano for so many years—I didn't want it to use up all that space, that sonic space. So much of the time, I think I had discussed this with you in the past, so much time is spent sometimes setting up mics for instruments, like drums especially, you end up carving out all the work you've done and you never needed it. And maybe you didn't know in advance that you didn't need it, or maybe there's just an assumption that you should always just get everything you possibly can, every frequency that exists, put it in there in case you need it. But so many times we didn't need it and it was just a waste. With the piano, we just realized we're not going for a super-rich sound. With my guitar, we're definitely not going for a rich, generous kind of sound. We really don't have the room for it when you factor in all the other instruments.

Do you record with that little acoustic you use onstage?

Yeah, it's a Goya [nylon string] with a Barcus-Berry inside it. It's put through the practice amp that Fender would give people—they'd give it for free when you bought a Fender Telecaster or Stratocaster; I think they still sell them for $70. The reason for that is not because we're trying to be cheap or use toy instruments, it's that all the other really nicer amps would create a lot of feedback with an acoustic guitar. So I thought, how about just using an amp that isn't capable of reproducing those feedback sounds. So that's what worked. So we used that same amp-and-guitar setup for most of the time when I play.

You guys have a pretty iconic sound. Sonically, this album reminds me of your old albums. The guitars are big and fat. The drums are mostly live, right?

Yes, almost 90 percent.

On *Pressure Chief*, there were more programmed drums mixed in.

Yeah, which really upset a lot of people. I think in some ways, it was our

> I wanted kind of a crappy sound. I didn't need all of the richness of the piano.

responding to the way we felt. It's hard to explain it. It seemed like things were increasingly mechanized in our culture and life experiences, and it seemed appropriate to include those kinds of sounds. So much time is spent on a cell phone. We wanted to sound like cell phones existed. We weren't trying to re-create what it sounded like to be in a garage band in 1968.

Miking drums is definitely not an easy thing.

Oh, I know, and really, we're not that great at it, but what we're getting is what we want. And it fits in geometrically with the other shapes and sounds within the recording.

Do you recall what you did, roughly how many mics you used?

We miked the kick. We used basically a kick, a snare, and an overhead [mic].

Just a single overhead?

Yeah, a single overhead. Crazy. If you listen to the mix, the drums are so diminished in a lot of ways; they're not out front. And texturally, they're pretty far back. The overhead doubled as a hi-hat mic. It was on the right and the snare was over on the other side, and then there was a kick. But if you think about it, that's more than what people did back in the 1940s. People enjoyed music back then. A lot of the recordings back then I like better than some of the recordings now. I think the assumption that you automatically need more—that more is better—is a flawed assumption.

You've done two self-recorded albums and four before that in commercial studios. When you compare the processes, aside from the money thing, do you like working at home more? Does it give you guys more of a relaxed feel?

It's a two-edged sword. There's something to be said for the adrenaline of feeling like this is it, we've got to make this work. We have a week to figure this whole thing out, and sometimes it can really work, especially if you're on crystal meth or something. You can really push yourself. No, we don't do crystal meth. What I'm saying is that you can rise to the occasion of the conventional, traditional studio pressure. Knowing this is costing thousands of dollars, I better be completely prepared.

Right.

And the other side of it is, and don't get me wrong, it still costs us money to record, we still have to pay everybody to be in the studio, and we did have an engineer help us mix and master, so it's not like it's free, but it's certainly not as tense as it would be if we were paying $300 an hour. The downside of that, of not having that pressure, it's easier to make mistakes. So that's a good thing and a bad thing. Sometimes it's great to feel like you can afford to make a mistake, so a lot of times that mistake won't be a mistake, it'll be great. At the same time, sometimes it's just going to be a mistake. The other thing—it's not a technical thing, but it's certainly about our process—this album was a lot more of a democratic process, which, as we all know, is an inefficient device.

Talk about your double-edged swords.

Exactly. That's a lot of the reason why we had to throw songs out. We couldn't come to consensus on things. Eventually, maybe we'll figure those songs out for another album. I guess what I'm saying is that we chose to do it more democratically because, number one, people can get more sense of investment in the process and in the end product when they feel like they are really part of it. I think we got that with this album. What we didn't get is a really fast, efficient process.


The band in their studio; McCrea is at right on the phone.

So it took two-and-a-half years?

Yes, sort of. One of the other things is we don't use an outside producer. We produce ourselves. So that also, whether or not we had democracy, would be making things take a long time because you get really invested in everything you do in the studio. It's very subjective. You dig down deep into whatever it is you're doing. You spend five hours getting the guitar tone just right, or changing the guitar part, until you're able to stand back far enough from that guitar part and realize that it sucks and the song sounds better without it. That's going to take you a good week away from that guitar part, after having put so much energy into it. The great thing about having an outside producer is an outside producer doesn't have as much sense of investment in every little detail and can be more honest and objective about what works and what doesn't. So doing it ourselves, it's a tall order and it takes a long time.

This studio is in one of your houses?

We've got this old junker house in south Sacramento, and we kind of gutted it and made it into a studio back during *Pressure Chief*, when we were getting ready to record *Pressure Chief*. We put carpet all over the place and made things dead and made things live and made different rooms for different things. With this album, we made it 100 percent solar so we can say that this album was recorded with 100 percent solar energy, which makes a lot of sense in California. I don't know where you're calling from.

I'm on the East Coast.

There, I think it actually makes more sense than people realize. We have toured in Germany a lot, and it's a kind of gloomy place. It's not a sunny, solar-energy kind of place you'd think, but it's actually the number one producer of solar electricity. It's because the government provided a lot of incentives there. They basically forced the market to purchase solar electricity, so everybody is motivated to put solar panels on their businesses and their homes. Realizing that Germany is number one, and realizing that our studio is in California, it just seemed wrong not to use solar electricity when there's just such an abundance of it. It ended up being a really easy conversion. We don't think about it. The only thing we think about is, because we are lucky enough to have a public utility and not a private utility in Sacramento, we actually get a check in the mail for our extra electricity.

Back to the recording. There were strings in "Italian Guy," were those synth strings or were they sampled?

Those are samples. We used the [Yamaha] Motif sampler keyboard for that. Sorry, they weren't real.

What did you use mic-wise for your vocals?

This is not an interesting interview for you.

Well, it would be if you used something not that great because it sounded good.

Yeah, it's [usually] just an SM58.

Wow, I think that is very interesting.

Well, again, I don't know if you can hear, my voice isn't this big, rich, sonorous Robert Goulet kind of sound. It's got a little bit of roughness to it, but it's not really big. It kind of works with the 58.

It seems like there aren't as many synth parts on this album as on past ones. Is that true?

Not quite as much. I think it sounds like there's not as much because there's no synthesized drones. Overall, it's weird how that's the one clinching thing that makes it feel more electric. Past that point, it's in another territory. Ultimately, fewer [synth parts], yes, but not that much fewer. I do think this album is a little warmer. There was just something about 2004 that just made me want to—I just thought there was just so much crap in the air. I thought, "Okay, it's not realistic unless we put some of that in our music." With this album, I'm just like, "Okay, enough crap. I miss the sound of guitars." ✱

Mike Levine is EM's editor and senior media producer.

FIG. 1: Assign the drums, keys, guitars, and vocals to subgroups, and route all the subs through the mix bus insert track.

Roughing It

How to quickly get a good-sounding rough mix together

By D. James Goodwin

When I was just starting out as an engineer, I hated the idea of rough mixes. I was never happy with them, and I always felt like someone would hear them and have second thoughts about my work. As my career continued, I began learning how to put together roughs that were flattering, sometimes to the point that they even were chosen for the record. Rough mixes can be innocent, less-thought-out, and as a result can have more energy and honesty. The flip side is that they can also be terrible. Time usually doesn't allow all the normal tweaking, and often you end up just throwing the tracks together to get it done and off to email or CD.

If you're just starting out, or maybe you're the tracking engineer but are trying to land the job of mixing on a particular project, sometimes the rough mixes can help give your client or artist a barometer of how well you mix. I've gotten several very good mixing gigs by doing just that. I was hired to do some overdubs, then I made sure my roughs sounded fantastic so that the client would consider me for the mix. It has worked for me many times over. In the next few pages, I'm going to share some of the things I do to make my rough mixes sound as good as possible.

The examples I'll be giving in this story revolve around Avid Pro Tools, but most other DAWs will share similar qualities so the translation should be straightforward. For the purposes of this article, I'll abstain from the use of a console or outboard hardware. I'll be working strictly in the box.

GET ON THE BUS

The first thing I do on a rough mix is to "Save Session As" and rename it with the date and nomenclature that clearly states it's a rough mix (which helps if you need to revisit it down the line). After that, I start setting up stereo subgroups so that I have the ability to make quick overall volume rides on, say, a drum kit or a group of rhythm guitars. The most important thing is to set up what I call a *mix bus insert*.

The mix bus insert is a stereo aux track that has its own dedicated input path (for example, bus 1-2). Its output is set to the main monitoring path. I prefer this to a master fader because any plug-ins I put on it will be pre-fader, and thus not affected by any fader moves such as a long fade. Also, I can send the output of this mix bus insert to a separate stereo track via sends to print the rough mix onto a new stereo track in the session itself.

Assuming I'm using bus 1-2 as the input path to the mix bus insert, I'll then set up a handful of stereo aux tracks for the subgroups. Typically for a rough mix session, I'll create between six and eight stereo aux tracks. Then I'll name them: Drums, Bass, Guitars, Keys, Vocals, and so forth. For each of these subgroup masters, I'll assign a separate input path. For example, Drums bus will use bus 3-4 as its input, Bass bus will use bus 5-6, etc.

Sometimes it helps to go into the session setup preferences and rename the bus paths to match the inputs. So instead of bus 3-4, you could rename it Drums Bus. This will help keep things nice and organized. When you go through your tracks, simply route their output to their appropriate subgroup bus. In other words, your drum tracks should all have their outputs set to bus 3-4, or Drums Bus, depending on what name you use. Finally, remember that each of these subgroups should be routed to the mix bus insert. Think of it as your final destination (see **Fig. 1**).

Setting up subgroups may seem like a lot of work, but you'll only have to do it once and it takes just a few minutes. You can import the settings into your next session so you don't have to go through this for every song, and it's the best way for you to do general volume moves without making it an all-night affair.

FIRST THINGS FIRST

After I've set up my subgroups and mix bus insert, I start adding plug-ins. In the real world, I typically use hardware inserts for a lot of my processing, but in this scenario, I want to be quick. Because I work on mostly pop and rock music, my main concern with a rough mix is that it has impact and clarity. I'm not overly worried about nuance. That's for the final mix. That said, I usually start by inserting three things on my mix bus insert: a compressor, an equalizer, and a brickwall limiter (see **Fig. 2**).

The mix bus compressor I choose most often is the Waves API 2500 plug-in. I use the hardware version daily, and I know it well. For my taste, I usually set up the plug-in in Old mode, with a 3:1 ratio, medium attack, medium-to-fast release, and with a high threshold setting to catch only the loudest peaks. I engage this compressor right away because I find that mixing into the bus compressor is the best way to make sure your mix holds up well. I like to hear its effect from the beginning.

Next, I insert a UAD Precision Equalizer. Any mastering-style EQ with high/low shelving and fairly wide bells in the mids will be fine. I'll keep it set flat for the time being. I only put this insert on in the event that I want to just shape the overall mix slightly at the end. I usually find that most roughs suffer from being a bit too dark as you typically don't sculpt individual tracks as much as in a final mix. A mix bus EQ can help with clarity in the low mids and up top where some "air" is needed, without requiring that you apply surgical EQ to individual tracks.

FIG. 2: Start the mix with a compressor, a brickwall limiter, and an EQ plug-in on the mix bus insert track.

Finally, I insert a brickwall limiter, purely for level. While I personally dislike the "volume wars" and what they've done to audio quality in the past few years, it has become necessary to add brickwall limiting to even a rough mix. My favorite for a few years now is the UAD Precision Maximizer. It sounds great, gets plenty of level, and doesn't destroy a mix like some others can. If you don't have access to it, I recommend using the most transparent brickwall limiter you have. As with the EQ, I keep the brickwall in bypass until near the end.

Before I start pulling up faders, I like to set up a reverb send and a delay/echo send. Back when we mixed on consoles, there was usually a plate reverb and an echo normalled to the console, so it was always available. This is my way of keeping up that tradition. I start by creating two new aux inputs, and calling them Reverb and Delay. I set the input accordingly, and the output to the mix bus insert. Lastly, I disable the solos on the

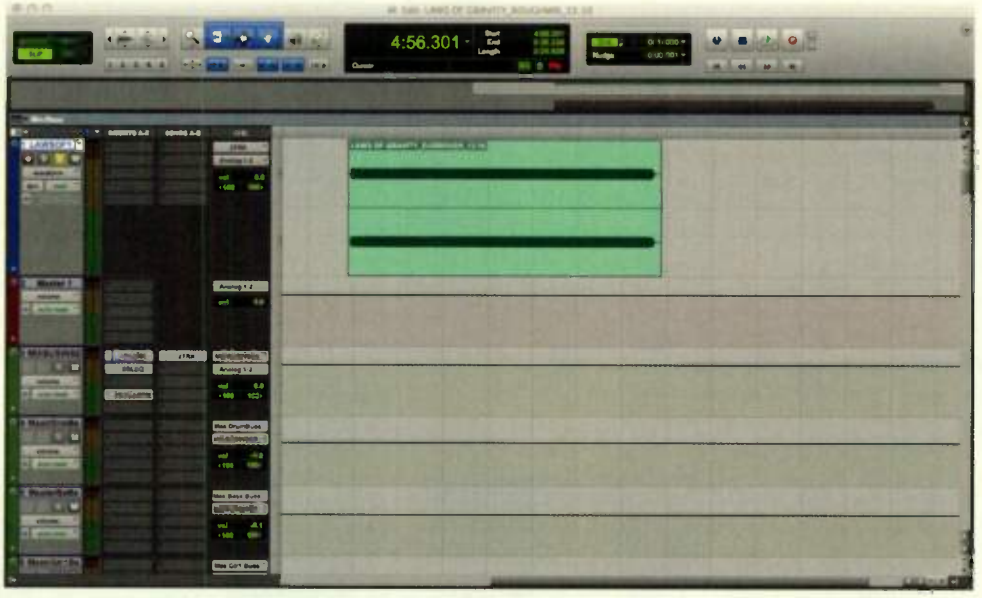


FIG 3: Printing to a track in the session keeps your mix inside the session, making later retrieval easier.

aux tracks so that they're in Pro Tools' Solo Safe mode. This makes it so that when I solo something, the aux input is still active, making it possible to adjust the reverb or delay in isolation. The reverb and delay on the aux inserts are set at 100-percent wet.

I almost always use the UAD Plate 140 plug-in for reverb. It's uncanny how close it sounds to a real EMT 140, and it's every bit as warm and lush. Any decent reverb will do here, but I prefer to use "plate" presets simply because of subtlety, habit, and familiarity.

For an echo, I'll choose something quick and easy, such as the Sound Toys EchoBoy plug-in. It lets me change tempo, sync the tempo to the track, or choose whether I want eighth-notes, quarter-notes, or any other subdivision—all on the fly. If I sense that the track will not need a long delay, I will sometimes set up a slapback-type echo, mostly for vocals. If this is the case, I will use it in mono because I prefer mono slapback to stereo.

THE RHYTHM SECTION

It would be impossible for me to give you specifics on treating balances, mostly because every track is different and each track requires its own set of fixes or complements. However, I'm going to give you some general tips on treating drums and bass quickly so that you can put together a respectable rhythm section mix within minutes. First, I start by getting a decent balance on the drums and bass, with no EQ or compression. Then I'll just try to deal with things that stick out or sound bad.

I generally start by putting a compressor on my drum subgroup. Often, this will be something like the Waves API 2500. I'll usually start fairly aggressive with about a 4:1 ratio, with a medium attack to keep the transients alive but a fast release to make sure the drums have impact and a little bit of pumping. This breathes some excitement into the drum group. I don't want to squash them; I just want to get a little life and sparkle out of the kit (see **Web Clips 1a** and **1b**).

If your kick drum lacks definition, try pulling out a few dBs around 220 to 240Hz. This will get rid of some of that mud. I don't do very much in the top end on a kick drum. I prefer to increase definition and punch by subtracting rather than adding. I also rarely compress a kick drum, unless it absolutely needs it. But don't be afraid to EQ the heck out of it if necessary. If it sounds right, it is right.

Snare drums in rock music tend to lack depth and low end to my ears, and they're often too "honky" or have too much midrange boxiness. That being said, I usually will pull out a little around 700 to 800Hz to pronounce the low end a bit. I may also just do a slight boost, somewhere around 5kHz. Also, check the phase with the overheads. I can't tell you how often I've gotten tracks to work on and the snare is out of phase with the overhead mics, making it lose impact. I'll also usually insert a compressor after the EQ. I set it to be pretty light, mostly grabbing the highest peaks, but I want to keep the transient alive, so my attack is set medium/slow and my release is set fast/medium. Most times, I'll also do a similar treatment with the tom tracks. Usually, they have a bit too much boxiness and not enough depth. So, again, I'll pull out some 400 to 600Hz, and give a slight boost to the upper mids for the attack. To be safe, I'll check the phase, too.

I'll usually do very little to the rest of the kit at this point. If I've tracked the project, I'm pretty confident that my sounds are there so I don't need to do much. It's all subjective, and you may find that you need to put a highpass filter on the overheads, or perhaps some compression on the room mics. These are common things. Don't be afraid to play around, but in keeping with the principles of a rough mix, don't labor over it. Just get it to sound good quickly and move on.

The whole time I'm balancing drums, I'm also keeping an ear on the bass to make sure it's working with the drums, specifically the kick drum. Look for the usual suspects here: Too much pick noise? Shelf the top end so it's not too bright. Not enough impact? Try pulling out some 300 to 400Hz to make the bottom seem more powerful. And, of course, try doing a slight boost around 90Hz for some thunder if you need it. If you're feeling good about the way the rhythm section sits and your levels are at a good spot, with no clipping on your subgroups, it's time to move on.

THE REST OF THE BAND

Next, I start pulling up faders on everything else except for vocals. This means guitars, synths, and anything that may be in the track. Because every song is different, my best tip is to maintain clarity by using your panning effectively. In other words, keep similar textures in different parts of the stereo field. For instance, if you have a synth and a guitar in the same general octave range, try panning them apart to maximize clarity and width. The goal of a rough mix is to have everything sit in a place where it can be heard. A lot of critical decisions will be made on the basis of a rough mix, so unless you know otherwise, try to make sure that you have clarity on the entire arrangement. Try to only spend a few minutes getting this balance together. It should be instinctual and not overworked. The biggest mistake people make is overworking the instruments.

You may want to apply an overall EQ to a guitar or keyboard subgroup. I do this to save time so that I don't have to go into individual tracks and apply treatment. In some cases, I'll insert an EQ on the guitar subgroup and engage a highpass filter to get rid of any rumble or mud in the guitars. You don't want to lose any power; you only want to clean it up so nothing unnecessary gets to the mix bus. I may also add a little boost in the upper midrange because that's where I can get some extra bite out of the guitars if needed. Because distorted guitars are inherently compressed, I typically opt not to use a compressor on them. However, I will sometimes put a compressor on the keyboard subgroup. Once again, my goal here is to simply catch any stray peaks and perhaps breathe a little excitement into overall group, if needed.

Finally, I'll get the vocals into the mix and do my best to get them sitting right without anything on them. Of course, that's easier said than done, and in pop and rock music, more often than not you'll be reaching for an EQ and a compressor. Here, I'll typically reach for a channel strip plug-in, most often the Waves SSL Channel. It lets me quickly set a highpass filter for removing rumble, EQ as necessary, and compress. Usually, I end up with a slight upper midrange boost and a lower midrange cut, unless the voice is nasally and thin.

With vocals, I always prefer very wide bell curves so that the voice doesn't sound overly processed. Then I'll set my compressor so that

> "The goal of a rough mix is to have everything sit in a place where it can be heard.

it grabs peaks, with a medium attack and a medium release. I may also insert another subtle compressor on the vocal subgroup to help it sit better in the mix (see **Web Clips 2a** and **2B**). I'll usually set this compressor to have a slightly faster attack and faster release, but I'll set the threshold much higher so it's just slightly hitting the signal.

There may be times when the use of Auto-Tune is necessary. In the case of a rough mix, however, the artist may prefer the vocals untreated to get an honest sense of the performance, especially if you're doing this for comping purposes. Otherwise, it's usually simple to insert Antares Auto-Tune, or something similar, and put it on the most conservative setting, using the Chromatic C scale so that all notes are active. If you set it right, you should barely see it doing anything, only occasionally grabbing the most egregious of notes. If it takes you more than a couple minutes to get it to sound transparent, then don't bother. This is a rough mix after all, and you likely have 12 more to do!

Through all this, remember that you have a reverb and a delay set up. I usually use them for vocals and maybe a guitar lead or synth lead. Either way, they're there for your disposal, and don't forget to consider them when you're focusing on the vocals.

PRINT AND FORGET

At this point, I usually have a serviceable rough mix together. In many cases, it shouldn't take more than 15 to 20 minutes to get to this stage. You're now ready to check your brickwall limiter and perhaps a little mix bus EQ to make it shimmer and sparkle a bit more.

The whole point of the brickwall is to increase your level without clipping. Depending on what plug-in you use, be sure that the mix doesn't go over 0dB and that your input to the limiter doesn't clip. If your mix bus insert is clipping the limiter, just bring all of your subgroups down equally until you're out of the red. Sometimes when you engage the brickwall, it can seem as though the drums get quieter or the guitars get louder. If so, it's handy to use your subgroups to readjust your overall balances to compensate.

When I'm satisfied, I create a new stereo audio track at the top of the session, and using a send route the mix bus insert to the new track, and set the send level to be at 0dB, or unity gain. I mute the new track while it's in record (to avoid feedback) and name it accordingly. This track will be for a print of the mix. The reason I print it in the session is so that I have a copy at all times inside the session, thus avoiding the need to hunt down a lost rough mix later on. It also lets me export different formats quickly without having to bounce in real time. I then arm the track, put Pro Tools into record, and print the mix (see **Fig. 3**).

When it's finished, I recommend opening up the session for the next song, saving it under a new name, and importing all of the subgroups, aux inputs, and the mix bus insert from the last session. If you have the same audio tracks for multiple sessions, you can also match tracks and import your track settings from the last session to save even more time. In most cases, after the importing you'll simply make sure your audio tracks are routed properly and you'll be in good shape. Adjust a few balances, and you're off to the races.

Rough mixing can be frustrating. Hopefully these tips will help you get better sounds more quickly, using broad strokes to sweeten things, and not getting caught up on details. As always, your mileage may vary, but use your imagination, and if you have the time, please experiment. Have fun with it! ✱

D. James Goodwin (djamesgoodwin.com) is a producer and mixing engineer, operating out of his own studio, The Isokon, in Woodstock, N.Y. His discography includes The Bravery, Norah Jones, Devo, and Natasha Bedingfield.

By Len Sasso

FIG. 1: This Numerology Stack uses an LFO (top) and a CV lane of a MonoNote sequencer (bottom) to modulate the sample start and loop boundaries of the speech clip in the Sample Synth module (middle). The MonoNote also sets the playback pitches and step lengths.

What's Your Number?

The step sequencer as a sound-design tool

My first encounter with a step sequencer was the 960 module in a Moog modular system. It provided three rows of eight knobs to create three control-voltage sequences that you could cable to affect other Moog modules. It also had individual trigger inputs (to select steps), gate outputs, an internal clock, and sundry special features. Creating note sequences was not the 960's primary purpose, and it presented some challenges, not least of which was tuning.

The note-based step sequencers in modern synths, samplers, and drum machines tend to obscure the considerable sound-design potential of step sequencing. The most robust environment for that on the Mac I've seen is Numerology 3 from Five12 (five12.com). It offers more than 50 modules for generating and processing notes and control signals. I've used Numerology in my examples, but you can adapt most of these techniques to any step sequencer that has controller lanes and flexible signal routing.

AH ONE, AH TWO

Numerology's Sample Synth is a good place to start. It holds a single sample, changes pitch by changing playback speed, and lets you modulate the sample start, as well as the loop start and end points. In addition to loading samples, you can create synthesized speech clips using Apple's OS X Speech Synthesizer; just click Sample Synth's Speak button, type in some text, and choose a voice.

Figure 1 shows a setup I often use to mangle speech clips (see **Web Clip 1**). A sample-and-hold LFO controls the sample start, a MonoNote sequencer plays the notes, and a CV lane of the same sequencer sets the loop end. The Scale and Offset module converts the loop-end position to an offset from the sample-start position, and the ParamMod module routes it to the Sample Synth.

The ParamMod module gives you instant access to almost any parameter in an AU plug-in or other Numerology module—no MIDI Learn or special setup required. To route a modulation source (LFO, envelope generator, sequencer CV lane, etc.), select the target and parameter from drop-down menus. (Many of Numerology's CV-generator modules have those menus built in.)

You can add ParamMod modules as needed to target multiple parameters. Use that to apply the same modulation to different AU synths and effects, as well as to multiple parameters of the same device (see **Web Clip 2**). Once you've set up a modulation scheme, save the Stack with or without the AUs. Then you need to re-create only the modulation routings to apply the same modulation in different setups.

You can forgo step sequencing and just use Numerology as a modulation source for instruments played in real time or from MIDI tracks in your DAW. You can route incoming MIDI directly to any AU plug-in. In addition, the MIDI-to-CV module extracts note number, gate, velocity, pitch-bend, aftertouch, and four MIDI CC streams from incoming MIDI data and provides them as separate outputs, which you can then route to control other plug-in or Numerology parameters. Use the gates to start and stop or advance CV sequencers or trigger envelope generators, and use the other outputs for direct control or to scale the effect of other Numerology modules. ✱

By Vincent di Pasquale

Master Bus Musings

Master bus processing is a key aspect of a mix and, if done properly, can give your tracks a more finished sound. Most people readily understand the overall concept of mixing, but not so many pay attention to what is happening on the master bus. This month, I'll discuss some of the important things to keep in mind when dealing with the master bus and share the processing chain that I use.

The master bus is the main stereo sum, or output of any mix. It is the last and final signal-flow point before your track gets printed to a final stereo mix. Whether you are working in the box or out, it is critical to know exactly what is happening on the stereo bus. You should be aware of such factors as the overall dynamic range and the frequency spectrum of your track when dialing in your finished mix.

PEAK VS. RMS

It's key to understand some principles regarding the overall dynamic range (i.e., the overall loudness) of your mix. In audio engineering, there are two types of loudness: peak and RMS. Peak represents the absolute loudest signals that are present. RMS is the average loudness of all the signals present. While it is important to know exactly both the peak and RMS levels of a mix, the RMS offers more insight as to how loud a mix will be perceived by the listener. Humans perceive loudness based on RMS because our ears and brains can't respond to every single frequency and sound in a complex waveform. It is also important to know what the overall peak level is to avoid clipping and distortion on your mix, especially when working completely in the box.

There is much debate about the dynamic range of today's popular music. Many believe a more natural sound with a wide dynamic range is best, while others try to smash the dynamics down so hard that the end result is an ultraloud, solid block of a waveform. I believe *both* have a place in modern music. However, most of the commercial projects and remixes that I work on end up being pretty loud. The main reason is that in the clubs, a track needs to hit as hard as possible. As a result, most club mixes are compressed pretty heavily.

I don't really consider master bus processing to be the same as mastering. Mastering is a separate process that not only deals with what is happening on a single track, but what is happening on a group of tracks, like an album or EP, to ensure that they all have consistency as a finished product. Still, not all projects I work on will immediately get mastered by a separate mastering engineer, so I always pay a lot of attention to master bus processing to make sure the mix leaves my studio sounding the way I want it to.

GETTING STARTED

It's never too early in a project to start paying attention to the master bus. As you go along, it's critical that you always leave some headroom, level-wise, for your master bus processing. Otherwise, you'll start turning everything up as you add other elements and you'll start clipping the master. This is one reason why I wait as long as possible to add the final automation to my mixes. I want to be sure that I can turn things down and adjust levels so as not to hit the master bus too hard before the final processing chain.

FIG. 1: Di Pasquale's master bus processors typically comprise Logic's Channel EQ and two Waves plug-ins: the SSL G-Master Buss Compressor and L3 Ultramaximizer (limiter).

I typically start making initial adjustments to the master bus chain (see **Fig. 1**) as I'm putting the track together, especially on remixes or productions that are done entirely inside the box. I'll instantiate the plug-ins and get some rough settings going early on. This prevents spending tons of time mixing things to sound a certain way only to discover that the sound changes significantly after the master bus inserts are added.

THE PROCESS

Once I have the final where I want it, I turn my attention to the master bus. I loop the loudest section of the song and work on that first. This may be a chorus coming out of the break or whatever section has the most energy. After I get the EQ, compression, and limiting set, I can listen to the whole song to make sure the levels are where I want them. In some cases, I will automate the master bus effects to keep a consistent sound throughout the whole mix.

The first insert in my master chain is a multimeter, which lets me check frequency and level (see **Fig. 2**). I either use Apple Logic's built-in multimeter or one from Waves or iZotope. This meter shows me the overall dynamic range of my mix as it hits the master. I study the frequency spectrum to see what the high end and low end of my mix are doing, and I make sure that the levels are not clipping. Set your meter to show both peak and RMS levels. (I also put a multimeter at the end of my master chain, which lets me check the dynamic range post-processing.)

FIG. 2: It's useful to have two multimeter plug-ins active: one at the beginning of the chain and one at the end.

Next, I insert an EQ and typically focus on EQ'ing three areas: First is the high end. I will always add a little high-end shelving to accentuate the overall brightness to the mix. Ultimately, the boost will start somewhere between 6 and 10 kHz, depending on what I hear in the mix. The next area I listen to is the 300 to 400Hz range. This is where a lot of muddiness can creep in. If it's muddy, I'll put in a sharp parametric cut to help reduce it. Finally, I listen to the low end. I like to add some subtle low-shelving EQ to help glue together the drums and bass.

The next insert is a stereo compressor, with which I aim for about 4 to 6 dB of gain reduction. If I use the Logic compressor, I will start with the "analog tape" preset, which really sounds good across the whole mix. Next up I might add a multiband compressor. I don't always use one, but it is really great when I want some extra punch in the low end. The last insert is a limiter, which applies the finishing touch to the mix. It's where the power of the mix comes out. My go-to plug here is usually the Waves Ultramaximizer, but sometimes I'll use Logic's Adaptive Limiter. Either way, I'll typically set the output ceiling to -0.1 and then adjust the threshold to get around 3 to 4 dB of attenuation.

OVER AND OUT

In the end, the main goal of my master bus processing is to make sure the frequency spectrum sounds balanced and looks that way on the post-processing multimeter. As for the overall dynamic range, I shoot for somewhere between -10 and -5 dbfs during the loudest sections of the song.

Once everything is sounding the way I want, I will bounce a pass and re-import it to listen to it and look at the waveform. If I know the project will be professionally mastered, I'll bypass the limiter and typically just use EQ and stereo compression. In these cases, I will just make sure the mix sounds exactly the way I want, with a little bit of headroom left for the mastering engineer. Like with everything in music production, the best method is to do what sounds good to you, and take time to experiment and get the results you want. ✱

Vincent di Pasquale (vcdstudios.com) is a producer/remixer who works out of his project studio and has remixed songs for artists including Madonna, Mariah Carey, Nelly Furtado, and many others. To learn more about remixing, check out his The Art of the Remix, *an interactive remixing course available at www.faderpro.com.*

REVIEWS

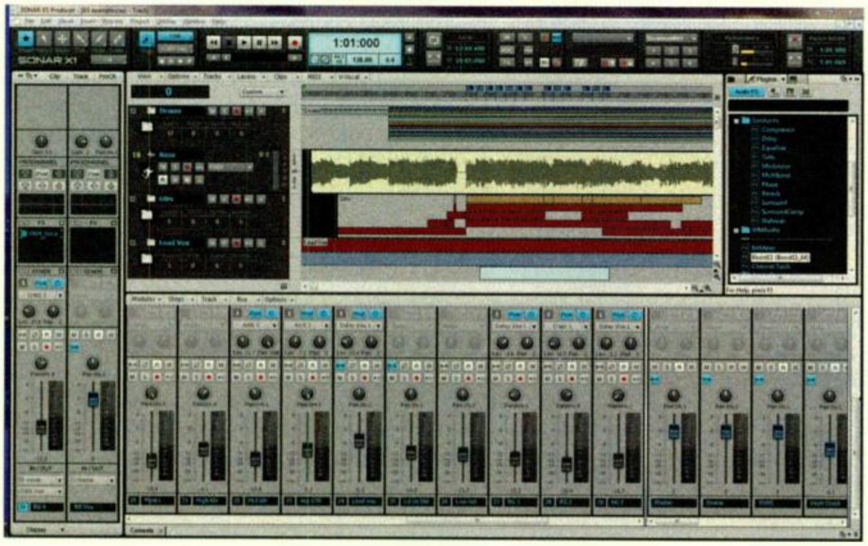


FIG. 1: Known as "Skylight," SONAR X1's interface redesign is highly configurable and includes an improved Inspector, consolidated Browser, streamlined Track view, and powerful MultiDock.

MAC

*** PRODUCT SUMMARY**

PC-BASED DAW SOFTWARE

PRICE: $399, street; $99, upgrade from 8.5 (download)

PROS: New ProChannel multi-effect. Effects chains. Major interface improvement. Streamlined dialog boxes and menus.

CONS: Nothing significant.

FEATURES: 4
EASE OF USE: 4
SOUND QUALITY: 4
VALUE: 4

RATING PRODUCTS FROM 1 TO 5

CAKEWALK.COM

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

Cakewalk SONAR X1 Producer

Interface redesign done right

By Brian Smithers

SONAR X1 is the biggest redesign of Cakewalk's flagship DAW since its evolution from Pro Audio 10 years ago. The development team has succeeded in bringing the program's ample feature set together under the control of a more streamlined interface. The majority of the improvements in this version are interface-related, with the most significant new features being the addition of ProChannel, an excellent integrated channel strip, and the ability to save and recall effects chains.

I tested SONAR X1a (an update released in December 2010) on a quad-core 2.67GHz Intel i7 920 with 6GB of RAM running 64-bit Windows 7 Ultimate. SONAR had no problems running in 64-bit mode with 64-bit audio processing.

SKY KING

SONAR's new interface goes by the evocative name Skylight, but it will still look like home to established users (see **Fig. 1**). Skylight brings together the traditional Track view, an improved Inspector, and an all-encompassing Browser with the powerful new MultiDock. These four views can be collapsed or, in the case of the MultiDock, maximized with convenient shortcut keys, so controlling the Skylight interface is a snap.

I was not a fan of docking windows when it was first introduced into SONAR, but the MultiDock has given the notion a new appeal. I can now dock various windows—such as the Piano Roll, the Event List, the console view, or the Matrix—and then with a few keystrokes maximize the MultiDock and switch to the view I need. You can also float the MultiDock on a separate monitor if one is available. If I take the time to create Screensets, I can speed the process further. Up to 10 Screensets can be saved per project and can be imported from another open session. The upshot of all of this? SONAR X1 is now easier to manage in a single- or multimonitor environment than previously. Even window-management functions I had previously sussed out with shortcuts and key bindings now feel more like an intended function than a kludge.

The Inspector now displays the output channel for the selected track. It lists more types of information than before, such as clip properties (including clip effects, groove clip properties, and AudioSnap properties) and track properties. It can also display the entire ProChannel for the selected track. For an instrument track, you can switch between displaying the audio output channel or the track's MIDI properties, including the arpeggiator, input quantize, and so forth.

Among other things, the Browser consolidates functions previously found elsewhere, allowing the import of audio and MIDI files, presets, step-sequencer patterns, track templates, and more. The most interesting new feature provided by the Browser is the ability to drag plug-ins—including virtual instruments—directly to a track. Simply drag a soft synth to the Track view to create an instrument track. Indeed, drag-and-drop functionality has been implemented or improved in various ways throughout SONAR X1.

SONAR has always had a context-sensitive "smart" tool, but X1's is a bit smarter and friendlier. If its primary functions are not what you need, you can switch to alternate tools either by pressing "T" to show the tool palette (called the Heads Up Display, or HUD) or by pressing F5 through F10 to switch directly to the desired tool. If you hold the relevant F key as you use its tool, the cursor automatically switches back to the smart tool when you release it. When multiple data types are displayed in a track, a new data filter lets you focus all edit functions on just what you need. You can also bring up the HUD by clicking the middle mouse button; moving the scroll wheel then activates the dropdown menu for selecting data types.

CHANNEL PRO

ProChannel—which is available from the Inspector as well as from each channel of the console view—combines EQ, compression, and tube saturation in a user-configurable chain that

FIG. 2: The ProChannel's emulations of major pro EQs and compressors are designed to bring big-studio sound to SONAR X1.

can be placed before or after the track's FX bin (see **Fig. 2**). The EQ offers six bands: two filters (high- and lowpass), two fully parametric bands, and two bands that can be switched between fully parametric and low- and high-shelf mode. The filters' slopes are adjustable between 6 and 48dB/octave in 6dB increments. The four parametric bands offer 18dB of gain or cut, and all four range from 20 to 20k Hz. The EQ can be set to Pure, Vintage, or Modern modes, each of which emulates the Q and level characteristics of a different famous EQ.

The EQ curve can be edited graphically, with Q adjustment via Alt-dragging. This is one seriously flexible equalizer with truly professional features. The EQ button also includes a Gloss button, which adds "air" to all modes.

The compressor stage emulates two well-known compressors—no bonus points for guessing what the PC76 U-Type and PC4K S-Type are modeled after. The controls of each, including the metering, are representative of those on the modeled devices. The PC76 even offers an extra Ratio button (labeled with the infinity symbol) that emulates its source's famous "all buttons in" mode. The PC4K offers a sidechain; the PC76 does not. Both feature a wet/dry knob, making parallel compression a no-brainer.

> The most exciting new feature in X1, ProChannel combines EQ, compression, and tube saturation in a user-configurable chain that can be placed before or after the track's FX bin.

The tube saturation circuit in ProChannel offers a wide range of colors. Two types of saturation are available (with no documentation on what the distinction is), and the input, drive, and output controls are perfectly intuitive.

Although SONAR previously had high-quality EQ and compression—the Sonitus:fx EQ, Vintage Channel, and LP64 Compressor and EQ are all worthy contenders—the addition of the ProChannel brings mixing in SONAR X1 to a new high. The inclusion of a clipping indicator at the top of each module is indicative of the professional mindset behind this new channel strip. Metering—especially between plug-ins—is an area that has not received as much attention in DAW interfaces as it should, so hats off to Cakewalk.

UNCHAIN MY HEART

Effects chains can now be saved and recalled, dragged or copied from track to track, collapsed within an FX bin, and named for clear reference. It's nice to see this feature finally migrated from Project5 to the mothership. Although such chains cannot include soft synths, you can accomplish that with a track template. Simply drag a soft synth from the browser to the Track view, insert any audio plug-ins you want on the resulting instrument track, dial in your desired settings, and save the track as a track template. You can then drag the track template from the browser any time you need that particular setup. Another related time-saver is the ability to store default plug-in presets; you can save multiple versions of a plug-in in the Plug-In Manager, each with a different default preset.

X1 continues to support grouping of mix parameters in the same way SONAR has implemented it for some time. However, "quick grouping" of parameters is now much easier. Simply select the relevant tracks and Ctrl-drag the desired parameters. Pan controls maintain their relative positions, as do mute, solo, record arm, and other status buttons.

The full list of improvements in X1 goes on and on, from performance enhancements to menu redesign. What used to be a nearly endless row of obscure icons in the Track view is now an orderly dropdown menu. What used to be separate global and project options are now consolidated in a Preferences dialog box. With respect to menus and preferences, my one lingering wish is for a meter options dialog box so that when I set up my metering preferences, I don't have to open and retrace the cascading menu multiple times to set meter mode, peak hold, and range for each meter type. Given the other improvements in this version, though, I can wait a bit longer.

If I had to pick something to gripe about, it might be that when SONAR gets cool new features, such as V-Vocal or Beatscape, they don't get updated and improved with each version. Or, I might quibble about the fact that the Publisher applet produces an audio player that I wouldn't put on any website with my name on it. Similarly, the CD-burning applet is so rudimentary it doesn't compare well to the app that was included free with your DVD-R drive, and it's a separate applet, not "integrated CD burning" as advertised. Sincere gripes all, but pretty trivial in light of SONAR X1's powerful audio production features now made more usable by an interface redesign done right. ✱

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

FIG. 1: The Main view of Nectar has easy, intuitive controls for each vocal preset.

MAC/WIN

iZotope Nectar

A cost-effective vocal-mixing contender with excellent features

By Steve Skinner

Most good mixes apply at least three plug-ins (or analog effects) to a vocal track. I often use as many as five, as well as one or two delays and one or two reverbs. In Nectar, iZotope has created one plug-in that offers a complete vocal-processing signal chain. Other plug-ins, such as the Waves Signature Series, offer EQ, compression, delay, doubling, and reverb in a single plug-in, but Nectar adds pitch correction, a de-esser, breath control, a gate, and saturation. And, it includes an exhaustive array of presets organized by vocal style.

TAKE CONTROL

The Main view has a dropdown menu of musical genres with submenus of styles within each genre (see **Fig. 1**). Simple adjustments for levels, space mix, size, color, saturation, pitch-correction scale, de-esser amount, breath target, and gate threshold are on the front panel. There is a somewhat tweakable graphic EQ display, as well as input and output meters and level controls. One section of the front panel changes its function depending on the selected style. These functions are sometimes given impressionistic names such as Girth or Psychedelic, and they usually control a unique element of the selected style.

The styles are put together well, and demos of many of the genres are available at the iZotope website. The style names are somewhat arbitrary; there's no reason why an R&B style, for example, couldn't work in a dance track. It's helpful for beginning mixers to see the settings laid out so clearly—both for ease of tweaking and to learn how certain vocal sounds are achieved (see **Web Clips 1** and **2**).

Nectar has two operation modes: Mixing and Tracking. Mixing mode employs a look-ahead function that improves sound quality but introduces latency and uses more CPU power. It's

PRODUCT SUMMARY

SIGNAL PROCESSOR PLUG IN
PRICE: $249

PROS: Excellent, quick vocal processing. Usable presets. Easy interface. Good price.

CONS: Sound quality is not quite pro level. High CPU load.

FEATURES: 4
EASE OF USE: 4.5
SOUND QUALITY: 3
VALUE: 4

izotope.com

for mixing situations where delay compensation is available. Tracking mode eliminates latency and reduces CPU load. Use that in tracking and live situations. (The Breath Control module is deactivated in Tracking mode.)

The HTML manual, which is stored on your computer, is clearly written; you access it by clicking the Help button. The advantage of HTML documentation is that it employs hypertext linking.

ADVANCED TWEAKING

You click the Advanced View button to access the individual components that make up each genre style (see **Fig. 2**). Components are displayed on the left, and when a component is highlighted, its parameters appear.

With the exception of pitch correction and the breath-control module, which always come first, you can reorder the components in the signal chain by dragging them up or down. Each component also has a Power button and a Bypass switch. In a nice touch, clicking the Solo switch leaves just the highlighted component on.

Whereas the Main view uses impressionistic labels for the functions, the Advanced view uses conventional audio engineering terms. This works well, although you sometimes have to experiment a bit to see exactly how Advanced view parameters correspond to Main view parameters. The Main view parameter Smash, for example, controls both settings on the Limiter module; it has no direct Advanced view equivalent.

You can automate every parameter on the Advanced view, but due to multiple correlations between parameters as just described, you can't automate the parameters on the Main view. In Pro Tools, the keyboard shortcut for automation (Control > Option+Command click) does not work so you need to access the automation from the plug-in page. Currently, no mono-to-stereo option is available in Pro Tools, but iZotope plans to address this in a subsequent release.

> In Nectar, iZotope has created one plug-in offering a complete vocal-processing signal chain.

GET THE PITCH

Nectar's Pitch Correction component is full-featured and competitive with other stand-alone pitch-correction plug-ins (see **Web Clip 3**). The main control is correction speed. At 0, you get the Cher effect, which at higher settings becomes less noticeable. At 50 and above, pitch correction is smooth and transparent. The scale choices are chromatic, major, minor, and custom. My one minor gripe here is that the black-key scales are all written as sharps—A# major instead of Bb major, for example. (I'm told this will be fixed).

On the Advanced view, an additional window (also called Advanced) opens for pitch correction. That's where you set the reference pitch (A=440Hz by default), formant shift, and formant scaling. You can also detach and enlarge a graphics display that works like the graphics displays on other pitch-correction plug-ins.

FIG. 2: Nectar's Advanced view has in-depth control of each parameter.

OUT OF BREATH

The Breath Control function is well-designed and, for some people, may alone be worth the price of admission. Its Threshold slider controls the breath-detection sensitivity, and its Target/Gain slider controls the amount of attenuation. Target mode reduces the breath to the target level regardless of how loud it was, whereas Gain mode reduces it by a set amount. A rolling graphic display shows the waveform and where the effect kicks in.

In a previous mix, I had to manually pull down every breath because after the vocal was compressed and EQ'd, the breaths sounded too harsh. Nectar's Breath Control achieved the same result quickly and with no noticeable artifacts (see **Web Clips 4** and **5**).

For some voice-over applications,

> The well-designed Breath Control function may alone be worth the price of admission.

such as telephony ("press 1 for customer service"), breaths need to be completely eliminated, which is quite time-consuming. Breath Control did this better than other de-breath plug-ins I've tried, although it sometimes missed a breath or cut off the tails of s's and f's (see **Web Clips 6** and **7**).

Nectar's breath control function could be a huge time-saver for telephony engineers, but its latency—26,623 samples at 44.1kHz sampling rate—is a big drawback. Delay compensation needs to be turned on in your DAW to use Breath Control as an insert. In Pro Tools, which has a maximum delay compensation of 4,000 samples, Breath Control must be used as an Audiosuite plug-in (if latency is a problem). Also because of latency, Breath Control is the only module that doesn't work in Tracking mode.

SPACE IS THE PLACE

The reverb module is a standard, algorithmic digital reverb. You get four plate and three hall presets. You'll find controls for pre-delay, damping, color, low-cutoff, high-cutoff, decay time, and wet/dry mix. I found the plates to be quite useful, with a lot of transparency. The hall presets, however, lacked the depth and warmth that you can get from a convolution reverb. For classical crossover or other styles where a large warm reverb is desired, I would recommend using one of Nectar's plates, then adding a separate large-hall convolution reverb plug-in.

The Delay module has a useful feature set: Digital, Analog, and Tape modes with a Trash mode for the last two, and it includes filter and modulation. I would like to see a send-to-the-reverb here.

The Doubler module is well-designed and offers up to four voices. You can set the pitch variation between 0 and 100 cents (a semitone), and you also get an octave-up/down option; intervals in between are not available. A Variation slider provides pitch and timing randomness. The sound is clean and comparable to other doubler plug-ins.

SATURATE, DE-ESS, COMPRESS

The Saturation module adds various types of harmonic distortion to the vocal. It ranges from slight brightening and thickening to moderate audible distortion. An Overdrive switch would be nice for those times when you want the vocal-through-the-Marshall sound. You have five types of saturation, as well as a mix slider that lets you add distortion while keeping the clarity of the original signal.

The De-Esser differs from a standard de-esser in that it does not have a threshold control; it attenuates the "s" down to the target level regardless of the program level. This is a nice touch.

The Compressor module offers the standard Digital, Vintage, Optical, and Solid-State modes. What makes it special is the optional Parallel mode, which brings in a separate compressor running parallel to the first. This has all the features of the first compressor with the addition of a shelf EQ to increase or decrease the highs and lows. If you brighten the parallel compressor, you can get the Motown sound. The ratio of the brighter compressor is set lower so that louder notes are somewhat brighter than softer ones.

The EQ section sports a 5-band graphic EQ. It can be edited in a limited way on the Main view and extensively on the Advanced view.

WISH LIST

Nectar is a great concept, and I applaud iZotope for tackling it. I'd love to see a few more items in the future. A multiband compressor, like Waves C4, would be great for those vocalists with too much midrange edge at high volumes. An Overdrive switch in the Saturation module would also be nice. A harmony generator would be a worthy addition. An auto-ride effect would round out the unit. With those things added, you could do a vocal mix in about two minutes.

Nectar is more taxing on the CPU than equivalent separate plug-ins. I'm told that this will be addressed in the first update.

For beginning and intermediate vocal mixers, I highly recommend Nectar. I would also recommend it for someone who is on a tight budget and doesn't have many plug-ins. The effects in Nectar are a step up from the effects that come within most DAWs, if only for their color and character.

For more advanced engineers, I would recommend it as a time-saver for rough mixes and for live use. You could save a signal chain for a particular singer and open it in seconds. Some modules in Nectar could become go-to effects for advanced engineers. I will definitely be using the Breath Control as an offline effect. The Saturation, De-Esser, and Plate reverb effects could also find their way into a pro mix. In general, the sound was not as crystal-clear as a high-end EQ followed by a top-quality compressor and reverb. *

Steve Skinner (steveskinnermusic.com) is a Grammy-nominated writer/producer/keyboard player/programmer living in New Jersey. He has mixed lots of vocals.

Figure 1: Drumagog's new GUI shows a wealth of information at once.

WaveMachine Labs Drumagog Platinum 5

MAC/WIN

A venerable drum-replacement plug-in adds even more features

By Michael Cooper

*

PRODUCT SUMMARY

DRUM-REPLACEMENT PLUG-IN

PRICES: Platinum, $359; Pro, $269; Basic, $149

PROS: Rich in features. Includes many excellent samples and IRs. Tight phase alignment. VSTi plug-in hosting. Improved file browser.

CONS: Buggy reverbs. Few stereo room and overhead mic samples included. Auto-Hi-Hat Tracking can't reliably trigger 16th notes.

FEATURES: 4
EASE OF USE: 4.5
SOUND QUALITY: 3
VALUE: 4

drumagog.com

Drumagog has long been regarded as the industry-leading drum-replacement plug-in. For those unfamiliar with drum replacement, input a crummy-sounding drum hit into Drumagog, and it will trigger one or more better-sounding drum samples from its built-in 5GB library (or from your own library) to replace it at the plug-in's output. Alternatively, you can layer replacement samples with the original sound to beef it up without totally replacing it.

Drumagog 5 adds promising new features and sample content to this popular plug-in. Three versions of the software—Basic, Pro, and Platinum—are available in AU, VST, and RTAS formats. I reviewed Version 5.03b of the flagship Drumagog Platinum AU version in Digital Performer 7.21, using an 8-core Mac Pro running OS X 10.5.8. I'll mostly cover what's new since Drumagog 4, the review for which you can read online at emusician.com.

A VERSION FOR ANY BUDGET

Drumagog 5 supports iLok and disc-authorization copy-protection schemes. All three versions of Drumagog 5 sport a new, enlarged GUI and an engine that can simultaneously trigger—with each drum hit—up to three samples derived from room, overhead, or other ancillary mics. In Pro and Platinum versions only, proprietary functionality (dubbed Auto-Align 2, not related to Sound Radix Auto-Align, reviewed on page 61) maintains perfect phase lock among all triggered samples and the original audio to preclude any phasiness or flamming.

Drumagog Pro adds other important features not available in the Basic version, including pitch control, a low-latency Live mode for use onstage, and a Stealth mode that processes only louder drum hits. Stealth mode prevents, for example, hi-hat bleed on the snare track from triggering samples. Drumagog Pro also includes a synthesizer tone you can tune and trigger to bolster, for example, the low end of kick drum hits.

Drumagog Platinum includes all of the Pro version's capabilities and much more. Automatic hi-hat tracking is designed to sense, in real time, whether the hi-hat you recorded is open (and by how much) or completely closed; it automatically substitutes a multisample's articulation that matches the real drummer's footwork. VI plug-in hosting lets you open any one VSTi (VST Virtual Instrument) plug-in such as BFD2, Kontakt, or Superior Drummer directly in Drumagog; this immediately expands the sample content available for triggering. Platinum's built-in convolution reverb includes a custom IR (impulse response) library and can also read third-party 16- or 24-bit IR files. If all this is still too tame for you, the new Morph|Engine feature continuously reshapes incoming drum beats to create wild effects.

GRAND VIEW

Drumagog Platinum's reorganized and expanded GUI provides access at once to the file browser, loaded samples, selectable options (such as dynamic tracking), graphic controls for fine-tuning triggering, and a section that shows, in turn, controls for the built-in synth, effects, hosted plug-in, and "main" functions (see **Fig. 1**). Main functions include sliders for adjusting pitch, articulation (called "position" in previous versions of Drumagog), blend (the mix of original sound and replacement samples), and relative levels for any overhead and room-mic samples—up to three stereo samples—included in the currently loaded Gog (Drumagog-format) file. You can use room samples from your own library. Visual controls for triggering samples (sensitivity, transient detail, and so on) have been consolidated into one window pane and function the same as in Drumagog 4. The synth section includes a new, automatable resonance filter.

Store your drum samples wherever you like—the new File Browser sees all your hard drives and keeps track of your favorite Gog files. Moving your samples into Drumagog's sample pool is much easier than it was with past versions: Shift-click multiple samples in the Finder, and then drag and drop the whole lot into Drumagog.

Drumagog Platinum's Auto-Hi-Hat Tracking induces latency that is not corrected by your DAW's automatic plug-in delay compensation. You must slide your hi-hat track earlier in time to compensate for the delay. Fortunately, Drumagog Platinum displays the exact amount of time you should slide your track, removing any guesswork.

The convolution reverb includes controls for selecting the desired IR and adjusting wet/dry mix, size (reverb decay time), pre-delay, and offset (IR start time; see **Fig. 2**). As you increase the offset, early reflections get progressively removed from the onset of the reverb, favoring the diffuse tail that follows.

The Morph|Engine section lets you select a type of dynamic "morph" processing and any of eight variations. You can adjust the frequency spectrum and wet/dry mix by moving a so-called Blob around X/Y axes in a miniature GUI; automate these controls to change your drum sound on every beat.

Each instance of Drumagog Platinum can load and trigger (in real time) one VSTi plug-in (see **Fig. 3**). You set the MIDI Channel and Note number that Drumagog sends to the VSTi plug-in. A Mix control blends the current Gog file with the output of the hosted plug-in, producing layered samples. Drumagog provides a mixer that adjusts the individual output levels and panning of VSTi plug-ins that have multiple outputs.

TESTING, 1, 2, 3

Drumagog's triggering is incredibly easy to set up and works reliably in almost every situation. However, it's not always possible to trigger all the nuances of a snare roll and weed out all mic bleed. In contrast, Slate Digital Trigger offers a leakage-suppression mode that directly addresses bleed issues, allowing for bombproof triggering, but using this mode requires a considerably more complex setup than Drumagog.

Figure 2: The GUI's Effects section includes controls for convolution reverb and the Morph|Engine.

Figure 3: Drumagog's plug-ins section can load any VSTi plug-in you own for direct triggering. The output mixer includes level and pan controls for the hosted plug-in's multiple outputs (if the VSTi includes these).

Used on a hi-hat track playing a simple quarter-note pattern, Drumagog's Auto-Hi-Hat Tracking reliably triggered matching articulations (see **Web Clips 1a** and **1b**). But when the hi-hat played spitfire 16th notes, Drumagog failed to trigger on most of the hits unless I turned Auto-Hi-Hat Tracking off (which defeated automatic switching of articulations).

Drumagog's room and overhead mic samples were apparently recorded in a relatively small space or fairly close to the source. While they fatten up the traps nicely, few of the included Gog files—I counted eight—include these additional samples. That said, Drumagog's close-miked samples generally sound outstanding, especially when lathered with one of the plug-in's better convolution reverbs. Many good electronic-drum samples are also included.

Drumagog Platinum came loaded with 30 IR files: various rooms, halls, plates, chambers, and ambience. Many of these reverbs sounded absolutely fantastic, while others virtually didn't sound at all—the output level was that weak. The convolution reverb needs an output volume control—and not just a Mix slider—to compensate for the wildly differing levels from program to program. The reverbs in V. 5.03b were also buggy: On roughly a third of the stock IRs, certain parameter settings produced a stuttering oscillation.

Selecting Toontrack Superior Drummer 2.2.3 (SD) from a dropdown menu in Drumagog's plug-in-hosting section made SD's GUI open in a separate window and automatically routed its outputs through Drumagog. Easy! With Auto Align 2 enabled, SD's three snare drum samples (top, bottom, and 1176) were perfectly aligned with Drumagog's currently loaded Gog file and the original snare track. I got similarly great results using Drumagog to host Kontakt 3.5.

The Morph Engine produces a wide variety of sounds, including whooshing noise, stuttering repeats, and resonant pitches. You likely won't be using this feature on your next country or rock mix, but you might for electronica. If you use the Morph|Engine, I advise you to follow Drumagog with a compressor to tame some of the excessively dynamic levels this processor can generate.

HIT OR MISS?

Drumagog Platinum V. 5.03b is a mixed bag. It includes more features than any other drum-replacement plug-in on the market, including Slate Digital Trigger. The plug-in hosting works brilliantly, and the Morph Engine provides nice spice for electronic music. But although its triggering performance gets high marks, it's not perfect, and the number of included room and overhead mic samples is paltry. WaveMachine Labs also needs to greatly improve the reliability of Auto-Hi-Hat Tracking and fix the buggy convolution reverbs. Judging by the company's excellent track record, I have no doubt Drumagog will eventually live up to its promise. ✱

EM contributing editor Michael Cooper (myspace.com/michaelcooperrecording) landed a principal acting role in the upcoming movie, The Wait.

B&H
PHOTO - VIDEO - PRO AUDIO

The Professional's Source™

TOOLS FOR CREATION

When it comes to the B&H sales staff, knowledge and experience are a given. We offer unparalleled expertise and solutions for the Pro Audio professional.

Visit our SuperStore in the heart of NYC, give us a call, or browse our newly expanded Web site featuring live assistance, and experience for yourself the most knowledgeable and helpful sales staff anywhere.

420 Ninth Ave, NYC
Visit Our SuperStore

800-947-5509
Speak to a Sales Associate

bhphotovideo.com
Shop conveniently online

QUICK PICKS

ROYER * R-101 RIBBON MICROPHONE

By Myles Boisen

Royer Labs has expanded its line with the R-101 ribbon transducer. At $895 list, the R-101 makes Royer quality available to those who aren't yet ready to shell out for the company's R-121 and SF-1-derived models.

The reduced pricing (about $500 less than the company's flagship R-121) is accomplished by taking a ribbon assembly similar to that used in the R-121 and installing it in an outsourced microphone body that's heavier and larger than Royer's American-manufactured models. The R-101 body is definitely chunky, weighing in at just more than a pound.

The R-101's ribbon assembly is similar to that used in the company's R-121.

The shockmount uses all metal parts and looks and feels very durable. The mic screws into a threaded ring at the bottom of an oversized protective basket that looks like it was designed to hold a condenser mic with a larger body. This basket assembly is suspended by elastic bands within an external ring. This ring is attached to a swivel mount, with an easy-to-grasp lever that loosens and locks the mount. When attached to a boom stand, the shockmounted mic is adjustable in any direction.

The kit comes in a foam-lined black-and-silver mic case. A black-fabric mic sock is included, embroidered with the Royer name. Unlike the more upscale Royer mics, a wooden storage case is not included. A downloadable manual with recording tips is on the Royer website, and includes an interesting tidbit: Royer's ribbon design makes it possible to get a brighter tonality from the mic by orienting the rear of its figure-8 ribbon toward the source. In this placement, a reversal of polarity is required to keep the mic in correct phase response.

I used the R-101 at a number of recording facilities. At San Francisco's Different Fur Studios while producing the Estamos Ensemble, I set up the mic on cellist Teresa Wong. Initially, we had used a vintage condenser mic on her instrument, but I found that the Royer conveyed a superior woody tone, as well as offered greater separation in a room with drums and electric guitar. At Broken Radio, also in S.F., Royer's new ribbon was used to capture the ambience of this Bill Putnam–designed room. During the session for local group Corpus Callosum, the R-101 sounded overly midrangy. But with some subtractive EQ, it was usable in the mix, adding beneficial ambience to a grand piano that was about 6 feet away.

At my Guerrilla Recording studio (Oakland, Calif.), I put the R-101 through its paces. As a room mic routed through a Sytek preamp, the Royer was again weighted toward the midrange in its response, but contributed a layer of classic jazz warmth to a recording of the Jayn Pettingill Quartet. On the group's trombonist, Rob Ewing, I achieved amazing separation by carefully orienting the figure-8 pattern with its null side toward a loud drum kit 6 feet away. The trombone's dynamics are a challenge for any mic, and the R-101 passed with flying colors.

The mic also saw action on trombone and bass trombone as part of a brass quartet recording for composer Aaron Novik. Paired with a Grace 101 preamp, the R-101 delivered bright trombone tones and mellow tonalities that blended perfectly but never lost their presence. In addition, I tried the mic on violinist Irene Sazer during a string quartet recording for Novik's project. The R-101 held its own again on this string session, which employed all ribbon mics, including pricey Royer and AEA models. I would characterize it as a bit bright in this context and probably more fitted for placement on viola than first violin.

I also did some comparison testing of the R-101 to other Royer mics during a rehearsal of the Club Foot Orchestra. Despite minor differences in high-end response, the two R-101s matched well enough for use as a stereo pair. At a distance of about 15 feet from the ensemble, timbre was smooth and natural, with clear highs and a tight low end.

The frequency response of the R-121 timbre was closely comparable to that of the R-101, with some differences on the upper midrange and treble ranges. Compared to a Royer SF-1, the R-101 sounded a bit thin and overly midrangy, although I have to say it had incisive presence and a more contemporary sound than the SF-1. The SF-1 embodies a flatter, classic ribbon signature, with more substantial and detailed response below 250Hz, and more airy highs.

BOX IT UP

Although priced significantly lower than Royer's R-121, the R-101 is a quality ribbon mic characterized by a "ready to mix" signature, with ample presence and less low end than a vintage ribbon. The accessory kit upholds high-quality standards and adds value to this affordable transducer. Those who find classic ribbon mics too dull or bass-heavy as compared to condenser mics, or simply too expensive, would be well-advised to check out this newcomer. ✱

Overall rating: (1 through 5): 4
royerlabs.com

QUICK PICKS

TANNOY ✱ REVEAL 501A

By Mike Levine

Tannoy has been a trusted name in studio monitor technology over the years, and the company recently reintroduced its Reveal line of near-field monitors. The new models, which are designed to offer quality and affordability, include an active 5-inch monitor, as well as active and passive 6-inch models. For this review, Tannoy sent me a pair of the 5-inch 501a models ($179 each).

DISCLOSING THE REVEAL

These monitors have a modern look, with black MDF (medium-density fiberboard) cabinets trimmed with silver. The front panel has an oval-shaped baffle at the bottom and a blue LED power-status light on the front. In addition to its 5-inch woofer, the 501a has a 1-inch soft-dome tweeter and an integrated 60W power amp. The crossover frequency is set at 2.3kHz. According to Tannoy, the overall frequency response for these monitors is 64 Hz to 30 kHz.

The Tannoy 501a, which features a 5-inch woofer, is one of three new models in the Reveal line.

On the back is a rotary volume control. I wish that monitor makers would detent these volume controls so that when adjusted to less than 100 percent, it would be easier to set the left and right speakers exactly the same. The back is also home to a three-position HF Trim switch, which is designed to help you tailor the 501a's high-frequency response to the acoustics of your space. The three-position switch adjusts the volume of the tweeter and gives you three choices: +1.5dB, 0dB, and -1.5dB. The other control on the back is the monitor's power switch. I prefer front-mounted power switches so that you don't have to reach around to turn them on or off, but it's not a major issue.

I began my testing by placing the speakers on the raised back shelf of my studio desk, about 8 inches above the desk surface (essentially meter-bridge height), on top of a pair of Primacoustic Recall Stabilizer speaker pads. As per the manual, I positioned them vertically, which Tannoy says provides the best dispersion. They were about 3 feet from my listening position.

The first thing I noticed when I fired them up was that they were quite bright; too much so for the acoustics in my studio. I turned the HF Trims to -1.5dB, which helped. I played a variety of music through them, and they sounded quite crisp, with the solid, unspattered sound that you usually hear in more expensive monitors.

Considering that they have 5-inch woofers, the bottom end on these monitors was impressive. Whether reproducing the thump of a kick drum or the tonality of a bass (both electric and upright), they did a very nice job. Most importantly, I found them useful for judging the bass level vis-à-vis the rest of the track, which is always tricky when mixing. I left the speakers in their desktop position for a couple of weeks and used them heavily in an acoustic-music mixing project. As you would expect, the more I became accustomed to their sound, the more valuable they were for judging mixes.

I next tried them on speaker stands (equipped with Auralex Mo Pads), positioned 90 degrees to the left of my mixing position, about 5 feet back from me when I turn to face them. I use this auxiliary listening position as a way to hear mixes on alternate speakers in a dissimilar part of the room acoustically. Again, I was impressed with their clarity and solid response. I liked their sound even more at this slightly greater distance.

Overall, despite being a tad on the bright side, the these speakers offer good sound and value. If you're looking for a low-priced monitor with a high-priced sound, they're a solid choice. ✱

Overall rating (1 through 5): 3.5
tannoy.com

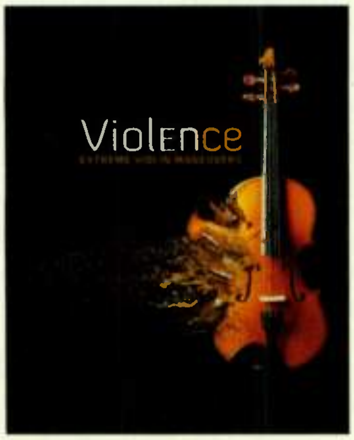
QUICK PICKS **VIR2 INSTRUMENTS * VIOLENCE** (MAC/WIN) By Marty Cutler

VIR2 Instruments' marketing suggests harsh, grating, mangled sounds, but even a cursory audition of the Violence sampler library reveals much of delicate beauty. It harvests sounds from a single violin that are simultaneously ethereal and otherworldly, yet earthy and folkloric. Sound designer Brendan J. Hogan plucks, taps, scrapes, fingers, and bows different parts of the instrument and often subjects the results to intense processing. To be sure, some sounds are ominous and cinematic, but just as many are intimate and warm, and would sit nicely in ethnic pieces and sparse pop arrangements. Even the most processed sounds retain traces of their acoustic origin.

Violence is available as a download from Big Fish Audio ($99.95, bigfishaudio.com). It installs with Native Instruments Kontakt 4 Player. It's also compatible with the full version of Kontakt 4, which I used to audition the sounds. Authorization uses the Native Instruments Service Center, which takes a few minutes at most.

BOW AND SCRAPE

Violence instruments are divided into four categories: Drum Kits, Melodic, Pads and FX, and Tempo Synched. You'll also find 15 Kontakt Multis combining several instruments.



VIR2 Violence is a library of musical but unusual sounds derived from a violin.

The half-dozen patches in the Drum Kits folder span several octaves. Some conform loosely to Roland's drum map, whereas others bear no resemblance to a traditional drum layout, and that's okay. Don't expect anything approaching a realistic drum kit; emulation is clearly not the point of this collection. You'll find kits played with chopsticks, a bow, and a filament. My favorite is the breathy, sometimes wheezy sound of the Bow kit. The instrument's patch panel reflects each kit piece's basic parameters, allowing you to edit as necessary. As always in Kontakt, a click on the Wrench icon opens the patch for lower-level edits.

You would be hard-pressed to identify all of the sources in the Melodic section. You might hear a trace of rosin at the onset of '80s Glass Pad Synth, but little else sounds like a violin. By contrast, Chopstick Zither sounds like chopsticks bouncing on violin strings. At first, this patch sounds tempo-synched, but it becomes obvious that the rhythmic pattern was performed live and with a loose, human feel. Claymation sounds a bit like an electric grand piano, until a breathy, flute-like tone begins to bloom. Metamorph sounds like a refugee from an early '60s sci-fi movie: electronic and acoustic at the same time. The patches make tasteful use of effects without slathering them with excess reverb or delay, and that goes a long way toward creating their overall warmth.

THE JOY OF SFX

You'll find the most outré and impressive sounds in the Pads and SFX folder. Focean is a slowly evolving pad with overtones suggesting muted brass, with a lower-frequency component of woodwinds and bow rosin—overtones come and go in a musical way. Despite the SFX label, these sounds are predominantly tonal, although there are a few atonal, harmonic clouds such as Industrial Labyrinth. Some sounds—After the Rain and the beautiful Desert Garden, for example—are pads with fluttering, rhythmic, and tonal elements. Lunar Gardens develops a chiming, internal melodic motif. Reservoir sounds like a muted-attack organ with some looping drawbar play.

The Tempo Synched folder has some nice pitched rhythms, along with purely percussive loops. There's an excellent mixture of natural and processed sounds here, although I found a couple of anomalies: Bounce Loop appears to play out of sync from G2 on down, although it changes tempo with MIDI Clock. It does allow for a few interesting polyrhythms if your timing is good. Likewise, I had to shift Tone and Overtone a few ticks early to get it locked in. Harmonic Alloy has no discernible looping rhythm, and is simply a duplicate of a patch in the Pads folder.

> Don't expect anything approaching a realistic drum kit; emulation is clearly not the point of this collection.

One of the great benefits of sampling is to provide acoustic sounds that have no equivalent in the real world. I'm greatly impressed by the wealth of fresh, mostly organic sounds in Violence (see **Web Clip 1**). The Multis run from eminently useful (Mallets and Strings) to awe-inspiring (Crop Circles and Hypnotic Rhythm). It's hard to believe that everything I heard came from a single violin. If you're looking for imaginative and organic sounds to round out your instrumental palette, you need to audition Violence.

Overall EM Rating [1 through 5]: 5
Vir2.com

By Mike Levine

There are plenty of inexpensive condenser mics on the market these days, so what does the Blue Spark ($199) have to differentiate it from the pack, besides its nifty red color and lollipop-style capsule? Actually, quite a bit. I've had one in my possession for the past month or so, and I've been constantly surprised by how well it does in just about every situation I try it in.

BLUE TURNS RED

Before getting into details on the mic's sound, lets talk about what you get when you purchase Spark. The mic comes in a fairly large (6 x 9.5 x 9.75-inch), red-stained wooden box with a sliding top and foam compartments for the mic and its accessories: a shockmount and metal windscreen. The shockmount is fairly large and has a threaded cylindrical base that you screw the microphone into. A large thumbscrew with the Blue logo on it adjusts the angle of the shockmount to the stand. On my review model, the thumbscrew didn't always stay tight, and occasionally it would loosen and the mic would slip out of position. According to Blue, the company has updated the shockmount design, which should solve that issue.

The pop filter comprises a metallic mesh screen inside a circular frame, and is just slightly larger than the capsule's surface area. Its lower section is a metallic hoop that sits at a right angle to the screen and slides over the capsule. The hoop section has a thumbscrew on the back that fits into a threaded hole on the rear of the mic, securing the filter in place.

The mic itself has a fixed cardioid pattern and its stated frequency response is 20 Hz to 20 kHz. There is no pad or low-cut filter. The only variable from a sound-pickup standpoint is what Blue calls the Focus Control, a small button on the back of the mic that dials in an alternate EQ curve. When it's engaged, the mic rolls off fairly steeply below 100Hz, which is intended to produce a sound that will sit in the mix better (more on that below). The Focus Control button resides in a recessed part of the mic, just above the hardware that screws into the shockmount, and you need to make a conscious effort to fit your finger into the slot to push the button. I'm guessing that Blue did this intentionally to reduce the chances of accidental switching.

Also included is a printed manual, designed to look like a giant matchbook, in keeping with the mic's fire-related name and color. When you read the manual, it becomes clear that Blue is aiming this mic in part at the novice recordist because there is useful tutorial content on mic placement and technique for both vocals and a number of other instruments. The manual is well-written and is a good example of Blue's attention to detail.

The Spark offers good sound and versatility at a low price point.

SPARK SONICS?

Overall, I was impressed with the sound of the Spark. While it's not going to make you ditch your high-end mics, it's very usable and can certainly hold its own with many mics costing twice its price. If you look at the frequency response graph, it has a boost in the 1kHz region, as well as one from 8 to 12 kHz. It's definitely a bright mic and puts forth a nice, present sound for vocals. There's a dip in the lower midrange, which reduces muddiness.

I tried the Spark on spoken and sung vocals, and liked it for both. I used it for voice-overs on a recent "EM Cast" podcast, and by getting up close and using the proximity effect, I was able to get a big voice-over sound.

I miked a variety of acoustic instruments with it, including acoustic guitar, dobro, mandolin, and shaker, and it provided me with solid results in virtually all cases. If I had to fault its sound, I would say that it lacked a little bit in warmth, at times feeling a tad metallic. For the most part, I was quite happy with the results on all sources on which I tried it.

I didn't find the Focus Control to be useful; activating it yielded thin-sounding recordings. I preferred to keep it off, which resulted in a much fuller sound. Personally, I'd rather EQ a signal after I've tracked it.

For just less than $200, the Spark is a bargain. It can give you solid results in a variety of applications. Blue has succeeded in producing a mic that performs beyond its price range.

Overall rating (1 through 5): 4

www.bluemic.com

By Barry Rudolph

When you record two or more tracks on a single source, the sound waves arrive at each mic (or DI box) at slightly different times. This allows you to obtain a sonic character that's impossible to get with a single mic. However, it can sometimes cause problems, especially with stereo sounds, because some frequencies will cancel out while others build-up unnaturally.

A graphical spectrum analysis (frequency vs. amplitude) of such a sound shows a distinct pattern of steep and regularly spaced peaks and troughs over the entire frequency range. It looks like the teeth of a comb and is called the "comb-filter effect." This phenomenon causes inconsistent and location-dependent frequency response. It occurs in theaters, studios, or living rooms—any place where sound from the loudspeakers reaches your ears at different times.

Sound Radix's Auto-Align ($149) is an AU/RTAS plug-in for Mac (a Windows version is coming) that gives you precise control over time alignment when combining two or more recorded tracks from the same source. When inserted on a track in your mix that you designate as the timing reference, it sends sample-accurate timing information over any one of its own eight internal buses to any other "satellite" tracks also running the plug-in, and then uses the DAW's delay-compensation engine to make the correction. It also detects and corrects phase differences and polarity reversal between the source and satellite tracks. *(Since this review was written, Sound Radix has added a switch to the GUI that makes it easier to compare the original and corrected sounds.)*

LET'S AUTO-ALIGN

I tested Auto-Align in Pro Tools 9 HD on a Mac Pro 8-core running in 64-bit mode. The plug-in installed quickly and was intuitive to set up and use. I tried it on a bass guitar recording comprising separate tracks for a DI and a miked Aguilar amp. I listened to the two tracks mixed equally without Auto-Align, and they sounded hollow and unusable (see **Web Clip 1**). I inserted the plug on the DI track (my reference) and sent out on bus 7. I also inserted it on the Aguilar track, set it to receive on bus 7, played the song, and clicked the Detect button. After a few seconds, it detected and corrected both time and phase differences with dramatic results: The mix of the two tracks was bigger, deeper, and clearer than just the DI or Aguilar track alone (see **Web Clips 2a** and **2b**).

Auto-Align allows you to correct time and phase alignment between two or more tracks recorded from the same source.

Auto-Align also has a polarity-reverse switch that lets you check how a track sounds flipped, but with the time correction applied. Another useful feature is the On/Off switch, which turns off all correction but leaves the phase correlation and spectral meters active. This is necessary because when bypassing on the plug's GUI, you'll hear its processing latency.

CORRECT THE PIANO

I also tested Auto-Align on a stereo Synthogy Ivory piano on which I'd compromised the stereo width to build more center-channel level (see **Web Clip 3**). With Auto-Align correcting phase only, I was able to maintain the original width while getting more punch due to a phase-accurate center image (see **Web Clip 4**).

The plug-in's GUI has a delay display that shows the approximate distance between the two mic sources (tracks), expressed in your choice of samples, milliseconds, or inches/centimeters. The handy Prev and Next buttons allow you to "slide" the timing of the satellite tracks in real time, relative to the reference track—even negative values are allowed. The Way Back Machine lives! To use Auto-Align as an effect, I'd like to see delay time also expressed in musical subdivisions relative to session tempo.

LOCK UP THE ROOM

Auto-Align will time-align a distant room mic on a drum kit to match the close mics. On a multichannel recording, I used the close snare mic as my reference and sent it to a stereo Auto-Align instance on the overheads and a mono instance on the room track.

Correcting the overheads removed a slight delay that smeared the overall kit sound. The room track had a noticeable delay in the kick drum's attack—a doubled attack that softened the overall kick drum sound. The room mic didn't sound bad on the snare so I used the kick drum track for reference and sent its information on another bus to the room track's Auto-Align plug. I ended up with a roomy kit sound with a sharper attack and focus on the kick.

Auto-Align is a wonderful tool for time- and phase-aligning multitrack recordings. It works accurately and automatically, and it takes the guesswork out of the time-consuming process of manually sliding tracks around the time line.

Overall rating (1 through 5): 4.5
soundradix.com

Berklee college *of* music

invites nominations and applications for the position of **Chair of the Electronic Production and Design Department.**

Located in Boston, Massachusetts, Berklee is the world's largest independent music college and the premier institution for the study of contemporary music. The Berklee community consists of 4,145 talented and diverse students, 538 internationally respected faculty, over 500 dedicated staff, and an extended family of alumni, whose 176 Grammy awards testify to their contributions to the music of our time.

The Electronic Production and Design major focuses on a variety of current and emerging musical technology art-making formats and provides immersive, yet diverse experiences in creating music and audio using original electronic sound sources. The creation and manipulation of all sonic elements used in electronic music and sound design is the primary emphasis. The program immerses students in topics such as synthesizer functions and sound/patch design, mastery with MIDI and digital audio production tools, electronic studio recording and mixing, interactivity and controllerism, and electronic production aesthetics. Students learn and apply a wide variety of technical and musical concepts and techniques to a range of creative projects representing career paths of electronic production and design graduates that typically includes music and sound design for video games/film/television, electronic music composition and production, interactive and mixed media, audio programming and software/hardware design, remixing, live performance, and education and graduate studies.

This position is a full-time, 12-month appointment with a preferred starting date of September 1, 2011, but the date is negotiable. Compensation will be both competitive and commensurate with the successful candidate's experience. **Application closing date is March 29, 2011.**

For more details on the position and how to apply, please visit www.berklee.edu/jobs

em FEATURING THE BEST OF

Remix

- Gear/Reviews
- How-To
- Managing Your Career
- Artists and Interviews

www.emusician.com

TO PLACE YOUR AD IN THE ELECTRONIC MUSICIAN MARKETPLACE OR CLASSIFEDS
please call 908-369-5810

EM CLASSIFIEDS

***Electronic Musician* Classified Ads** are the easiest and most economical means of reaching a buyer for your product or service. The classified pages of *EM* supply our readers with a valuable shopping marketplace. We suggest you buy wisely; mail-order consumers have rights, and sellers must comply with the Federal Trade Commission as well as various state laws. *EM* shall not be liable for the contents of the advertisements. For complete information on prices and deadlines, contact **Paul Leifer** at **Paul.Leifer@penton.com** or **908-369-5810**.

ACCOUSTIC PRODUCTS

MASTERING

MIXING

SOFTWARE

ANALOG SYNTH

FURNITURE

SOFTWARE

TAPES, RECORDS, CDS

Create your all-in-one MOTU studio

Run DP7 on the latest updated multicore iMacs, add a MicroBook for studio-grade audio I/O, then customize your system with the latest desktop studio technology.

Digital Performer 7 'Plasma' theme

Apple iMac

The ultimate all-in-one. Turbocharged.

With new Dual-core Intel Core i3 and Core i5, and quad-core Core i5 and Core i7 processors, plus advanced graphics, the new iMac is the fastest, most powerful iMac yet. New Turbo Boost technology dynamically increases the speed of available cores for processor-intensive applications like Digital Performer, virtual instruments, and effects plug-ins for extra performance on demand — in an all-in-one computer that's as beautiful as it is powerful.

XLR mic cable adapter is included.

MOTU MicroBook

Studio-grade I/O that fits in your pocket

Welcome to studio-grade personal recording that fits in your pocket. Plug in your iMac, mic, guitar, keyboard, speakers, and iPod, and enjoy pristine audio performance that rivals expensive interfaces costing hundreds more. You get all the I/O you need for a complete personal studio.

Shure KSM44A

Large-diaphragm, multi-pattern condenser mic

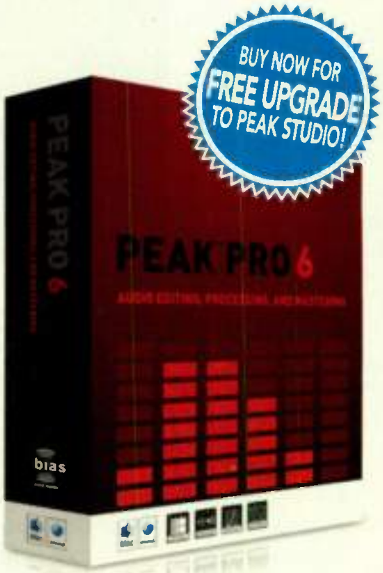
The flagship of the KSM line — and the new must-have mic for any MOTU studio. The KSM44A is a go-to mic you can count on for great results in just about any DP project, with extended frequency response specially tailored for critical studio vocal tracking. The dual-diaphragm design lets you select among cardioid, omnidirectional, or figure-8 pickup patterns.

MOTU

McDSP Emerald Pack v5

Premium plug-in processing for DP7

McDSP's world-renowned emulations of vintage equalizers, compressors, tape machines, and channel amplifiers, combined with custom console modeling, and the latest technology in convolution reverb and mastering limiters, are now available for Digital Performer 7 in the AU-compatible Emerald Pack v5. Winner of a 2009 *Electronic Musician* magazine Editors' Choice!



BIAS Peak Pro 6

Mastering, restoration, and delivery for DP7

Whether you're a musician, a sound designer, an audio editor, a multimedia producer, or a mastering engineer, Peak Pro 6 offers more creative potential than ever before. Used side-by-side or launched directly from within DP7, Peak Pro 6 offers comprehensive restoration, mastering, and delivery tools to streamline your workflow with industry-renowned sonic quality and precision. Peak LE and Peak Pro XT are also available — call your Sweetwater Sales Engineer today!

Genelec 8040A

Active bi-amplified studio monitor

With performance comparable to much larger systems, but in a compact package, the bi-amplified Genelec 8040A is ideal for use in many MOTU studio situations where wide frequency response is needed but space is limited. Use the 8040A for nearfield monitoring in project/home studios, edit bays, broadcast and TV control rooms, and mobile production vehicles.

Expert advice

Call your Sweetwater Sales Engineer today for expert, up-to-the-minute advice regarding every aspect of your MOTU desktop studio. No one knows MOTU better that Sweetwater!

Value Added Reseller
Professional Audio

Sweetwater®
Music Instruments & Pro Audio

(800) 222-4700
www.sweetwater.com

By Mike Levine

photo by Anna Weber

Sandy Vee considers himself a "plug-in geek."

Q&A: Sandy Vee

A French producer hits the ground running in the U.S.

If you charted the progress of French producer/songwriter Sandy Vee's career, you'd see that during the past couple of years, it's been on a remarkable upward trajectory. Last year alone, he had Top 10 *Billboard* hits with Katy Perry, David Guetta, and Rihanna. He also recently produced Pitbull featuring T-Pain, Nee-Yo, and Taio Cruz. Vee is booked solid with projects for months to come.

He gets involved in all parts of a production, from songwriting through the final mix. "I'm a musician first," he says, "and what I love to do is songs."

He's also a gear aficionado, a self-proclaimed "plug-in geek," with endorsement deals from companies such as Waves, GForce, and Softube.

Vee moved from Paris to New York City in 2010, which has helped spur his career growth. "It's like a rebirth," he says about living and working in New York. "It's like a dream, really."

Talk about your studio in New York.
I'm really lucky because I got a really huge loft in Soho. I built a big, big booth; it's totally soundproof. So I'm very happy for now; it's great in the booth. Next year I will buy my own apartment and build a big, big studio.

What DAW do you use, Pro Tools?
I use Nuendo 5. I have Pro Tools 9 for the compatibility. Here in the U.S., everybody is on Pro Tools. What I like in Pro Tools 9 is that I can open it on my laptop, and I can connect everything, all the audio. Then I export AIFF, and in my Nuendo arrangement I have all the parts. And then I can start to arrange, tweak, add, and mix. I feel like Nuendo is so powerful and so stable—I really love it.

What kind of plug-ins are you using?
For mixing, I use two UAD Quad cards with all the plug-ins. They're really great. Also, I'm endorsed by Waves and they gave me the Mercury [bundle]. I love those plug-ins. I'm also endorsed by a company that I love, Softube. They're amazing; they do some great, great, plug-ins. Rob Papen is a really cool guy. I'm working really close [with his plug-ins]. I love SubBoomBass and Predator. And the last one I'm testing for the last few months for my friend Dave from GForce is the ImpOSCar 2. I was just crazy with the ImpOSCar 1. But with ImpOSCar 2, it's the big next step in plug-ins.

Typically, how long does it take you to mix a song?
It depends on the song, but it can be between three and five hours, generally three to four hours.

Do you ever get to the point when you're mixing when you say, "I can't tell anymore, I've got to take a break and come back to it?"
Generally, when I finish a mix. Let's say I mix a song in three hours. I stop and then I want about three days off [from it]. I won't listen to the song for three days. And then I listen to the song and generally don't spend more than 20 minutes [doing final tweaks].

What's the music scene like in Paris?
France is a small country. We have some great talent, some very talented people with good vibes, but the market is really down. A friend told me yesterday that in a week, a Top 10 single in France sells just 500 copies a week. That's just crazy. It's very different working here.

Do you like being in New York?
The first time I arrived in the center of Manhattan, I said, "This is the place I want to live." And today, I couldn't go back to France, I couldn't go back. ✱

World Radio History


MOTU
motu.com
HYBRID
TECHNOLOGY
Introducing the 828mk3 Hybrid
28 x 30 Firewire • USB 2.0 recording system
The new 828mk3 Hybrid audio interface has both FireWire 400 and USB 2.0, so you can connect it either way to any current Mac or PC and enjoy pristine audio quality, zero-latency on-board mixing, studio grade effects, exclusive V-limit™ clip protection, rock-solid DDS™ clocking, bullet-proof drivers, renowned reliabillity, and award-winning sound.
Learn more at motu.com/828mk3
Hybrid technology in a mobile interface...
The UltraLite-mk3 Hybrid
10 x 14 FireWire • USB 2.0 audio interface
motu.com/ultralite-mk3
Hybrid Firewire 400 USB 2.0
16-bus mixer for monitoring and live mixing
Modeled EQ, compression, and reverb
Stand-alone operation with LCD control
Tuner and advanced audio analysis tools
AudioDesk® workstation software
©2011 MOTU, Inc. • 617.576.2760 • 617.576.3609 fax • www.motu.com • info@motu.com • MOTU and AudioDesk are registered trademarks of Mark of the Unicorn, Inc. All other trademarks are the property of their respective owners.