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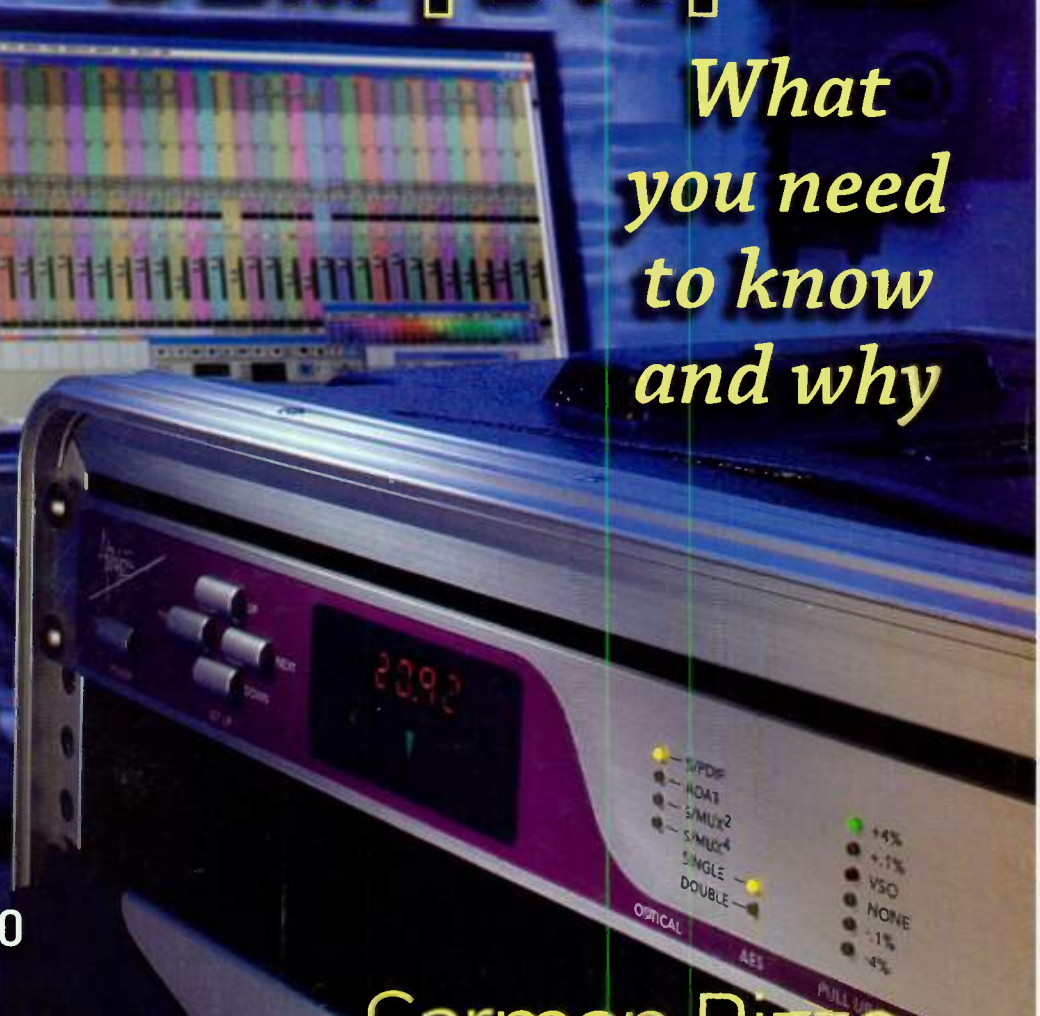

Carmen Rizzo
on film scoring and studio productivity



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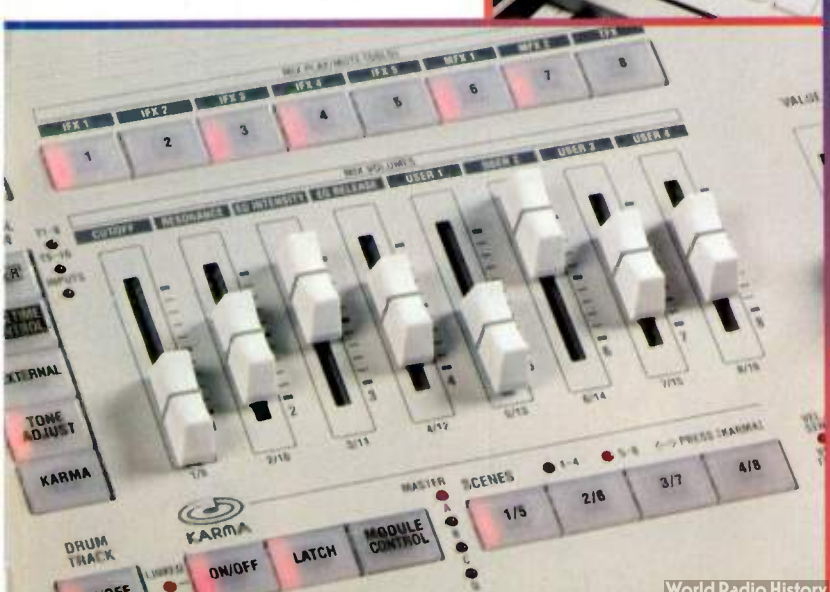


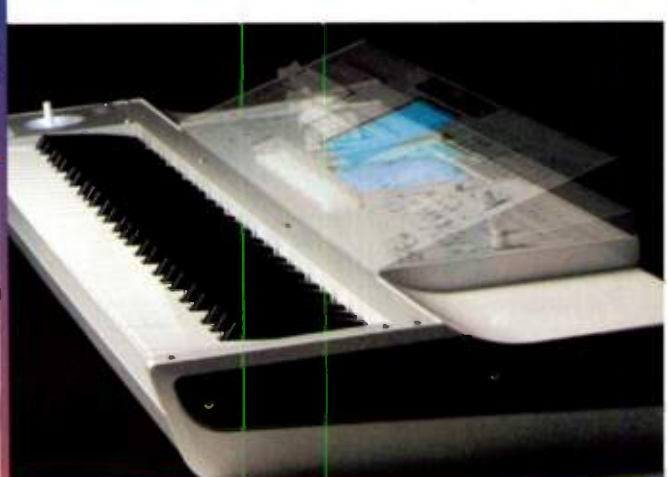

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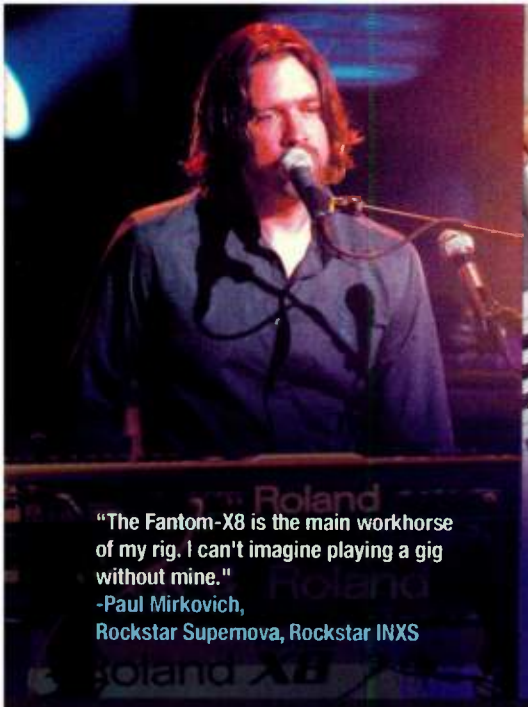

"The Fantom-X8 is the main workhorse of my rig. I can't imagine playing a gig without mine."
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Rockstar Supernova, Rockstar INXS

"The Fantom-X7 has become a main-stay in my keyboard arsenal – I love the old school synths, rhodes' and clavs."
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"The last two Madonna world tours have seen my keyboard rig built around the X7 – it's my meat and potatoes, and it makes programming a dream."
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Roland Fantom X8

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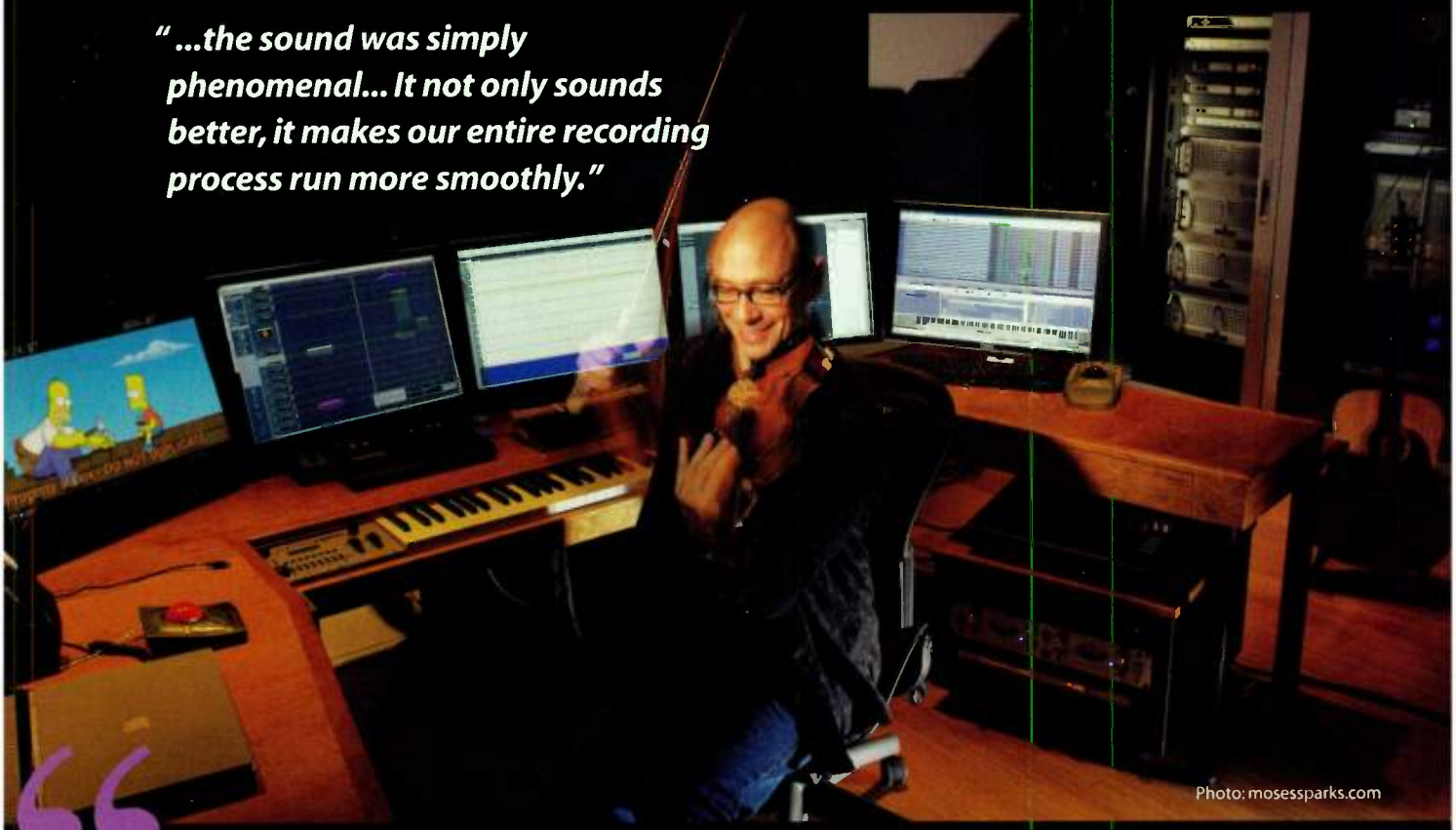
# A Symphony that Scores.

***"...the sound was simply phenomenal... It not only sounds better, it makes our entire recording process run more smoothly."***



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## Michael Levine

Film and Television Composer

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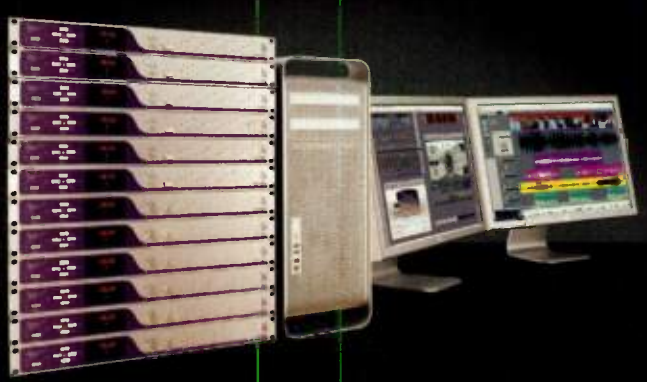

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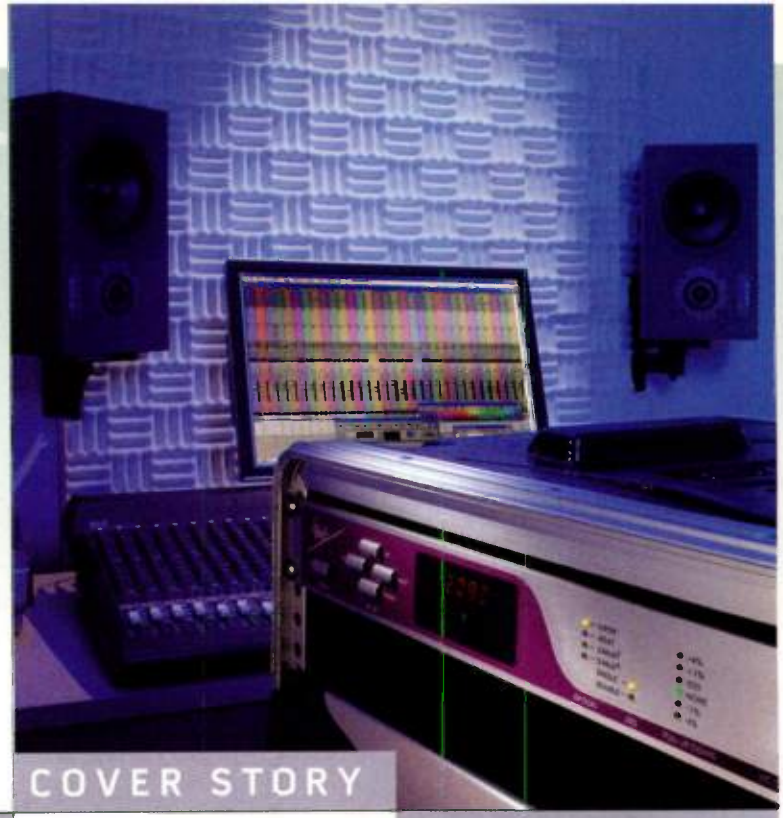

ROBERT PERRY

COVER STORY

## FEATURES

### 29 PRODUCTION VALUES: RENAISSANCE MAN

From producing to engineering to remixing to playing, Carmen Rizzo does it all. His diverse résumé includes clients from Paul Oakenfold to Coldplay to Kate Havnevik. In this interview, Rizzo talks about his production style and gear and offers tips for studio productivity. By Mike Levine

### 34 TIMING IS EVERYTHING

To many personal-studio owners, the subject of digital clocking remains a mysterious black art. EM pulls back the curtain to reveal the fundamentals that will help you understand this essential topic and get your digital gear in sync. By Larry the O

### 51 MASTER CLASS: MIXING EFFICIENCY

The secret to focusing on the creative aspects of a mix is to avoid distraction by being prepared. We provide some tips and tricks for increasing your efficiency when mixing in Pro Tools. By Brian Smithers

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## DEPARTMENTS

Electronic Musician
August 2007

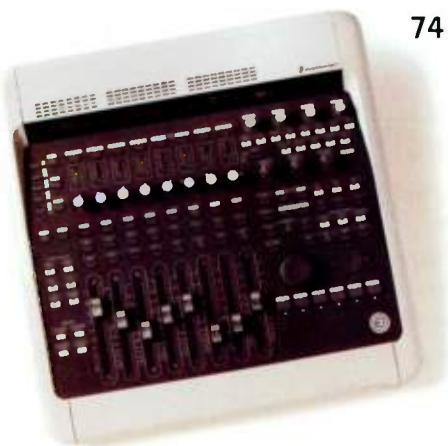
114

# INSIDE

## COLUMNS



74



## REVIEWS



96

96 **QUICK PICKS**

**Audio-Technica ATM450** condenser microphone

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# Happy Trails

"Happy Trails" was the famous theme that Roy Rogers and Dale Evans sang at the close of each episode of *The Roy Rogers Show*. Penned by Evans, it's a song of hope, renewal, and friendship that promises that although we must part, we will see each other again in the next episode. As a lifelong fan of western movies and TV shows, I thought of "Happy Trails" because after nearly 20 years of working on the EM editorial staff and more than 9 years as the magazine's editor, it's time for me to embark on a new adventure, and I want you to join me in the next exciting episode.

Recently, Penton Media Audio Group editorial director Tom Kenny took the next step in a long-planned reorganization of our editorial department. (Penton is our parent company, and the Audio Group includes *Electronic Musician*, *Mix*, *Remix*, and *Music Education Technology*.) Some of the reshuffle is for internal purposes: we now share some editorial resources that were previously being duplicated. But the primary goal is to allow us to dramatically increase our focus on the Web, including a major increase in rich media, such as video and audio. You will soon see the first of many such improvements on our emusician.com, remixmag.com, and mixonline.com Web sites, as well as some entirely new Web products that we're not quite ready to unveil yet.

JULIE BIRUM

Naturally, we'll employ a lot of new technology to create, deliver, and track all this online content, and the staffs of our various magazines will need to coordinate more closely than in the past. Faced with these challenges, Tom decided to appoint, in essence, a "technology czar" to coordinate with our corporate technology folks and to sort out and direct how we cover technology in print and online, what technology we use, and how we can employ that technology to best effect. It's a big job, and as you may have guessed by now, Tom gave it to me. So I'm the new tech czar around this place, with the title Director of Technology, Audio Group.

Of course, I couldn't take on this new and rather daunting job and also run EM's day-to-day editorial operations. Something had to give. So although I still retain the title of editor in chief, and I will continue to be available to the EM staff, I'm turning my day-to-day editorial responsibilities over to newly minted EM editor Gino Robair. I've been helping Gino prepare to take over the reins for the past few years, and he's definitely ready. Gino has some very interesting ideas for improving the magazine and Web site, and he now has the opportunity to implement them. Starting with the September issue, he will take over this column, and he'll share his editorial vision with you directly.

I'm excited about this new opportunity, but I am going to intensely miss being the hands-on leader of EM's editorial team and having a special relationship with our readers. I started with EM and *Mix* in a very different era, before any of the U.S. music-tech magazines were desktop published and before the Web existed. (The Internet had been around for decades, but not the Web.) For years, EM's only email address was my personal email account. When I started working here in November 1987, electronic-music technology was in a relatively primitive state compared with what it is today, and personal studios were strictly for making demos. Now, top pros create highly successful musical projects in personal studios.

I've changed a lot too in the past 19-plus years. Since I arrived, I have held virtually every editorial post at EM, from editorial assistant to editor in chief. I've been writing the "First Take"

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Creativity First

column (formerly called "The Front Page") since the January 1998 issue and have headed up the editorial department longer than any two of my three predecessors combined. I've brought in new people and

SUSAN WILSON

**FIG. 1: Attila the Hun.**

ideas, and I've had a lot of fun doing it. I hope I've been able to help you in your musical endeavors and perhaps given you a chuckle or two along the way; after all, who else around here would publish their dog's picture in the editorial column, as I have occasionally done (see **Fig. 1**)? Now it's Gino's turn.

Lest you get the wrong impression, this may be my last "First Take" column, but it's by no means good-bye, because I'm still very much part of the team here. True, EM's journey continues with a new trail boss; however, I'll be active behind the scenes, helping the EM, *Mix*, and *Remix* staffs create and deliver great print articles, videos, Podcasts, blogs, online activities, and new and exciting products that I can't discuss yet.

Still, my relationship with you is changing. To each and every EM reader, thank you for your support, your feedback, and your loyalty during the past two decades. It has been a great ride; now it will continue in a new direction, and I hope you'll join us. Happy trails to you, till we meet again—online.

**Steve Oppenheimer**
**Editor in Chief**

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LAURA WILLIAMS

# Letters

## Ow, My Freakin' Ears!

In the letter "Lassoing Laskow" in the June 2007 issue of EM, Nik Tyler said, "People can go to somebody's basement and have some hack engineer tear their recording apart and make it sound like utter crap."

Or, they can go into an expensive studio and let a hack like Nik make their music sound like crap. I visited Nik's killer Web site (www.myspace.com/playworkproductions) and couldn't listen to more than a few seconds of anything posted. It was extremely unkind on my ears—harsh, no depth at all, no dynamics. The fact that this is the best his production team can get out of a 15-foot SSL or vintage Neve 8088 room is the best argument I've heard for DIY.

Normally I wouldn't be so brutal, but he brought it on himself with his clueless tirade. It's not about the gear. It's not about the size of the studio. It's all about the person behind the controls. Nik's statements just broadcast his ignorance. I could go on, but I invite EM readers to check out the aforementioned MySpace page and listen for themselves.

**Don Cameron**
**Pittsburgh, Pennsylvania**

## Making Sense of Logic

I have worked with, sold, taught, and authored books on DAW production, and I wanted to thank Len Sasso for attempting to demystify Apple Logic Pro for the rest of us (see "Master Class: Think Different" in the June 2007 issue of EM).

Sadly, he failed, but I don't blame him. The one sure thing about Logic Pro is that Steve Jobs is not even aware that his company puts out such a powerful, flexible, but ultimately user-hostile piece of software.

When Apple bought Emagic, everyone hoped the company would be able to rein in the mad scientists who had so much power over product complexity. Somehow, the sweet but illogical folks at Emagic have managed to fly under Apple's radar and continue to make the most powerful DAW in the world, without understanding that their program is completely unusable to the average audio engineer. And just trying to teach it is another story! As Tony Soprano would say, "Fuhgeddaboudit."

Steve Jobs, please take a tour of Emagic, Apple-fy the software, and make Len Sasso's job easier, or at least a bit more logical.

**Nathan "Adan" Adam**
**Nashville, Tennessee**

## UAD Plugs In

First, welcome back, Larry the O!

I love my UAD plug-ins. I mixed my album with almost nothing but. The vintage EQ and compressor models are distinct and powerful, and the variety makes me feel like I have a world-class studio in my Mac. (For a review of Universal Audio's UAD-1e Expert Pak, see the June 2007 issue of EM.)

However, the DSP card is a major limitation. On the one hand, Larry the O points out the boundaries of the card's latency and processing power, but on the other, the title of his review includes the line "Take a load off your CPU." Well, my Mac Pro tower has four 2 GHz processors. The UAD-1e has a single 1 GHz processor. My CPU doesn't need the help, thank you very much. The only thing slowing me down is this card.

Another gripe is that the UAD-1e conflicts with the Mac Pro's ability to deep sleep. Universal Audio is aware of the issue but has suggested the ball is in Apple's court. That may be—when I boot my Mac Pro in Windows XP, it sleeps with the UAD just fine.

The plug-ins are amazing, but the card is a pain in the neck. Give me native UAD plug-ins!

**Noah Stone**
**Los Angeles, California**

*Larry the O replies: Thanks for your thoughts, Noah. Although my computer is a 3 GHz quad-core Intel Mac, I still appreciate the help the UAD-1e offers. But even though CPU power no longer imposes a significant restriction on track count for most practical purposes, virtual instruments and many plug-ins continue to make considerable demands on the processor. For me, the UAD-1e means that I can run computer-intensive native plug-ins like Audio Ease Altiverb without overtaxing the CPU, as well as throw around "less-expensive" plug-ins (such as basic EQ, compression, and effects) like candy.*

*Your comments suggest that you do not think UAD plug-ins would pose a significant challenge to your CPU. Only Universal Audio really knows the truth of that, but judging from how hard some of the plug-ins work the UAD-1e card, I would have to see some data indicating that to be the case before concluding that we would be better off with native versions.*

*I have no comment on the sleep problem, as I did not encounter it.* EM

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JOHN DILIBERTO

## EMspotlight

### Man vs. Machine: Conny Plank

German producer-engineer Konrad "Conny" Plank helped forge the innovative sounds of Kraftwerk, Cluster, Ultravox, the Eurythmics, and numerous other influential electronic bands. In this interview from the EM archives, Plank talks about composing with industrial percussion, using natural ambience, and exploiting the musical tension between men and machines. By John Diliberto. emusician.com/em_spotlight

## SHOW REPORT

The 2007 Summer NAMM show is one of the largest annual musical-instrument expos in the United States. Visit emusician.com for Associate Editor Geary Yelton's report on the exciting new recording gear, music software, and electronic musical instruments unveiled at this year's show.

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# MRS. MILLS IS COMING TO YOUR REASON RACK

This is Mrs. Mills. She's an ancient Steinway upright piano with a characteristic, out-of-tune sound you've most definitely heard before. Mrs. Mills is one of many unique instruments found at Abbey Road, the world's most famous recording studio.


AbbeyRoad

## Introducing Abbey Road Keyboards

The Abbey Road Keyboards ReFill brings the sound of legendary Abbey Road Studios into your Reason rack. Recorded on location using Abbey Road's recording rooms and original vintage equipment, this ReFill gives you access to seven time-honored Abbey Road instruments, instruments whose sound can be heard on countless immortal Abbey Road recordings.

The Instruments:
Mrs. Mills Piano, Challen Piano, Hammond RT-3 Organ, Mannborg Harmonium, Premier Tubular Bells Schiedmayer Celeste, and the Mellotron M400.



propellerhead SOFTWARE

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By Geary Yelton

## Moog MF-107 FreqBox

First announced in January, the latest addition to the Moogerfooger line of effects modules is now available from Moog Music (www.moogmusic.com). The MF-107 FreqBox ($359) contains a unique voltage-controlled oscillator (VCO) that responds to instrument- and line-level input signals. Instead of modifying the sound of an external audio source, you can use a synth, guitar, voice, or other source to modulate the sound of the FreqBox. Audio signals can apply hard sync or frequency modulation to the VCO, or the VCO's frequency or amplitude can track the input signal's dynamics.


The FreqBox's VCO has a continuously variable waveshape, and it can produce sounds ranging from fuzzlike distortion and bell-like tones to harmonic sweeps and timbral morphing. The unit's rugged stompbox chassis has control-voltage inputs for expression pedals, analog sequencers, and other sources, as well as outputs for interconnecting additional Moogerfoogers.

## Starr Labs Ztar Z7-S

Since 1991, Starr Labs (www.starrlabs.com) has been designing, building, and improving MIDI controllers for guitarists. The newest is the Ztar Z7-S ($1,495), the most affordable Ztar yet. Unlike guitars fitted with guitar-to-MIDI converters, the fully programmable Z7-S doesn't have a traditional fretboard. Instead, it has a touch-sensitive 24-by-6-key fingerboard with 32 independent zones, each with its own channel, transposition, and sensitivity settings. Each key can be tuned separately, and the six Velocity-sensing StringTriggers offer programmable alternate tunings. The Z7-S's software functions are identical to those of higher-priced Ztars.

In addition to duplicating traditional guitar-playing techniques such as picking, strumming, hammer-ons, and pull-offs, the Z7-S has a layout that makes it suitable for triggering electronic percussion or loops. You can use the instrument's embedded arpeggiator to control specific fingerboard zones, and an onboard ribbon controller can be used like a whammy bar for a variety of effects. The Z7-S also has a 2 × 40–character LCD, a programmable volume pot, and jacks for a sustain switch and a volume pedal. A number of options are available, including USB connectivity, a breath controller, a sound card, and custom finishes.

## Korg M3

Korg (www.korg.com) has announced its next generation of workstations by unveiling the M3 (61-note, $3,000; 73-note, $3,475; 88-note, $4,000), a synthesizer and sampler with 120-voice polyphony, 256 MB of waveform ROM, and up to four layers per sample. The 61- and 73-note models have a new semi-weighted keyboard designed by Korg, and the 88-note model incorporates the same RH3 weighted keyboard as the OASYS 88. In addition to tilting the M3's control panel, you can detach the entire module and use it as a freestanding synth; the module is also available without the keyboard as the M3-M ($2,375). The 73- and 88-key models have space for an additional M3-M or a Radias-R. Standalone and plug-in editing software (Mac/Win) is included.

The M3 boasts many features inherited from the OASYS, such as similar voice architecture, low-aliasing filters, AMS mixers, and a multifunction control surface. For real-time performance control, the second-generation KARMA section provides eight Velocity-sensitive pads, eight sliders, and eight switches. The 16-track MIDI sequencer has 480 ppqn resolution and stores 128 songs, each with 150 preset patterns and 100 user patterns. The M3 gives you five insert effects, two master effects, and one Total Effect—all in stereo. Assignable controllers include a joystick, two switches, and a ribbon, as well as switch, pedal, and damper inputs. The 320 × 240–pixel color touch screen also functions as an *x-y* controller. Digital connections include optical S/PDIF, three USB ports, and MIDI In, Out, and Thru; FireWire is optional. Another option is the EXB-Radias expansion board, which gives the M3 an additional 24 notes of Radias-style synthesis and vocoding.

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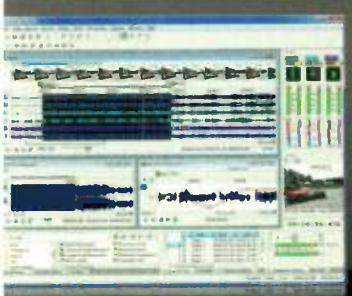
SONY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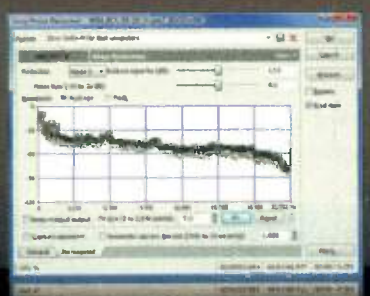
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## PRECISION, STABILITY, EASE, AND SPEED

Sound Forge software is the go-to tool for audio production professionals, and has been since the dawn of digital audio editing. Why? Because it's fast, it's precise, and it's rock-solid dependable. New version 9 is now a full-on audio production suite with included software for CD design, effects processing, and mastering. The Sound Forge 9 suite also has a long list of new features including multichannel audio recording and editing, and Dolby Digital AC-3 encoding. Put it to work in your studio and see why after more than a decade, Sound Forge is still the most comprehensive and efficient audio editing platform you can buy.

**sony.com/soundforge9**

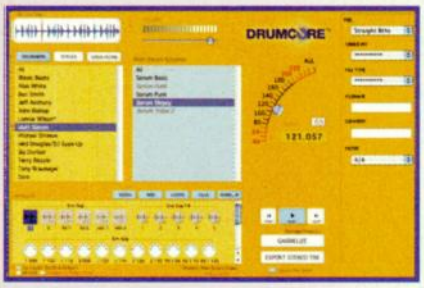
Works with Windows Vista™

## Vir2 VI.One

Vir2 (www.vir2.com) has begun shipping VI.One (Mac/Win, $399.95), a Native Instruments Kontakt Player 2 instrument that does it all. Encompassing 21 GB of sample data on three double-density DVD-ROMs, VI.One supplies more than 2,000 instruments in a wide array of categories, making it an outstanding choice for just about any musical genre you can think of. Because it's built on Kontakt Player 2, it runs standalone and as a plug-in for AU, DXi, RTAS, and VST hosts.

VI.One's content runs the gamut from the world's most popular instruments such as grand piano and guitar to obscure world instruments such as katem and iliili. Construct your own ensembles, from intimate rhythm sections to full-blown orchestras, and save them as Multis. Twenty factory Multis make the most of VI.One's capabilities. You also get a complete General MIDI sound set comprising 128 instruments and 8 drum kits. A complete collection of modern and vintage synths lets you choose from classics like the Jupiter-8, Matrix-12, and Memorymoog. Tempo-synced loops have been arranged into eight categories, and premium drum kits have been individually miked in three positions. All parameters respond to MIDI CCs for hands-on control.


## Submersible Music DrumCore 2.5




Submersible Music (www.submersiblemusic.com) is shipping a new version of DrumCore (Mac/Win, $249), an application for creating audio and MIDI tracks using kits and performances recorded by real drummers. New features include pad swapping, which allows you to assign individual instruments to each pad in a kit; for example, you can combine Alan White's snare and Sly Dunbar's kick with Jeff Anthony's cymbals and John Bishop's toms. Also, you can now play DrumCore as a standalone drum module without using ReWire; just connect your MIDI keyboard or pad controller and play kits from Matt Sorum, Terry Bozio, or one of a dozen other drummers whose content is included. You can queue up loops and fills so they play back-to-back, edit pads and kits in real time, create kits from scratch using WAV and AIFF files, and download additional free content and a demo from Submersible Music's Web site.

To accompany DrumCore 2.5, Submersible Music is also publishing new expansion packs and bundles. If you want traditional Latin and Afro-Cuban drumming, you can choose *LuisPack II* ($79.99), the second DrummerPack from Sergio Mendes percussionist Luis Conte. For hip-hop and R&B styles, there's *Urban GT* ($49), and for rock and funk, Bryan Mantia's *Brain: One Stroke Done* ($79). All DrummerPacks are also compatible with DrumCore LT, a free downloadable player.

## IK Multimedia AmpliTube Jimi Hendrix

IK Multimedia (www.ikmultimedia.com) has launched AmpliTube Jimi Hendrix (Mac/Win, $249; $199 upgrade or crossgrade), its first artist-specific software for guitarists. This signature version of AmpliTube delivers computer models of the amps and effects—some of them rare and collectible—that Hendrix used in the '60s to attain his legendary sound. Created in cooperation with Authentic Hendrix, a company owned by the late guitarist's family, the software furnishes a collection of presets that instantly recall the unique tones heard in songs ranging from "Purple Haze" to "Red House."


AmpliTube Jimi Hendrix gives you five modules that emulate amp heads, speaker cabinets, mics, and more. Nine stompbox effects are based on vintage gear that includes the Univox Uni-Vibe, Roger Mayer Octavia, Vox Wah V846, and Dallas Arbiter Fuzz Face. Four virtual amp heads model a 1959 Marshall JTM100 and Fender Bassman, a Dual Showman, and a Twin Reverb. Seven simulated cabinets include open and closed 2 × 12, 4 × 12, and 2 × 15 models. You get four simulated rack effects too: stereo reverb, parametric EQ, tube compression, and rotary speaker. You also get two discrete signal paths, SpeedTrainer, a built-in tuner, and all the other features that make AmpliTube so versatile. AmpliTube Jimi Hendrix runs standalone or as an AU, RTAS, or VST plug-in.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## Wave Arts Master Restoration Suite

Almost every studio occasionally faces the task of bringing old recordings back to life. To that end, Wave Arts (www.wavearts.com) has released Master Restoration Suite (Mac/Win, $499.95), a collection of plug-ins designed specifically for cleaning up vinyl and tape recordings. The suite comprises four separately available plug-ins that tackle different aspects of restoration, and a combo plug-in that does it all. Master Restoration Suite supports AU, DX, MAS, RTAS, and VST formats at sampling rates as high as 192 kHz; it also supports 32-bit processing with 64-bit EQs.

MR Noise ($349.95) is a broadband noise-reduction processor that minimizes artifacts. With latency low enough for live situations, it removes noise while you monitor the input, noise, and output signal visually and aurally. MR Click ($99.95) provides filtering algorithms to remove clicks, pops, crackles, and scratches. Offering spectrum analysis and brickwall filtering with adjustable frequency bands, MR Hum ($99.95) is optimized to remove rumble, hums, and buzzes. MR Gate ($99.95) is an expander/gate with adjustable look-ahead and comprehensive metering. The Master Restoration plug-in combines the functions of all four individual plug-ins and is available only as part of Master Restoration Suite.

## Download of the Month

### SONICPROJECTS OP-X AND OP-X PRO

OP-X (Win, $99) and its sibling OP-X Pro (Win, $149) are excellent VSTi re-creations of the late-'70s classic synth, the Oberheim OB-X. OP-X Pro even gives you virtual trim pots emulating the touchy oscillator-tuning pots on the original circuit boards. Both soft synths faithfully reproduce many of the charming limitations of the OB-X, such as the reverse-polarity modulation lever, On/Off and Half/Full switches for mixing the oscillators, and a single Depth knob for the amount of LFO modulation applied to the filter and oscillators. If you want to know what using the OB-X was really like, these synths are a good place to start.

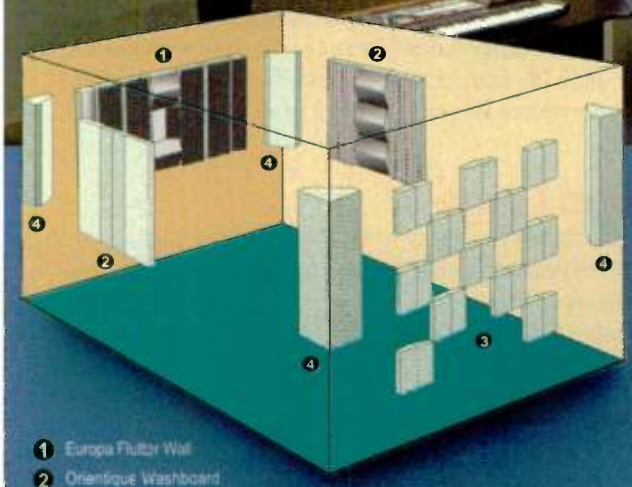

OP-X and OP-X Pro bring a few welcome additions to the original feature set. You can turn on the sawtooth and square oscillator waveforms simultaneously, and when both are off, you get a sine wave, which is useful for frequency modulation. OP-X Pro also features a morphing multimode filter. You can control most of the new features, such as ring, pulse-width, and filter-envelope modulation, using buttons at the bottom of the control panel that replace the original preset buttons (presets are now managed by the VST host). Another example is the arpeggiator, which is actually a cleverly implemented 6-step sequencer.

Unlike the original OB-X, the OP-X models come with hundreds of presets and are fully MIDI compatible. Although there is no MIDI Learn implementation, MIDI Control Change messages have been mapped to all OP-X parameters, and the mapping adheres closely to the Native Instruments Pro-53 mapping, which has become fairly standard. Preset banks cover classics ranging from the OB-X and other vintage synths to presets that show off the new VI features (see **Web Clip 1**). Native Instruments Reaktor 5 users can get a Reaktor Ensemble version of OP-X (Mac/Win, $49), and both synths are now Muse Receptor compatible. You can download demos as well as purchase the synths from Sonicprojects' Web site at www.sonicprojects.ch.

EM WEB CLIPS

*—Len Sasso*

# Take Control of Your Room!℠


1 Europa Flutter Wall
2 Orientique Washboard
3 Scandia Scatter Blocks
4 Australis Bass Trap

**London Z14 Studio**

Face it. Most project studios and post-production rooms are built in typical rectangular rooms. You spend thousands of dollars on gear, only to battle standing waves, flutter echo and all the hash that makes it difficult to get a good mix. Battle no more.

Introducing Primacoustic – a new concept in broadband acoustical treatment that is easy to install, affordable and has the look and performance of an architecturally designed studio.

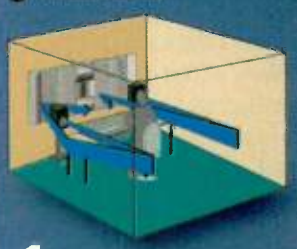
Primacoustic is based on a modular concept whereby precision cut acoustical absorbers are combined to resolve the four main problems common to all square rooms. The Europa Flutter Wall controls 'front-to-back' flutter and works with the Scandia Scatter Blocks to reduce standing waves. The Orientique Washboards reduce side wash and powerful primary reflections. The Australis Bass Trap is a corner wedge that tightens up bass and brings balance into your room.

## London Calling for Under $849 *

The London Z14 is one of more than a dozen studio packages available. London combines all four acoustic systems into one affordable package (London Studios start at $660*). Other packages include the New York Voice-Over Booth, Rio Video Suites, and Montreal Studios. With complete room treatments starting at $425*, no other acoustic treatment is as affordable or so effective – we even include the glue!

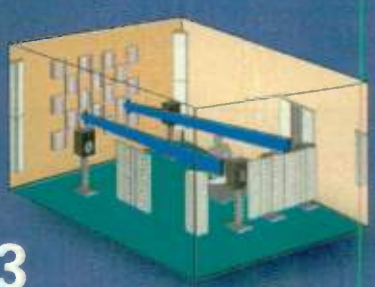
For more information visit www.primacoustic.com or see your local pro-audio shop and...

## Tell 'em you want to take control!

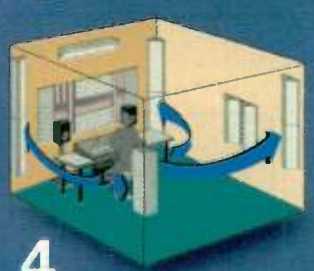


**1**
**Europa Flutter Wall**
Reduces 'front-to-back' echo, standing waves and monitor foldback. Dozens of creative patterns to choose from.
Prices start at $189**.

**2**
**Orientique Washboard**
Absorbs primary reflections and side-to-side flutter. Eight creative patterns to choose from. Priced from $189**



**3**
**Scandia Scatter Blocks**
Affordable alternative to diffusion. Keeps room live and reduces standing waves. 12 creative patterns to choose from. Prices start at $149**



**4**
**Australis Bass Trap**
Effective down to 45Hz, tightens up bass and reduces smear. Can be used in corners or on walls.
Priced at $219** a pair.

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* Street price USD
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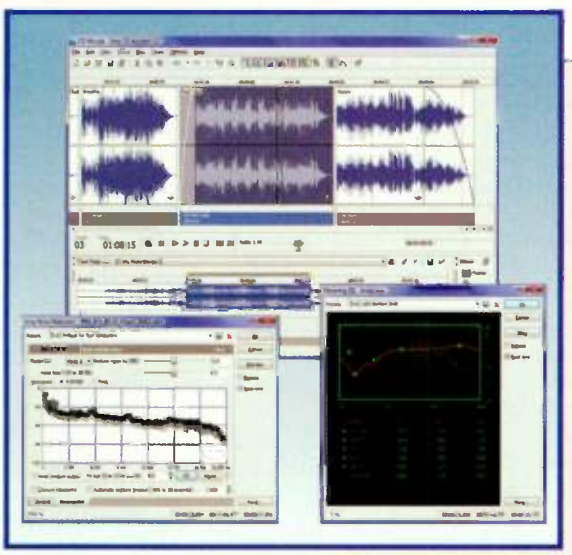
**www.primacoustic.com**

## Dangerous Music D-Box

One solution to the classic dispute over whether analog sounds better than digital is the new D-Box ($1,699), a multipurpose analog-summing device from Dangerous Music (www.dangerousmusic.com). Combining the functionality of a portable monitoring system with an 8-channel summing bus, the D-Box incorporates elements of several products in one rackspace. It has a talkback section, two ¼-inch headphone outputs with independent level controls, and two channels of 24-bit, 96 kHz D/A conversion.

For mobility and convenience, the D-Box is a single-rackspace unit with all controls on the front panel and all connections (except headphones) on the rear. Two analog audio inputs, two stereo digital audio inputs, two analog sum outputs, and four speaker outputs are all on balanced XLR jacks. The D-Box's digital inputs handle coaxial S/PDIF or AES3 signals. The analog inputs let you select either +4 dBu or –10 dBV levels, and the speaker outputs are switchable in pairs. The summing input has a 25-pin D-connector that connects to any 8-channel TDIF (Tascam Digital Interface) source. The cue/talkback section furnishes a built-in mic, a momentary switch, and a TRS input for an external switch. The D-Box is an all-in-one solution for a compact studio setup.



## Sony Sound Forge 9

Well over a decade after its introduction, Sony's flagship audio-editing application (www.sonycreativesoftware.com) is in its ninth incarnation. Sound Forge 9 (Win, $399.95) serves up a feast of features for recording, editing, sweetening, processing, and encoding digital audio and burning audio CDs. Sound Forge now supports multichannel audio files, so you can record multiple sources at the same time, work with 5.1 surround mixes, and drag-and-drop data between channels. Visually monitor your tracks with multichannel spectrum analysis, and easily mix multichannel files down to stereo or mono.

Among Sound Forge 9's new tools and effects is Noise Reduction 2.0, an audio-restoration suite for cleaning up tapes and vinyl. You also get iZotope's Mastering Effects Bundle, comprising four mastering plug-ins for reverb, compression, limiting, and parametric EQ. Other enhancements include GUI color customization, updated markers and rulers, new crossfade options for effects, a greater assortment of onscreen meters, and Dolby Digital AC-3 export. In addition, Sound Forge 9 gives you a direct link to Sony Music Studios Internet Mastering, an online service offering professional mastering by top engineers, starting at $99 per track.

## Apogee Symphony Mobile

If you're an electronic musician migrating to the MacBook Pro platform, you're going to need a fast, high-quality audio interface. And if you want 32 channels of 24-bit, 192 kHz audio, your choice of hardware is extremely limited. Apogee Electronics (www.apogeedigital.com) can fulfill your needs with the new Symphony Mobile ($595), a multichannel ExpressCard designed specifically for the MacBook Pro. With only 1.6 ms of latency at 96 kHz, from analog source in to analog source out, the Symphony Mobile delivers the sound quality and performance of Apogee's Symphony PCI-ExpressCard for desktop Macs.

When installed in your MacBook Pro and connected to any combination of Apogee Rosetta or X-series converters fitted with an optional X-Symphony card ($200), the Symphony Mobile can form the basis of a portable recording system on a par with full-size Mac Pro setups. For compatibility with any Core Audio application, the Symphony Mobile includes Apogee's Maestro software. It also comes with VBus, software that allows virtual routing between any Core Audio applications. EM

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*—Ernie Rideout, KEYBOARD*

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# Mind Control

By Scott Wilkinson

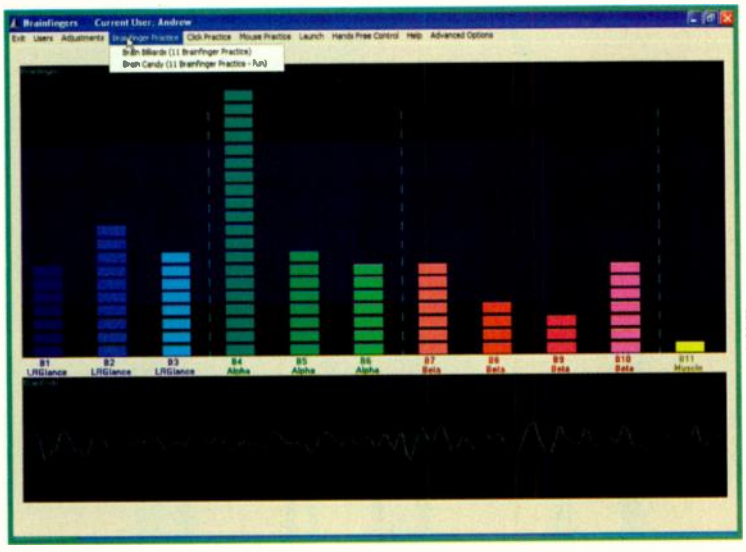
## Hands-free control lets the mind sing.

In the May 2007 issue of EM, you'll find a feature entitled "Overcoming Adversity" written by Joanna Cazden and myself. In that article, we examined some of the ways in which musicians with various disabilities use technology to make music despite their limitations.

Not long after that article appeared in print, I came across another technology that could be of great benefit to anyone with impaired mobility wishing to use computers, including musicians. Developed over the last 17 years by Andrew Junker and available from his company, Brain Actuated Technologies (www.brainfingers.com), Brainfingers is a remarkable system that opens a whole new world of computer access that is normally denied to those with limited or no use of their hands.

The hardware includes a headband with three sensors that make contact with the user's forehead. The sensors are connected to an interface box that amplifies, filters, and digitizes the signals and sends them to a Windows computer via USB. A software driver interprets the signals and uses them to move the cursor, click on icons, invoke macros, and perform other tasks normally accomplished with a mouse and keyboard. The product is designed to be used with a wide variety of third-party software.

The headband sensors detect three types of biometric signals: eye movement (electrooculographic, or EOG), facial-muscle movement (electromyographic, or EMG), and brain waves (electroencephalographic, or EEG), primarily alpha (8 to 12 Hz) and beta (>12 Hz). Patented algorithms separate the different types of signals and further divide the brain-wave signals into distinct frequency bands. Each signal type and frequency band is then used as a virtual finger, or "brainfinger" (see Fig. 1), that can be mapped to control a particular software trigger or continuous parameter.



COURTESY BRAIN ACTUATED TECHNOLOGIES

FIG. 1: Each colored bar represents the signal strength of a particular brainfinger, and the waveform along the bottom is a composite of all three biometric signals. In this example, the lowest-frequency alpha brain wave is the strongest, suggesting that the user was in a meditative state.

Muscular and eye movements can be controlled relatively precisely. For example, Junker's colleague, Chris Berg, a composer and music professor at the University of Cincinnati, programmed the system to use the pressure of his tongue on the roof of his mouth (which involves some facial muscles) as a signal to change notes played on a MIDI synth.

However, users must practice using the system to gain proficiency and precision with it, so the software includes a training program. Once the nuances are mastered, users can often improve on their best reaction times. In a study conducted by the United States Air Force, subjects' reaction times to visual stimuli with the Brainfingers EMG "button" were 15 percent faster than with a physical button.

For musical applications, Junker and Berg have developed several ideas, such as triggering short MIDI phrases with different brainfingers and controlling the tempo of prerecorded MIDI and digital audio segments by flexing the jaw muscles. At the 1999 South by Southwest music and film festival and conference, the lead singer of a Native American group called A Band of Indians wore a Brainfingers headband and affected the playback of some MIDI synths while singing.

According to Junker, the most important thing about Brainfingers is that it measures both brain and body neural impulses, reflecting the complex and intimate interaction between volitional muscle movements and mental/emotional states. Junker claims that when applied to music, the effect can be quite profound for performers and listeners alike. The system can also be used as an advanced biofeedback device, inducing calmer mental states as soothing music is played in response to relaxing the facial muscles and lowering brain-wave frequencies. The potential of this technology is vast for both mobility-impaired and able-bodied musicians who want to explore how music, mind, and body can interact in entirely new ways. EM

# Creative Explosion

This is Live 6, the latest version of Ableton's award-winning software that composers, producers, DJs and musicians worldwide have taken to heart. Live now includes a versatile, comprehensive collection of sounds ready to play and inspire - from faithfully sampled acoustic and electric instruments to impressive electronic creations. Pre-configured controls let you play expressively without worrying about technical intricacies, or, if you prefer, you can dig deeper and explore endless possibilities for creating your own unique and personal sounds.

**Check it out at www.ableton.com.**

version 6

live

ableton

# SOFT SYNTHS vs. HARD SYNTHS


# THE DEBATE ENDS HERE.

Plug-ins give you endless virtual instruments and effects, but they punish your CPU, and playing live with a computer is risky.

Hardware synths and modules give you fast-access presets for playing live, but you're stuck with one set of sounds.

Why choose? Receptor is a reliable hardware unit that lets you store, mix and play plug-ins with or without a connected computer.

**Instant access for live sound**

Plug in a MIDI or USB keyboard, or plug a guitar into the instrument input, connect Receptor to a mixer or amplifier and you're good to go. Use Receptor's 16-channel mixer to sculpt your plug-ins into patches you can recall instantly using the front panel controls.

**More processing power for your studio**

Connect to your computer's DAW via MIDI, analog, S/PDIF, and/or ADAT, and your plug-ins are available on 10 outputs and 4 inputs. Or use UniWire™ to get 16 channels of I/O using a single Ethernet connection, which lets you open plug-ins right inside your DAW. Either way, your computer breathes easy while Receptor does the heavy lifting.

**Plays pro plug-ins like Ivory, Atmosphere, EWQLSO Platinum, and Komplete 4***

Every Receptor comes with $400 worth of plug-ins free, and you can also order your Receptor loaded with Native Instruments Komplete 4. Receptor runs plug-ins from Applied Acoustics, Spectrasonics, Synthogy, East West, Toontrack, IK Multimedia, FXpansion, Big Fish Audio, and Garritan* to name a few.

**Hardware or software control—your choice**

Control Receptor's mixer from the front panel buttons or via the graphic mixer on your computer. You can even plug a VGA monitor, keyboard and mouse directly into Receptor.

**Find out more at www.PlayReceptor.com or visit an authorized dealer today.**

## MUSE RESEARCH RECEPTOR

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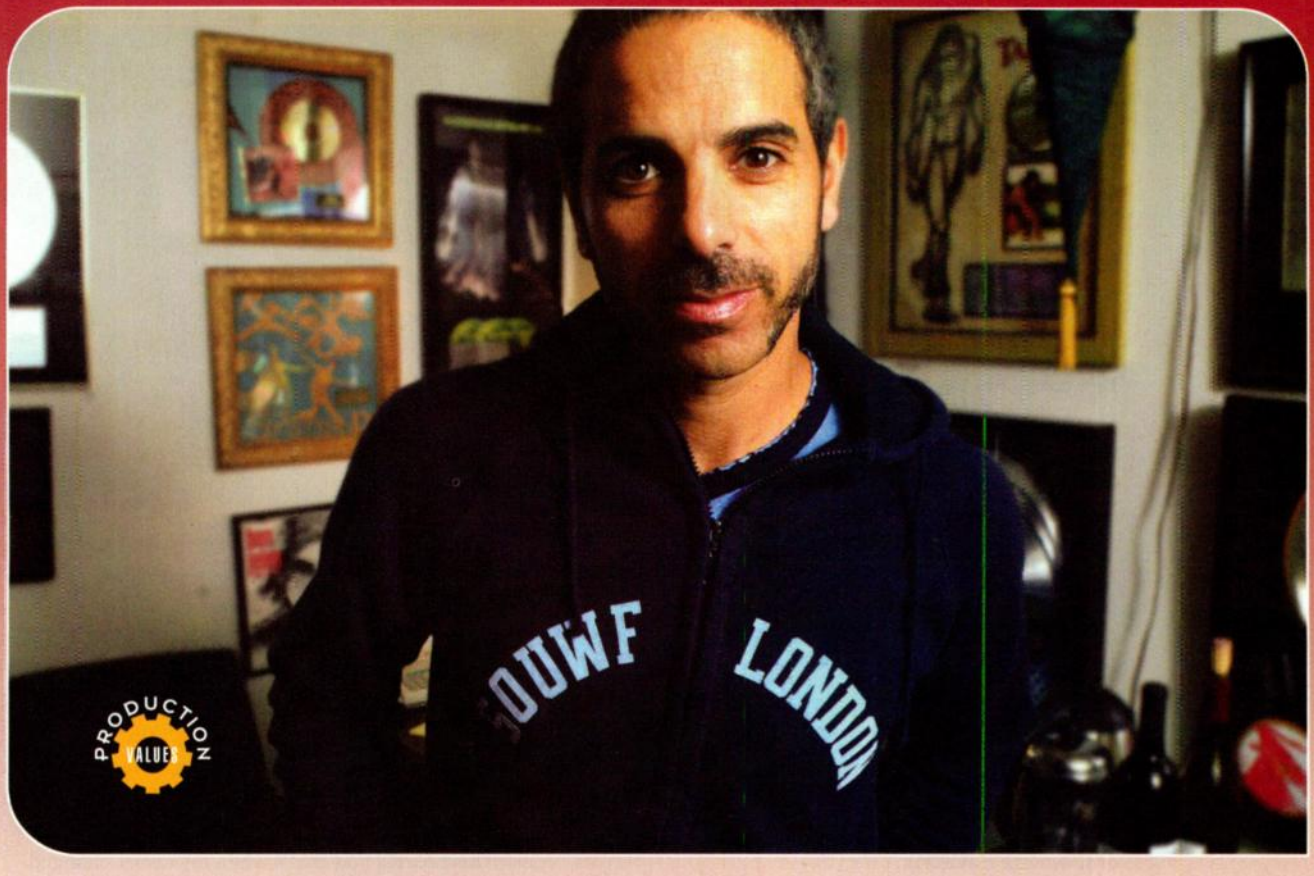



JASON VAUGHN

# Renaissance Man

By Mike Levine

Whether producing, mixing, playing, or programming, Carmen Rizzo excels.

Producing, engineering, mixing, remixing, playing keyboards, programming—you name it, Carmen Rizzo does it. What's more, the genres that the multitalented Rizzo works in are as diverse as his job titles. Electronica, world music, rock, R&B, and pop are all styles he's comfortable with. His eclectic credit list (see "Carmen Rizzo: A Selected Discography") includes artists ranging from Paul Oakenfold to Coldplay to Grant Lee Phillips to Kate Havnevik to Jem to Seal; he once even mixed a Ray Charles project.

Rizzo has a solo career as well. His latest CD is called *The Lost Art of the Idle Moment* (see Fig. 1). In 1999 he put out a CD called *Life in Volcanoes*, along with singer Christina Calero, under the band name of Povi. In his "spare time," Rizzo is on the Advisory Council of the Producers and Engineers Wing of the Recording Academy.

When I interviewed Rizzo by phone for this story, he was somewhere in New Mexico, on the road with Niyaz, a world-music-influenced group he belongs to that also includes Persian vocalist Azam Ali and multi-instrumentalist Loga Ramin Torkian "We've toured on and off for the last two years from all over Europe to India to Tokyo to Turkey to Montreal. We've done some really amazing gigs," Rizzo says. "I'm very, very proud of the project. It's sort of a Middle Eastern electronic project." But he adds that he's "trying to fade away from it" because it takes up too much of his time and he's got so much else going on.

**You've worn many hats so far in your career. What's your favorite one?**

I would say that producing is really how I make my living, and it's what I enjoy the most. That's parlayed into remixing a lot of records, and participating as a musician and programming and

such. But it's evolved into being a recording artist. The reason I made the Povi CD, and my current solo record, was not to hopefully become a rock star or anything; it was really just to get noticed more as a producer. So people would notice my music as an artist, and a producer, and would hopefully give me more work as a producer.

JASON VAUGHN

Rizzo recommends that personal-studio owners get as much RAM and disk space as they can to take full advantage of the power of their computers.

**But your solo career turned into more than that.**

It kind of started out when I was doing the Paul Oakenfold record *[Bunkka]*; my manager at the time said, "You know, you're doing all these tracks, and this and that. You should do another artist record." And it just kind of evolved that way. I'm fortunate enough that I have a very eclectic career. I've kind of purposely handcrafted that, and I've met a lot of wonderful people, done a lot of cool records, and so when it came time to do my artist record, it was convenient and easy to reach out to people that were credible, like a Grant Lee Phillips, Esthero or a Ladybug Mecca, or even Jem or now Kate Havnevik.

**When did you start working professionally in music?**

Well, I would say that I probably got my start in 1984 at Westlake Studios [in Los Angeles] as sort of a gofer. But I was working professionally in '89.

**So you were there in the early days of MIDI.**

Oh, absolutely.

**And you saw all the technological changes, and the evolution of digital audio.**

Yes, I'm 43 now. I showed up in Los Angeles in 1984 with $1,000, and I didn't know anybody. I was lucky to get the job at Westlake Studios, which of course was very primitive. I was fortunate to be around, seeing the first 3M digital tape machine, being schooled on analog tape machines, all the obvious old consoles. I can remember seeing a [Yamaha] DX7 for the first time, a sampler, all of that stuff. I feel old when I talk to kids.


FIG. 1: One reason that Rizzo put out his latest solo CD, *The Lost Art of the Idle Moment*, was to call more attention to his production skills.

**So in the beginning you were an engineer?**

Yes. I was schooled as an engineer, and then I was fortunate to start to mix records. And then where I think my career changed was when I started working with [producer] Trevor Horn. In the early '90s, I was working for Seal and Trevor, and I was going to London a lot. Trevor was way ahead of the curve. So I was involved with Euphonix consoles and samplers and all these wonderful keyboards, and that's when my career changed.

**Was Horn your mentor as a producer?**

Yes, because I was around him for so many years engineering and programming and mixing for him that I was seeing how he made records. I was such a fan of his work.

**Is there a signature aspect to your productions?**

For one thing, I pride myself on instrumentation. And sound selection is very important to me. I've really tried to create a very cinematic, lush sound, but one that is very complementary of modern electronics and organic instrumentation. And I think that that's something that I learned from Trevor. If you listen to a lot of his records, whether it's Frankie Goes to Hollywood, or Seal, or Yes, or any of those, there's a good texture of modern electronic instrumentation and beautiful organic instruments. And that's something I've tried to model my style after, which is to be very creative in the sound selection.

**Are most of the people you work with individual artists, rather than bands?**

Well, yes. I would say that. I don't work with as many bands as other producers do. I definitely have, though, but usually I'm sort of the person to go to when somebody wants to be a little bit edgy or a little bit different.

**So you're running the show as far as the arrangements and instrumentation?**

Absolutely. If somebody wants a safe record, they don't go to me. I pride myself on doing something a little bit different where the instrumentation is different, the arrangement is different. Something almost a little European sounding instead of U.S. sounding. Another great thing that I bring to the table is that I'm well traveled and I've done a lot of world-music albums. I think that's something that people recognize now in my work, from making a lot of records in France: a lot of world-music elements. So when I choose the instrumentation of records that I produce, I try to bring in instruments that the artist might not know of, or they've never heard, or they wouldn't put those two instruments together.

**If you're mixing electronic and organic, you have a large palette to choose from.**

Absolutely. And I think that was why with Niyaz, the three of us worked so well. You know, trying to blend and to make a sort of fusion of East versus West. Often, those types of records can be really cheesy. It was not easy to make it tasteful.

**What are the most challenging situations you run into when producing?**

Choosing the right songs. That's something that's not easy. The first thing a good producer does is to choose the best songs for the artist. That's something that's always a challenge: trying to convince the artists that certain songs shouldn't be recorded.

JASON VAUGHN

FIG. 2: Inside Rizzo's project studio, which is located in a commercial building in Hollywood. To enforce separation between his work and family life, he opted against having a studio in his home.

**So typically, they'll come to you with demos and you have to pick from them.**

Yes. They might say, "These are my demos." And I'm somebody who will be the first to say, "I'm the wrong guy for this song. I think you'd be better suited with someone else." Another thing that's hard to relay to an artist is when they say to you, "I need a radio hit" or "I need a chart hit." And I always say, "You're talking to the wrong guy." Because that's not what I do. I do what I feel is the best for that song for this artist. If it happens to be a hit, then great.

**Do you have a home studio?**

I refuse to *[laughs]*. My philosophy is: when I go home, I'm at home. I've got children. I work enough as it is, so when I come home I want to just be at home. So I have a studio outside my house [see Fig. 2; also see the **online bonus material** at www.emusician.com] that's in a wonderful building in Hollywood that's becoming a music mecca. There are quite a few people in the building. Dave Stewart [from the Eurythmics] is right below me; Glenn Ballard; Michael Danna, who's a big film composer. It's right in the middle of Hollywood, and I get inspired walking out on Hollywood and Vine.

**I understand you recently did your first film-scoring project.**

## CARMEN RIZZO: A SELECTED DISCOGRAPHY

Azam Ali, *Elysium for the Brave* (Six Degrees, 2006); producer, musician, cowriter

Rachel Fuller, *Shine* (Candycat Ltd., 2006); producer (with Pete Townshend), programmer, mixer

Kate Havnevik, *Melankton* (Continentica Records, 2006); mixer, producer, programmer, musician

Coldplay, "Speed of Sound" from *X&Y* (Capitol, 2005); engineer

Niyaz, *Niyaz* (Six Degrees, 2005); band member, producer, cowriter, musician

Carmen Rizzo, *The Lost Art of the Idle Moment* (The Lab/Universal, 2005); musician, writer, producer, engineer, mixer

BT, "The Great Escape" from *Chillout—A Nettwerk Escape* (Nettwerk, 2004); remixer

Delerium, *Chimera* (Nettwerk, 2003); producer, cowriter, engineer, programmer

Alanis Morissette, *Under Rug Swept* (Maverick, 2002); programmer, keyboardist, engineer

Paul Oakenfold, *Bunkka* (Maverick, 2002); producer, writer, programmer

Ekova, *Space Lullabies and Other Fantasmagore* (Six Degrees, 2001); producer, engineer, mixer, programmer

Perry Farrell, "Song Yet to Be Sung" and "Shekina" from *Song Yet to Be Sung* (Virgin, 2001); mixer

Grant Lee Phillips, *Mobilize* (Zoë/Universal, 2001); producer, engineer, mixer, programmer

*Planet of the Apes* theme (20th Century Fox, 2001); remixer (with Paul Oakenfold)

Supreme Beings of Leisure, *Supreme Beings of Leisure* (Palm Pictures, 2000); producer

Povi, *Life in Volcanoes* (Nettwerk, 1999); band member, producer, cowriter, musician

Robbie Robertson, *Contact from the Underworld of Red Boy* (Capitol, 1998); engineer, coproducer, mixer, programmer

Seal, *Seal* (Sire, 1994); cowriter, engineer, programmer

Ryuichi Sakamoto, *Beauty* (Virgin, 1992); engineer, mixer

I did. It was a film called *The Power of the Game* (Pathé Pictures International, 2007), directed by Michael Apted, who's a big British director.

**What was the work flow like for that project?**

They would send me a QuickTime [movie], I would load it into Pro Tools, and I would score it to picture and then send an MP3 to them for approval. Once it was approved and I would score any sort of adjustments, then I would post it on my server as an AIFF file, and it was done. The only negative thing I would say was that it was before I had my Xeon Intel Mac. And I had a G5, 1.8 GHz single. Boy it was hard, because I was running picture, virtual instruments, audio, live MIDI, and recording at the same time. I would say a prayer every time I hit play. It was not easy. That poor Mac, God bless it, was just huffing and puffing.

**You've noticed a big difference with the Intel Mac?**

Oh my God. Praise Intel for getting in bed with Apple, because it's made such a huge difference.

**So you never have to worry about how many tracks and plug-ins you have going?**

No. With my Intel Mac and my two Universal Audio cards, I'm like a kid in a candy store. I've got my TDM stuff, I've got my UA stuff, I've got my RTAS stuff—I couldn't be happier. And it's all thanks to Intel.

**If you had to give a few pieces of advice to people who are producing songs in their own studios, what would they be? In terms of production techniques, things to watch out for or home in on, etc.**

One would be monitoring. I think people don't monitor well enough.

**You mean like having good acoustic treatment so that they hear things accurately?**

Having a good set of monitors and trying to position yourself in a good listening environment. I think that's something that's overlooked. The second thing is, if you make music through or with a computer, you have to invest in power. People think they just need to get a computer, but then if they have no [extra] RAM, no drive space, they're shooting themselves in the foot.

**Power in the computing sense.**

Yes. Having a powerful engine. The third thing, which is something that I've always prided myself on, is, you have to take every artist seriously. I could give you countless stories of discovering artists who could not get arrested, who I took seriously because I believed in them, and who have become very successful stars. One example is an artist called Jem, who is very successful, probably sold a million records around the world. I couldn't get her arrested, I couldn't help get her a record deal. But I worked with her because I believed in her.

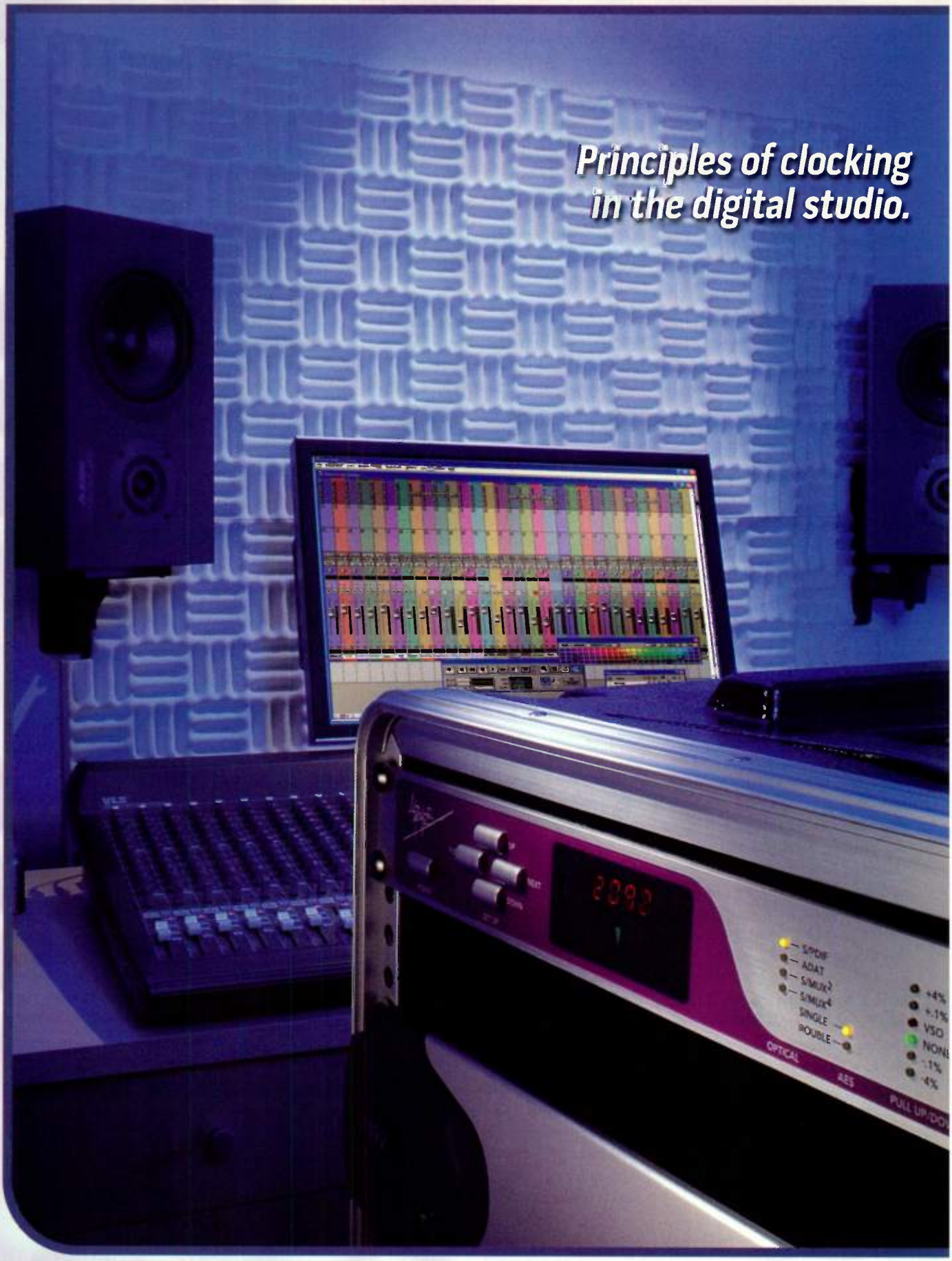
**Any other advice?**

Know your equipment the best that you can. I find often, a lot of my colleagues have a museum of gear, but they don't know how to use any of it. They have all this gear, but they end up using one thing. If I look back at my past, I made my best records when I had an MPC, a keyboard, and a sampler. And I knew those three boxes like the back of my hand. Know the gear as well as you can before you buy more gear. If you're on a desert island with a drum machine and a keyboard, you'll be making great music. But a lot of these guys have, like, 50 plug-ins and all this stuff, but they don't know how any of it works. EM

To hear an additional interview with Carmen Rizzo, go to the Podcasts page at www.emusician.com.

*Mike Levine is an EM senior editor.*

# Principles of clocking in the digital studio.




ROBERT PERRY

By Larry the O

# Timing is Everything

Although digital clocking is not a particularly sexy subject, it is certainly an important one. Most project studios today are built around a computer running multitrack recording software with audio interfaces and other outboard devices connected to it. Computers operate on a step-by-step basis, and precisely timed execution is an essential element of digital recording. Making sure hardware and software all march in lockstep is crucial, and that's what clocking is about.


Despite clocking's importance in the digital studio, its fundamentals are unclear—or just plain voodoo—to many recordists. One of the main objectives of this article is to convey enough about the subject of clocking that you feel equipped to contact a manufacturer and ask the right questions to get the information you need to maximize your studio setup. Although I probably can't make the entire subject simple for you, I will try to make it at least understandable. I can provide information that is important for you to have but that is not readily available elsewhere.

To help get the lowdown on this broad and tricky subject, I spoke with several experts. I'd like to recognize and thank them here (in alphabetical order): independent consultant Joe Bryan, formerly Universal Audio's head of engineering and architect of UA's 2192 Master Audio Interface; Jim Cooper, director of marketing, and Jon Foley, technical-support supervisor, at MOTU; Doug Ford, engineering supervisor at Skywalker Sound; Gannon Kashiwa, Digidesign's market manager; Igor Levin, sync guru at Antelope Audio; Dan Phillips, product manager at Korg R&D; and Lucas van der Mee, senior design engineer at Apogee Electronics. All contributed great insight and technical detail, without which I could not have formulated a coherent picture of the world of digital clocking.

## Time to Begin

As with most digital technology, clocking involves numerous subtleties about which experts have differing opinions. In most cases, though, grasping a handful of fundamental principles is sufficient. I will present these principles here and fill in some other details along the way.

The first and most important principle you need to know about using clocks in digital studios is that there can be only one master, to which every other device must be slaved. Uncompressed digital audio plays at a fixed rate. If two devices in a signal chain are set to the same sampling rate, but each runs off its own internal clock,

they will sample at the same rate, but the odds are infinitesimal that they'll do it at the exact same time. Digital audio streams are delicate; if discontinuities result from two devices having different ideas about when sample time is, the outcome is likely to be a pop or glitch.

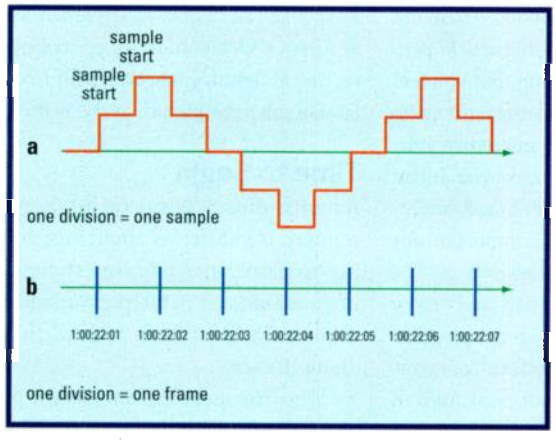
If, however, every device in a system is locked to a single clock source, then samples will be taken, passed around, and output in beautiful synchronicity, and audio will play without ugly artifacts. When the kids play nice together, life is sweet, but of course, almost nothing is ever quite that easy. Things can still go wrong, and I'll examine some common problems once I've explained the basics.

The second fundamental principle of clocking is the distinction between rate synchronization and location synchronization, a frequently misunderstood concept (see Fig. 1). The word *clock* usually indicates something that tells the time, but a clock in the digital studio marks only the rate at which samples should occur. Like a metronome, it carries no information—nothing to distinguish one clock cycle from another. With no concept of elapsed time in the signal (beyond the duration of a single sample period), it is impossible to state a location for any clock cycle relative to any other. Clocks in the digital studio are strictly timing related. In contrast, location synchronization systems like timecode embed an address for each frame of data, making it possible to locate to any specified point.

The third important principle is that digital audio clocks are crucial only in real-time contexts. Information being distributed in real time requires timing to be preserved, but by definition, timing is not important in non-real-time transmission. Thus, for example, a simple file copy from one hard disk to another does not entail clocking.

## Self-Clocking Systems

There are two kinds of systems for delivering clock signals to digital devices in a studio: self-clocking systems and distributed systems, better known as word clock. Self-clocking schemes embed timing data into a real-time digital audio stream. The self-clocking interfaces most familiar to readers are AES3 (also called AES/EBU), S/PDIF, and ADAT Optical (also called Lightpipe). S/PDIF can use the same Toslink optical ports used by Lightpipe, but the data streams are totally different. In this article, *S/PDIF* refers to transmission over unbalanced coaxial and not optical lines.




CHUCK DAHMER

FIGS. 1a and 1b: In 1a, sample start times (and therefore rate) are defined, but nothing differentiates one sample from another. In 1b, a timecode address identifies each frame; thus, positional clocking allows location to any frame.

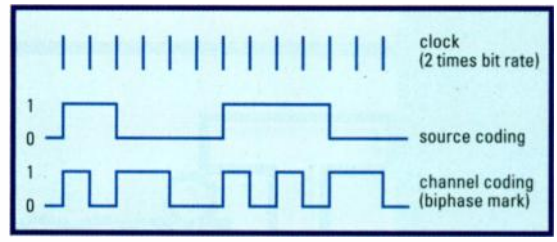



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FIG. 2: A chart from the AES3-2003 standard illustrates how data and clock are encoded in a self-clocking interface. Here you see the clock signal on top, the audio data in the middle, and the resulting biphase mark coded data on the bottom.

In self-clocking systems, the receiving device must extract the embedded clock signal from the digital audio. This is a critically important process with significant impact on the amount of *jitter* (distortion caused by irregular timing, which I'll discuss later in detail) in the system. Self-clocking systems usually run at some multiple of the sampling rate; for example, AES3 runs at 64 times the sampling rate (see Fig. 2).

AES3 and S/PDIF incorporate the clock signal into the data stream as part of the encoding scheme, which is called *biphase mark code*. Without going into a full explanation of this coding, the start of every bit in the stream is marked by a state transition. These transitions (it makes no difference whether a transition is from high to low or low to high) are detected and extracted to create the clock. Lightpipe, however, uses NRZI (non-return-to-zero, inverted) coding, which does not have a transition at the beginning of each bit, making clock recovery more difficult.

Some experts describe audio sent over FireWire (aka IEEE 1394) and USB as self-clocking, but there's a major difference between those two systems and the others I've described. In AES3, S/PDIF, and Lightpipe, samples are streaming in real time, with the clock marking the start time of each sample. FireWire and USB, on the other hand, break the stream of samples into packets of data and transmit them sequentially. The packets are received by buffers in the receiving device, reassembled into the original sample stream, and then sent from the buffer at the appropriate time, as determined by the device's sample clock—a process called *reclocking*. Because the samples are reclocked, FireWire and USB are not dependent on embedded clocks for timing.

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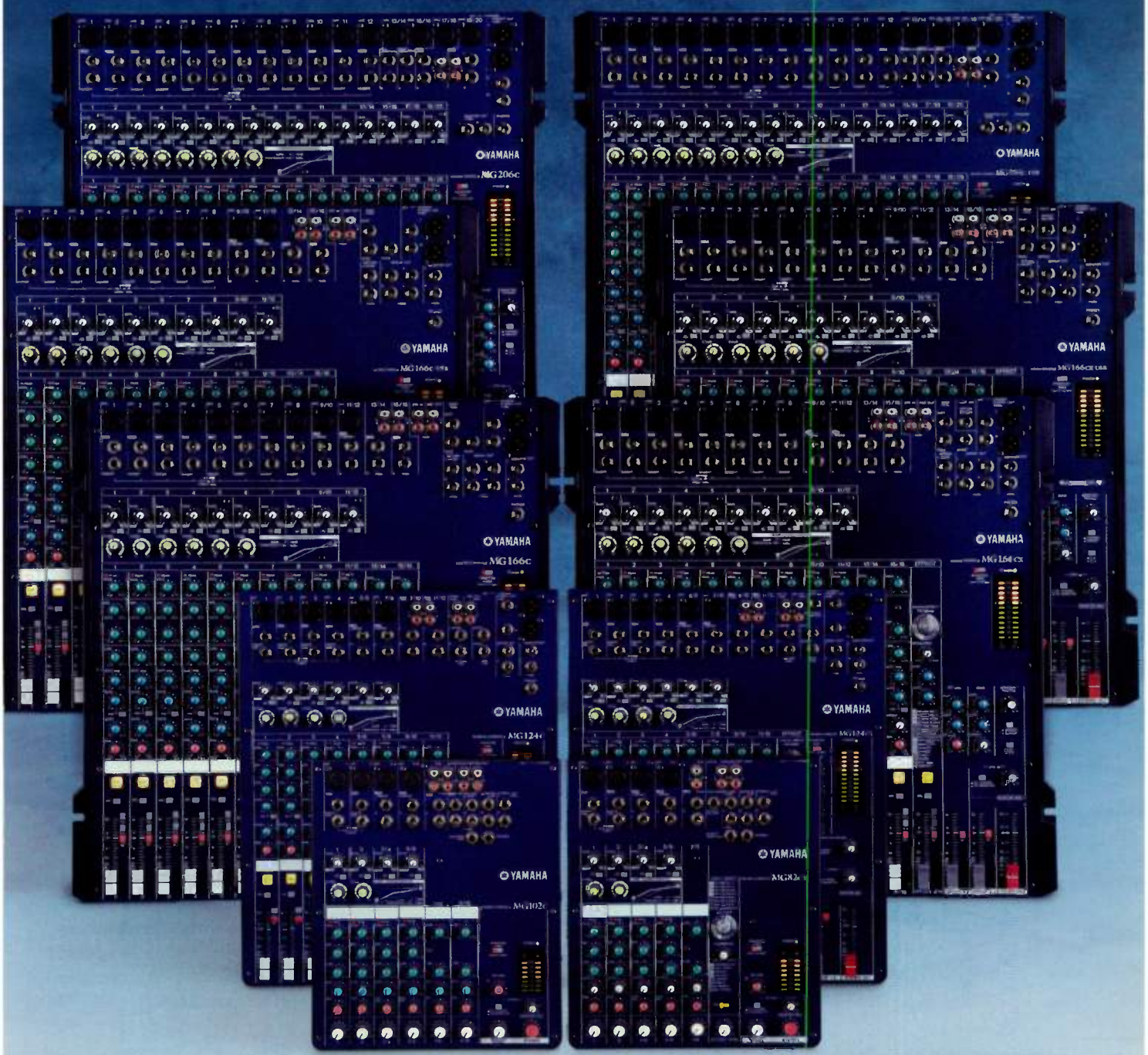


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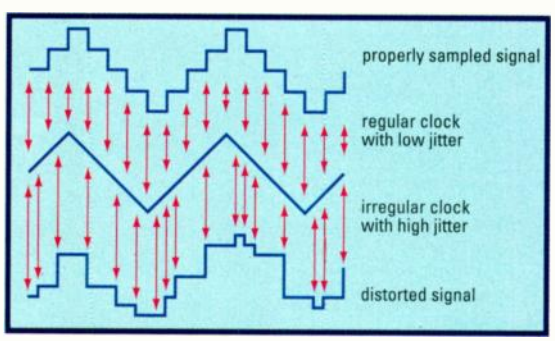
However, Levin points out that there are still sources of jitter in these interfaces. For instance, FireWire and USB send packets that are synchronization messages, but the architecture of the interfaces can introduce small variations in the transmission time of these packets. When a phase-locked loop (more on this later) at the receiving end tries to extract the clock from the slightly irregular timing messages, jitter is created.

## Word-Clock Systems

Distributed (word-clock) systems deliver, or distribute, a clock signal that is entirely separate from audio data. It most commonly appears as a simple square-wave signal generated by a device designated as the master and routed to each of the other devices. Word clocks usually run at the sampling rate and take the form of unbalanced signals on BNC connectors. That format is a legacy of the first digital audio-clocking system, which was SDIF (Sony Digital Interface, not to be confused with S/PDIF, which came later). SDIF set the mold of using BNC connectors and a square wave running at the sampling rate.

At the time that SDIF appeared, analog-to-digital converters (ADCs) and digital-to-analog converters (DACs) did conversions directly to and from PCM audio. Some early devices that accepted word clock even used that clock directly to drive the sample clock, meaning that any irregularities in the incoming clock were propagated into the device as jitter.

Since those earliest days, word clock has been a common method of clocking, even in some early multichannel interfaces that didn't use BNC connectors. For example, the TDIF format used by Tascam's digital recorders carries a word clock on one of the conductors in the 25-pin D-sub connector employed by the interface. And though Lightpipe is self-clocking, Alesis also released ADAT Sync at the same time. ADAT Sync is a companion interface that carries a word-clock signal on a 9-pin D-sub connector, thus eliminating the need to do the clock extraction required when syncing to Lightpipe and avoiding the jitter that could incur.




CHUCK DAHMER

FIG. 3: This is an example of jitter-induced waveform distortion in A/D conversion. Note how timing irregularities in the jittery clock create distortion, which is exaggerated to illustrate the point.

Digidesign Pro Tools systems have always been able to slave to an external clock. For many years, however, Digidesign used its own form of word clock called Superclock, which runs at 256 times the sampling rate. Current Pro Tools systems use standard sampling-rate word clock, and several connection schemes are possible for clocking them, but the most common (when using more than one interface) is Digidesign's Loop Sync system, which I'll describe later.

Once AES3 came into widespread use, engineers realized that AES3 streams containing audio data do not result in as clean a clock as those with no audio data. In a stream with no audio data, the waveform is quite regular. Therefore, waveform distortion from capacitance or termination problems should affect each clock the same, and the waveform, while distorted, should still be regular. A stream containing audio has many more irregular transitions (a characteristic of biphase mark code), which means more noise in the system and ultimately a less stable clock.

The AES11 standard specifies the use of an AES3 stream containing no audio data as a reference clock that can be distributed to all devices in a system. In effect, AES11, also called *AES black* (a reference to *black burst*, a composite video signal without picture information used as a master timing reference in broadcast and video-production facilities), acts as a word clock, turning a self-clocking interface into a distributed clock interface. AES11 is used by some high-end and multiroom production facilities.

## A Clock to the System

Project studios working to picture use QuickTime or AVI movies as their video source, but professional postproduction and video facilities still use black burst as the house-sync source to which everything is locked. Their reliance on black burst creates one of several clocking challenges that post houses deal with. Word clock is commonly derived from the horizontal sync pulse in the black-burst signal, at a rate of the number of video lines per frame multiplied by the frame rate. For NTSC color, that works out to 15.734 kHz. The mathematical relationship of this frequency to typical audio sampling rates is just plain ugly and difficult to properly resolve, but most master clock devices and many professional digital audio devices can do it.

Whenever film and video are both used in a production, the issue of *pull-up* and *pull-down* crops up. Explaining these is beyond the scope of this article, but I'll just say that relating pull-up and pull-down rates to digital audio clocking is yet another hurdle to overcome. Because problems occur in the types of facilities that deal with black burst, it is not surprising that professional audio-for-video devices and master clock devices are usually equipped to resolve such oddities against word clock, AES3, and S/PDIF. Such devices sometimes handle Lightpipe and other references too.

Clocking can become a real issue in audio post-production for film and video. Problems often arise in audio post because of decisions made during production without coordinating with the post house. Agreeing on clocking systems before production gets under way will avoid off-speed clocking issues that can cause sound and picture to slip out of sync.

High-end professional situations can also involve audio networking systems such as MADI—the AES multichannel standard—or CobraNet, EtherSound, and Optocore. MADI is structured similarly to AES3, except that it carries a separate distributed clock signal in addition to embedded clock information, so either can be used. CobraNet, EtherSound, and Optocore all transmit audio in packets, accompanied by a real-time word clock in the form of timed network messages. While it may seem like a self-clocking interface because the clock travels with the data, it is carried by time-stamp messages and does not have to be extracted from the data. Individual nodes synchronize their local clocks with the master clock.

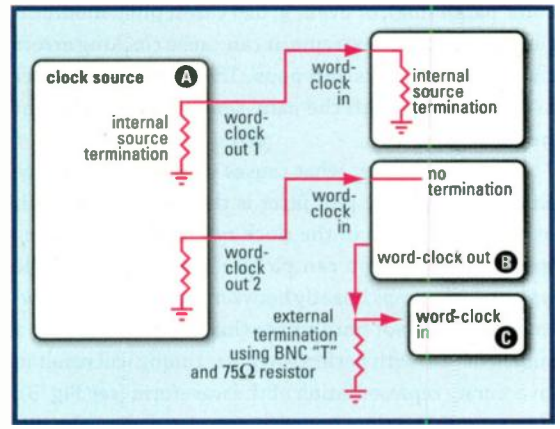



FIG. 4: In this transmission line, the clock source (device A) is internally terminated. Device B has no termination and daisy-chains the clock to device C, which also has no termination but uses a BNC T-connector with termination on one leg.

## Jitter: Too Much Caffeine?

By now, you may be wondering how clocking affects your sound. When clocking is not done well, the audible effects can range from very subtle to infuriatingly blatant. You've probably at least heard of jitter, and it is a significant topic. The effects of jitter can take the form of loss of clarity and definition, a reduction in the soundstage (that is, less width and depth in the

sonic panorama), or even, in bad cases, pitch modification. At the worst extreme, it can cause clocking errors that result in clicks and pops. The audibility of jitter artifacts varies with the ears' sensitivity to different frequencies.

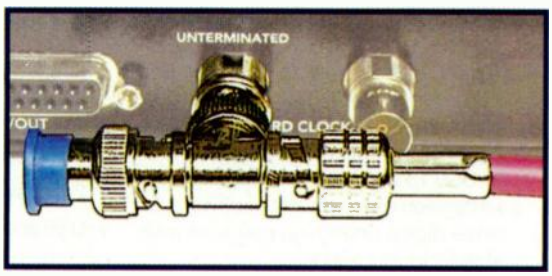
So what is jitter, what causes it, and how can it be eliminated? Simply put, jitter is the measurable result of small variations in the clock rate that cause timing irregularities. If you can picture sampling a triangle wave, which ramps linearly between its highest and lowest values, it's not hard to see that samples taken at a sufficient rate with perfectly regular timing will result in an accurate representation of the waveform (see Fig. 3). However, if the time between samples varies, the difference between two successive samples will result in waveform distortion. Because such timing variations are quite small in most cases, the resulting distortion may be subtle, but it is still there.

Because we live in an imperfect world and every clock has some amount of jitter, totally eliminating jitter just isn't possible. The goal is to minimize the amount of jitter that occurs in your system. Random jitter is caused by crosstalk and other induced sources, whereas periodic jitter can be signal related, or the result of two clocks in the system beating against each other.

Periodic-jitter-induced distortion occurs because jitter modulates the audio data. The jitter's amplitude and the signal's rate of change determine the degree of modulation. Periodic jitter is a form of frequency modulation and therefore produces sidebands. Thus, the more high-frequency content in the signal, the more susceptible it is to jitter-induced distortion. Jitter can also be cumulative, so a clock that passes through several devices can end up with considerably more jitter than it began with. For example, cumulative jitter could come into play if a signal passes through a poorly designed digital router (an electronic, not a physical, patch bay), which can add considerable jitter.

Another fundamental principle, and the most important fact to know about jitter, is that it is of consequence only in devices in which conversions occur—either ADCs, DACs, or real-time asynchronous sampling-rate converters (ASRCs). An ADC or ASRC encodes audio for storage or processing by converting it from an analog to a digital signal. Remember, analog audio is a continuous waveform, and a sample is a snapshot of that signal at the moment of sampling. Variation in the regularity with which the signal is sampled results in a distorted capture of the waveform. Distortion introduced by jitter on an ADC's clock is recorded as part of the signal and can never be removed; the same applies to poorly clocked ASRCs. Once jitter occurs in the recording process, you're stuck with it, which is one reason why high-quality ADCs are important.

On the other hand, jitter in a DAC creates distortion in playback: you will hear it, but it has no effect on the source data. In that situation, reducing jitter will improve the sound. In fully digital transfers, data is reclocked after it has been received; consequently, as long as none of the data is corrupted during transfer, jitter in the clocking will have no effect. Therefore, the rest of this discussion will not pertain to digital-to-digital transfers.




COURTESY APOGEE ELECTRONICS

FIG. 5: This is properly terminated word-clock input using a BNC T-connector and a 75Ω terminator.

## All That Jitters

The three most common causes of jitter are a poor-quality clock source, a bad transmission line (that is, the clock connections between devices), and a poor-quality receiver in the slave device. I'll discuss these causes one at a time.

Assuming that a clock is able to generate a good square wave, its stability determines its quality. If the source clock does not have precise and regular timing, nothing down the line can improve the situation. In earlier days of digital audio, clock quality was quite variable and many devices had very poor clocks. Current digital audio devices generally have much-better-quality clocks, though the operative word here is *generally;* some devices still do not. Mostly, they are inexpensive devices for which the manufacturer can't or won't spend the money to develop a high-quality clock or purchase more-expensive parts.

So how can you tell which devices have good clocks and which don't? Several common oscillator architectures are commonly used for digital audio clocks, each with benefits and drawbacks that include their jitter characteristics. But without specialized test equipment and sufficient knowledge, a user has no easy way to know for sure how good a clock is.

When possible, making A/B/C comparison auditions in your own studio can reveal what is most important: the differences in how devices sound. But it can be difficult to acquire numerous pieces of equipment to audition; often, the best you can do is approach companies you feel are technically competent and concerned with quality, and question them closely based on the material in this article to get a feel for the quality of their clocking.

## Playing by the Rules

Assuming that the device providing the master clock for your studio contains a stable clock, your next concern is the connection that conveys that clock to the various devices in the studio. The next principle I'll discuss is the concept of a transmission line. A transmission line must be properly terminated at each end but nowhere in the middle (see Fig. 4). Ideally, the transmission will degrade the signal as little as possible. Termination and length both matter. I will attempt to explain these ideas with a colorful analogy.

Imagine that you're playing a game of catch with a friend. If your friend throws the ball too hard or too soft, it will be difficult for you to catch correctly. If it is windy, rainy, or in some other way nasty outside, it will affect how the ball travels, again making it difficult to catch correctly. If the ball is wet, it will be tougher to catch than if it were dry. In addition, the person catching the ball has to be properly equipped. If you're trying to catch a baseball without a mitt, it could be painful and you might drop the ball. If you try to catch it with something made of metal or some other rigid material, the ball will bounce away from you. If, however, you have a proper mitt, the environmental conditions are reasonable, and the ball is thrown well, then everything will probably go fine—unless either of you is simply an idiot.

Now let's translate that analogy into a clocking situation. Throwing the ball correctly and using a suitable mitt translate into proper termination—that is, having the correct impedance at each end of the transmission line. For BNC word clocks, 75Ω is the proper termination, and a resistor of that value can provide it quite simply. S/PDIF also requires 75Ω termination, and AES3 uses 110Ω termination. Ensuring that your clock lines are properly terminated at the ends takes some thought and care, but it is key to minimizing jitter, especially for AES3 and S/PDIF, which operate at much higher rates than word clock. You have to assume that a clock output has the proper impedance termination at the source, so your real concern will be with terminating the far end of the line.

Many devices provide internal termination; others, like the MOTU interfaces in my studio, do not. The device's user manual should tell you whether word-clock inputs are terminated, but sometimes it is necessary to contact the manufacturer to find out. Some devices have internal terminators that can be switched in or out, to allow the device to be either at the end (terminated) or the middle (unterminated) of a clock chain. If a device has such a switch, it may appear on the back panel next to the BNC

connector, as a software setting, or as a jumper that must be changed inside the unit. Again, consult the manual.

If a device has no internal termination, there are simple options for proper external termination. The simplest is to get a BNC T-connector, which has two female and one male connection. The male goes into the receiving equipment, the clock connects to one female connector, and the other female connector should get a BNC connector with a 75Ω resistor soldered between the middle (signal) pin and ground. If you can use a soldering iron, you can do this yourself (see Fig. 5).

If the transmission line is not properly terminated at the receiving end, it is analogous to a metal mitt: the signal can reflect back into the transmission line, corrupting the clock signal and introducing jitter. It is important that clock lines not be overterminated or underterminated; either causes waveform distortion that can bring problems. A few clock manufacturers provide diagnostic LEDs to indicate whether clock outputs are properly terminated at the far end.

## Stable Cables

The rain and wind in my analogy refer to the principle of cable capacitance. Now you'll learn why you shouldn't use mic cables for AES digital connections, hi-fi cables for S/PDIF, or even cheap BNC cables for word clock.

Every cable has some capacitance, but the amount varies greatly with the cable's materials and construction. Capacitance occurs between the cable's signal line and ground and acts as a lowpass filter. If you've ever connected a synthesizer to an oscilloscope and looked at a square wave as you lowered the lowpass filter's cutoff frequency, you've seen the waveform's edges start to round off; the presence of high frequencies make the edges sharp. Excessive cable capacitance rounds the edges of the clock signal, making it less clear whether it is a one or a zero. That uncertainty results in jitter.

Of course, cable has inductance too, which acts opposite from capacitance. Van der Mee says that the art of making a good digital cable is in controlling the characteristic impedance, which comes down to keeping a proper ratio between capacitance and inductance. When the two are in proper balance in a cable, the artifacts created by it approach simple delay, with less phase distortion than when one or the other predominates.

The solution to capacitance and inductance problems is to use the proper impedance cable. Even the best-quality mic cables are designed for frequencies of 1 MHz or less. That's plainly sufficient for word clock, but an AES3 signal is about 3 MHz at a 48 kHz sampling rate. At audio frequencies, you are contending more with the resistive component of cable impedance, but in the megahertz range, the reactive component comes more into play, and even supercool mic cables could have enough reactance to cause the clock signal harm.

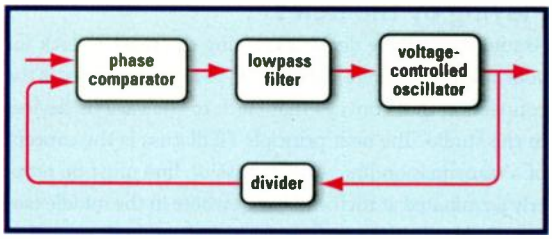



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FIG. 6: Here you see the basic architecture of a phase-locked loop.

Regular hi-fi cable with RCA connectors can't handle the bandwidth of S/PDIF for the same reason. Cable aside, the RCA connectors themselves are not made to work with 75Ω lines. Also remember that good shielding is more crucial for word-clock connections, which are unbalanced, than for AES3, which is a balanced line feeding a differential receiver.

Even with the proper cable, cable runs of excessive length can result in enough losses to cause problems. Looking at an AES3 signal on an overly long cable run, for instance, you will see a flattening of the waveform and unclear transitions. The AES3 specification states that you can run cables as long as 100 meters without any equalization or special treatment, whereas S/PDIF cables have a maximum length of 10 meters, and you should generally keep BNC word-clock cables to no more than 5 meters. (However, van der Mee reports good results using cable as long as 300 meters.) Optical fiber cables should be kept to about 15 meters if they're plastic, but glass cables can run much farther—over a mile, in fact. Note that optical connections such as Lightpipe are vulnerable to jitter resulting from dispersion of light in the optical fiber, which smears the waveform edges similarly to capacitance.

## Phasers Locked

Now I'll address the idiot receiver in my analogy. Even if you use a low-jitter clock source, a low-capacitance cable is in place, and the line is properly terminated at both ends, there is still the matter of how the receiving device processes the incoming clock signal.

Each device already has its own internal clock, which must be synchronized to the incoming clock in order to slave the device. In the old days, when slaved to an external word clock, some devices clocked directly off the word clock appearing at the BNC word-clock input jack. Those devices were extremely sensitive to jitter on the external clock. Today, internal clocks are synchronized to an external clock through the use of another clocking fundamental, the *phase-locked loop (PLL)*. A PLL is a servo system that compares the incoming clock to the internal clock and applies a correction signal as needed to the internal clock to match its frequency and phase to that of the incoming clock (see Fig. 6).

That may sound like a tricky process, and it is. There

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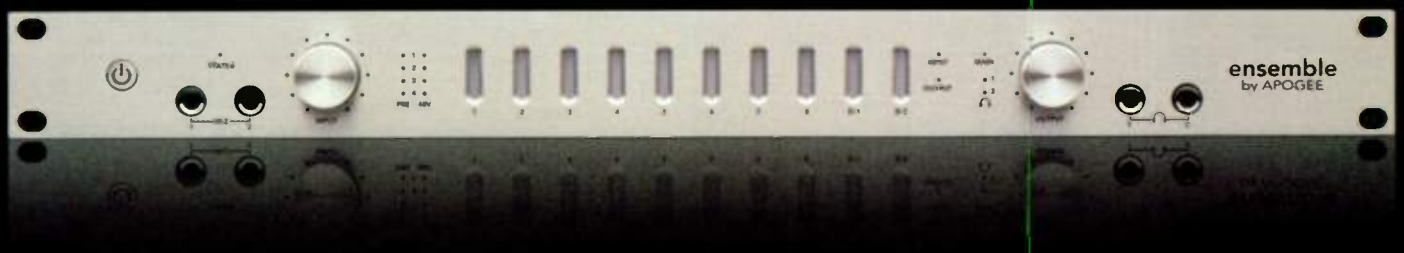


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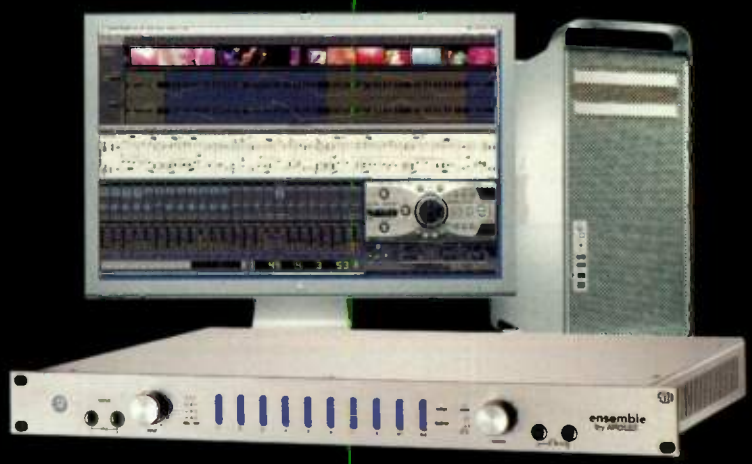


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are many ways to implement PLLs, from cheap and dirty to expensive and clean. If the receiving device has a low-quality PLL, it is an idiot and all the trouble you've gone to in getting a stable source and a low-capacitance, properly terminated line is for naught. The quality of the PLL in a digital device is hugely important.

Though PLLs can be weak links in the chain, they also have some very powerful benefits. One is that a well-designed PLL can do an excellent job of cleaning up even substantial amounts of jitter from an incoming clock (within limits, of course). Further, almost all of today's digital audio devices use sigma-delta ADCs, which generate high-bandwidth 1-bit streams that are later downsampled and converted into multibit PCM samples. Because word clock runs at the sampling rate, in order for a sigma-delta converter to sync to word clock, the frequency of the word clock must be multiplied up to the 24 MHz frequency of the sigma-delta converter's clock, called the *M clock*. This multiplication is performed by a PLL.

Given all of that, you obviously want to know how good the PLL is in any device. As with determining clock quality, a thorough analysis of jitter in a PLL is probably beyond the scope of most EM readers; however, Bryan suggests a way to get at least a reasonable take on PLL performance using regular audio-editing software with frequency analysis.

Simply record a 10 kHz sine wave at 0 dB into your ADC and perform a frequency analysis. A perfect sine wave should show energy at only one frequency, so measuring the amplitude of any sidebands you see will give you a good indication of how much jitter-induced frequency modulation is happening. In general, says Bryan, no sideband should exceed the ADC's stated signal-to-noise ratio.

## Watching the Clock

Beyond that, the best strategy is to buy good equipment from companies that appear to know what they're doing.

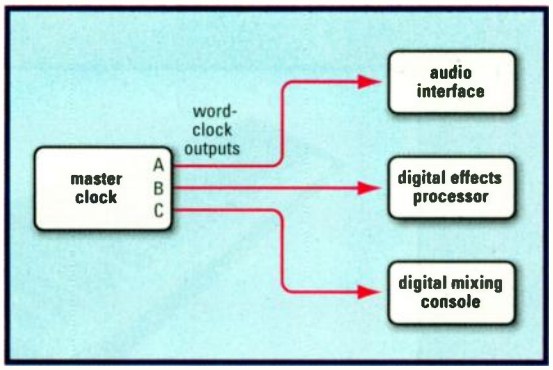



FIG. 7: To minimize transmission-line complications, a star configuration sources all clocks from a single high-quality master clock with multiple outputs. Note that proper termination must be applied at each slave device's clock input.

All the companies I spoke with for this article care about doing clocking well, and some, like Apogee Electronics and Antelope Audio, specialize in clocking devices. Many other companies, of course, are striving to make high-quality equipment with stable clocks; Mytek Digital, Benchmark Media, Lucid Audio, Rosendahl Studiotechnik, Brainstorm Electronics, and Lavry Engineering are only a few of them, and there are lots more.

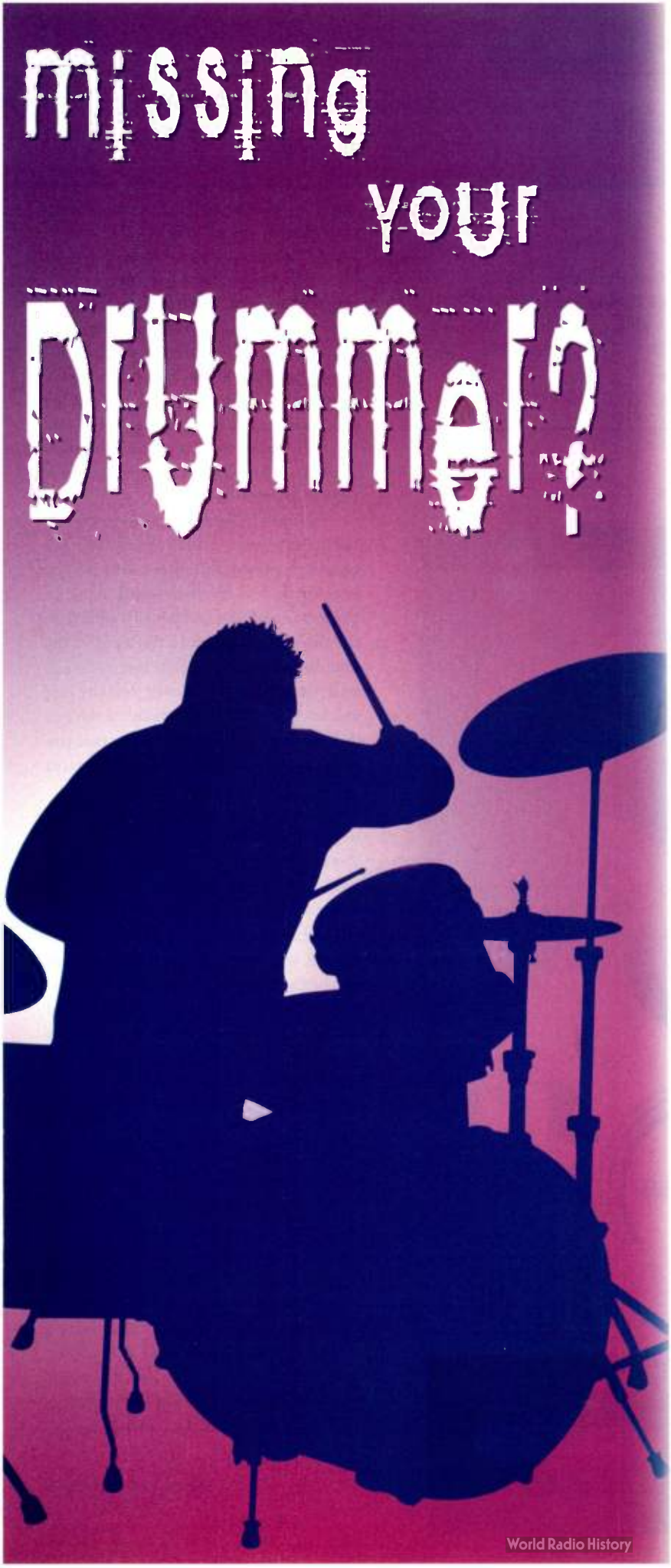
The fact that such critical information is unavailable to users poses a real problem in knowing whether you're assembling a well-clocked system. But it gets even worse: there is no standard defining the voltage level or phase of a BNC word clock, and manufacturers have different ideas about how devices should behave. This lack of standards can lead to all sorts of troubleshooting nightmares that will reduce your method to just trying every configuration you can think of until, hopefully, you find one that works. It's not very scientific, but that's how it is if things don't work right. Fortunately, they more often do.

Even with the entire transmission line in place, you're not done yet. You still have to make sure that every device is "listening" for the external clock. Some devices will automatically switch to an external clock when they detect one present; they will also detect the clock's sampling rate and switch to that. Other devices require the user to change a setting manually to enable external clocking. Audio interfaces for computers often have software driver settings that must be changed before they will clock from an external source. If you change the application you are running, you may need to recheck the clocking to make sure the application did not set the driver to sync to a different clock source.

Finally, although a device set to receive an external clock should detect the sampling rate and switch its own sampling rate accordingly, not all of them do—again, including some computer audio interfaces. Assumptions will lead to trouble; be sure you know that each device is not only enabled to receive external clock, but also is set to the correct rate.

Most devices will simply mute if they cannot auto-switch to an incoming clock at a different rate than the one set internally, but some may switch to their internal clocks and not resync when the external clock reappears. As with any situation in which devices are not locked to the same clock, ugly artifacts can occur. For example, when using a master clock device, you need to be vigilant in explicitly switching its frequency every time you use a different sampling rate.

To paraphrase 19th-century abolitionist Wendell Phillips, eternal vigilance is the price of good clocking. Often, 48 kHz and 44.1 kHz are close enough in frequency that some devices set to one rate will accept a clock and run at the other without complaint. For

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instance, consider a file playing at 48 kHz and recorded to an external device inadvertently set to 44.1 kHz. When played back at 44.1 kHz, all the samples would be present, but they'd play back too slowly, resulting in the pitch being lowered.

## Who's in Charge?

Once you understand that one device must be the clock master, which one should it be? The answer probably depends on the context, but in all cases, it should be a good-quality, stable clock. Devices like CD or DVD players make very jittery clock sources.

Because the most critical point for clocking is the ADC, an ideal scheme is to use a high-quality ADC's clock as the master. One of the features that make such a device high quality (and more costly) is having a good clock. In simple systems, you should be able to clock from the ADC, and that's usually a good idea.

Studios with digital mixers often use them as clock masters for a couple of reasons. When you load a project and the mixer resets to the session's sampling rate, any devices being clocked from the mixer that autodetect sampling rate will automatically change. Another reason is that some mixers just don't seem to like being clocked externally. In other studios, the DAW audio interface is used as the master to attain the same level of convenience.

Pro Tools HD systems can use a scheme called Loop Sync, in which the word-clock output from each interface in the system is connected to the word-clock input of the next, with the output of the last in the chain looping back to the input of the first. In software, any of the interfaces can be designated as the master, and the rest will slave to it, making it easier to slave to digital inputs in different formats that you have connected to the different interfaces without physically repatching.

Another very popular solution is to use a dedicated master clock device. These devices are designed to have very stable clocks, but they generally also provide multiple outputs, usually including a few different formats (AES3, word clock, S/PDIF, and so on). Further, they provide functions needed in postproduction situations, where it is necessary to resolve word clock to other incoming signals, such as timecode, AES3, or black burst.

Regardless of what device provides the clock, there is the question of how to distribute it to all the other devices. Most devices that have a word-clock input also have a word-clock output or thru. A word-clock-thru connector makes it easy to daisy-chain the clock from one device to another. Though a very simple method of clock distribution, it is potentially problematic. The first issue is that the length of the transmission line is increased. Even though you're using multiple cables, they act as one long cable as far as capacitance is concerned. Another issue is that daisy-chaining makes it critical to know whether

the word-clock input of a device in the middle of the chain is terminated; if it is, havoc will ensue.

High-quality devices from some manufacturers, such as Apogee, do not simply hardwire the clock input to a thru connector, but actually regenerate the incoming clock before sending it out. This is exemplary behavior that really improves system clocking. However, Kashiwa and van der Mee both caution that every PLL has its own jitter signature, which is added to the clock; consequently, passing a signal through several devices that each reclock could result in a troublesome accumulation of jitter.

The perils of daisy chaining are best avoided by wiring in a star configuration whenever possible (see Fig. 7). Master clock devices are useful in larger systems with a number of components that need clocking, because they have multiple outputs that can usually drive several units directly. If you are clocking instead from an ADC or other device, a word-clock distribution amplifier can take the master clock and generate a number of outputs to fan out to your studio's devices.

## Clocking Out

Although analog audio is still alive and kicking, digital audio dominates most production today, bringing with it the need to understand a different set of basics than that in the analog world. Clocking is absolutely one of the most important elements in a digital audio system. At times the concepts can be a little slippery, but clocking is a technical underpinning of modern studios that is ignored or misunderstood at the user's peril. (For more information about clocking and a recommended reading list, see the **online bonus material** at www.emusician.com.)

The basic foundations of digital clocking are not hard to understand or implement. Once you have a good system set up and know how it works, hopefully you will notice a difference in the sound of your studio. At the very least, you should be able to eliminate (or greatly reduce) ugly clicks and pops. You may also notice greater definition in your sound.

Always keeping fundamental principles in mind will guide you through most clocking questions. The bright side is that once a well-thought-out clocking scheme is in place, it should remain stable for as long as your system setup is consistent, leaving you free to concentrate on the project at hand. EM

*Larry the O will soon embark on a studio-building project that you may well be reading about this time next year.*

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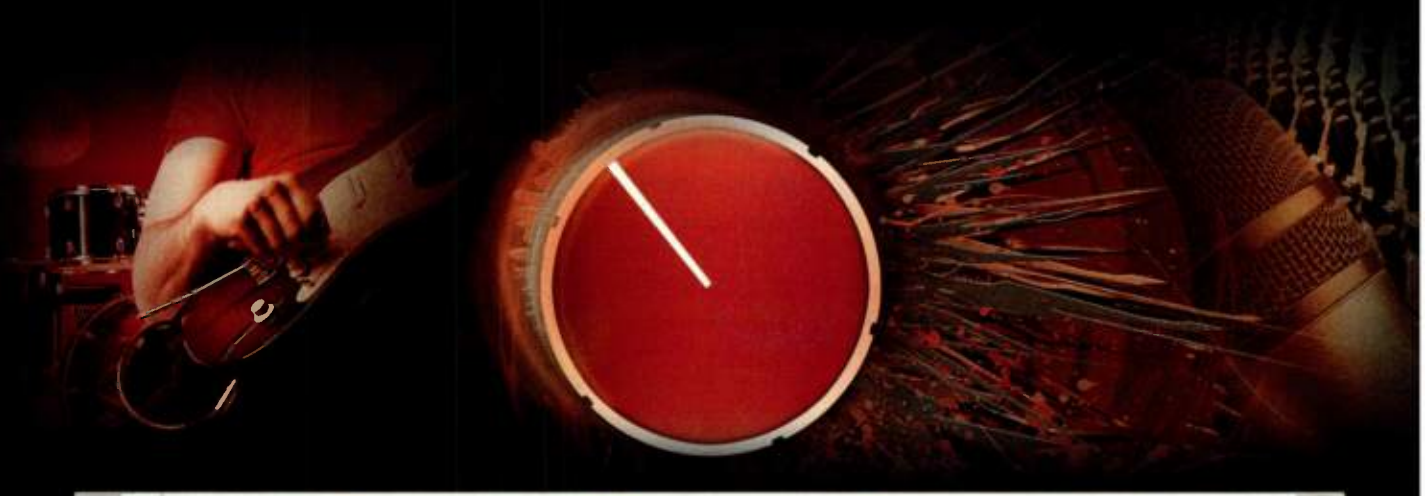

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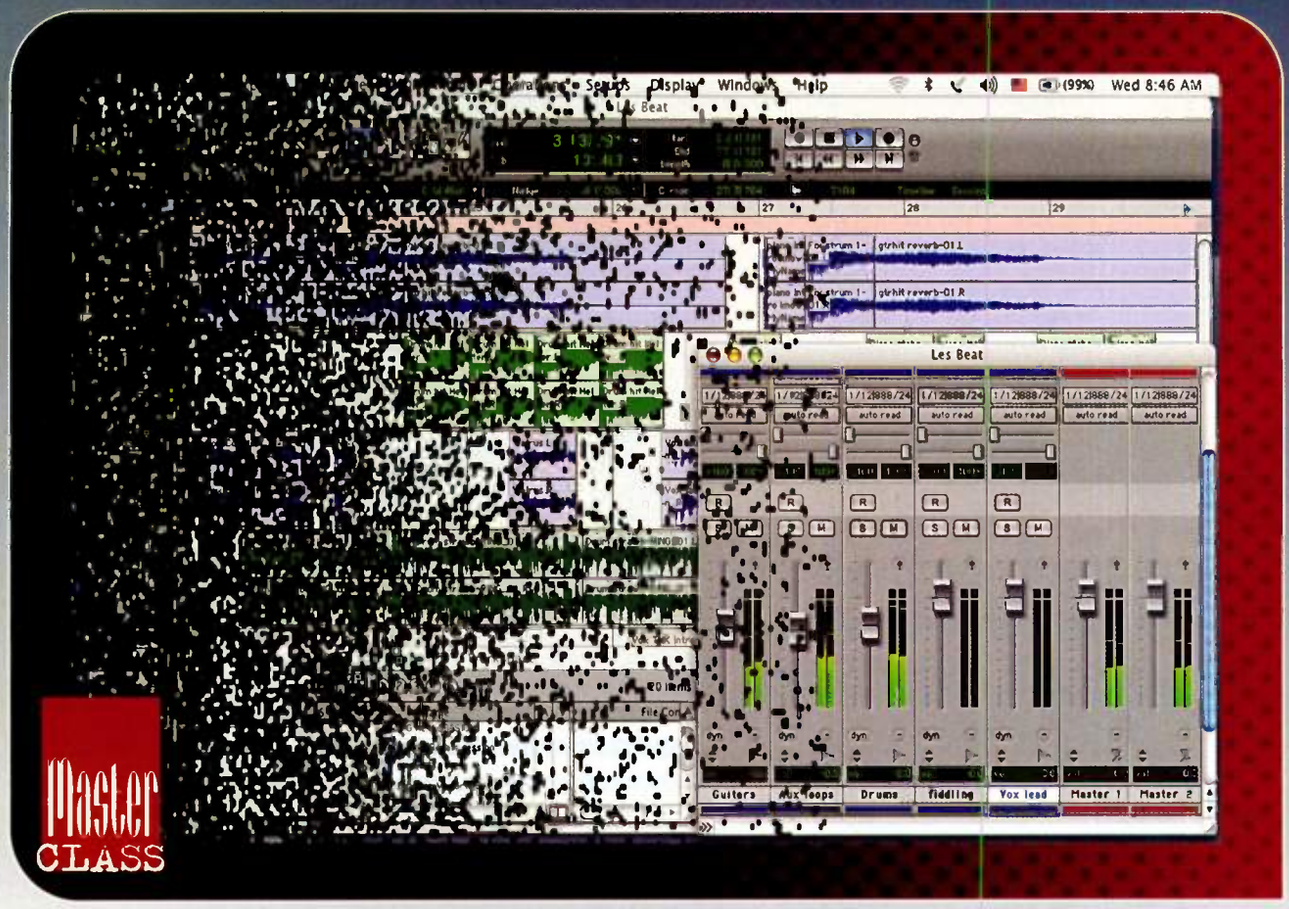



DMITRY PANICH

# Mixing Efficiency

By Brian Smithers

Take better advantage of Pro Tools' creative potential.

As an acoustic musician, I have spent a major portion of my life in the practice room, squaring away all the left-brain technical stuff so that when I perform I can be entirely in right-brain expressive mode. As an electronic musician, however, I am too often guilty of trying to be both artist and technician at the same time, with sheet music on one knee and a software manual on the other. The predictable result is that I am less effective at both roles.

I don't expect to persuade anyone to sit down and practice at the computer the way one plays scales and arpeggios, but clearly some thoughtful preparation can pave the way for more-effective creative time. Big-league engineers ordinarily have assistant engineers to set up the console, patch in their favorite processors, organize tracks, and document everything so they can focus on the music instead of the technology. For the rest of us, it pays to be our own assistant engineers and spend some time preparing a session for the real creative work.

In this article, I will focus on ways to set up a session for a more organized and efficient mix. In addition to thinking ahead so you don't have to interrupt the creative flow, you can use certain tricks and underutilized functions in sequencers to help smooth out the rough spots in a mix session. Although I will use Digidesign Pro Tools 7.3 for my examples, each of the major DAWs and digital audio sequencers offer similar shortcuts and time-savers. All of my suggestions work in HD, LE, and M-Powered systems unless otherwise noted. Being efficient in Pro Tools means using lots of shortcut keys—I'll use Windows modifier keys and note their Mac equivalents in parentheses (see **Web Clip 1**).

## Conquering Space and Time

In preparation for mixing, you may listen to a rough mix or simply throw the faders up to unity and listen through once or twice to get the big picture. As you make mental notes of each new section, press Enter on the numeric keypad to drop a marker. In the Edit Memory Location dialog box that appears, type a name (intro, verse, chorus, and so on) and click on Marker (see **Fig. 1**). Be sure that all other options are deselected and Reference is set to Absolute, then press Enter to close the dialog box. No matter how long it takes you to type *cantankerous contrabassoon cadenza*,

Mixing Efficiency

colors of the tracks' small color bars to the entire length of the channel strips, hold Ctrl + Start + Alt (Ctrl + Option + Command) as you click on any color in the palette, and drag upward until you achieve the desired intensity.

## E-I-E-I/O

Label your inputs, outputs, inserts, and buses. All can be renamed via a right-click menu, or you can manage them all from I/O Setup (see **Fig. 3**). It's more efficient to assign a send to Verb Bus than to wonder whether it's supposed to be Bus 13-14 or Bus 15-16, and it's much easier to remember what you've done with a track if you can see it has sends to GtrDelay and ShortVerb. When you patch in a hardware processor, name the insert after that device.

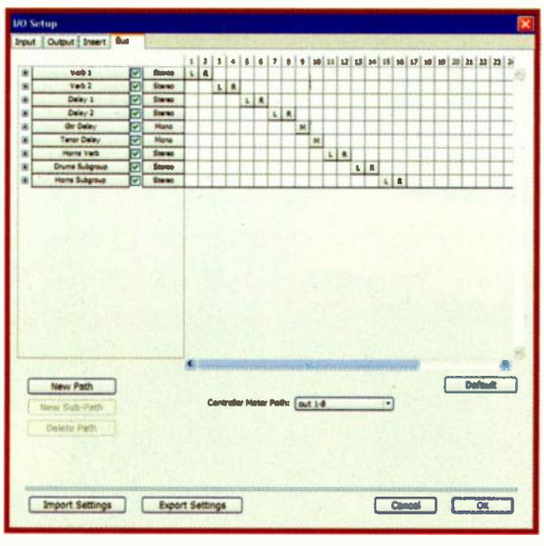
Export and reuse your I/O settings. Open a well-documented session and click on Export Settings in the I/O Setup window. Choose a fitting name and save the file to the default location. When you create a new session, choose this setup from the New Session dialog box. All of your helpful names will be immediately available in that session. You can import your favorite mix I/O settings into an existing session by using the Import Settings button in the I/O Setup window. You'll need to reassign any existing I/O paths, but if you do this as you start a mix, it should save you more time than it costs.

## Group Hug

The notion of a fader group (a mix group in Pro Tools) is intuitive: move one fader, and its buddies will move along with it. In Pro Tools, simply select multiple tracks and press Ctrl + G (Command + G). The Create Group dialog box underwent some major revisions in version 7.2 (7.3 for LE), and Pro Tools HD's group behaviors became substantially more flexible (see **Web Clip 3**).

If you want to adjust a grouped track individually, you can either Start-drag (Ctrl-drag) its fader or suspend the group. The standard way to suspend a group is to click on its name in the Groups list—groups are active only when they are highlighted. In the Mix view, it's far quicker to press the group's ID letter, at least for the first 26 groups (*A* to *Z*). When creating groups, choose an ID of *D* for your Drums group and *K* for your Keyboards group, and you'll have no trouble remembering which key to press.

FIG. 3: It's much easier to create and understand your session's signal flow when paths have relevant names, such as Horns Verb and Drums Subgroup.




Even if you don't find yourself using fader groups that often, groups have other important uses. Click on a group's ID in the Groups list to select all members of that group, and Shift-click on another group's ID to add its members to the selection. This works even if the group is not currently active. Start-click (Ctrl-click) on a group's ID to display its member tracks while hiding all other tracks. Shift-Start-click (Shift-Ctrl-click) on another group's ID to show its members as well. Start-click (Ctrl-click) on the All group to show all tracks.

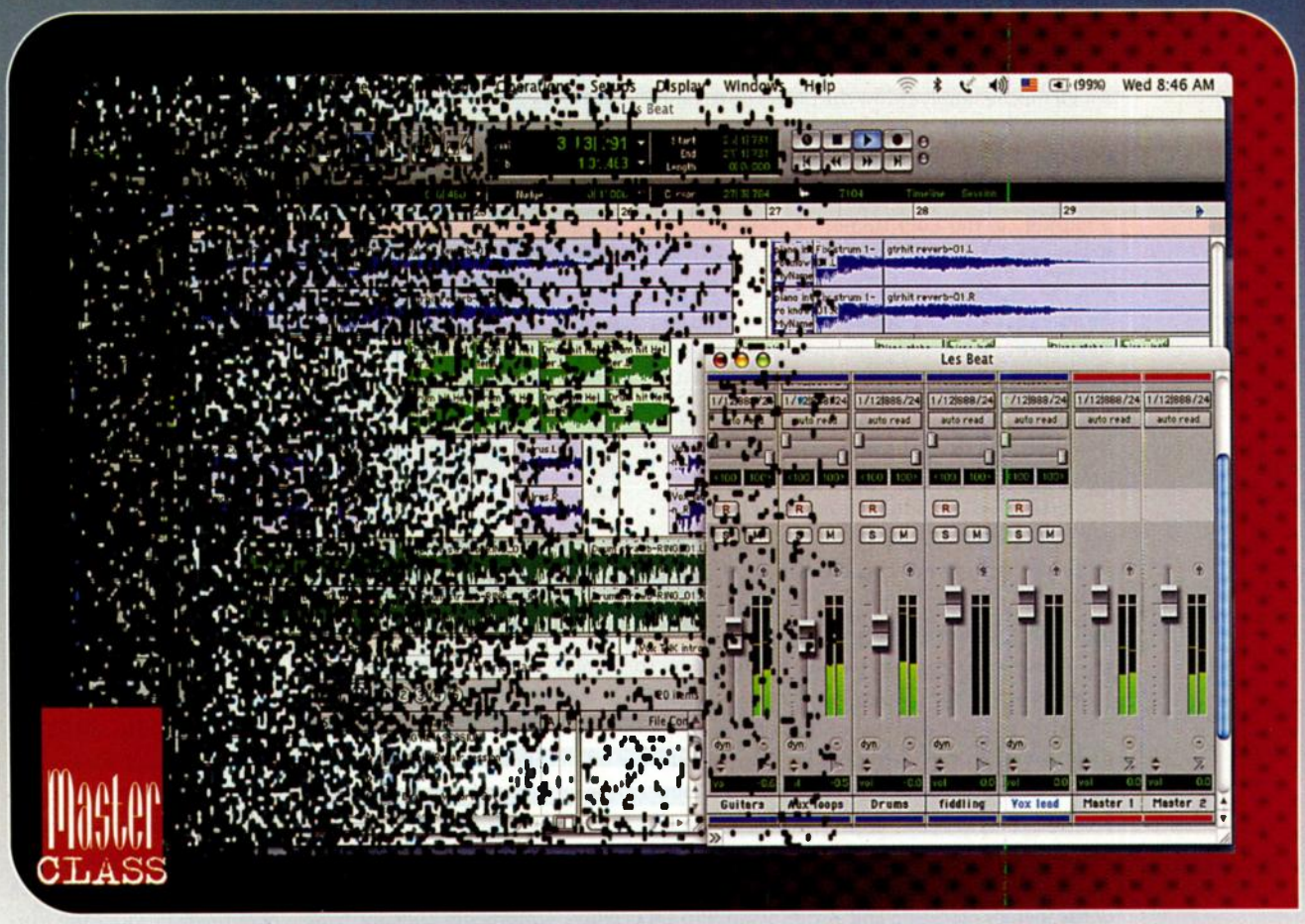
Although some functions, such as assigning sends and inserts, don't follow groups, you can use groups to apply them efficiently. Click on a group's ID to select its members, then Shift-Alt-click (Shift-Option-click) to assign a send or plug-in to one member track. The send or plug-in will be created across all members of the group with the same channel format. Remember that in Pro Tools, the modifier combination Shift + Alt (Shift + Option) means "apply to all selected tracks." To apply an action to all tracks, hold only Alt (Option) while clicking.

## Making Memories

Be sure to take full advantage of memory locations. In addition to storing markers, memory locations allow you to recall zoom settings, track show/hide settings, track heights, group enables, selections, and more. Many repetitive tasks can be sped up by using memory locations. You know how to use mix groups to show and hide grouped tracks—create memory locations to show and hide tracks that are not ordinarily grouped. Create a memory location that shows only your effects returns, only your subgroups, or only your instrument tracks.

You don't need to create memory locations in the default numeric order. Make all your show/hide memory locations start with 30 and your group-enable memory locations start with 50. To create a memory location with a specific number, press the Period key, the number of the memory location, and Enter on the numeric keypad. With 999 memory locations available, you can organize them as you wish.

Create a pair of track-height memory locations to toggle between Small and Jumbo so that when you go into the Edit view to edit mix parameters graphically, you can jump in to do detailed editing. If you find yourself returning to specific sections and looping them, such as looping a short drum fill to tweak EQ on the toms, save the selections as memory locations. Be sure you deselect all General properties (zoom, group enables, and so on) so when you recall a selection, you don't change any




DMITRY PANICH

# Mixing Efficiency

By Brian Smithers

Take better advantage of Pro Tools' creative potential.

As an acoustic musician, I have spent a major portion of my life in the practice room, squaring away all the left-brain technical stuff so that when I perform I can be entirely in right-brain expressive mode. As an electronic musician, however, I am too often guilty of trying to be both artist and technician at the same time, with sheet music on one knee and a software manual on the other. The predictable result is that I am less effective at both roles.

I don't expect to persuade anyone to sit down and practice at the computer the way one plays scales and arpeggios, but clearly some thoughtful preparation can pave the way for more-effective creative time. Big-league engineers ordinarily have assistant engineers to set up the console, patch in their favorite processors, organize tracks, and document everything so they can focus on the music instead of the technology. For the rest of us, it pays to be our own assistant engineers and spend some time preparing a session for the real creative work.

In this article, I will focus on ways to set up a session for a more organized and efficient mix. In addition to thinking ahead so you don't have to interrupt the creative flow, you can use certain tricks and underutilized functions in sequencers to help smooth out the rough spots in a mix session. Although I will use Digidesign Pro Tools 7.3 for my examples, each of the major DAWs and digital audio sequencers offer similar shortcuts and time-savers. All of my suggestions work in HD, LE, and M-Powered systems unless otherwise noted. Being efficient in Pro Tools means using lots of shortcut keys—I'll use Windows modifier keys and note their Mac equivalents in parentheses (see Web Clip 1).

## Conquering Space and Time

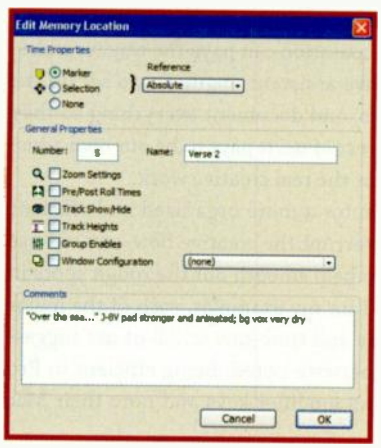
In preparation for mixing, you may listen to a rough mix or simply throw the faders up to unity and listen through once or twice to get the big picture. As you make mental notes of each new section, press Enter on the numeric keypad to drop a marker. In the Edit Memory Location dialog box that appears, type a name (intro, verse, chorus, and so on) and click on Marker (see Fig. 1). Be sure that all other options are deselected and Reference is set to Absolute, then press Enter to close the dialog box. No matter how long it takes you to type *cantankerous contrabassoon cadenza*, Pro Tools will drop the marker where you first pressed Enter.

Later, you can add comments to the markers to indicate moods, lyrics, objectives, and so forth. Comments can be displayed in the Memory Locations window, and they'll show up in the tool tips that appear when you hover the cursor over a marker. (Enable Details under the Tool Tips options in the Display Preferences window.)

After you've marked the song's road map, locate to each marker in turn and click on Play. If you were off-target, zoom in and drag the marker to a more precise location. Instead of locating by clicking on markers in the Memory Locations window, use the shortcut of pressing Period (.) on the numeric keypad followed by the number of the memory location and Period again. Pro Tools relocates efficiently enough that you can use this technique to experiment with the arrangement, skipping or repeating sections without editing.

If the song was not recorded to a Pro Tools click track, the grid will not align with the song's tempo. Tempo-based effects, such as filter LFOs and note-based delay times, will not correspond to the song. Take the time to create a tempo map of the entire song, using either Identify Beat or Beat Detective. Even if you have no intention of using Beat Detective to time-correct any parts, the tempo map will be useful.

FIG. 1: Markers are a special type of memory location. Use them to delineate a song's form and annotate the flow of the mix.




Work in 4- or 8-bar segments, starting from the beginning of the song. Be careful to make very precise selections and listen to each selection in Loop Playback to confirm that it is accurate. Set up a click track and listen to whether it aligns well with the audio tracks. You will almost always want to base the tempo map on the drums.

## Up the Organization

You should have named your tracks when you created them—Pro Tools (and other DAWs) names

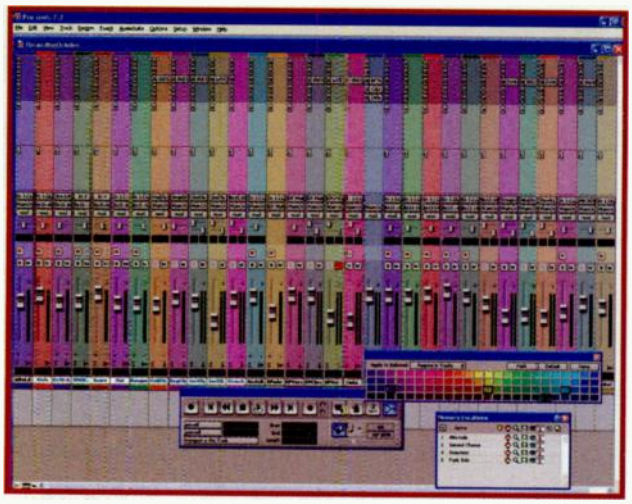


FIG. 2: Pro Tools, like other software, supports color-coded tracks. Using colors consistently across projects is a great way to help identify tracks, especially in a mix containing many tracks.

recorded or processed files after the tracks in which they were created. If you didn't, do it now so you know which fader is which. Although Pro Tools is fairly intelligent about shortening long track names, you're always better off keeping them short to begin with—you're more intelligent by far. You have room for only eight to ten characters of average width, so use them wisely.

Be sure your Comments field is visible and use track comments liberally. Note any outboard processing, corrective action, musical function (lead, background, and so on), or other tidbits. Assume that you will forget everything, and use comments to help you remember.

Group your tracks by instrument type or musical function. Obviously, the drum tracks should be next to each other, and you probably want the rest of the rhythm-section instruments to be near the drums. Don't be dogmatic, but do be consistent from session to session. You won't have to waste time searching for the guitar-solo track if you know it always goes immediately to the right of the electric and acoustic rhythm-guitar tracks (in that order). Consider adopting the Recording Academy's recommendations (see **Web Clip 2**).

If you have a lot of tracks, reorder them more efficiently by dragging them up and down in the Tracks list. If you need to move groups of tracks, do it from the Edit view. Alt-click (Option-click) on any track's Track Height Selector and choose Small to fit as many tracks as possible onscreen. (Mini or Micro will fit more, but you'll have a harder time reading track names.) Select the tracks you need to move, and drag them up or down as a unit.

Most DAWs allow you to color code tracks, and this can be a great help in finding your place in a lengthy set of tracks. In Pro Tools, you can automatically colorize tracks and regions by function (under Display Preferences) or individually assign colors from the Color Palette, which is found under the Windows menu (see **Fig. 2**). To apply the

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colors of the tracks' small color bars to the entire length of the channel strips, hold Ctrl + Start + Alt (Ctrl + Option + Command) as you click on any color in the palette, and drag upward until you achieve the desired intensity.

## E-I-E-I/O

Label your inputs, outputs, inserts, and buses. All can be renamed via a right-click menu, or you can manage them all from I/O Setup (see **Fig. 3**). It's more efficient to assign a send to Verb Bus than to wonder whether it's supposed to be Bus 13-14 or Bus 15-16, and it's much easier to remember what you've done with a track if you can see it has sends to GtrDelay and ShortVerb. When you patch in a hardware processor, name the insert after that device.

Export and reuse your I/O settings. Open a well-documented session and click on Export Settings in the I/O Setup window. Choose a fitting name and save the file to the default location. When you create a new session, choose this setup from the New Session dialog box. All of your helpful names will be immediately available in that session. You can import your favorite mix I/O settings into an existing session by using the Import Settings button in the I/O Setup window. You'll need to reassign any existing I/O paths, but if you do this as you start a mix, it should save you more time than it costs.

## Group Hug

The notion of a fader group (a mix group in Pro Tools) is intuitive: move one fader, and its buddies will move along with it. In Pro Tools, simply select multiple tracks and press Ctrl + G (Command + G). The Create Group dialog box underwent some major revisions in version 7.2 (7.3 for LE), and Pro Tools HD's group behaviors became substantially more flexible (see **Web Clip 3**).

If you want to adjust a grouped track individually, you can either Start-drag (Ctrl-drag) its fader or suspend the group. The standard way to suspend a group is to click on its name in the Groups list—groups are active only when they are highlighted. In the Mix view, it's far quicker to press the group's ID letter, at least for the first 26 groups (*A* to *Z*). When creating groups, choose an ID of *D* for your Drums group and *K* for your Keyboards group, and you'll have no trouble remembering which key to press.

Even if you don't find yourself using fader groups that often, groups have other important uses. Click on a group's ID in the Groups list to select all members of that group, and Shift-click on another group's ID to add its members to the selection. This works even if the group is not currently active. Start-click (Ctrl-click) on a group's ID to display its member tracks while hiding all other tracks. Shift-Start-click (Shift-Ctrl-click) on another group's ID to show its members as well. Start-click (Ctrl-click) on the All group to show all tracks.

Although some functions, such as assigning sends and inserts, don't follow groups, you can use groups to apply them efficiently. Click on a group's ID to select its members, then Shift-Alt-click (Shift-Option-click) to assign a send or plug-in to one member track. The send or plug-in will be created across all members of the group with the same channel format. Remember that in Pro Tools, the modifier combination Shift + Alt (Shift + Option) means "apply to all selected tracks." To apply an action to all tracks, hold only Alt (Option) while clicking.

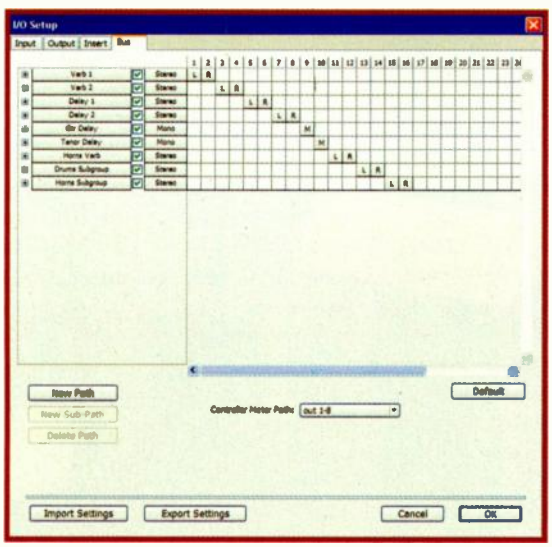


FIG. 3: It's much easier to create and understand your session's signal flow when paths have relevant names, such as Horns Verb and Drums Subgroup.

## Making Memories

Be sure to take full advantage of memory locations. In addition to storing markers, memory locations allow you to recall zoom settings, track show/hide settings, track heights, group enables, selections, and more. Many repetitive tasks can be sped up by using memory locations. You know how to use mix groups to show and hide grouped tracks—create memory locations to show and hide tracks that are not ordinarily grouped. Create a memory location that shows only your effects returns, only your subgroups, or only your instrument tracks.

You don't need to create memory locations in the default numeric order. Make all your show/hide memory locations start with 30 and your group-enable memory locations start with 50. To create a memory location with a specific number, press the Period key, the number of the memory location, and Enter on the numeric keypad. With 999 memory locations available, you can organize them as you wish.

Create a pair of track-height memory locations to toggle between Small and Jumbo so that when you go into the Edit view to edit mix parameters graphically, you can jump in to do detailed editing. If you find yourself returning to specific sections and looping them, such as looping a short drum fill to tweak EQ on the toms, save the selections as memory locations. Be sure you deselect all General properties (zoom, group enables, and so on) so when you recall a selection, you don't change any

of those attributes. Conversely, be sure your Time properties are set to None for your show/hide and track-height memory locations so they don't relocate the playback cursor.

Pro Tools can now save up to 99 window configurations, and these can be recalled as part of memory locations. Window configurations can recall the entire layout of all windows—which ones are open, how they are configured, and how they are arranged onscreen—or the display settings of the Mix, Edit, or Transport windows.

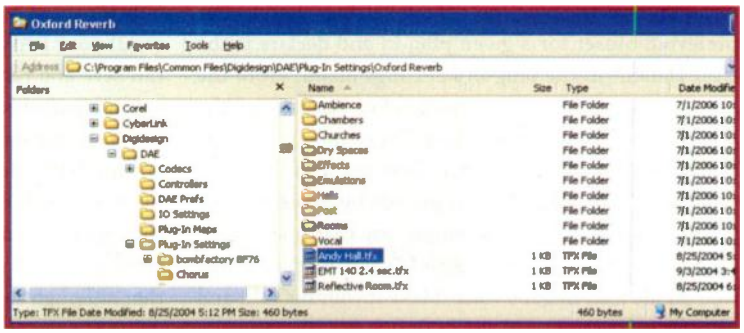


**FIG. 4: Plug-in settings submenus are simply folders. Plug-in settings files that exist within the plug-in's Root Settings folder appear at the top of the settings list for quick access.**

Suppose you use sends A through E for global sends, such as your main reverb, while reserving sends F through J for individual sends, such as a guitar delay. Show just A through E and press the Period key followed by the number 11 and then the Plus key on the numeric keypad to create a window configuration showing just those sends. Check Mix Window Display Settings in the Edit Window Configuration dialog box, and don't forget to give the configuration a meaningful name and description. Create a window configuration with the number 12 that shows just F through J. Press Period, then 11, then Asterisk (*) on the numeric keypad to recall the A through E view.

Create window configurations for any common changes you make to the Mix view, such as opening and closing the Tracks list (or even changing its width) and displaying or hiding sends, inserts, comments, and I/O. If you simply click in the Window Configuration List to change views, you are saving yourself mouse-clicks each time. Better still, recall them from the numeric keypad.

## Plug It In

There are several ways to optimize your use of plug-ins. If you reach for certain presets regularly, make Pro Tools

recall them as the plug-ins' default settings. Recall your preferred preset for a given plug-in and declare it the User Default in the plug-in's Settings menu. From the Settings Preferences submenu, tell Pro Tools to set the plug-in's default to User Setting. Every time you insert that plug-in, it will launch with these settings.

Even if you do this on a project-by-project basis, it will save you time. For example, say that on the first song of a project, you create an EQ setting that works well on the lead singer's voice. Make it your default, and as you work your way through the rest of the songs on the album, you will always have a good starting point. Be sure, though, that you have saved this setting to the Root Settings folder, not the Session Settings folder.

You should make a habit of preserving the settings on all your plug-ins by saving presets in the Session Settings folder. Should you encounter a corrupt session file, you could re-create your session from the existing audio files, but you would have lost your mix. If you have saved your plug-in settings, however, you could recall them easily. Saving them to the Session Settings folder prevents them from cluttering the Root Settings folder—you can always import settings from one session's folder into another session if you want.

When a plug-in has multiple folders (submenus) of presets, there are probably a few you use more than others. Copy those preset files from their subfolders to the plug-in's Root Settings folder so they appear at the top of the list (see **Fig. 4**). You'll need to go submenu diving only for those settings you use less often.

If your DAW supports effects chains, whereby you can save and recall multiple plug-ins in order and with specific parameters as a single preset, invest the time to set up and save a few. Pro Tools does not currently do this, but it does allow you to drag a plug-in settings file from the Workspace browser directly to an insert, thereby instantiating the relevant plug-in with those settings in a single action. The one obstacle to this is that although the browser is great at searching for files, it is not very efficient at navigating to known folders. To get around the problem, create a folder at the root of any volume and copy your favorite settings there. They will then be only a click away in the Workspace browser when you want to insert them.

There are no doubt a handful of plug-ins on which you depend frequently. Make these into "favorites" so they appear at the top of the plug-in list, and you can get to them more easily. Ctrl-click (Command-click) on an insert and choose a plug-in from its usual folder, and it will always display above all your plug-in submenus. Repeating the process clears a plug-in from the favorites list.

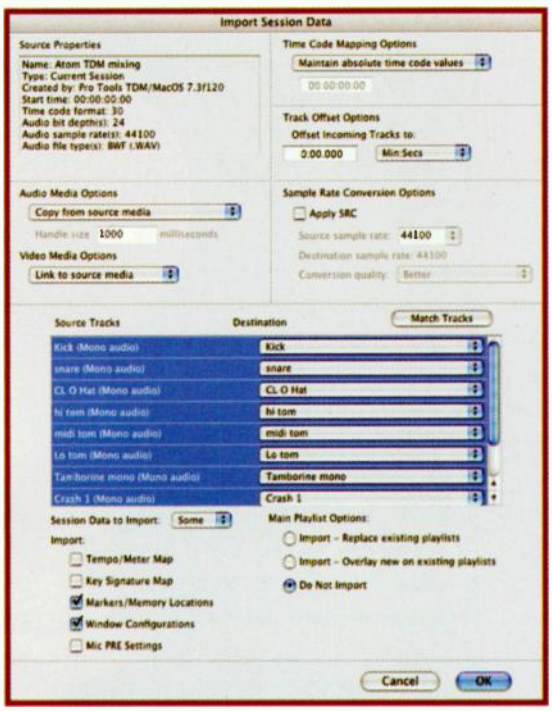


FIG. 5: Import Session Data in Pro Tools HD can be used to import any combination of track and session parameters from one session to another.

## Changing the Reels

This tip is exclusive to Pro Tools HD. (A work-around for LE and M-Powered users is available online; see **Web Clip 4**.) Although it's useful in myriad situations, its power is immediately evident when you're working on several songs by the same band. You work your way through the first song and get it sounding just right, and then it's time to move on to the second song. In a tape-based studio, you remove one reel and put on the next, leaving your console, patch bay, and effects alone. If you used the same track order on the second song as you did on the first, song 2 is half-mixed already. In a DAW, you might feel as though you're starting over from scratch when you open song 2.

In Pro Tools HD, you can simply import the mix from one song to another using Import Session Data (see **Fig. 5**). Once known as Import Tracks, this powerful function can selectively import plug-in and I/O assignments, automation playlists, and many other aspects of a session into another. In Pro Tools LE, only the name has changed—it can still only import entire tracks, including audio regions.

Within the second song, go to File→Import→Session Data and navigate to the first song's session file. If you used the same track-naming conventions, Pro Tools can automatically correlate the appropriate tracks; otherwise, you can assign them manually. From the Session Data To Import drop-down list, choose which parameters to import, making sure not to import alternate playlists. Under Main Playlist Options, choose Do Not Import so that none of the audio regions from the source session are imported. In this way, you can apply

any or all of song 1's mix to song 2 (and 3, and 4 . . .).

Import Session Data can also be used in either HD or LE to import memory locations and window configurations, so your prep work on one session can speed another even more. Show/hide and group-enable memory locations are track specific, so they won't translate, and neither will selections or markers. Track height, zoom level, and pre-/postroll will translate perfectly, however.

## Templates

You may find it useful to create one or more template sessions tailored to your most common mix scenarios. You can create a blank generic session from scratch or take a representative existing session and strip out its edit playlists and regions. Set the template to be read-only on the PC, or declare it to be a stationery pad under File Info on the Mac. This will prevent it from being changed easily. If you ordinarily follow projects from tracking through mixing, you'll probably want to start each session from a template, but if you often get called in to mix projects that were created elsewhere, you'll simply import the mix attributes from the template.

If you're impatient for Digidesign to implement effects chains, add some tracks to your template session that have your favorite plug-in sequences and presets on aux tracks. Make the tracks inactive so they don't use any resources. When you import the plug-ins or drag them from the dummy tracks to active tracks, they will be activated.

## Get Busy

Of course, obsessing over organization is anathema to creativity, so don't get too carried away. Still, the more you can do to preempt any unnecessary interruptions to the creative process, the better. Pick the techniques discussed that seem most valuable to your work flow and use them as a springboard for your own innovative efficiencies.

At the conclusion of a project, when you're burning CDs and backing up sessions, take a few minutes to reflect on the procedures that ate up valuable time or disrupted your creative flow. Figure out ways to streamline the process—a new template, a different way of organizing memory locations, memorizing a useful shortcut—and implement the cure right away. Next time you start a project, you'll be glad you planned ahead. EM

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*Brian Smithers would like to thank Andrew Hagerman of Digidesign and his colleagues at Full Sail Real World Education for their insights and inspiration.*

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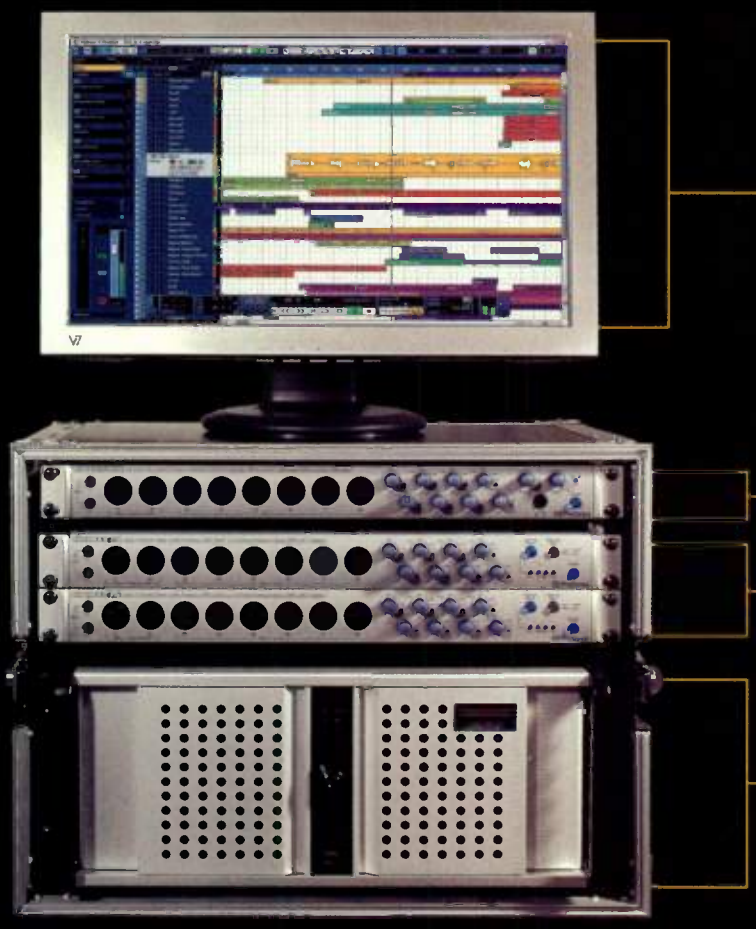

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# Removing Vocal Artifacts

By Dave Darlington

## Make your lead vocals shine with microediting.

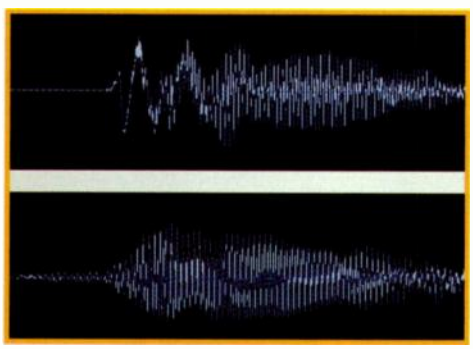
Your lead vocal can make or break your production. The goal of any producer or engineer is to present that vocal in the best possible light, enabling the lyric, musical performance, and emotional content to shine. Lead vocalists not only provide emotionally charged performances, but they also throw in gasps, popping *p*'s, sizzling *s*'s, and annoying lip smacks for good measure. Fortunately, most DAWs offer a number of tools to manage these artifacts and eliminate distracting nonmusical sounds while preserving the song's musical content. Here are six steps I find useful for removing unwanted vocal artifacts (see "Step-by-Step Instructions" on p. 62).

It's easier to locate audio problems when your transport is in tape-emulation mode, in which the cursor remains in place when you stop and continues forward when you resume playback. If your DAW has such a mode, engage it and set one bar of preroll and a generous postroll. Then when you stop the transport, the cursor will be located conveniently one bar before the problem.

### Search and Destroy

Solo the vocal track without EQ or compression and zoom in tight. Listen to the soloed track, and any time you hear any sibilance, breath, or plosive problem, stop the transport. Depending on your preferred work flow, you can deal with the problems as they appear or mark them for later repair and continue on. You will see obvious anomalies in the waveform for plosive *p*'s and *b*'s, as well as telltale sawtooth waveforms for sibilant *s*'s (see **Fig. 1**). Loud breaths are less obvious, but you can use this method to check crossfades between phrases and make the breathing sound natural.

FIG. 1: Before (above) and after shots of a corrected plosive *p* (see Web Clips 1 and 2).



Drawing in volume automation is an easy and elegant solution to many vocal problems. Sibilants and plosives require a sharp, narrow dip, whereas breaths require you to attenuate larger areas. Don't be shy about the amount of volume reduction; these narrow dips pass very quickly (see **Web Clips 3 and 4**). Center the point of the dip right on the problem area, and ramp sharply down and back up so as not to affect the rest of the audio. Don't use sharp right angles, because these will often cause audible pops. Keep auditioning the spot as you tweak the dip, creating automation that is effective but not obvious.

Sometimes the volume automation is too audible to be ignored. In that case, you must alter the offending area of the waveform using your audio editor. This method works especially well on plosives because the problem area of the wave is easy to recognize. Highlight the problem area and include a small amount of extra audio on the right for a crossfade. Using a gain plug-in, reduce the level by as much as 15 or 20 dB. Then pull back the splice point to where the plosive ends and the vocal syllable begins. A tiny crossfade to smooth the transition should now make your edit inaudible to the listener.

### Pull the Plug

Another good way to deal with momentary nonmusical sounds is to automate a plug-in to do the heavy lifting. You can see from a spectral analysis that a plosive is a very low-frequency sound. To rid the plosive of all that low-frequency energy, try inserting a 24 dB-per-octave highpass filter with a 150 to 200 Hz cutoff frequency. Automate the plug-in's bypass control, enabling the plug-in for only the tiny area containing the offending plosive. By tweaking the bypass automation and the filter's cutoff frequency, you can often make the plosive diminish or even disappear, leaving a natural-sounding *p* or *b* consonant. You can similarly automate the bypass of an aggressively set de-essing plug-in, keeping it bypassed except for the most offending regions while leaving the rest of the vocal untouched.

If your volume automation is too clunky to be useful, then let your DAW's preset fades do a bit of fader riding for you. This trick works especially well on overly long *s* sounds. Separate the region near the artifact, trimming some of the sibilance if necessary, and then use the fade tool to draw a fade-in. With a little experimenting, you can get the region and fade lengths just right to make the *s* sound natural without removing it completely.

### The Last Resort

If none of these volume tricks solves a specific problem, then it may be time for minor cosmetic surgery. I

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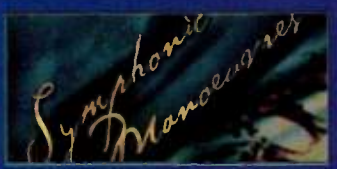


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## STEP-BY-STEP INSTRUCTIONS

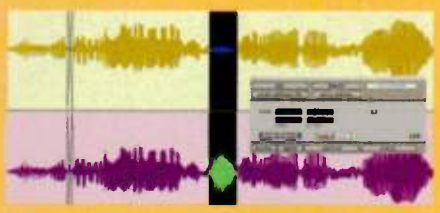
1

Mark and isolate problem areas for future repair.

2

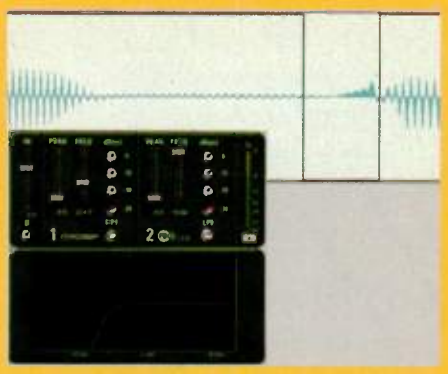
Draw short volume-automation curves to handle sibilants and plosives wherever possible.

3



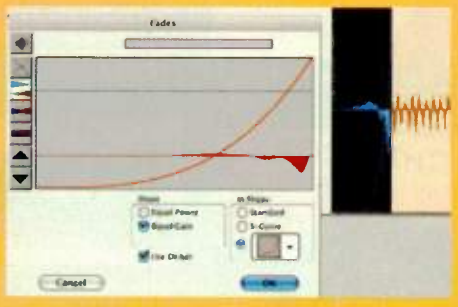
When the automation is too obvious, use an audio editor to reduce a plosive, then pick an appropriate splice point and crossfade to the edited version.

4



Use a 24 dB-per-octave highpass filter plug-in to squelch low-frequency plosives.

5



When riding the volume fader is too obvious, let your DAW's fade-in tool do the work for you.

6

When all else fails, try replacing a sibilant or plosive with one copied from another part of the track.

have had success searching the vocal take for a clean, nonpitched *s* or *t* sound and pasting it over a problematic one. This method is extremely time-consuming, and you need to crossfade the regions carefully to mask the transition, but radical measures are sometimes called for.

Recheck your vocal in solo after all the tweaks. It often takes two or three passes to finesse the various fades and automations. EM

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*Dave Darlington won a Grammy in February for engineering the Brian Lynch/Eddie Palmieri Project album* Simpatico *(ArtistShare, 2006). He has just finished mixing albums for Vesta Williams, Deniece Williams, and Maysa Leak.*

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# Speak to Me

By Jim Aikin

## Getting creative with FL Studio's Speech Synthesizer.

One of the unexpected extras that come with FL Studio 7 is the Speech Synthesizer. If you're doing a track that calls for a phrase from an alien or robotic voice, then you'll probably find that one of the 20 speech models in the menu does the trick. Just type your phrase in the box, and you can hear it spoken or sung by a man, a woman, a munchkin, a nerd, or a robot, among others.

When you click on the check mark to accept what you're hearing, FL Studio asks you to save a .speech file, which contains the phrase as separate words. The program then creates a Fruity Slicer Generator (see **Fig. 1**), in which each word is assigned to a separate key, as well as a piano-roll window from which the words can be played back with the original rhythm.

There are many ways to extend the power of the Speech Synthesizer by rearranging and processing its output. Here are a few of my favorite techniques.

### A Choir

Use the Speech Synthesizer three or four times, entering the same phrase each time but choosing different Personalities. Select Monotone as the Style for each phrase and enter a different pitch. Choose the pitches to create a chord.


FIG. 1: The words produced by the Speech Synthesizer appear in the Fruity Slicer Generator.

Assuming you didn't switch to a new step-sequencer pattern during the process, you'll now have a pattern that will play back a choir singing your phrase as a chord. The sense that it's an ensemble will be enhanced by the differences among the Personalities, but for the same reason, you may find that the choir doesn't have good rhythmic cohesion. You can solve this by quantizing or hand editing the notes in the piano roll so that they have the same start times. Rerendering one of the voices with a different Rate setting may give you a better rhythmic match.

Each note in the Slicer's piano-roll data can be tuned up or down in the event editor strip. Moving the note pitch by more than a couple of half steps will introduce timbral changes, but if you choose initial pitches in such a way that each of them needs to be transposed up or down by only a few semitones, it's quite practical to produce a synthetic choir with moving harmonies (see **Web Clip 1**).

### Massaging a Word

FL Studio's new audio editor, Edison, provides some nice tools for processing samples, including time-stretching, a convolution reverb, and a blur tool. Edison can't load .speech files, but there's an easy work-around. After bringing up the Channel Settings window for the Slicer that's playing your phrase, click on the Plugin tab and then on the second button where you see a file folder. From the drop-down menu, select Save Original Sample. This command can export the .speech file as a WAV file, which you can then load into Edison with markers for the words.

If you want to process individual words in a way that would change their length, create a second Edison. Select the word(s) you want to process, click and hold the Drag button (the one at the right end of the Edison toolbar), and drag into the second Edison.

Drag the resulting file into a new Sampler Generator to trigger it in patterns. Before you do the drag-copy, however, open the Edit Properties window for Edison (the shortcut key is F2) and switch off Tempo Sync. In the early-release version of FL 7, the Sampler Generator has problems playing tempo-synced files.

### Scratch Me Up

From the Channels menu, create a Wave Traveller and drag your synthesized word(s) from Edison into the Wave Traveller. Here you can create a different scratching sound for each MIDI key. A Wave Traveller can contain many scratches but only one sound file. If you need to scratch two or more different sounds from the keyboard, then create a Layer Generator, assign several Wave Travellers to the Layer Generator, and choose Split Children from its menu.

FL Studio is capable of many creative effects, from ring modulation to envelope-controlled morphing EQ. Using the Speech Synthesizer is a great way to start experimenting with them. EM

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*Jim Aikin writes regularly for EM and other music-technology magazines. Visit him online at www.musicwords.net.*

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# Sonic Mayhem

By Brian Smithers

## The down and dirty facts about distortion.

Isn't it ironic that a perfectly accurate account can completely miss the essential truth of an event, while fiction often reveals the most profound truths? As electronic musicians, many of us dance freely between these two realities, sometimes documenting a performance literally and sometimes fabricating a performance out of whole cloth. Often we find ourselves capturing an event as carefully as possible, just to transmogrify it thereafter for our own nefarious purposes.

The concept of distortion, then, is near and dear to our yin and anathema to our yang. In a guitar amp, distortion is treasured, whereas in a mic preamp it is most often reviled. In this article we'll take a close look at distortion. What is it, and what types exist? How is distortion described, and how is it measured?

### Deviant Behavior

Simply put, distortion is any change to a signal other than a simple gain change or delay. Given that definition, one could reasonably look at compression and equalization as forms of distortion, but it's probably more useful to take them at face value and consider any unintended consequences to be distortion. For example, many EQ circuits change the phase of filtered frequencies relative to unfiltered frequencies—*linear phase* equalizers are designed to avoid this phase distortion.

Distortion ordinarily arises when any processor exhibits *nonlinear* behavior (see **Fig. 1**). In a hypothetical distortion-free device, for every change in the input signal, there would be a corresponding and proportional change in the output signal. In a guitar amplifier, for instance, playing twice as loud would result in twice the volume from the amp. Although this may in fact be more or less what happens most of the time, real-world devices always deviate from this behavior at least to some small degree.

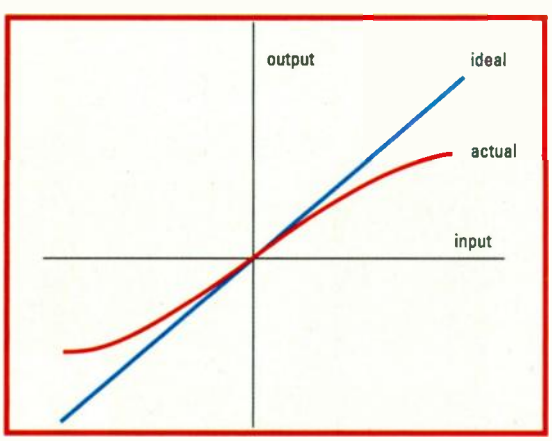


FIG. 1: In an ideal device, output would change in direct proportion to input, but in real-world devices, the relationship between input and output is not linear, resulting in distortion.

All devices have their limitations, and distortion most often occurs as you approach these limitations. At more moderate levels there may be negligible distortion, but at the device's operational limits it misbehaves. If you push too much signal through a device, it will overload and distort. Crank up the volume on an amplifier too much, and it will overdrive and distort. These are amplitude distortions, wherein the output-to-input relationship changes at different volumes.

Usually, if a piece of gear's performance varies according to input frequency, we describe that behavior with a frequency-response curve. However, if that gear's frequency response varies with input level, that's frequency distortion.

### Harmonic Convergence

The best-known form of distortion is *harmonic distortion*. This is a distortion in which the output contains additional components occurring at harmonics of the input signal. Harmonics, or *overtones*, are whole-number multiples of a fundamental frequency. If a 100 Hz sine wave were fed to the input of a device, harmonic distortion would exhibit itself as one or more additional tones at the output, at frequencies of 200, 300, 400, or 500 Hz and possibly other such harmonic frequencies.

If you think this sounds like terrific fodder for synthesis, you're right. Distortion techniques such as waveshaping allow the creation of complex waveforms from relatively simple oscillators. However, excessive distortion in a mic preamp is usually unwelcome.

When a device *clips*, the waveform is flattened (see **Fig. 2**). Analog tape tends to flatten the top of the waveform gently and progressively—in moderate amounts, this distortion is known by the pleasant name *saturation*—whereas digital converters and processors chop the top off abruptly, sharply, and unceremoniously. If this flattening happens symmetrically, the distortion

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# Q&A: Daryl Friedman

By Mike Levine

## Musicians and manufacturers clash over technology.

In the April 2007 issue of EM, this column featured an interview with Michael Petricone of the Consumer Electronics Association (CEA), a trade group that represents technology manufacturers. Some of Petricone's comments raised the hackles of members of the Recording Academy, a group that not only awards the Grammys each year, but also represents the interests of musicians, songwriters, producers, and engineers.

The most controversial statement from Petricone's interview was when he cited the major labels' lawsuit against XM Radio—regarding the recording features in some of its new receivers, which allow satellite radio users to record and organize songs off the air—as an example of how the music establishment is hindering technology that could benefit musicians. Petricone argued that because satellite radio opens up more opportunities for independent musicians to get on the air, a suit that harms it could negatively affect those musicians. The record industry establishment feels that these devices exceed the limits of the Audio Home Recording Act and should incur additional licensing fees.

COURTESY THE RECORDING ACADEMY

FIG. 1: Daryl Friedman is the Recording Academy's VP of advocacy and government regulations.

To give the Recording Academy's side of this issue and others relating to the ongoing battle between technology manufacturers and content producers, I interviewed Daryl Friedman (see Fig. 1), vice president of advocacy and government regulations for the Recording Academy.

**Could you elaborate on what your issues are with the CEA's positions about music technology, and in particular Petricone's comments in the earlier column?**

One is the implication that the CEA represents the interests of artists. That's disingenuous. It's not uncommon, though; everybody loves to say they're representing the interests of artists. The CEA used to say that they represented the interests of consumers; now they say they represent indie musicians. But in fact, the CEA and Petricone represent the technology manufacturers, whose job is to use music in a way that makes money for CEA members. On the other hand, the Recording Academy speaks for tens of thousands of artists and creators. And it does happen that, on the specific issue of satellite radio and XM's recording device, we're actually on the side of the record labels. So one disagreement I have with Michael's comments is regarding the general concept that the CEA speaks for musicians. The other disagreement I have is on the specifics of the XM technology. We believe that what that device does goes beyond the license of a performance for XM to broadcast, and, instead, that it is actually a distribution requiring an additional license.

**In other words, because this device records the stream off of satellite radio, people can then use it in a way that the satellite radio's current license doesn't cover?**

It's not just that it records off the air. We're okay with that. We're actually huge fans of satellite radio. We believe it's opened up great opportunities for our members, and we think the fact that satellite radio is broadcasting and paying a performance fee for broadcasting is great. It's even fine with us (and here I'm not speaking for the labels, I'm speaking for the Academy) that there are devices out there that can record blocks of time. But this device goes beyond that and manipulates individual tracks. If you record for one hour, for example, you won't just have a one-hour block of time on the device, you'll have a list of individual songs and artists. You can then do anything you want with those tracks. You can disaggregate them, you can delete the ones you don't like. You can playlist them in a different order, you can playlist them by artists—however you want to do it. That goes beyond the Audio Home Recording Act, and that, we believe, requires an additional license.

**Petricone also argued that because satellite radio opens up broadcasting to a wider range of**

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**musicians than terrestrial radio, adding additional financial pressure to those broadcasters could harm indie musicians.**

Well, satellite radio certainly does open up avenues. It allows independent artists to be heard more, and we love that. We love to partner with satellite radio, and we have. But what Michael is saying is that independent artists would be better served if we let the user take that track and store it on a device without paying anything extra. Would the independent musician be better off in that scenario than if users went to iTunes and paid 99 cents for each track? We don't think so. We strongly believe that musicians deserve to be compensated. If Michael and the CEA are really concerned about artists, they should want to see that those artists, who are basically the backbone of these industries, are being compensated for every use of their music. And the other thing, this notion that content is stifling technology, is really completely out of line with reality. In fact, it's content that's driving all these technologies.

**I think there's a perception out there that your group marches in lockstep with the RIAA [which represents the major labels], but in fact, your group is entirely independent, right?**

Yes. We have different aims and a different constituency. Representing creators is very different from representing the labels. Very often there are tensions between artists and labels; that's part of the history of the recording industry. But on these particular issues, we happen to be united with the labels. What we're talking about is how much users of our music should pay to use our music to build their technologies and build their companies. We may argue over how we split that money once it comes in, but we'd rather have more money coming in to fight over than to take the CEA's view, which is that uses of our music by certain technologies should not be compensated for at all.

**What about the issue of Podcasters playing copyrighted material? Some say the publicity helps the artists.**

If the artist owns their copyright and the artist wants to let it be used for free for publicity, the artist should absolutely have the right to do that. But what's happening a lot is, the artists aren't making these choices. The technology companies and the Podcasters are making the choices for them. And that's where we think it's wrong. A lot of Podcasts are licensed—that's wonderful. Any new avenue that connects our music-creator members to their fans is great. We're in favor of it. All we ask is that there be a compensation for each use.

**Are there other new technologies that your membership views as potentially problematic?**

There's certainly the potential for an issue with high-definition radio like there is with the XM device. As HD comes out, there's potential for the same sort of functionality to be on receivers of high-definition digital radio. The industry and the broadcasters are working on that issue as the technology is developing. But probably the most important upcoming issue is the matter of royalty payment by traditional, terrestrial radio broadcasters for performances. We might want to get into this a little bit here because, you may be surprised to hear, it's an issue where the Recording Academy is actually on the same side as the technology companies. Webcasters have to pay the songwriter and the artist performance royalties for broadcasting their work. So does satellite radio. In other countries besides the United States, traditional, over-the-air radio also has the same rule: they have to pay the artist and the songwriter. But here in America, broadcasters have an exemption on that payment. They pay songwriters, but they don't pay artists. It's a huge anomaly where the United States is out of step with the rest of the world. This is an issue we're going to be fighting for, and this is an issue where we will be on the same side as the Webcasters and satellite radio. Because they want a level playing field. They're paying, properly, royalties to both artists and writers. And yet their competitors, represented by the NAB, Clear Channel stations, etc., only pay songwriters, not artists, and that creates an unfair disadvantage for the new technologies.

**Overall, there seems to be a contentious state of affairs where music technology—especially on the consumer side—and commerce intersect. There are a lot of competing interests.**

Yes. One thing we're doing at the Recording Academy is to follow up on our president's keynote speech at Recording Arts Day in Washington last September, which called for a truce between the technology and content industries. Recently, there's been this sort of war on Capitol Hill, with PR campaigns and a lot of rhetoric going back and forth. At the Recording Academy, we believe that the technology and content industries actually have more interests in common than we do interests that divide us. Neil Portnow, the Academy's president, is engaged in a couple of activities that are working toward that goal of moving both industries forward. One of them is a retreat this summer that's going to put leading members of music technology, along with labels and artists, in a closed-door retreat in the Bay Area—hopefully, away from the spotlight, away from the media, and away from the policy makers. The hope is that industry leaders sitting down together can begin to come up with solutions that will help us all to move forward. EM

*Mike Levine is an EM senior editor.*

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# DIGIDESIGN 003

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## A venerable product gets more than a makeover.

By Nick Peck

Digidesign has released a successor to the Digi 002, its popular midrange Pro Tools LE system. Dubbed the 003, this system is more evolutionary than revolutionary, maintaining the Digi 002's principal design ideas while updating a few features and giving it a significant cosmetic face-lift (see Fig. 1).

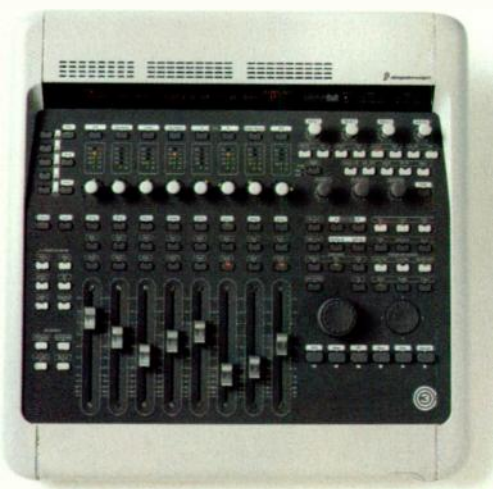


FIG. 1: The 003's control surface includes a jog/shuttle wheel and dedicated automation-mode buttons (middle left).

The 003 comes with Pro Tools LE, which is capable of handling up to 32 audio tracks at 24-bit, 96 kHz resolution, and 256 MIDI tracks. (The track count is expandable to 48 stereo tracks with the purchase of Music Production Toolkit [$495] or DV Toolkit 2 [$1,299].) Because LE is a native system, the limitations of your CPU may restrict the number of tracks you can actually use.

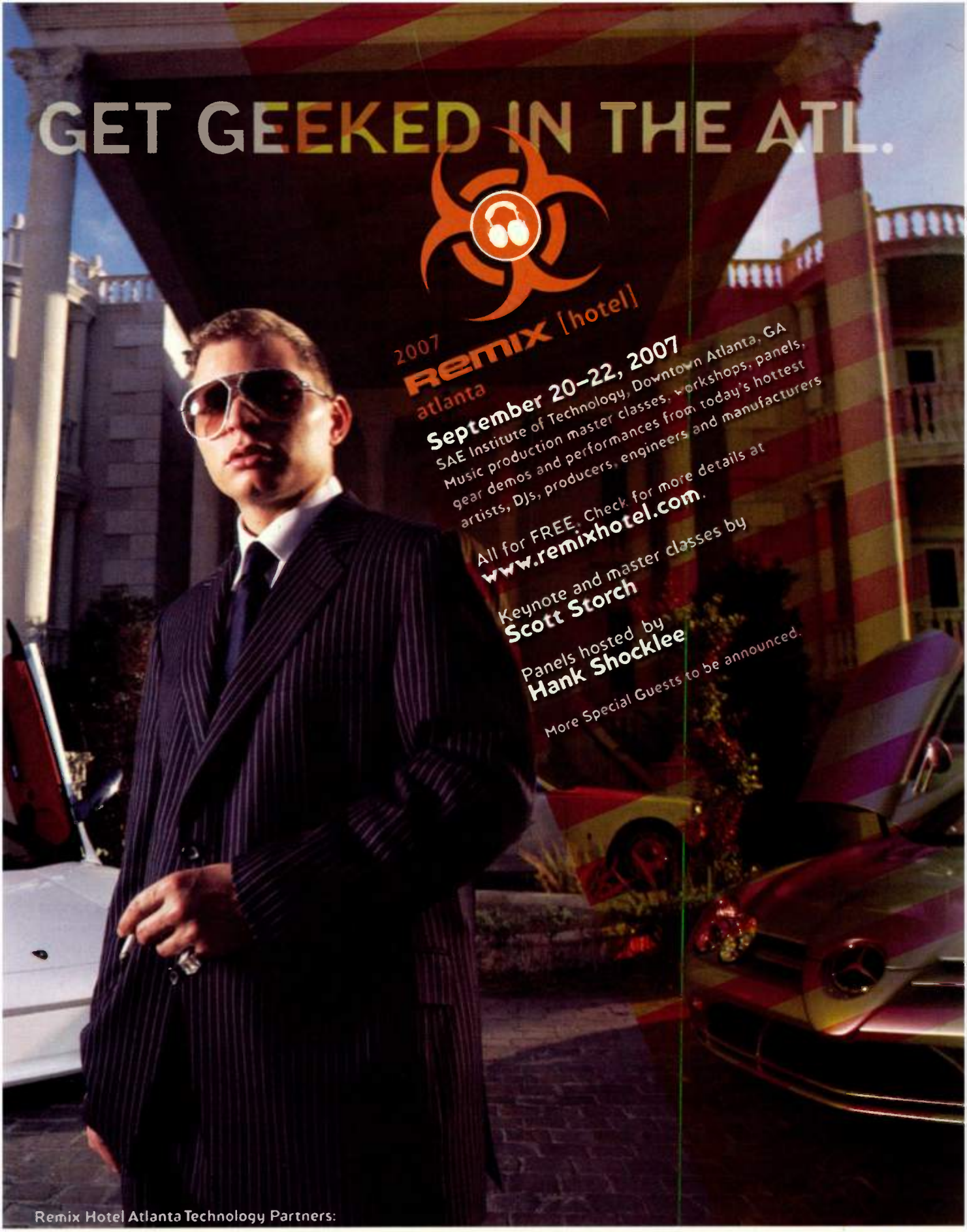
The 003 comes in three flavors: Factory ($2,495), Rack Factory ($1,695), and Rack ($1,295). The Factory model is a desktop design, with a control surface on top and I/O around the back. Both rack models are 2U designs with gain and monitor controls on the front panel, and a rear panel that is nearly identical to the desktop version's. The Rack version comes bundled with the standard DigiRack plug-ins and the Ignition Pack 2 set of effects, virtual instruments, and sound libraries. The Factory and Rack Factory include the Ignition Pack 2 Pro set, as well as the Factory plug-in bundle. All these extras are enough to keep you busy for quite a while, and specific details can be found on Digidesign's Web site.

### The Ins and Outs

As an interface, the 003 can handle 18 channels of combined analog and digital I/O: 8 analog channels, 8 digital over ADAT Lightpipe, and 2 digital over S/PDIF. In the analog realm, you have eight line inputs and four mic inputs to choose from. Mic inputs 1 through 4 are on XLR connections, with a 48V phantom power

**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

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button for each pair (see Fig. 2). Line inputs 1 through 4 are labeled as DIs and are optimized for instrument-level sources, though you can use them with line-level sources by adjusting the input gain. Mic/line switches, gain knobs, and 75 Hz highpass filters for the first four inputs are located on the front panel.

Inputs 5 through 8 are fixed-gain, line-level inputs on balanced ¼-inch TRS jacks, with individual +4 dBu/–10 dBV switches. A pair of +4 dBu aux inputs on ¼-inch TRS jacks are included, and the balanced connectors suggest they are intended for pro-level sources. (On the Digi 002, the corresponding inputs were –10 dBV on RCA jacks, which is appropriate for consumer-level audio gear.)

The monitoring section consists of Main and Alt outputs, which can be used to feed two separate pairs of speakers. The two sets of monitor outputs are an improvement over the Digi 002, where line outputs 1 and 2 doubled as the main monitor outs. The monitor gain knob and Main/Alt monitor select buttons are on the front panel, as are Mute and Mono buttons.

The 003 also has eight fixed-gain line outputs on balanced ¼-inch TRS jacks, running at +4 dBu, and a pair of ¼-inch stereo headphone jacks, each with its own level knob—a welcome addition. As with the Mbox 2 Pro, outputs 3 and 4 can be assigned to the second headphone jack. The monitoring system is simple overall, but comprehensive enough for a small studio. Unfortunately, the Digi 002's standalone audio mixer mode is not available on the 003.

However, the 003 can be used as a standalone MIDI controller. Each fader, rotary encoder, and button can be mapped to any MIDI channel and continuous controller value. MIDI presets can be named and saved to flash memory, and you can create up to eight MIDI maps to use with programs such as Propellerhead Reason, Ableton Live, and Apple GarageBand. A MIDI input and two separate MIDI outputs are also included.

Two-channel S/PDIF I/O is available on coaxial and optical ports, offering 24-bit, 96 kHz resolution. The optical I/O can also be used as an ADAT Lightpipe interface to send and receive eight channels of 24-bit, 48 kHz audio. A footswitch jack is available for punching in and out, and the power connector is the standard IEC type.

For those with complex digital studios, the 003 has an important addition to its I/O: word clock in and out on BNC jacks. This allows the 003 to slave to a master clock, or to be used as the master clock for devices that sync to word clock at rates up to 96 kHz.

The 003 has two FireWire 400 ports for computer connectivity, and it passed the bandwidth test: I recorded, edited, and played back 32 simultaneous channels of 24-bit, 96 kHz audio to a single FireWire drive daisy-chained from the 003. Loads of edits and fades didn't faze it, and playback was nice and solid. I am a bit surprised that the 003 does not have a FireWire 800 port to keep up with current computer hardware design. However, the FireWire 400 port clearly has enough bandwidth to handle the maximum number of tracks at maximum resolution.

PRODUCT SUMMARY

DIGIDESIGN 003

DAW interface/control surface
003 Factory, $2,495
003 Rack Factory, $1,695
003 Rack, $1,295

| | |
|---|---|
| FEATURES | 4 |
| EASE OF USE | 3 |
| AUDIO QUALITY | 4 |
| VALUE | 3 |

RATING PRODUCTS FROM 1 TO 5

**PROS:** Ability to assign automation modes, I/O, sends, and plug-ins from front panel. Word-clock I/O. Improved LCD. Two headphone jacks on front panel. Enhanced MIDI controller mode.

**CONS:** Mushroom-shaped knobs are uncomfortable, seem fragile, and are spaced too close together. Jog wheel feels flimsy and is not easy to use. Standalone mixer mode from 002 has been removed.

MANUFACTURER
Digidesign
www.digidesign.com

## Bird's-Eye View

Like the Digi 002 and Command|8, the 003 has eight channel strips, each consisting of a 100 mm motorized fader; mute, solo, and select buttons; and a rotary encoder ringed by LEDs. The transport controls and Pro Tools–specific function buttons are laid out similarly to other Digidesign control surfaces, and I was happy to see the inclusion of a dedicated Save button.

The Digi 002's 4-character LED display has been replaced in the 003 by a 110-character, 2-line display, which allows you to see longer channel names and parameter values simultaneously. Inputs, outputs, inserts, and sends are now all assignable and removable from the control surface. Dedicated automation-mode buttons, located on the left side of the faders, are a welcome addition and were previously available only on Digidesign's high-end products, such as the Control|24.

The company has also added a jog/shuttle wheel. The outer ring allows you to shuttle back and forth at various speeds, while the inner wheel lets you scrub the audio for fine-resolution editing. Holding down the Nudge button while moving the jog wheel rapidly scrolls the track assignments across the faders.

## New Mic Preamps

Digidesign touts the 003's improved mic preamps and A/D/A converters. I did an A/B test, recording a male voice and a Martin acoustic guitar through the Digi 002 and 003, using a Neumann U 87 mic plugged directly into the mic preamps (see **Web Clips 1 through 4**).

For an all-in-one box, the Digi 002's mic preamps make me perfectly happy: they are quiet, detailed, and fairly neutral. But I have to agree that the 003's preamps are a subtle, though worthwhile, improvement.

FIG. 2: The rear panel, which is similar to that of the 003 Rack, includes balanced aux inputs and word-clock I/O.

The low midrange feels a bit fuller, the top end is not quite as bright, and the overall sound is somewhat smoother. Digidesign mentions that the dynamic range of the 003's mic preamps has been improved by 6 dB over the Digi 002, and that total harmonic distortion is extremely low.

## Ergonomics

My beef with the 003 is in the component choice for the user interface. Although I like the unit's white color and rounded edges, I have problems with its knobs and buttons. Veering away from a more traditional design, the knobs resemble a mushroom cap, where a thin stem is crowned with a thicker, rounded top. Everyone I spoke with who has used the 003 had the same opinion: the knobs are difficult to grasp and feel fragile.

The overall impression is that the first time something falls on the 003 or someone reaches over the unit and catches their sleeve on it, the knobs will snap off. Because of their shape and position, there is little room between the eight rotary knobs, and my fingers would not fit comfortably between them.

In addition, the 003's transport buttons do not feel as solid as those on the Command|8 or Digi 002. The four arrow buttons on those latter units have been replaced with a single 4-directional pad, which feels loose and cheap.

I'm a big fan of scrub/shuttle wheels in digital audio. Unfortunately, the 003's scrub wheel is too small and feels fragile. Rather than having a circular knob to grasp or an index finger indentation to put your finger in, the wheel has a series of small dots that are meant to create traction with your finger. However, I can't see using this method on a regular basis. And although most scrub wheels are fairly large, metal, and slightly weighted, the 003's wheel is small, plastic, and lightweight.

## New or Improved?

I was expecting the 003 to be a bigger move forward from its predecessor. Although it offers some new features and improvements over the Digi 002, the core capabilities, as well as the price point, are essentially the same. Digidesign has acknowledged the 003's modest innovation, referring to it as a "refresh" of the Digi 002, not as a full-fledged new product. Consequently, the 003 is aimed not at users wanting to upgrade from the Digi 002, but rather at new Pro Tools users, as well as Digi 001 and Mbox owners who want to upgrade their hardware.

To be fair, though, the Digi 002 is a tough act to follow. While I don't like the 003's knobs and some other design elements, these are balanced by better mic preamps, an improved LCD, and the ability to assign automation, send, and plug-in data directly from the control surface.

If you already own a Digi 002, I don't see enough in the 003 to justify a change. However, if you are just starting out in Pro Tools or are upgrading from a legacy LE system, the 003 is worth investigating. EM WEB CLIPS

*Nick Peck (www.underthebigtree.com) is a composer–sound designer–vintage keyboardist in the San Francisco Bay Area. His new CD,* Fire Trucks I Have Known, *will be released in 2007.*

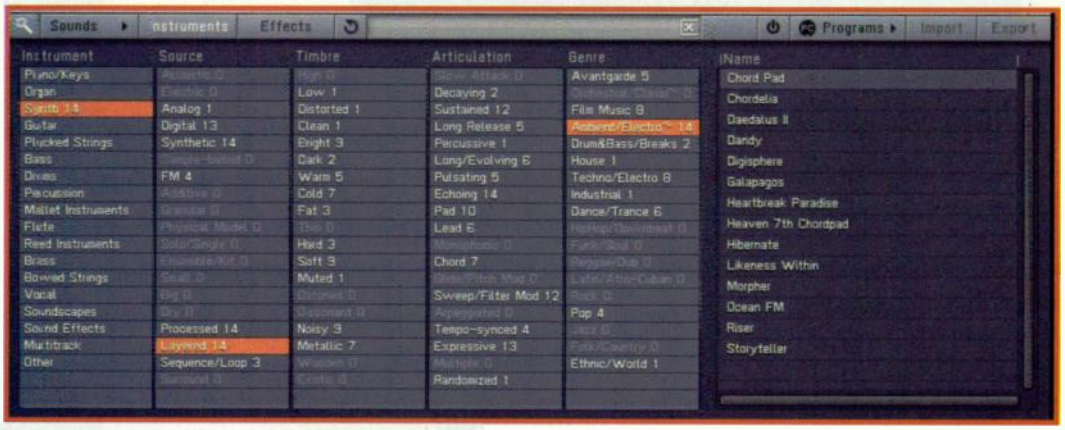


FIG. 1: The new Kore-style browser filters presets according to five attribute classes.

# NATIVE INSTRUMENTS Komplete 4 (Mac/Win)

## Three major synth upgrades make this bundle a must.

By Len Sasso

Since its introduction at the 2003 AES show, the Native Instruments Komplete bundle of virtual instruments and effects has grown to include most Native Instruments offerings, and the price has remained consistent. Komplete 4 contains all of the company's synths and effects except Massive, the latest in its product line. Of particular note, Akoustik Piano has been added to the bundle, along with major upgrades to three of Native Instruments' most appreciated synths: Absynth 4, Battery 3, and FM8. Intakt and Kompakt, which were part of the Komplete 3 bundle, are discontinued products and not part of Komplete 4. (For a closer look at the changes in Absynth 4 and Battery 3, see the **online bonus material** at www.emusician.com.)

The Komplete Care option available in previous years has been replaced by a generous upgrade program. Owners of Komplete 2 or 3 can upgrade to Komplete 4 for $339. That is an incredible bargain considering the addition of Akoustik Piano and the quality of the three upgraded synths. Owners of two or more of the synths (not effects) in Komplete 4 can purchase the full bundle for $1,149, or they can buy the bundle along with Kore, Native Instruments' hardware and software plug-in hosting environment, for $1,708. (You'll find a full review of Kore in the October 2006 issue of EM, available online at www.emusician.com.)

### At the Kore

In addition to the aforementioned products, Komplete 4 includes the instruments B4 II, Elektrik Piano, Kontakt 2, Pro-53, and Reaktor 5 and the effects Guitar Rig 2, Vokator, and Spektral Delay. These come with a huge library of multisampled-instrument, synth, and effects presets. All instruments and effects come as standalone applications as well as plug-ins in the usual Mac and PC formats. Furthermore, their presets have been integrated into the Kore browser.

Both Absynth 4 and FM8 have a browser similar to Kore's (see Fig. 1). Factory presets are assigned attributes in five categories—Instrument, Source, Timbre,

Articulation, and Genre—and you use the browser to display presets matching the attributes you select. A very handy option displays the number of hits for each remaining attribute once one or more have been selected. You can change any preset's attributes and, of course, categorize your own presets. Searchable text attributes are also available, but the two types of searches—category and text—cannot be used simultaneously.

## Keynotes

If you're a keyboard player, Komplete 4 has you covered with sampled acoustic and electric pianos and an emulated Hammond B-3 organ with Leslie. These instruments' sounds and highly evolved user interfaces set them apart from their competitors. And when you want to reach beyond authenticity, you can load Akoustik Piano and Elektrik Piano presets into Kontakt 2 for more-extreme processing.

Akoustik Piano won an EM 2007 Editors' Choice Award, and it ranked at the top of EM's survey of standalone sampled pianos, "Software Eighty-Eights," in the October 2006 issue. It contains four exquisitely sampled pianos: Bechstein, Bösendorfer, and Steinway concert grands and a Steingraeber upright. The user interface is clearly laid out and uncluttered. The controls are pianistically meaningful, which makes tweaking the pianos a breeze. Although no sampled piano measures up to the real thing in solo performance, Akoustik Piano can hold its own onstage and in the studio.

Elektrik Piano (reviewed in the December 2004 issue of EM) starts with samples from four classic electromechanical keyboards: the Fender Rhodes Mark I and Mark II, Wurlitzer 200A, and Hohner Clavinet. The user interface is minimalist and most closely resembles a Fender Rhodes. Four knobs on the left change function depending on the preset; for example, controlling an ADSR envelope for one preset and tremolo, chorus, and reverb for another. The sound is great and the standalone version is certainly gigworthy.

From its inception in 2000, Native Instruments B4 was the preeminent organ emulation. B4 II added a bunch of new tonewheel sets, more speaker and amp setups, and improved MIDI playability. You'll especially appreciate it if you're an organist with back problems. (You can find a review in the July 2006 issue of EM.)

## Synth Zone

Emulations of classic hardware synths are common these days, but Native Instruments was first with Pro-5, an emulation of the Sequential Circuits Prophet-5. That has evolved into the third-generation Pro-53, which, although enhanced with all the conveniences of software emulations, remains a reasonable facsimile of the original. (Its predecessor, Pro-52, was reviewed in the February 2001 issue of EM.) Among other things, you're not limited to the Prophet's arcane 8 × 8 preset scheme;

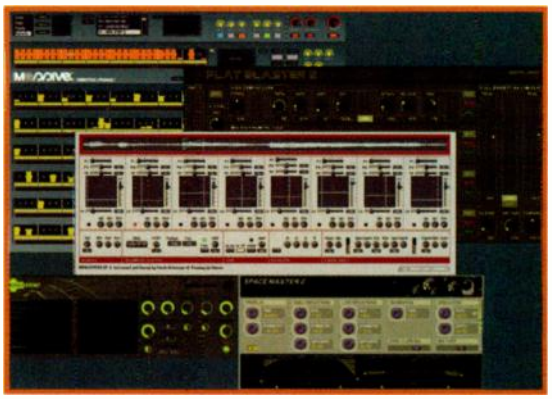

FIG. 2: Reaktor 5 includes dozens of virtual instrument and effects Ensembles ranging from standard fare to off-the-wall.

you get full-featured plug-in host automation and MIDI remote control with prebuilt mappings for popular control surfaces; and you can import or create your own microtuning scales. You can even process external audio through the Pro-53 filter, amplifier, and effects sections. You might use that, for instance, to add the great Prophet filters after your favorite Yamaha DX7 patch in FM8 (see **Web Clip 1**).

Native Instruments' next classic-hardware target was the DX7. FM7, released in 2001, was a faithful replication of the original. The new FM8 takes things several steps further with some important enhancements. An Easy Page gives quick access to essential operator parameters. Morph Square (also found on the Easy Page) lets you seamlessly morph between four presets using automation, MIDI, or the mouse. The new effects rack adds 11 effects. In addition to the original factory sounds, the library now includes FM7 Sounds Volumes 1 and 2 and roughly 200 presets showing off the new FM8 features.

Arpeggiator/step-sequencer hybrids are the current rage, but the FM8 version is particularly robust. As with any arpeggiator, the note pattern is based on the held notes, but the order in which they're played, the number of steps, which steps are accented and tied, and octave and semitone transposition are all set on a step-by-step basis. You can randomize any or all of these elements for a kaleidoscopic approach to pattern generation.

## Overreaktion

The original version of Komplete included Reaktor Session, a player for instruments and effects (called Ensembles) created in Reaktor. All subsequent versions of Komplete contain the latest full version of Reaktor—currently Reaktor 5. This is a great thing if you want to create or modify Reaktor Ensembles. But even if you don't have the time or inclination to get under the hood with Reaktor, don't overlook it as an Ensemble player. The dozens of factory and thousands of user Ensembles arguably offer more novel sounds and unusual effects

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Music Instruments & Pro Audio

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Introducing the 003™ and 003™ Rack: the next generation of professional Pro Tools® personal studio systems from Digidesign. Combining superior hardware designs, industry-standard Pro Tools LE® software, and the all-new Ignition Pack 2 instrument and effects plug-ins and tools, the 003 and 003 Rack empower you to achieve professional studio results wherever you create.

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than everything else in Komplete 4 combined.

The Reaktor Ensemble Massive (not to be confused with the new synth of the same name) is among my favorites (see **Fig. 2**). It's a 6-track drum computer that uses granular resynthesis, among other techniques, to mangle each track's sampled source material. The GrainStates Ensemble from the Reaktor 4 library is another favorite. It also uses granular techniques, this time to create a kaleidoscope of evolving timbres. SpaceDrone is another ambience generator. It starts with 96 noise sources and then uses bandpass filters and panning to spread them across the frequency and stereo spectra. **Web Clip 2** combines those three instruments with compression and reverb effects Ensembles in a hands-free ambient collage.

## Making Kontakt

Kontakt 2 is to sampling what Reaktor is to instrument and effects building. It's a full-featured, modular sampler that even has its own scripting language. Kontakt 2 (reviewed in the November 2005 issue of EM) can import presets from most hardware and software samplers and supports all popular audio-file formats.

Kontakt 2's modularity sets it apart from other samplers. All Kontakt 2 instruments start with a Source module that you can configure for standard playback from RAM or streaming from disk, for granular resynthesis using one of three algorithms optimized for pitch-shifting or time-stretching, or for loop slicing and playback (think REX or Acid files). The Source module is followed by racks for insert and send effects into which you drag various DSP modules from the Kontakt 2 browser. The browser also has a tab from which you drag modulators such as LFOs and envelope generators to the knobs you want them to control.

Kontakt 2 comes with a 15 GB library of multisampled instruments. That includes a special edition of Vienna Instruments from Vienna Symphonic Library, a variety of presets illustrating the Kontakt Script Processor (KSP), individual instrument collections, an assortment of presliced loops, and multitimbral setups containing several instruments mapped to different MIDI channels. The KSP presets are especially interesting. For instance, KSP Elektrik Guitar chooses samples from different guitar strings depending on context, which makes it possible to play much more realistic guitar parts from a keyboard.

PRODUCT SUMMARY

NATIVE INSTRUMENTS Komplete 4

virtual instruments and effects bundle
$1,499
$1,149 (upgrade for owners of two or more included virtual instruments)
$339 (upgrade for owners of Komplete 2 or 3)

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

RATING PRODUCTS FROM 1 TO 5

PROS: Huge palette of top-quality sounds. Excellent integration of instruments and effects with Kore hosting environment (not included). Tremendous discount over combined individual prices. Generous upgrade path to future versions.

CONS: Requires a fast CPU and lots of disk space.

MANUFACTURER
Native Instruments
www.native-instruments.com


FIG. 3: Guitar Rig 2 reaches beyond mic-amp-cabinet simulation to include a wide variety of stompbox and studio effects.

## The Right Effect

Three top-notch effects round out the Komplete 4 bundle. Guitar Rig 2 is much more than a mic-amp-cabinet simulator, though it offers a lot in that department: eight fully parameterized amp models. It comes with a bevy of effects covering distortion, modulation, EQ, compression, and reverb (see **Fig. 3**). Two tape decks (one intended for playback and the other for overdubbing), together with a Loop Machine, provide all possible combinations of recording, overdubbing, and looping. All three modules allow you to save the recorded material to disk.

The Crossfader splits the signal for separate processing of high and low frequency bands, whereas the Splitter enables separate processing of the left and right channels of a stereo signal where possible. (Some components are stereo; others, such as cabinets and amps, convert the signal to mono.) Modulators, which you can assign to any component parameter, include LFOs, envelope generators, step sequencers (both analog- and digital-style), and an envelope follower. (Guitar Rig 2 was reviewed in the June 2006 issue of EM. For non-guitar applications for Guitar Rig 2, check out "Making Tracks: Rigged Up" in the January 2007 issue.)

The remaining two effects, Spektral Delay and Vokator, work in the frequency domain—they split

the signal into frequency bands using a process called short-time Fourier transform (STFT), manipulate the frequency bands independently, and then convert the signal back to the time domain for playback.

As the name implies, Spektral Delay applies a separate delay to each frequency band. In short, it's a feedback-delay line on steroids. The results tend to be otherworldly and digital, but you can tone them down a bit by using Spektral Delay's Dry/Wet control. (The dry signal is automatically delayed to compensate for the latency inherent in the STFT analysis-resynthesis process.) Each channel of the stereo signal is processed separately and allows predelay processing by enigmatically named effects such as Jelly Roll, Lime Twist, and Horse Tail. (Spektral Delay was reviewed in the December 2001 issue of EM.)

Vokator, as you might guess, is a vocoder, but again STFT analysis is used to analyze the modulation and carrier signals (typically speech and tonal material, respectively) and then put the imprint of one on the other. Because of the symmetric nature of its processing, you can use either input as the carrier or the modulator, and Vokator simply labels them A and B.

Either Vokator input can be taken from one side of a stereo-input signal or from an audio file, and the B input can come from a built-in morphing synthesizer or granular sample player. The output is a mix of A vocoded by B, B vocoded by A, and a direct mix of A and B. For input effects, you get dynamics, EQ, feedback delay, and the special effects found in Spektral Delay. (A review of Vokator appears in the November 2003 issue of EM.)

## Komplete Success

Komplete 4 gives you a lot of bang for your synthesis buck. The bundle price represents a savings of roughly 60 percent. However, it's a savings only if you have use for many of the components.

If you use virtual instruments in your music and are continually on the lookout for new sounds, it's a no-brainer—Komplete 4 has it all. If your music is more mainstream, and you're looking for usable keyboard instruments, it is still a great deal. For sampling and a wide variety of somewhat wacky instruments, the combination of Kontakt 2, Battery 3, and Reaktor 5 is almost the same price, so you might as well buy the full bundle.

EM WEB CLIPS

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


FIG. 1: One of a new breed of handheld recorders, the Korg MR-1 is the first to record 1-bit, 2.82 MHz audio. It includes a small stereo electret microphone.

KORG MR-1 and MR-1000

See Product Specs @emusician.com

# These two recorders are just a bit different.

By Geary Yelton

With more portable stereo recorders available than ever, recordists have some tough choices to make. The past couple of years have seen hardware innovations from a variety of manufacturers, from Fostex and Sony to Edirol and M-Audio. Two of the newest machines to stir up some excitement are from Korg, a company known for its multitrack digital recorders.

Eschewing the flash memory that competing models embrace, the pocket-size MR-1 and book-size MR-1000 contain fixed 20 GB and 40 GB hard drives, respectively. They offer a comprehensive selection of audio formats, the most notable being 1-bit Direct Stream Digital (DSD). They also record 16- and 24-bit linear PCM audio at sampling rates from 44.1 to 192 kHz and store those files in Broadcast WAV (BWF) format. The MR-1 can record MP3 files at a fixed rate of 192 Kbps, and both recorders can play MP3s at all standard rates. The bundled software application AudioGate (Mac/Win) imports audio files by means of the recorder's USB 2.0 port and converts them from any supported format (except for MP3) to any other (see the **online bonus material** at www.emusician.com).

## Bit by Bit

Although most EM readers have lots of experience using multibit digital recorders, you may be unfamiliar with single-bit audio. It's easy to assume that 24 bits must be 24 times better than 1 bit, but that isn't necessarily true. The MR-1's maximum sampling rate (2.82 MHz) is 64 times that of a standard audio CD, and the MR-1000's maximum rate (5.64 MHz) doubles that. Another 1-bit advantage has to do with the way A/D converters process PCM audio. (However, because your computer's audio interface can't play 1-bit audio files, you'll need to play them through the recorder's outputs if you want playback at full fidelity.) Rather than trying to explain the theory behind 1-bit audio's potential superiority, I suggest checking out "Future Proof Recording Explained" on Korg's Web site, which details the differences between single- and multibit formats.

One-bit recording has been around for more than 15 years, and it has served as the basis of the DSD Interface

Format (DFF or DSDIFF) used by Super Audio Compact Disc (SACD) players since 1999. In addition to DSDIFF, the MR-1 and MR-1000 record and play two other 1-bit formats: DSD Stream File (DSF), supported by some Sony VAIO computers, and Wideband Single-Bit Data (WSD), a format used by the 1-Bit Audio Consortium, a manufacturers' group.

## Small Is Beautiful

The chrome-colored MR-1 isn't much larger than a full-size Apple iPod, and it contains a factory-installed, rechargeable lithium-ion polymer battery, just as the iPod does (see **Fig. 1**). Its user interface also bears some resemblance to the iPod's. You turn a data wheel (called the parameter dial) on the right side panel to scroll through menu choices, press the same wheel to select them, and press a Menu button to step backward through the menu hierarchy.

Two additional buttons on the right side panel increase and decrease the playback level, and holding down a sliding switch for two seconds turns power off and on. Sliding the switch in the opposite direction locks it to prevent accidents. On the left side panel are a connection for the 5 VDC lump-in-the-line power supply (which, with cable, weighs much more than the unit it's powering) and a miniature USB port like those on digital cameras.

On the top panel, line and headphone outputs are unbalanced stereo minijacks, and the left and right inputs are minijacks that accept balanced or unbalanced sources. Although that arrangement offers some flexibility, it precludes using a stereo mic with a TRS miniplug unless you rig a splitter. Phantom-powered mics are out of the question without a separate mixer or preamp. Also on the top panel are a switch to select either mic or line inputs, and another to enable +3V "plug-in power" for the included CM-2M stereo electret microphone.

On the front panel, the MR-1's graphical 160 × 104–pixel LCD has a backlight you can set to remain on either all the time or for between 2 and 60 seconds when you press any button. In addition to displaying menus, the LCD shows level meters and status, information about selected files, and various parameter settings. Below the display are five transport buttons for playback and pause, record, stop, rewind, and fast-forward. When you're recording, pressing the parameter dial conveniently toggles to the Rec Level screen, which is calibrated from –95.5 to +31.5 dB—an ingenious solution for instantly changing levels.

Korg says that the MR-1's battery is good for about 2.5 hours of continuous operation, depending on the data format you choose. Of course, starting, stopping, and using the backlight will shorten battery life. That figure falls short of recorders that rely on flash memory because of the need to power the MR-1's internal hard drive. And the battery takes about 5 hours to reach a full charge with the MR-1 connected to AC power. For recording without an AC source for more than 2.5 hours, then, you'll need an external battery pack; Korg plans to introduce one that holds four AA batteries soon after you read this, and numerous third-party 5V battery packs are currently available. When the MR-1's battery will no longer hold a charge, Korg recommends taking it to a service center for replacement. Doing it yourself should be a simple operation; however, the battery sells for more than $80.

## Bigger Is Better

The MR-1000 is the MR-1's tabletop counterpart (see **Fig. 2**). It doubles the DSD standard's sampling rate and contains a hard drive twice the size of the MR-1's. The MR-1000 has a sturdy aluminum chassis and comes with a handy padded carrying case. Power is supplied by the included 12 VDC lump-in-the-line supply (which ironically is more lightweight than the MR-1's) or by eight AA batteries, which yield four hours of mobile power.

The MR-1000's backplate has ample room for full-size audio connections (see **Fig. 3**). Two inputs on XLR/TRS combo jacks share a phantom power switch for the XLRs and a low/high-gain switch. Balanced outputs are on XLR jacks, and unbalanced outputs are on RCA jacks. A limiter on/off switch, standard USB 2.0 port, and DC power connector complete the rear panel. A front-panel TRS jack accommodates stereo headphones and has its own level knob. All that's lacking are minijack inputs, which would be handy for certain compact mics (like the CM-2M).

The MR-1000 has the same display as its smaller sibling, but thanks to a larger front panel, it furnishes dedicated, concentric Rec Level knobs instead of the MR-1's Rec Level screen. The MR-1000's larger transport buttons, power and Menu/Esc button, and parameter dial all have an excellent feel. You can reach and see all the controls when the recorder is hanging from your shoulder in its carrying case, which has a thoughtful hinged Velcro panel that affords easy access to the input jacks.

The MR-1000's ability to record 1-bit audio at 5.64 MHz makes it unique among recorders. Korg suggests that the best use of this ability is to capture and archive your master recordings, and I agree. Recordings of live performances at the highest 1-bit rate sound closer to the original than anything I've ever heard, including analog tape.

PRODUCT SUMMARY

KORG MR-1

portable stereo recorder
$899

| | |
|---|---|
| FEATURES | 4 |
| EASE OF USE | 3 |
| AUDIO QUALITY | 5 |
| VALUE | 3 |

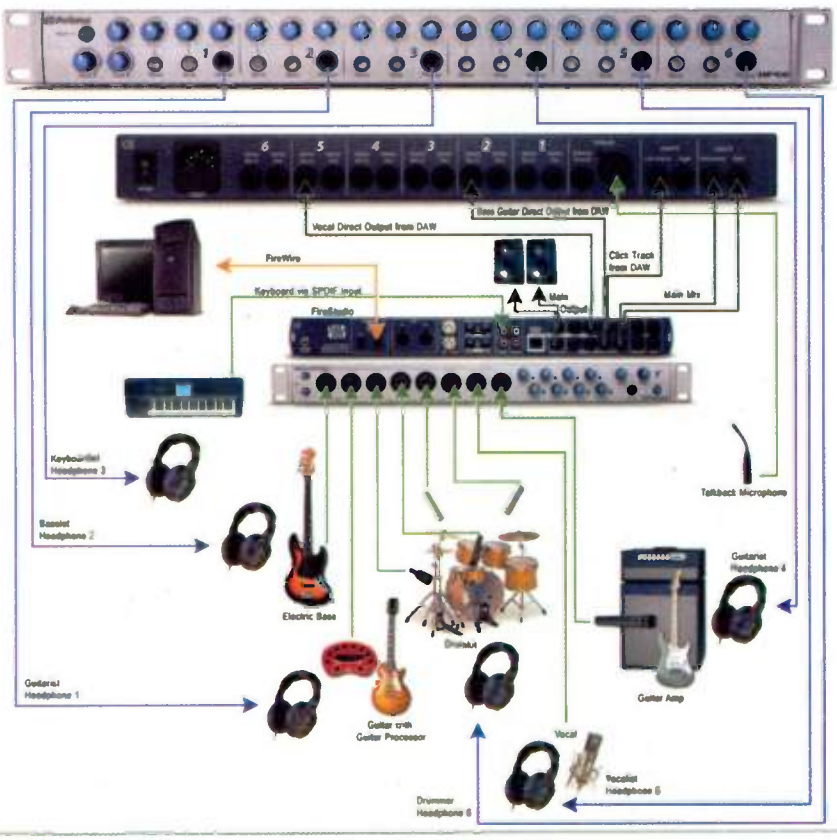
RATING PRODUCTS FROM 1 TO 5

PROS: Excellent sound. Supports many audio formats. Very portable. Includes a sensitive stereo mic.

CONS: No digital audio I/O. Audio I/O on minijacks only. Disappointing battery life. Expensive battery.

MANUFACTURER
Korg
www.korg.com


# HP60 Six-Channel Headphone Mixing System




**"CAN I GET MORE ME?"** Everyone in the band wants a different headphone mix. Give the drummer more click, the singer more vocal, the guitarist more guitar, the bassist more bass, all without effecting each other's mix!

It's easy with the **HP60** since you have three stereo or mono inputs to mix between for each of six headphone outputs! Send the main mix to Input A, the click to Input B and each individual musician's output to their direct input on each channel or any other combination you can dream of. Use the HP60's talkback to communicate with the musicians at the push of a button. And as with all PreSonus products, you can be sure that your HP60 will be loud and clear!

FEATURES

- Six independent ultra low-noise and HIGH OUTPUT headphone amplifiers
- Two sets of stereo or mono inputs (A and B) with balanced TRS connectors
- Stereo external input point on each channel for "more me" control
- Mix control between inputs A and B
- Talkback with external XLR microphone
- Direct stereo line output on each channel
- Mono and mute on each channel
- **$299 (estimated USA street price)**

*The recording experts at Sweetwater have been helping pro and project studio owners put together their perfect recording rig for decades. They know that when a musician has the right headphone mix, they turn in an inspired performance. Call Sweetwater today and pick up an HP60. Your sessions will never be the same again!*

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REVI

## Out Here in the Field

I captured a variety of ambient and musical sources using both recorders. For remote recording with the MR-1, I used the included CM-2M stereo mic. The mic comes with a bracket that serves as a tiny mic stand, and the bracket has a threaded hole for mounting on a camera tripod if needed. The mic performed well, though it was extremely susceptible to handling noise, and recording outdoors would have been impossible if I hadn't rigged up a windscreen. Nonetheless, the CM-2M's high sensitivity proved advantageous in some circumstances and even allowed me to make microscopic, close-miked recordings of insects buzzing and water dripping.


FIG. 2: With a maximum 1-bit sampling rate of 5.64 MHz and a maximum 24-bit rate of 192 kHz, the compact MR-1000 boasts plenty of professional features.

For recording with the MR-1000, I used a matched pair of sE Electronics SE3 small-diaphragm condenser mics. They have a cardioid polar pattern and come with shockmounts and a bar for mounting both mics on a single stand. I preferred the MR-1000 for recording live musical performances for two reasons: the phantom-powered XLR inputs allowed me to select mics most appropriate for the job, and 1-bit recordings made at 5.64 MHz were so lifelike. Of course, using superior mics made a difference. Which machine you choose will depend on your budget, though, and either one offers advantages for a host of applications, from Podcasting



PRODUCT SUMMARY

**KORG MR-1000**

portable stereo recorder
$1,499

| | |
|---|---|
| FEATURES | 4 |
| EASE OF USE | 4 |
| AUDIO QUALITY | 5 |
| VALUE | 4 |

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent sound. Supports many audio formats. XLR I/O. Phantom power. Uses AA batteries. Real input-level knobs. Carrying case included.

CONS: No digital audio I/O.

MANUFACTURER
Korg
www.korg.com

to taking a feed direct from your mixing board.

On either machine, you can add markers by simply pressing the Record button as you're recording. Markers are added automatically if you pause and continue recording, and pressing the Record button during playback also adds markers.

New recordings are automatically named by their format followed by a sequential number (WAV_0001, for example). You can rather quickly rename files with up to 16 characters by scrolling through lists of upper- and lowercase letters, numbers, and symbols with the parameter dial and selecting them one at a time.

## Get It to Go

A pocket-size recorder like the MR-1 is desirable for many situations, like when you want to record discreetly or you don't want to carry mic cables or adapters around. The MR-1 is well suited to those occasions; its 2.5-hour battery life can be a drawback, but work-arounds exist if you don't mind carrying an extra power source. I wish there were some way to attach the included stereo mic to the MR-1 itself, but noise from the internal drive would be a problem. Still, the MR-1's size and fidelity definitely make it an ideal companion for capturing sounds you might otherwise miss.


FIG. 3: On the MR-1000's rear panel, you'll find connections for analog audio I/O, USB 2.0, and 12 VDC power, as well as switches for phantom power, a limiter, and high or low gain.

I used the MR-1 for a couple of months before I received the MR-1000, and though I appreciate the MR-1's portability, I'm more impressed by the MR-1000. I do wish both recorders had digital audio ports, however. Although 1-bit recording on the MR-1 sounds amazing, the MR-1000's top rate sounds twice as amazing. Either way, the awesome fidelity and performance of 1-bit recording have become affordable.

*EM associate editor Geary Yelton has been fascinated with capturing sounds ever since his older brother brought home a reel-to-reel tape recorder in 1962.*

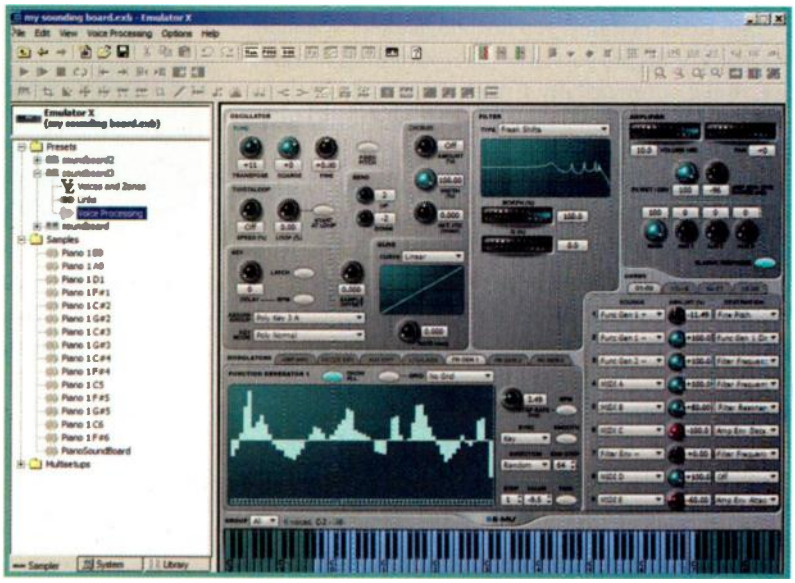

FIG. 1: Emulator X2's Voice Processing page provides access to many powerful sound-programming options. The interface, shown here with the Tree view on the left, is clean and well laid out.

## E-MU Emulator X2 (Win)

# More power, new tools in this high-end sampler.

By Dennis Miller

E-mu has released Emulator X2, a major update to its high-end Emulator software sampler. Among the new features in X2 are the ability to use the software with any third-party audio interface, automated hardware-device sampling, custom modulation control sources, and a host of powerful sound-design features.

Unlike the original Emulator X sampler, which required one of E-mu's professional audio interfaces, X2 requires only the included Xmidi 2x2 USB MIDI interface to run. That means you can use it with any audio interface you choose. You can also purchase X2 in an add-on version if you already own an Xboard USB MIDI series controller or any of the full range of E-mu audio interfaces. At a list price of $399 for the Xmidi version, X2 is among the least expensive of the major Windows software samplers (Steinberg HALion 3 lists for $399, Native Instruments Kontakt 2 is $449, and Tascam GigaStudio is $599, for example).

We covered the original Emulator X in the February 2005 issue of EM (available online at www.emusician.com), so after a quick overview, I'll focus on the main features of the X2 version. I'll also give a preview of the forthcoming Platinum upgrade, which should be released around the time you read this (see the sidebar "Go for the Platinum").

## Going Soft

Emulator X2 is modeled on the sound engine in E-mu's highly regarded Emulator hardware, which has been under continuous development since the early 1980s. The software version provides a vast number of professional sampling features, and unlike many other software samplers, it actually samples (stereo only,

Once you get comfortable with X2, you might consider unloading all that hardware.

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however). X2 has a host of options for tweaking your recordings, such as the ability to detect individual, isolated hits within a single recording and convert them into multiple samples. Once the file is split into samples, X2 will automatically build presets that use them. It will also automatically normalize each sample and detect and eliminate silence at the beginning of a recording. You can use the same set of features on any sound file (WAV and AIFF) that you load from disk.

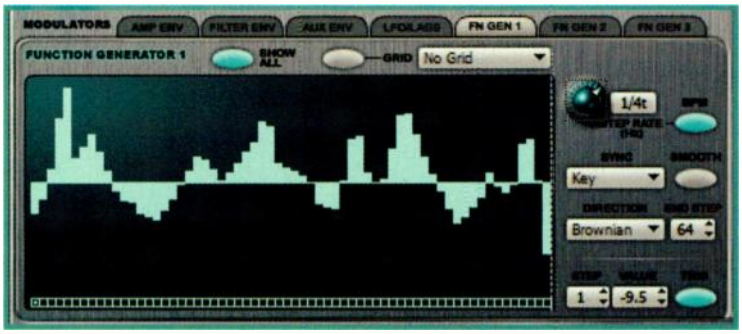


FIG. 3: Using the new multifunction generators, you can draw control data for use with any modulation destination. A variety of controls are available to tweak the data before it reaches its destination.

Emulator offers a vast number of sound-programming options, many of which are found in the Voice Processing page (see Fig. 1). Voices are the components that make up Presets, and you can have as many Voices in a Preset as you want. The Voice Processing page provides access to a large number of filter settings and filter presets, as well as to the LFOs and new function generators (more on those in a moment). You'll also find glide and chorus controls and effects and aux settings in this screen.

The Voice Processing page is home to the modulation matrix, where you assign up to 36 Cords (automation pathways) to connect a modulation source to a destination. Each Cord has an Amount control that lets you further tweak the interaction. Equally useful are the Modulation Processors, which allow even greater manipulation of the source data before it reaches its destination. Using the Quantizer, for instance, you could turn a smooth, continuous control signal like an LFO into a quantized, stepped pattern. Other Processors let you mix, offset, amplify, invert, or perform a variety of transformations to your data.

Emulator retains one archaic feature from its hardware days. Rather than providing a MIDI Learn option to assign an external MIDI control source to a parameter directly, it requires you to first assign the external source to an internal routing system, which uses the labels MIDI A through MIDI P for the routing pathways. You then assign one of the internal MIDI sources to the desired destination (a number of routings are set by default). Even then, there's no option to let Emulator autodetect the external source (some software will detect and assign a control source if you simply move the hardware slider or knob that you want to use). But you can save the mappings for use across all your banks, and if you don't change the hardware source of the controller messages, then the routing setup is a one-time deal.

One of the advantages that many software samplers have over hardware devices is the ability to layer multiple presets on a single MIDI channel. Using the Links function, you just drag as many individual presets as you want from the Tree view on the left of the screen to the work area on the right, and they will all trigger when you play a note on the current channel. This lets you build huge, multilayered sounds with very little effort. You can tweak volume, pan position, and coarse and fine tuning for each preset, and if you use the same preset multiple times, you can quickly build chorused sounds by simply altering the tuning of each layer.

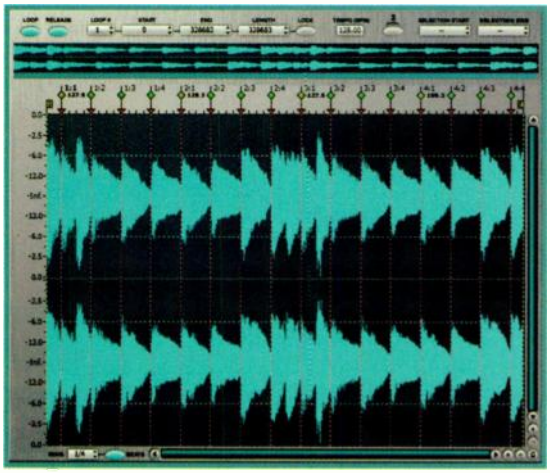

FIG. 2: The Beat Analysis tool does an excellent job of detecting even minor tempo changes in a sample. Note the changes shown at the beginning of measures 2, 3, and 4.

## Twist and Shout

X2's new TwistaLoop feature is intended primarily for people doing beat-oriented music, but it has the potential for more-esoteric sound-design applications as well. TwistaLoop analyzes a sample and provides a tempo map of the pulses, then allows you to transform individual notes, beats, or entire measures (see Fig. 2). There are a large number of settings you can adjust to produce good results for many different types of material, though like other tools of its ilk, TwistaLoop works best with music that has clear transients.

Once the sample is analyzed, you can set an unlimited number of loop points within it (in the previous version, you could have only one loop per sample). From

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there, the sky's the limit: you can jump from any loop to any other while playing back by simply selecting the loop number from a drop-down menu or assigning loop regions to specific keys or MIDI controllers on your keyboard. Other TwistaLoop features, well covered in an online tutorial, let you override the individual tempos of multiple loops and conform them to a single master tempo or change pitch without changing tempo. Individual loop regions can be edited (reversed, time-stretched, or faded, for instance), and you can configure even more playback parameters once you create a preset around your sample.

TwistaLoop did an excellent job of analyzing the beats in the examples I threw at it, even detecting very slight tempo changes within a single measure. On my 3.7 GHz Pentium 4 machine, a 12-second loop took less than 10 seconds to be analyzed, though times will vary depending on what settings are used for the beat analysis.

## Swipe a Sound

If you own a lot of synth hardware, you'll appreciate the new SynthSwipe feature. SynthSwipe will send out a series of MIDI Note Ons to trigger your external devices, then record the sound generated by each trigger in sequence. You determine how long each Note On lasts, whether SynthSwipe will record every note or only some interval that you choose (every perfect fourth or fifth, for instance), the duration (in seconds) between notes, and the Velocity level for each trigger. Press Record, and then SynthSwipe takes over.

As cool as that is, though, SynthSwipe works only with the currently loaded program—an option to send out 128 Program Changes and record *all* the sounds in a bank would be even niftier. And the ability to transmit a Bank Select message would allow you to grab everything in your synth at once. Of course, once you get comfortable with X2, you might consider unloading all that hardware.

PRODUCT SUMMARY

**E-MU Emulator X2**

software sampler
$399

| | |
|---|---|
| FEATURES | 4 |
| EASE OF USE | 4 |
| QUALITY OF SOUNDS | 4 |
| VALUE | 4 |

RATING PRODUCTS FROM 1 TO 5

PROS: Large number of sound-design features. Intuitive sampling and sample-processing options. Several sample libraries included.

CONS: Inadequate documentation. SynthSwipe feature could be more efficient. No tool tips. No MIDI Learn feature.

MANUFACTURER
E-mu
www.emu.com

## Talkin' 'Bout My Generators

In addition to its two multishape LFOs, X2 provides three new user-programmable multifunction generators that you can use as modulation sources for any available destination. You draw the shape you want the function to generate (up to 64 steps), then apply a variety of controls including speed, direction, and sync method (Key, Free Run, or Channel) to tweak the data before it reaches its destination (see **Fig. 3**). You can also determine whether the function's output adds to, subtracts from, or alternately adds to and subtracts from the value of the destination it is intended for.

Convolution is all the rage these days, and X2 gets into the game with its Transform Multiply feature. Though there's only a single parameter (Intensity) to adjust, you can use TM to impose the room ambience of a real space onto a recording of your choice or convolve any two audio files (a drum loop and a vocal sample) for cross-synthesis purposes. (See **Web Clip 1** for an example of a montuno crossed with a dense piano cluster.) Simply copy all or part of a sample to the Clipboard, access a second sample, and click on Transform Multiply, and the deed is done. This is an area that I'd really like to see expanded.

## Referential Treatment

X2's 328-page manual is mostly for reference purposes—it mentions a vast number of things the software can do but is not clear on how to do many of them. (Some usage tips are scattered throughout, though.) However, E-mu has recently released an Advanced Applications Guide that covers a wide range of common and more-creative tasks. The guide is a must-have if you want to get the most out of the software. But it would also be helpful if E-mu implemented some type of pop-up help system for the various buttons and knobs on the interface.

X2 ships with several sample libraries—more than 3 GB in all. Though you won't find some of the more advanced logic-driven programming features of megabucks libraries, such as automatically changing the selected sample based on what range of MIDI notes you are playing, there are a number of "intelligent" features built into the newer collections. As expected, many of the included libraries are aimed at popular contemporary styles (Hip Hop Producer and Beat Shop, for example), but you'll also find the entire 1,000-plus sounds

## GO FOR THE PLATINUM

E-mu's Emulator sampling engine continues its evolution with the forthcoming release of X2 Platinum. Platinum, which should be available by the time you read this, will provide a number of new features, such as streaming sample playback, multiprocessor support, and new beat-slicing options. A massive library containing more than 20 GB of sounds will be included with the release, along with a 64-bit version of Emulator X2.

from the Proteus 2000 tone module as well as many other esoteric offerings.

E-mu releases new collections on a regular basis—for example, most recently a 7 GB (with included 1 GB and 200 MB versions) grand piano, a small (2-CD-ROM) collection of historic instruments, and a 10 GB orchestral library (see the Sound Central area at E-mu's Web site for a complete list). You can also use the included sample conversion program to convert samples in SoundFont, GigaStudio, HALion, Akai, and other formats that you might have on hand.

I've been a longtime Kurzweil K2X00 user and must admit that I really had no intention of making the switch to a software sampler. But the ease of integrating X2 into my work flow, whether standalone or as a VST plug-in, was really too tempting to pass up. If you own a lot of synth or sampling hardware, the SynthSwipe feature can ease the transition to a soft environment, and even if you haven't worked with a lot of sampling hardware, you'll find X2 to be a very intuitive program.

Moreover, the ease with which you can alter the included sounds (or your own) is striking. With very little effort, I removed the pitched component from a piano multisample, leaving only the sounding-board element, which I proceeded to modify with several interacting multifunction generators (see **Web Clip 2**). The great-sounding filters and extensive set of sound-design features provide nearly unlimited potential for shaping your sounds, and the extensive modulation matrix offers enormous opportunities for creating evolving, time-varying sounds no matter what style of music you work with.

I did run into one rather serious issue when using X2 as a plug-in in Cakewalk Sonar 6.2: I was unable to get it to respond to Program Changes that were inserted into a sequence (Program Changes that I entered in the Track Properties field were sent when playback started). You can work around this by building a Multisetup in X2 and assigning each patch to its own channel, and then using multiple tracks in Sonar for each different part. (Cakewalk acknowledged that this is a problem with Sonar and says it expects to have it fixed in a future version.)

When E-mu released the first hardware sampler under $10,000 in 1981, no one could have foreseen an all-soft version appearing even in the most distant future. With the release of X2 and its very minimal hardware requirements, the cycle has been completed.

EM WEB CLIPS

*Associate Editor Dennis Miller is a composer and animator. Check out his work at www.dennismiller.neu.edu.*

# QUICK PICKS

## AUDIO-TECHNICA

### ATM450

By Larry the O

Audio-Technica has completely redesigned its Artist Series line of performance-oriented microphones. One worthy model, the ATM450 ($369), is a small-diaphragm, side-address, cardioid electret condenser mic. It is a well-built, 5-inch-long charcoal stick with recessed switches on the body for an 80 Hz highpass filter and a –10 dB pad. Although no power supply is needed to polarize the permanently charged capsule, 11 to 52V phantom power is required for powering the onboard preamp.

The ATM450's stated frequency response is 40 Hz to 20 kHz, but the specifications don't mention tolerances for that range, and the frequency plot's resolution is 10 dB per division. Consequently, you're left to rely on your ears to judge the mic's real-world response. The specs also say that the ATM450 can handle SPL levels up to 152 dB peak and has a signal-to-noise ratio of 69 dB. Though the ATM450 is useful in the studio, this indicates that it is really a stage mic and not a mic for critical, high-gain recording situations.

Audio-Technica's ATM450 is a side-address, cardioid condenser mic that's small enough to squeeze into tight spots, making it especially useful on drums and percussion.

#### The World's Address

The side-address orientation is somewhat unusual for a small-diaphragm condenser, and obviously changes how the mic is placed. In many live situations, especially on drums and percussion, it simplifies getting the mic into a tight spot. Other applications require techniques more often associated with large-diaphragm mics, such as XY stereo-miking configurations, where two ATM450s must be placed nose-to-nose with their bodies in-line with each other, as opposed to at the 90-degree angle you'd expect with front-address mics. That isn't a problem, just a bit different than usual, though placement did prove a little tricky when I tried using an overhead XY technique to record vibraphone.

The placement versatility gained from the ATM450's being a side-address mic is further enhanced by the included ATM8471 mount, which worked in every position I needed to squeeze it into. Sometimes the small things count for a lot. The side-address configuration means that there is somewhat more capsule exposure than with front-address small-diaphragm condensers. Though that might make the capsule more vulnerable to damage in a live performance, the ATM450 appears well constructed, so it wouldn't surprise me to find ATM450s in the field with dents in the grille but sounding fine. Still, you can't hand one to a roadie with the same impunity you could if it were a Shure SM58.

#### Mic Check

I tried the ATM450 on a variety of sources, especially percussion, acoustic guitar, and vibraphone (one of my favorite torture tests for any microphone or mic preamp). Overall, my impression was that it sounded truer than many other stage instrument mics, which are often strongly hyped in the mid-high frequencies. The ATM450's frequency response plot does show a significant high-end bump, however.

Acknowledging that the ATM450 is not positioned as a studio mic, I have been doing some 24-bit, 192 kHz recording work and have found that format to reveal a lot about microphone performance. I set up a little comparison test with the ATM450, an Earthworks SR77, and a Shure KSM141, both of the latter costing at least twice as much as the ATM450.

The ATM450 had a very appealing quality that struck a balance between the "documentary" sound of the Earthworks mic and the "in-your-face" sound of the Shure. For instance, shakers were clear, clean, and agreeably nonaggressive, and a few different hand drums all had nice body but also a clear presence of skin on skin (hands on the head).

If I were miking a solo or otherwise very exposed acoustic guitar, I'd be very likely to reach for the ATM450, which had nice body and shimmer. For a strummed acoustic guitar in a band, though, the KSM141's more forward mid-high frequencies would make that my choice. I wouldn't say the ATM450's frequency response is necessarily flatter than the KSM141's, but it does have a gentler quality to it that could be very welcome when heard through a typical horn-loaded club P.A., which can add edge to many instruments.

In the last 15 years or so, performance microphones have come a long way in terms of fidelity. Audio-Technica's Artist Series is an example of this trend, and the ATM450 is a superb specimen—versatile enough to cover a variety of different sources in live situations and a useful addition to a studio. It is an excellent, affordable, all-around utility choice for the mic cabinet.

**Value (1 through 5): 4**
Audio-Technica
www.audio-technica.com

## ALESIS

### ControlPad

By Jim Aikin

If you need to augment your trap kit with a MIDI percussion controller,

# "My #1 Country Hit Started With a Phone Call to TAXI"

***Elliott Park – TAXI Member***

*Photo:* Elliott (left) with publisher, Michael Martin

I used to think that living in Clyde, Texas (Population 3,345) really limited my chances of ever having success in the music business. But all my friends and family members live here, so I've never wanted to move to Nashville.

Although I love to write songs, I felt isolated when it came to getting them heard by anybody in the music business. Then a friend told me that TAXI would bring real opportunities for my music right to my front door.

## I Used a 4-Track

I signed up and sent in songs that I demoed with my digital piano in my little home studio. The A&R people at TAXI liked my songs and began sending them off to some pretty high-level people in Nashville.

All the sudden, doors started opening. With the connections I made through TAXI, I began to have meetings with some of Country Music's top executives, and signed a staff writer deal with a great publisher in Nashville.

## Tim McGraw, Rascal Flatts and Faith Hill Put My Songs on Hold

Over the next three years, my songs were considered by a Who's Who of Country Music, but the "big cut" eluded me. I learned to be patient and worked even harder on my songwriting.

Then, my publisher hooked me up with veteran songwriter, Walt Aldridge. Together, we wrote a song called, 'I Loved Her First,' and finally, I hit pay dirt!

## #1 Hit on Two Charts!

The group 'Heartland' cut our song and released it as a single. It started out slowly, then gained momentum, and eventually made it all the way to the Number One spot on the Billboard *and* R&R Country charts.

Could that have happened without TAXI? Probably not.

Although there were many people that helped me once I signed my publishing deal, it was TAXI that made that all important first connection for me. And I didn't have to leave my hometown to do it.

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Alesis would love to talk to you about the ControlPad ($399.95). With eight rubberized pads that are meant to be played with sticks, the ControlPad is an effective performance tool for drummers who want to trigger sampled sounds but don't need a bunch of fancy features.

The ControlPad can transmit MIDI over the traditional 5-pin DIN cable or via a USB link to a computer. The controller can be powered by USB or from the included wall-wart power supply. A generic USB driver is used for both Windows and Macintosh computers, which means there is no special driver to install: just plug in the included USB cable, turn on your computer, and launch your favorite percussion software.

### Launching Pad

The ControlPad has 21 memory slots for kits, and there is no Save button; any changes you make to a kit will be there the next time it's called up. A kit defines a MIDI Note Number and MIDI channel for each pad, as well as a Program Change number that is transmitted each time the kit is loaded.

Each pad also has its own settings for sensitivity, Velocity curve (four options, including fixed), and triggering threshold. However, these settings are global for the instrument and are not stored separately with the various kits.

The ControlPad's default settings worked fine for me. I encountered no false triggering, double strikes, or other problems. While I'm not much of a drummer, I felt the ControlPad responded realistically to variations in stick velocity. The output Velocities were a bit lower when I played near the edge of each pad—though not by much—but there is a generous sweet spot in the center.

The user interface is simplicity itself. It includes a function selector button (with LEDs that let you know which of the seven functions you're editing), increment/decrement buttons, and a 3-digit numerical LED to show the data values. The buttons are big enough to poke with a drumstick and have concave tops so the stick won't slide off—very thoughtful.

The ControlPad is 9.5 inches deep by 15.75 inches wide, and the boundaries between pads are a bit curved. The pads are about 4 inches by 5 inches, and the boundaries between the pads are dark, recessed grooves, which could make it difficult to play the pads accurately in dim stage lighting conditions.

Rather than including a built-in mic-stand mount, the ControlPad has holes in the bottom that accept third-party mounting brackets such as the Roland APC-33. For tabletop use, the ControlPad's rubberized feet grip surfaces well, so I doubt scooting will be much of a problem as you play.

Around the back of the unit are two footswitch inputs for triggering additional MIDI notes, two trigger inputs for drum sensors, a TRS dual-footswitch input that duplicates the increment/decrement buttons, a USB port, and MIDI In and Out jacks. When the kit select function is enabled, the dual footswitch will work as a kit selector, which is bound to come in handy onstage.


The Alesis ControlPad is an easy-to-use MIDI percussion controller.

### Big Deal

The ControlPad comes with a copy of FXpansion BFD Lite 1.5 (Mac/Win), a software drum module that can be used as a VST plug-in or as a standalone instrument. BFD Lite lets you mix and match instruments, chosen from 1 GB of included sample content, to create your own setups. Using the program's mixer, you can blend the close-miked sound with room and overhead mics.

BFD Lite can also function as an auto-accompanist, thanks to its MIDI groove playback features and generous supply of built-in drum patterns. However, this feature cannot be used with the ControlPad, because it relies on a MIDI key being held down while the pattern plays. The ControlPad sends a Note On message followed immediately by a Note Off, with no possibility of sustaining a note.

### Hit Me!

Overall, the Alesis ControlPad is a simple but effective MIDI percussion controller. Once you factor in the discount pricing available from online retailers and the included copy of BFD Lite, you'll find it to be a sweet deal.

**Value (1 through 5): 4**
Alesis
www.alesis.com

## SONY

## MDR-7509HD

**By Mike Levine**

Sony's MDR-7509HDs ($265) are high-quality headphones aimed at audio professionals. They are designed to offer flat response over a wide frequency range and to be comfortable to wear for extended periods. The *HD* in the name indicates that the headphones' frequency response (5 Hz to 80 kHz) is wide enough for clean reproduction of high-resolution audio.

### Ear to Ear

The MDR-7509HDs feature a circumaural (closed-back) design to keep outside noise to a minimum. Sony clearly paid attention to the details on these headphones. I was impressed with touches like the markings on the adjustable-length side pieces, which allow for precise size adjustment; the cable that terminates in a minijack with a ¼-inch adapter on top of it, giving you the flexibility to connect to both consumer and pro gear; and comfortably cushioned earpieces that can be folded up for storage or flipped around for one-ear, off-the-head listening.

Like on its other pro headphones, Sony chose to use a coiled cable with the MDR-7509HDs. According to a company

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that have distinct right- and left-hand components, such as congas. If you use a Zendrum percussion controller, you can download SysEx data containing custom maps for it. You also get a dedicated, noneditable version of PSPaudioware's Nitro filter plug-in with custom presets. *FHP* supplies impulse responses created by Voxengo for Kontakt 2's built-in convolution reverb. Be warned, however: the Kontakt 2 convolution reverb can impose a sizable tax on your CPU, as it did on my older dual-processor 1.42 GHz Mac G4.

### Bang for the Buck

Instruments range from traditional Latin percussion such as congas, timbales, and bongos to djembe and udu from Africa and naal from India. *FHP* also supplies less conventional instruments such as a boomwhacker and a Morphosis trap kit made from found objects, noise shaping, and FM synthesis. The instruments sound natural and resonant, with beautiful, natural decays.

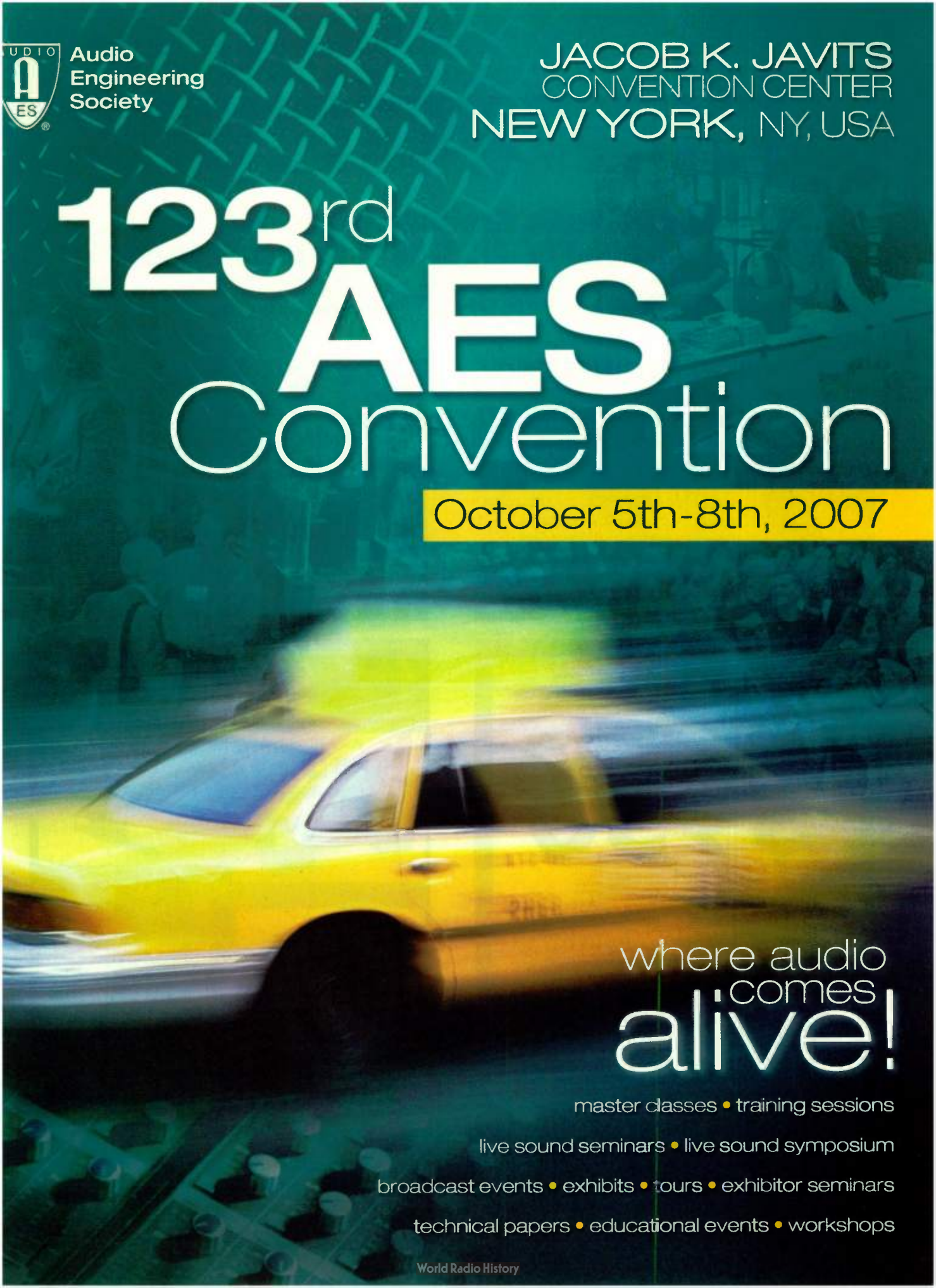
Independent of knobs and sliders, *FHP* delivers lots of real-time sonic control, as you'd expect from a hand-percussion instrument. Transitions between Velocity layers feel smooth and continuous, adding a realistic playing experience to *FHP*'s sonic authenticity. The PDF manual does a terrific job of describing each instrument family and its playing technique. It provides a detailed view and description of each instrument's keymap and assigned controllers, even suggesting playing techniques.


***FlyingHand Percussion* is a massive 24-bit, 44.1 kHz sample library for Kontakt 2, focusing on hand drums and handheld percussion.**

*FlyingHand Percussion* is one of the most detailed, lifelike collections of sampled hand-percussion instruments I've found. Innovative and thoughtful application of Kontakt 2's features breathes a remarkable amount of life into performances. *FHP* gets my highest recommendation, but you don't need to take my word for it—listen to the demos and download a couple of free samples and patches from www.flyingnandpercussion.com. EM

**Value (1 through 5): 5**

HandHeld Sound/SoniVox (distributor)
www.sonivoxmi.com



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## SOUND EFFECTS

# Give Your MOTU Studio 8-core Performance

**Your MOTU desktop studio performance just doubled again with the 8-core Mac Pro tower and a host of exciting new third-party virtual instruments, effects plug-ins and affordable, world-class studio hardware.**

## 8-Core Intel Processing

The new 8-core Apple Mac Pro tower delivers the power of two Quad-Core Intel Xeon "Clovertown" processors running at 3.0GHz. That's eight 3 GHz processors performing as one, allowing you to run Digital Performer 5, a stunning array of virtual instruments and all the powerful plug-ins you need, all from the supreme convenience of a single computer desktop. Just a few years ago, a system with this much real-time audio processing power would have cost you many tens of thousands of dollars. Now you can own it with plenty left over to equip your MOTU studio with the latest and greatest virtual instruments, plug-ins and studio hardware.

## Eight mic inputs via FireWire

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


## Waves native processing

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



The MOTU experts at Sweetwater can build the perfect DP5 8-core studio for you. We'll help you select the right components, and we can even install, configure and test the entire system for you. Why shop anywhere else?

## Jimi Hendrix Guitar Tones

IK Multimedia AmpliTube Jimi Hendrix™ is the first authentic software recreation of the complete Hendrix guitar amp and effects rig in a single product, offering all the rare vintage stomps, amps, cabinets and mics that contributed to make Hendrix's legendary sound. Based on IK's award-winning AmpliTube 2 and made in cooperation with Authentic Hendrix, the official family-owned company, AmpliTube Jimi Hendrix™ features 9 vintage stomp effects, 4 vintage amp models, 7 cabinets and 5 mics, putting a complete collection of extremely expensive and hard-to-find gear right at your fingertips, all carefully modeled with proprietary AmpliTube 2 DSM™ (Dynamic Saturation Modeling) technology and accurate craftsmanship from actual gear of the '68s. Get that classic Jimi Hendrix tone, plus the best vintage guitar effects and amp gear ever offered in software.



## Mastering at its finest

Waves Master Perfection Suite – Six stunning new audio plug-ins at a breakthrough price. Repli-Q provides unparalleled spectral matching for improving an improperly equalized track, mastering for different delivery formats, or applying the EQ of one mix or track to another. Squeez-3 and Squeez-5, two powerful multi-band compressor/limiter/expander plug-ins, feature advanced new linear phase equalization filters. PitchCraft offers super natural pitch correction/transposition and manipulation with minimal artifacts. Reveal integrates seven analysis tools in a single plug-in — Oscilloscope, Peak & RMS Power History, Spectrogram, Pan Power, Spectral Analysis, Lissajous Phase Scope and Peak & RMS Level Meters. Supafreq offers ultimate 4,6, 8 and 10-band paragraphic EQ. GrooEx provides high-quality gating and downward expansion. Available separately, or with Peak Pro XT, the Master Perfection Suite is an indispensable addition to your Digital Performer desktop studio.


## Vintage EQ/Compression

The Focusrite Liquid Mix is another Focusrite first and a true one-of-a-kind. Based on the award-winning Liquid Technology, the Liquid Mix provides 32 channels of simultaneous DSP powered vintage and modern EQ and Compression plug-ins into your DP5 mix without affecting your host computer's CPU. 40 classic compressors and 20 timeless EQs are included. Each EQ and Compressor emulation is painstakingly created though a process called Dynamic Convolution which is a huge step beyond modeling. Through Dynamic Convolution, every frequency at every possible combination of settings is perfectly sampled. That means that you get the true sound and feel of a vintage or modern classic. Tens of thousands of dollars of gear is now right at your fingertips! Think how great your Digital Performer tracks will sound with the Liquid Mix.



## Komplete control

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



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# Give Your MOTU Studio 8-core Performance

**Keystation Pro 88 features**

- 88-key hammer-action, velocity-sensitive keyboard
- Powered via USB bus (cable included) or optional 9V power supply
- 24 MIDI-assignable rotary controllers
- 22 MIDI-assignable buttons
- 9 MIDI-assignable Alps faders
- MIDI-assignable pitch bend and modulation wheels

## 88 Weighted Hammer-Action Keys

Digital Performer 5 gives you unprecedented control over your MIDI and audio tracks. And what better way to take advantage of this hands-on control than the M-Audio Keystation Pro 88. Regardless of whether you're a seasoned pro or just ready to take your music to the next level, these hammer action keys are so expressive that you just won't want to stop playing! The Pro 88 could easily become your sole keyboard in the studio or onstage. Yet the Keystation Pro 88 weighs only 47 lbs. — half of most weighted-action keyboards! And the Pro 88's extensive features make it the most comprehensive and competitive product of its kind!


## Control room monitoring

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


## Lexicon effects processing

With the Lexicon MX500 you can have, in a purely digital form, the rich Lexicon sound that has defined studio reverb for three decades, with the convenience of an automated plug-in right inside your Digital Performer studio environment. The new MX500 takes Lexicon's "Hardware Plug-in" feature to the next level by actually streaming dual stereo audio via FireWire. Like the MX200, MX300, and MX400, all parameters of every reverb, delay and dynamic effect can be controlled from a plug-in window in Digital Performer. The MX500 houses 17 classic Lexicon reverbs, delays, and modulation effects plus dbx compression and de-essing. Front-panel control is easy, too, with a generous LCD display and big comfy knobs you can get your hands around. A wealth of routing options from quad mono to dual stereo are possible. You get 99 Factory programs and 99 user programs for Stereo mode, another 99 Factory and user programs for Dual-Stereo mode, and 25 Factory/25 user surround programs. Also included is Lexicon's intuitive MX-Edit™ Editor/Librarian software, for convenient yet advanced programming.

## Professional pad controller


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

**The MOTU experts at Sweetwater can build the perfect powerhouse DP5 8-core rig for you. We'll help you select the right components, and we can even install, configure and test the entire system for you. Why shop anywhere else?**

## New Mackie monitoring

The high-resolution Mackie HR824mk2 active studio reference monitor sounds as smooth as it looks. The new Zero Edge Baffle™ minimizes diffraction for a crystal clear image of your Digital Performer mix, and controls sound waves for wide, even dispersion. The rear-firing, mass-loaded passive radiator ensures tight, detailed bass extension, down to 35Hz. And thanks to remarkably linear frequency response, you always get accurate mix translation. Acoustic Space, LF roll-off and HF controls let you tailor the sound to suit your MOTU studio space—and your taste. With all this and more, the HR824mk2 turns your Digital Performer desktop studio's sweet spot into a full-on sweet zone.

## New hands-on control for DP5

The new Mackie Control Universal Pro control surface gives you ultimate hands-on control of your Digital Performer desktop studio. Nine motorized, touch-sensitive Penny + Giles faders, eight V-Pots and more than 50 master buttons let you tweak parameters to your heart's content. Unlike generic MIDI controllers, the MCU Pro employs a sophisticated communication protocol that delivers ultra-precise control, makes setup easy - no mapping required - and enables you to see your mix in action with real-time visual feedback via the huge backlit LCD and eight LED rings. Apply the custom overlay for Digital Performer for dedicated labeling of DP-specific functions. The MCU Pro is the ultimate way to mix in DP5!

## Multi-pattern condenser mic

The flagship of the KSM line — and the new must-have mic for any large multitrack studio, the Sure KSM 44 multi-pattern condenser microphone has an extended frequency response specially tailored for critical studio vocal tracking. Its ultra-thin, externally biased, large dual diaphragms provide precise articulation; extremely low self-noise (7 dBA) ensures that the KSM44 captures only the sound of the performance. Inside, the three polar patterns — Cardioid, Omnidirectional, and Bidirectional — offer greater flexibility and uniformity in a wide variety of critical recording applications. Class A, transformerless preamplifier circuitry provides extremely fast transient response and no crossover distortion for improved linearity across the full frequency range.




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# With Special Guests

By Mike Levine

## Strange Angel's debut features legendary players.

For years, Bryan Bell worked behind the scenes as a stage and studio tech and consultant for artists such as Herbie Hancock, Santana, Neil Young, INXS, and Michael Jackson. Then about four years ago, a songwriter named Jim Baxter asked Bell to produce his album. Bell, who's also a guitarist, liked the music so much that he ended up forming a band with Baxter called Strange Angel. "Once we decided that we were making a record as a band, rather than just a demo, I got out my Rolodex, called up all my clients and my friends, and really got serious," recalls Bell.

The result was *Labor of Love* (Synth-Bank, 2007), which features Baxter and Bell's songs and vocals, an eclectic mix of musical influences, and, thanks to Bell, a roster of guest musicians to die for. "The electric band was Graham Lear [Santana] on drums, Freddie Washington [Steely Dan] on bass, and Steve Porcaro [Toto] on keyboards," says Bell, who, along with Baxter and Jay Koder, played guitar. "We had two background singers and an African master percussionist, Obo Addy. Vicki Randall from the *Tonight Show* band also played percussion and sang background."

Then there was the acoustic band: "Herbie Hancock, Ron Carter, Wallace Roney, and Branford Marsalis," says Bell, rattling off the jaw-dropping lineup. "For the acoustic group, the concept was if Bob Dylan had opened for the Miles Davis Quintet and then forgot to leave. So a singer-songwriter backed by the Quintet. And then the electric band was kind of like Herman's Hermits backed by the L.A. Express *[laughs]*—pop sensibilities, dual lead vocals, and a smoking band."


ANDREA BELL

*Labor of Love*/Bell (left) and Baxter at Avatar Studios.

### RIFFS

**Strange Angel**

**Home base:** Portland, Oregon

**Sequencers used:** Digidesign Pro Tools, MOTU Digital Performer

**Recording formats:** 24-bit digital and 2-inch analog

**Web site:** www.synthbank.com/strangeangel

Another story line from the making of *Labor of Love* was the hybrid recording process. "We worked at Falcon Recording Studios in Portland, Oregon, and we recorded to 2-inch for the main rhythm tracks," says Bell. "We recorded the drums, the bass, and the guitars analog, and then we bounced to [Digidesign] Pro Tools and MOTU Digital Performer."

Next, Bell put together a laptop-based mobile studio and hit the road, visiting the various guest artists and overdubbing their parts either in their studios or in local studios. "We literally made a 22-inch rolling suitcase that was an entire studio, and that blew everyone's minds when they saw and heard it," Bell says. Its contents included an Apple laptop, a Digi 001 interface, BSS direct boxes, a pair of vintage Neumann U 87 mics, and cables and other accessories.

"We went into the home studios of Freddie Washington and Steve Porcaro, patched in my Pro Tools rig, and overdubbed all of their parts," says Bell. "In Herbie's studio, we used my directs and went straight in." Ron Carter's bass parts were recorded at New York City's Avatar Studios. Final vocals were cut at Bell's home studio.

With the remote sessions done, the tracks were brought back to Falcon Studios, bounced into Digital Performer, and mixed on a large-format console. The finished mixes were mastered by Paul Stubblebine in San Francisco.

The irony here is that Bell, who spent so many years developing music technology and helping other artists use it, finally got a chance to apply it to his own music. "The combination of this new, lower-cost, more portable technology and access to some of the world's greatest musicians," he says, "allowed Jim and I to realize a vision of our songs that would never before have been possible." EM

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- Stand-alone mixing
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- 10 TRS analog outputs
- Separate main outs
- Main volume knob
- Headphone volume knob
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- Metering for all inputs & outputs
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# Studio to go

## UltraLite

**Compact bus-powered 10x14 FireWire audio interface**

The award-winning 828mkII FireWire audio interface turns your Mac or PC into a top-notch desktop studio. Born from the same innovative design, the award-winning bus-powered UltraLite lets you take your studio to go. And it's the only half-rack audio interface that offers stand-alone operation with programmable mixing from its unique backlit front panel LCD, plus all the analog, digital and MIDI I/O that you need.


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