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JULY 2007

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...and How to Avoid Them

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from the Ground Up

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REVIEWS
SSL Duende, Radikal Spectralis, Digidesign Mbox 2 Mini, *and 6 more*

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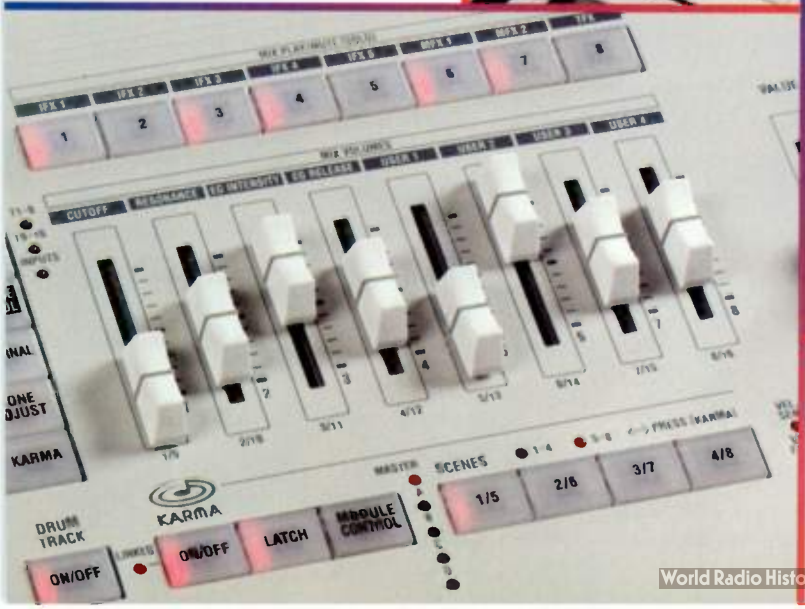

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July 2007

INSIDE

DMITRY PANICH; LOCATION: NEW, IMPROVED RECORDING, OAKLAND, CALIFORNIA

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Setting up a personal studio is not a trivial exercise: you need to consider work flow, equipment location, and financial resources. We offer tips on how to design your new studio so that it fits your needs now and in the future. By Nick Peck

36 TWELVE COMMON MIXING MISTAKES

Mixes that need help often suffer from the same problems, such as uneven levels throughout the range of bass frequencies, large swings in spectral balance, a lack of punch, and an edgy, fatiguing sound. Fortunately, you can avoid or correct such shortcomings with these simple techniques. By Michael Cooper

51 PRODUCTION VALUES: PUSHING THE RIGHT BUTTONS

Bill Bottrell has produced memorable CDs for artists such as Sheryl Crow, Michael Jackson, Madonna, the Traveling Wilburys, and Shelby Lynne. In this interview, he attributes much of his success to his ability to motivate musicians in the studio. The veteran producer also criticizes the music-business establishment, with which he's feuded over the years, and lambastes pop music for its digital perfection. By Paul Tingen

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Digidesign Mbox 2 Mini digital interface

Vir2 Instruments Acoustic Legends HD (Mac/Win) sample library

Abbey Road Studios/Chandler Limited EMI TG12413 Limiter Pack limiter plug-in


Roland RD-700SX

Be Prepared

I was a Cub Scout dropout, so I can't credit the Boy Scouts' "Be prepared" motto with my penchant for planning. I learned about preparation during my years on the road, where one is vulnerable to anything and everything. In such an environment, if you aren't ready for trouble, you are sure to find it.

If you want to produce music, "Be prepared" applies big-time. If you are so eager to get on with tracking and mixing that you rush through the planning phases, you are asking for trouble.

A few months ago, Senior Editor Gino Robair and I started planning an article on how to avoid or fix many problems that occur with mixing. EM contributor Michael Cooper was our first choice to write the story for many reasons, among them the fact that he often mixes tracks that his clients have recorded themselves, and masters projects that were mixed by others. In doing so, he has developed a good understanding of the ways that musicians tend to mess up their mixes. Drawing on his wealth of experience, Cooper has delivered a first-rate story on solving—and avoiding—common mixing errors (see "Twelve Common Mixing Mistakes" on p. 36).


JULIE BIRUM

The deeper that one delves into the problems Cooper has identified, the clearer it becomes that fixing them in the mix is an undesirable solution. It makes a lot more sense to prepare properly and heed Cooper's advice in order to avoid these problems to begin with.

Similarly, EM readers and others have often bent my ear about their dissatisfaction with their studios. Perhaps their room doesn't sound right, they don't trust their cables, or their gear is inconveniently arranged. If you are unhappy with your studio, you are unlikely to be fully productive. In many cases, trouble arose because the studio owners didn't prepare their plans carefully before beginning construction; they just started building.

When author Nick Peck relocated his studio, he was faced with several challenges. The new space was significantly different from the old, and some of the nature of his work had changed. He needed to reexamine his goals and rethink his studio setup. This planning process inspired and informed his article "Starting Over" (see p. 29).

The lessons that Cooper's and Peck's stories have in common may seem obvious, but enough people mess up their mixes and studio setups that a few of these lessons are worth highlighting. A crucial point is to be clear about your goals and keep them in mind when making decisions. With the goals established, take the time to thoroughly plan your project before you begin implementing it. If you plan well and employ some elbow grease, you can do a lot on a modest budget. Don't take the fastest solution just because it's quick and easy; make sure it's the right one for your purpose. And remember, it's not about having the sexiest gear; it's about having the appropriate gear for the job.

Whether you are building a mix or a studio, be prepared for the unexpected, because studio construction and mixing rarely go exactly as planned. And that's the simple truth—scout's honor!

Steve Oppenheimer
Editor in Chief

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LAURA WILLIAMS

Letters

Calculating Latency

In the response to the letter "A Missed Opportunity?" in the May 2007 issue of EM, Jim Aikin said, "I know of no direct relationship between sampling frequency and system latency." Steve Oppenheimer followed with, "I also agree with Aikin that latency is not related to sampling rate." Actually, sampling frequency is one of two important components in calculating latency.

Ignoring some negligible factors, latency can be derived simply in terms of buffer size and sampling rate: latency = driver buffer size (in samples)/sampling frequency (in samples per second). For example, at a sampling frequency of 96 kHz with a 256-sample buffer, latency would be 2.7 ms (256 samples/96,000 samples per second = 2.66 ms). With the same buffer size at a sampling frequency of 48 kHz, the latency would double to 5.3 ms (256/48 = 5.33).

Therefore, buffer size and sampling rate are equal factors in reducing system latency. Given that higher sampling rates typically have the added benefit of increased audio quality, you could even argue that sampling rate is where you should look first to lower latency.

Aikin does make a valid point that increasing the sampling rate will increase CPU usage, to the point that it could choke a system running a complex project. However, when your system can handle it, increasing the bit rate can be a very handy way of both lowering latency and improving fidelity. This would be true of many of today's higher-end DAWs (for example, those that run on AMD Athlon 64 X2 Dual-Core or Intel Core 2 Duo processors) in which the CPU is not pushed to the limit and the input buffer size is at or near its lowest configurable setting.

John Pittas
via email

Author Jim Aikin replies: Your equation is obviously correct, John. I think I was using the term system *a bit loosely to mean not just the computer, but the computer and everything it's doing.*

In my own system, for instance, I typically run a lot of soft synths, so my CPU usage is normally up to 75 percent. I simply couldn't double the sampling rate, because the CPU would choke. (Okay, I could freeze a lot more synth tracks earlier in the composition process and then unfreeze them one at a time in order to edit them, but working that way is a pain in the butt.)

By this definition of the term system, *I can't reduce my latency any further by doubling the sampling rate, because I can't double the sampling rate. The real-world, practical latency of my system as a whole (including my soft synths) is at an irreducible minimum.*

We should have devoted a bit more space in the article to this topic. Thanks for helping to clear up the confusion.

John—Thanks for the correction. I should have had a cup of coffee to ensure that I was awake before answering that letter! —Steve O

Deliver Music Properly

When I read the interview with Michael Petricone (see "Music Business Insider: Q&A: Michael Petricone" in the April 2007 issue of EM), my first reaction was to throw out the magazine and cancel my subscription.

I have heard Petricone speak, as I am on the Audio Board of the Consumer Electronics Association (CEA). I do appreciate and respect that the CEA has stated that piracy is bad, because at any given time over 20 million people on P2P networks are sharing music mostly on devices sold by CEA-member companies.

In regard to their recordings, most musicians I know care about at least these two things: (1) getting paid when people are moved by sounds they create, and (2) giving the listener the best possible audio experience.

Sirius and XM propose to give their subscribers the ability to store a terrible-sounding version of recordings for free. They broadcast at roughly 56 Kbps, which is lower quality than most Webcasts are.

We Welcome Your Feedback

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When I hear either service, I hate the way it sounds. People argue that this quality is okay for listening in the car, but if one were to play a CD in the same system, one would hear how good audio can sound in that car. Sirius and XM argue that they are promoting our works by playing them, which is the same excuse that radio stations use to avoid paying sound-recording fees. This is what Petricone is supporting on behalf of XM, Sirius, and the CE companies that make the devices that consumers use to listen and record.

I assume that people do not want to give away all their music for free and have it sound awful. My viewpoint is most likely the same as 100 percent of EM readers.

My concerns are for the community of artists, songwriters, producers, and engineers who are dedicated to the art of making music. What would record companies, consumer electronics audio companies, and broadcasters do if the creative community stopped producing the music that made those companies' products and services possible?

Elliot Mazer
via email

Elliot—Thanks for your comments. I'm glad you didn't cancel your subscription! In August's "Music Business Insider," we will interview a spokesperson for the National Academy of Recording Arts and Sciences whose views are markedly opposed to those of Mr. Petricone. —Steve O

Error Log

May 2007, "Picture Window," p. 56. It was incorrectly reported that Digital Performer's visual cues can be output through FireWire. That feature is still in development, although MOTU expects it to be implemented in the near future. EM

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MARK C. DAVIS

EMspotlight

Maurice White: Funk's Shining Star

Maurice White helped mastermind numerous hits for Earth, Wind & Fire before launching his own production company in 1994. In this interview from the EM archives, White covers the finer points of tracking a rhythm section, crafting an arrangement, and working with vocal talent. By Michael Molenda. emusician.com/em_spotlight

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By Geary Yelton

M-Audio IE Series

M-Audio (www.m-audio.com) is shipping the IE series of in-ear reference monitors, which comprises three models designed for audio professionals. Designed in cooperation with Ultimate Ears, maker of high-end custom-fitted earphones, the IE-10 ($129), IE-20XB ($249), and IE-30 ($299) each have a lightweight polycarbonate housing and a detachable and replaceable 46-inch cable. All three models include a universal-fit kit with an assortment of silicone and foam ear tips for optimal isolation and comfort. Hanging loops hook over and behind the ears to minimize snags.

Designed for live and mobile monitoring, the IE-10 is a single-driver model with −26 dB of isolation. It includes a soft travel pouch and a cleaning tool. The IE-20XB has extended bass response and −16 dB of isolation. Its dual-bore design, along with a high-frequency armature driver and low-frequency dynamic driver, delivers highs and lows through separate canals. Another dual-bore model, the top-of-the-line IE-30, has high- and low-frequency armature drivers and −26 dB of isolation. The IE-20XB and IE-30 come with a cleaning tool, an in-line attenuator, and a gold-plated 1/8-to-1/4-inch adapter.

Roland VG-99

Roland (www.rolandus.com) has rolled out the latest incarnation of its V-Guitar system, the VG-99 ($1,399). Featuring two COSM-based processors and a collection of expressive performance controllers, the VG-99 lets you dynamically switch, layer, and combine guitar and amp models. The rackmountable tabletop unit modifies your guitar's sound to emulate electric, acoustic, and bass guitars, synths, amps, and effects. With two modeling engines, you can simultaneously layer two virtual guitars and amps with your real guitar and switch between them using foot controllers, buttons and knobs, or even your picking dynamics. You can instantly call up alternate tunings that you've saved, and a new Freeze feature sustains notes and chords indefinitely.

The VG-99 requires a guitar outfitted with the optional Roland GK-3 pickup ($199) or another hexaphonic pickup. You can use the unit's built-in guitar-to-MIDI converter to control MIDI hardware and software instruments. The VG-99's dual GT-Pro-class processors offer an array of effects, such as reverb, flanger, delay, and pitch-shift. USB connectivity lets you send audio directly to your computer and use the included editing software for Mac and Windows. The VG-99 has twin D Beams, a ribbon controller, and S/PDIF, XLR, and ¼-inch outputs.

Alien Connections ReValver Mk II

ReValver Mk II (Mac/Win, $219.92) is a unique guitar amp, cabinet, and effects simulator from Alien Connections (www.alienconnections.com). It runs standalone or as a plug-in for AU and VST hosts. ReValver supplies a virtual rack with more than 60 modules that you can configure in any order. Its amp simulation goes beyond other software by furnishing virtual tubes, transistors, resistors, capacitors, and other components, essentially allowing you to design your own amplifier. All the knobs, faders, and buttons are MIDI controllable in the standalone version, and most parameters respond to sequencer automation.

ReValver's selection of amp modules ranges from virtual Fender, Vox, and Marshall heads to home-brewed electronics and various preamps and power amps. You can build your own virtual cabinet by selecting a convolution speaker simulation preset or by using ReValver's Speaker Construction Set, which lets you specify cabinet dimensions, type and number of speakers, and microphone parameters. Fifteen stompboxes supply everything from overdrives and compressors to multitap delays and automatic wah. Additional effects include 8-band parametric EQ, a stereo widener, and even a module that hosts third-party VST plug-ins. You also get tuners, frequency analyzers, signal splitters, an oscilloscope, and other virtual tools.

BRING ON THE HEAT.

TUBEFIRE8

Eight Channel Tube Preamplifier with Firewire

The ART TubeFire8™ delivers the best of all worlds in one tube driven digital audio interface package. Ideal for any recording application, the TubeFire8™ adds eight incredibly warm tube driven microphone or line inputs and eight balanced outputs to any FireWire equipped computer.

Complete FireWire Based Studio Solution

Designed as a complete studio package, the TubeFire8™ is shipped with Steinberg's Cubase LE 48-track for both Mac and Windows operating systems making it a truly plug and play recording solution, although it is compatible with many popular ASIO and Core Audio based applications.

Class-A Tube Design

ART's TubeFire8™ packs eight quality second-generation discrete Class–A vacuum tube microphone preamps in a single rack space audio interface with balanced I/O and FireWire connectivity.

Versatile I/O

ART's microphone preamps provide clean quiet gain while maintaining incredible transparency through the input stage. The eight balanced outputs of the TubeFire8™ can be driven from either the analog microphone preamp inputs making the TubeFire8™ an in-line eight channel tube preamp, or from the internal high quality D/A converters making it a high quality multi-channel audio output for your PC.

Full Input Control & Low Latency Monitoring

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- **1 x 1/4-inch TRS headphone jack**
- **8 x Input Gain / Channel Level / HPF / Phase Invert**
- **8 x Channel Metering**
  (4 bar led graph w/ clip indicator)
- **44.1KHz, 48KHz, 88.2KHz, 96KHz Sample Rates**
- **24-204KHz External sample rate**
- **44.1K, 48K, 88.2K, 96K, 176.4K, 192K Internal sample rates**

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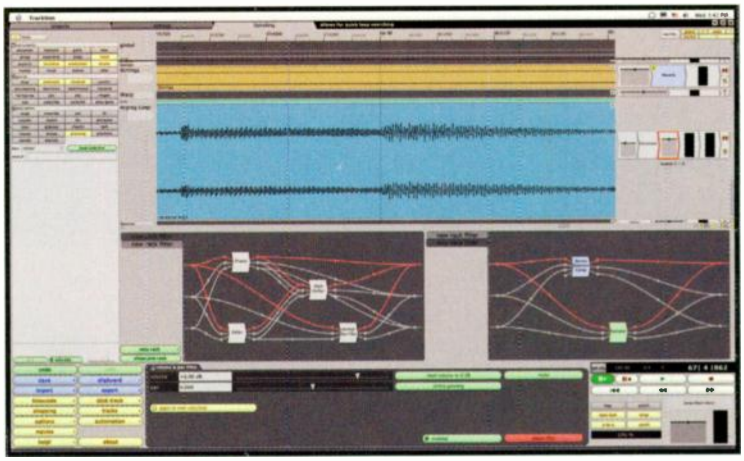
Mackie Tracktion 3

Mackie (www.mackie.com) has released an upgrade to its digital audio sequencer that introduces more than 150 new features and improvements. Tracktion 3 (Mac/Win) is available in two versions: the Project Bundle ($129.99) and the Ultimate Bundle ($319.99). New features include pitch-shifting, time-stretching, and a loop browser that supports Acid, Apple Loops, and REX formats. You can organize loops by instrument, genre, key, and descriptor, and search for them using keywords. Additional enhancements include Folder Tracks, Marker Tracks, Collection Clips, VCA filters, and user-creatable templates. You also get a streamlined MIDI editor and expanded control-surface and multicore-processor support.



Both bundles contain the same full version of Tracktion 3 with a complete suite of mixing and mastering tools. The Project Bundle comes with a collection of software instruments such as CronoX 3 Lite, SampleTank 2 SE, DrumCore TK Diet, and GPO Lite, as well as the Sonic Reality T3 loops collection and the Tracktion 3 quick-start video. The Ultimate Bundle ups the ante with the full CronoX 3, twice as many SampleTank presets, an 820 MB GPO Lite library, 2 GB of DrumCore content, four times as many Sonic Reality loops, over four hours of Mastering Tracktion videos, and more. Previous Tracktion owners can upgrade to the Ultimate Bundle for $259.99.

Sound Advice

Soundware developer **Bass VI** (www.basssix.com) has just begun shipping its first product, a massive electric bass sample collection in EXS format. *Anthony Jackson Contrabass Guitar Library* ($400) comes on ten DVDs and takes 38 GB of hard-disk space to install entirely. Jackson, a renowned master of the 6-string bass, has recorded hits with the O'Jays, Paul Simon, Chick Corea, Steely Dan, Roberta Flack, and many, many others—more than 500 albums in all. The library captures the signature sound of his Fodera contrabass guitar, delivering multisamples of flat-picked, open-finger, and palm-muted variations, each with a choice of DI box or a miked speaker. Some presets let you use the mod wheel to switch string and fret positions in real time. Jackson performs numerous articulations, with up to five Velocity layers for each note, all the way down to the low B string. *Anthony Jackson Contrabass Guitar Library* will soon be available as a Tascam GVI instrument (Mac/Win, $425), so that no other sampler will be necessary.

From **Sony Creative Software** (www.sonycreativesoftware.com) comes **Poptronica** ($59.95), a 531 MB collection of loop construction kits that Sony describes as "mind-bending musical widgets and ingenious sonic tweakery." Produced by Aaron Mellinger, creator of *Electrocution* and *Aural E: Eclectic Electronica*, the new collection provides a total of 15 Acid 6 projects in the avant-pop genre. With titles like "Flying Fantastic Fan," "Superspy," and "Slam You Off the Edge," each project folder supplies plenty of synth leads and pads, bass, percussion, and other instrumental loops you can open in any program that imports Acidized WAV files. The CD-ROM also includes a copy of Sony's Acid Xpress 5 (Win).

Korg R3

New from Korg (www.korg.com), the R3 ($799) is an 8-voice synthesizer with an onboard vocoder and a 37-note, Velocity-sensing USB/MIDI keyboard. The R3 puts the Radias MMT (Multiple Modeling Technology) synthesis engine in a more affordable and portable package. Each of 128 onboard programs contains two timbres, a master effect, and an 8-step arpeggiator, and each timbre has two insert effects, 2-band EQ, and a modulation sequencer. LCDs show parameter names above each of five knobs, much as they do on the microKontrol and Kontrol49.

Each R3 voice has two oscillators, two multimode filters, three ADSR generators, and two LFOs. Six virtual patches let you connect 12 modulation sources with 15 destination parameters. The oscillators generate noise and analog-modeled waveforms, as well as DWGS (Digital Waveform Generator System) waves. The R3 further expands the timbral palette with waveshaping, cross-modulation, variable phase modulation (VPM), and more. One filter is switchable from 24 dB-per-octave lowpass to 12 dB-per-octave lowpass, bandpass, or highpass, with continuous morphing between responses, and the other offers an additional comb response.

The R3's 16-step, looping modulation sequencer can apply time-varying changes to almost any parameter, and motion recording can capture and replay real-time knob movements. The 16-band vocoder can record 7.5 seconds of Formant Motion data into each of 16 internal locations. The R3 comes with an XLR gooseneck microphone and a standalone editor-librarian (Mac/Win).

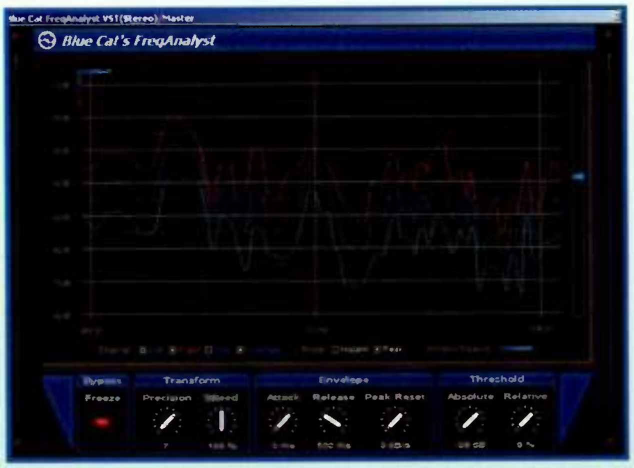
Download of the Month

BLUE CAT AUDIO FREEWARE PLUGINS (WIN)

Blue Cat Audio (www.bluecataudio.com) is a developer of custom audio plug-in effects for the PC. Its plug-ins emphasize audio analysis and repair, with phasing, flanging, and chorusing thrown in for good measure. Individual plug-ins are in the $30 to $60 range, and several bundles are available at bargain prices. You can have everything the company offers for $370.81, and you can get a free taste by downloading its Freeware Plugins Pack. Don't be fooled by the price; there's a lot to like in the free bundle.

At the top of the list are three great-sounding time-based effects: chorus, phaser, and flanger. The chorus and flanger come in mono and stereo flavors, with the latter adding stereo panning of separately processed channels. The chorus and flanger are LFO-modulated delays with maximum delay times of 10 and 30 ms respectively, and the flanger has separate feedback and feed-forward controls. The LFO rate ranges from 0 to 20 Hz with a maximum modulation depth of 100 percent. The phaser is a multistage notch filter with up to 32 stages and similar LFO modulation. Each of these effects is capable of subtle as well as extreme processing (see **Web Clips 1 and 2**).

For analysis and repair, you get a sophisticated frequency analyzer and a suite of plug-ins for gain control. The gain plug-ins include peak metering, overall gain control, and separate gain control for stereo and mid-side channels. The frequency analyzer has a huge window for displaying instant- and peak-frequency spectra in real time. You can zoom in the display vertically and horizontally to zero in on a specific frequency and level range. One nice touch is the window's Opacity control, which allows you to see your sequencer tracks running behind the GUI. The Threshold control is another unusual feature; you



can force the curve to update only for peaks above a threshold that you set.

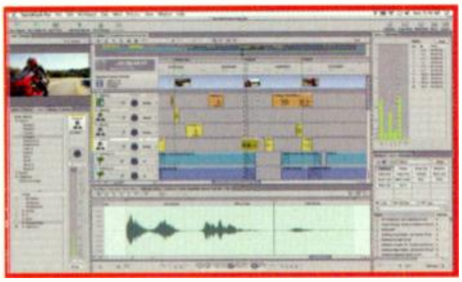
Of course, the purpose of giving away these plug-ins is to encourage you to buy their more full-featured brethren. But each plug-in is useful, and the frequency analyzer is exceptional. Furthermore, they all come with PDF manuals, onscreen help, basic presets, and multiple skins.

EM WEB CLIPS

—Len Sasso

Apple Final Cut Studio 2

Apple (www.apple.com) has debuted Final Cut Studio 2 (Mac, $1,299), an upgrade to its comprehensive audio and video production suite. It comprises six programs for creating and editing media projects: Final Cut Pro 6, for editing virtually any video format; Soundtrack Pro 2, for editing and mixing stereo and multitrack audio; DVD Studio Pro 4, for DVD authoring and mastering; Motion 3, for creating three-dimensional animation that responds to sound; Compressor 3, for encoding in a variety of formats; and a new addition, Color, for color adjustment, grading, and finishing.



Soundtrack Pro 2 is the suite's audio postproduction application. Its streamlined user interface provides graphical tools for working in 5.1 surround, processing or mixing tracks, or doing audio restoration. A new spotting display lets you precisely align sound with picture. The integrated audio-conform process automatically adapts your audio project to match changes you make when editing picture and highlights the modified clips for your approval, without the need to build new and manually work through edit decision lists (EDLs). A surround panner gives you a visual representation of the sound field, and numerous surround plug-ins are included. Final Cut Studio 2 comes with a library of 5,000 sound effects, including more than 1,000 surround effects and multichannel music tracks.

Big Fish Audio Drums Overkill

Big Fish Audio (www.bigfishaudio.com) is shipping Drums Overkill (Mac/Win, $299.95), a Kontakt Player 2–based instrument paired with more than 2.6 GB of content. The collection divides hundreds of kits into ten categories ranging from Analog Syn Percussion to Mixed Dance Effects. The Percussion category includes talking drums, mouth percussion, whistles, and loads of instruments from around the world. The Drum Machines & Rhythm Boxes category features analog and digital beatboxes from days gone by; alongside classics like the Roland TR808 and Maestro Rhythm King, you also get vintage electronic drums such as the Korg Wavedrum, the Simmons SDS-7, and even the Mattel Synsonics. Banks of individual drum hits deliver hundreds of snares, kicks, toms, cymbals, and more.

Because it's built on the Kontakt Player 2 platform, Drums Overkill runs standalone or as a plug-in for AU, DXi, RTAS, and VST hosts. A second DVD provides the same content in WAV format, along with read-to-play instruments for Native Instruments Kontakt, Apple Logic EXS24, Steinberg HALion, and Propellerhead Reason. All told, the collection gives you about 1,200 kits in 16 musical styles.

Zplane Vielklang

Berlin-based software developer Zplane (www.zplane.de) is shipping an innovative audio-harmonization plug-in called Vielklang (Mac/Win, $249). The AU- and VST-compatible processor can generate four voices from a mono source with a minimum of formant-shifting artifacts. Chord-progression and voice-leading models control the synthesized voices, resulting in musically intelligent harmonization with little or no user configuration. Beginning with a melodic phrase, Vielklang can automatically generate natural-sounding, harmonically correct vocal and brass sections, for example.

Vielklang opens in your host program as an instrument plug-in. When you load an audio file, Vielklang automatically provides a piano-roll display of the harmonized output and allows you to control each voice's audio properties. You have the option of transposing any voice to any interval, and you can modulate a major scale to one of ten other scales with a single click. In addition, the VST version of Vielklang will convert harmonized voices to MIDI data that can trigger other instruments. Vielklang can store and reload as many as 32 harmonization snapshots and trigger them with MIDI Notes. EM


M-AUDIO
"Trigger Finger takes my performance options to a whole new level." – Joe Hahn, Linkin Park
M-Audio Trigger Finger is the drum pad controller of choice for pros like Linkin Park, who can use any gear they want. That's because Trigger Finger is about more than just the ability to trigger sounds, samples and video clips—it's about total performance and total control. Other drum pads use piezo sensors that are susceptible to vibration during performance, which can lead to false triggers. Trigger Finger uses superior FSR (Force-Sensing Resistance) technology for rock-solid performance and dependability. Add that to the onboard knobs and faders and we're talking ultimate creative control—on stage and in the studio.
• 16 velocity- and pressure-sensitive FSR pads > precision triggering
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MOR Is Better

By Scott Wilkinson

Algorithms that make computers seem almost human.

Computers have become essential tools for many composers, who use them for a wide variety of musical purposes, such as trying out ideas, producing demos, and notating scores and parts. But when you step-enter music note by note, as one commonly does when composing using a notation program, playback can sound mechanical and lifeless.

That problem could soon be solved if Gershon Silbert has his way. In 2004, after four years of researching musical expression and cognition, he founded a company in Herzliya, Israel, called Silpor Music (www.silpormusic.com).

The company's core technology, a suite of algorithms called MOR (Music Objects Recognition), will be available first as a plug-in for Finale 2005 and 2006 running under Windows 2000 and XP. The company expects to have a Macintosh version soon, as well as support for Finale 2007. The plug-in should be commercially available soon after you read this article, and it will support the Music XML file-interchange format, which will be used to build translators for other notation programs and a standalone version of the software.

MOR is unique in its ability to imbue step-entered and computer-generated music with convincingly humanlike performance characteristics. The suite includes two major functional elements: the Analysis Engine and a set of Performance processors. The Analysis Engine looks at a Finale .mus file and identifies all musical objects, such as notes, chords, dynamics, and expression markings. In addition, it identifies higher-level objects, including cadences, modulations, dissonances and their resolutions, rhythmic structures, melodic structures, and voice hierarchy (for example, melody versus accompaniment).

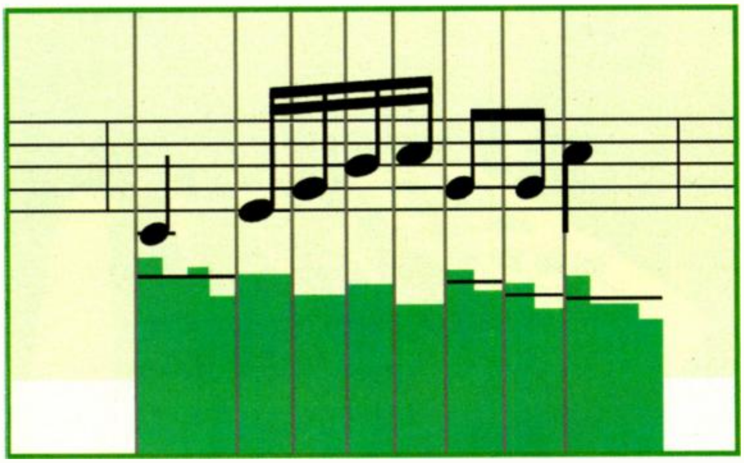
COURTESY SILPOR MUSIC

FIG. 1: To simulate a human performer's nuance, MOR's Tempo groove processor makes slight adjustments to the tempo of each beat subdivision (16th notes in this example). When a note's duration extends beyond the specified subdivision, the tempo values are averaged, as indicated here by the horizontal lines under each quarter and eighth note.

Once the analysis is complete, the Performance processors modify the performance parameters to produce a more human feel as the music plays back. Each processor affects a different parameter (such as tempo, MIDI Velocity, and duration) as well as MIDI controllers (such as Sustain Pedal). These parameters can be applied globally and to individual notes and groups of notes, such as chords and melodic phrases, depending on the analysis of voice hierarchy.

Users define what the Performance processors do by establishing presets in an editor. You can specify a range of values, and that range can be limited by a user-defined number of beats, notes, or milliseconds. You can also stipulate which objects will be used by the processor to modify those values; Velocity, for example, can be tied to pitch and duration. Finally, you can compress the values of parameters such as Velocity and tempo to fall between two user-defined limits, and you can apply various filters that restrict the processors to specific MIDI channels, durations, and/or notes. Two groove processors—Tempo and Velocity—further humanize the playback (see **Fig. 1**).

Unlike the humanize function in many sequencers, MOR does not randomize events in an attempt to simulate human inaccuracy. Instead, says Silbert, "It is based on humanlike understanding of the musical context and, by extension, possible meanings or emotional messages, which can enable superhuman expressive perfection, free of any mistakes. In other words, MOR has no technical limitations when processing expressive parameters, as opposed to most human performers' inaccuracies, which are mostly due to technical difficulties."

The Silpor Music Web site includes examples of pieces performed with and without MOR, as well as the same pieces performed by human musicians. You can also hear examples at www.emusician.com (see **Web Clips 1 through 3**). This technology is a real boon for composers who want to hear their work played by a computer with a convincingly human touch. EM

EM WEB CLIPS

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So when you're ready to hear what your music sounds like, you're ready for EXO. Find out more at www.abluesky.com and ask your local dealer for a demo today.

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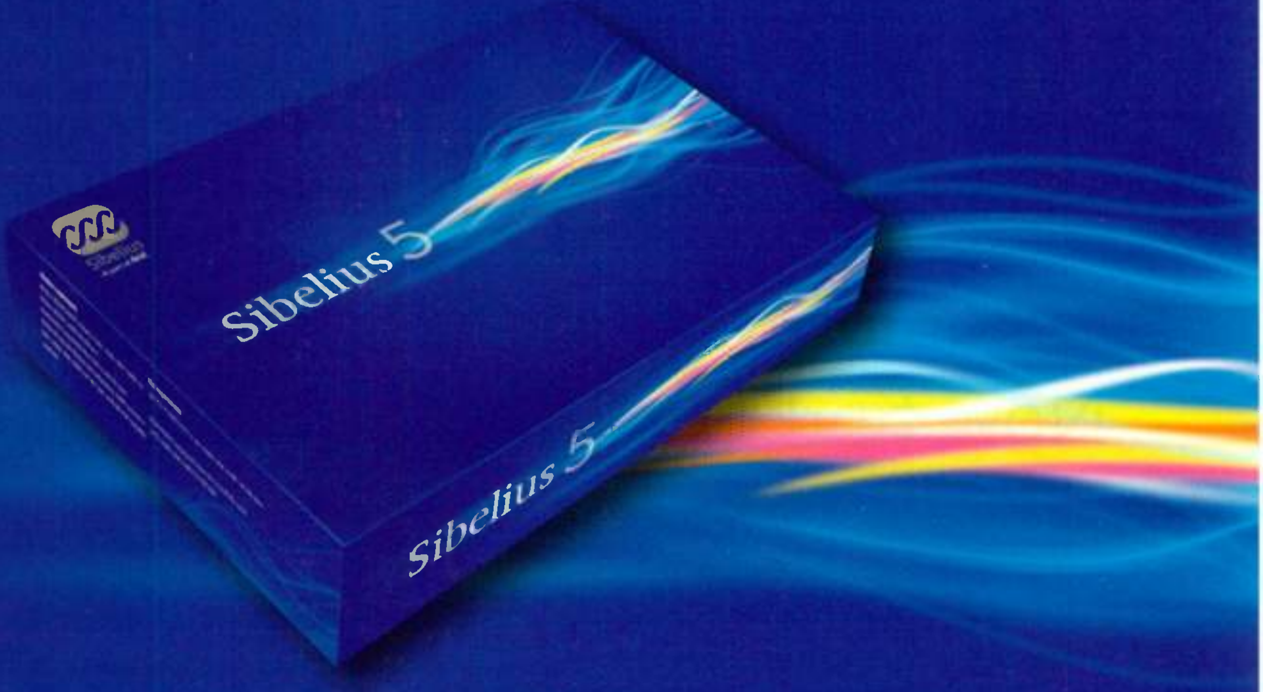
Sibelius 5



"Sibelius helps me keep the fire throughout the writing process, breaking through the slow isolation that is a part of writing music. Writing an orchestral score takes months, sometimes years – and the first time you hear it might be even longer.

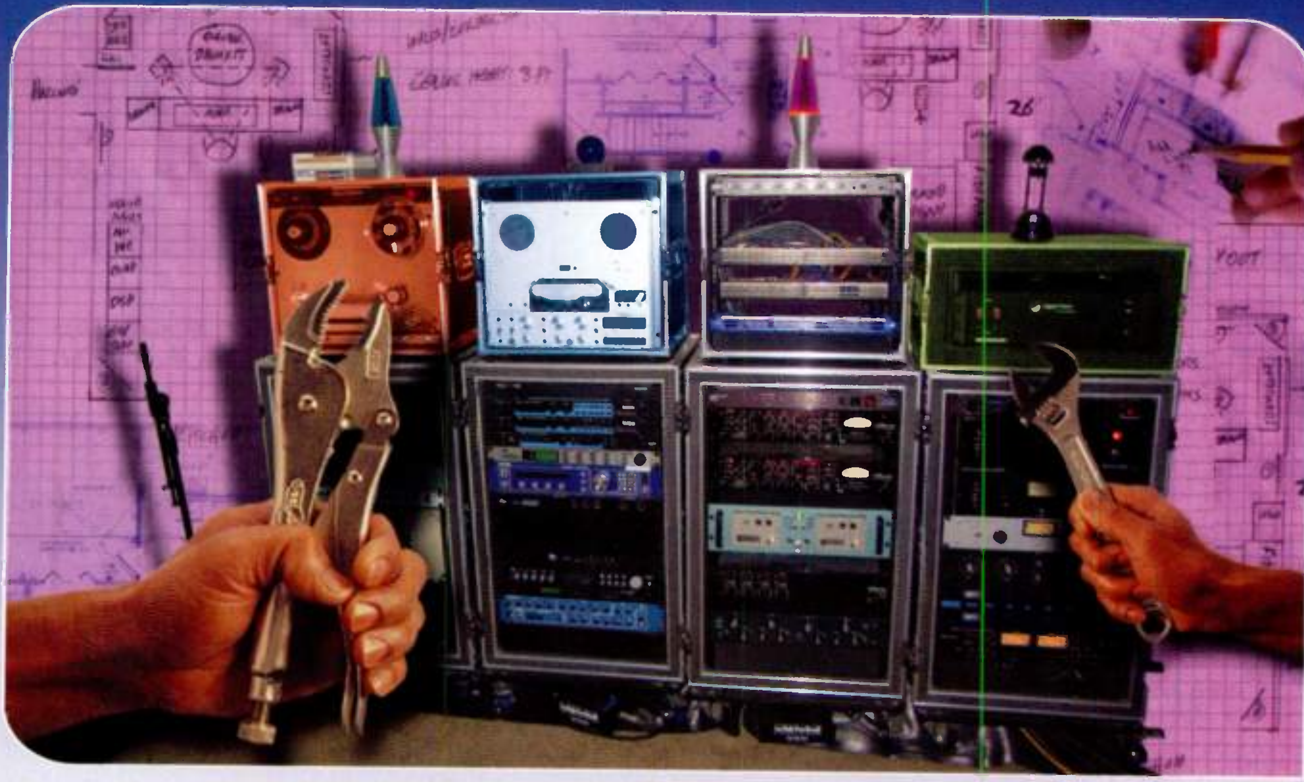
Now, I can immediately play back what I've written, creating the illusion of a dialogue between what I've written, and myself as the performer.

The bottom line... Sibelius keeps me inspired!"

Esa-Pekka Salonen
Music Director, Los Angeles Philharmonic

www.sibelius.com

Sibelius



LAURA WILLIAMS

Starting Over

By Nick Peck

What to consider when redesigning your studio.

Eight years ago, I built a large, complex project studio to do freelance film and game sound design as well as music production. The studio was a success. But sometimes the unexpected happens: opportunity knocked and I answered, taking a position as audio director of a video game company.

Unfortunately, the commute was brutal, so I eventually sold my house and studio and moved my family to a rental home closer to work. Because I no longer needed to do sound design in my studio, I could repurpose it to focus on recording music exclusively. My new recording space is smaller than my previous studio, which forced me to rethink the layout and my priorities. I was thus faced with the daunting but fun task of reinventing my studio, streamlining it to fit my new life.

This experience taught me two valuable lessons: first, that nothing is permanent, and second, that as you grow as a person, your studio should change to accommodate your changing needs. If you are ready to rethink the design of your own recording space, here are some tips on how to alter it for the better.

Starting from Scratch

Before you rebuild (or build) your studio, sit back for a moment and ask yourself an important question: What are my goals for this room? Although this question may seem simple, the answer will provide a context that informs the answers to subsequent design questions, such as: Are you interested in recording your band? Will you mix other people's music? Will you do sound design for postproduction, or voice-over work for radio? Is it going to be a one-man MIDI studio? Do you plan to take on outside clients? Knowing which of these functions your studio will fill can help you make the proper equipment and design choices.

For example, the primary design considerations for my new studio were sonic excellence, analog solutions wherever practical, and portability. A key point was to be able to take the studio down in one day and set it up elsewhere in another. To that end, equipment is housed in touring racks or sets of drawers on casters whenever possible for you. In the end, the goal

Starting Over

Cable positioning always seems to be a hassle. Rule number 1 is to avoid running cables through foot traffic areas. In the past, I've built simple cable troughs out of ABS drainpipe cut in half and attached to the walls. My previous studio had cable troughs built into the foundation under the floor.

Putting a cable run behind equipment and furniture is not a problem, but occasionally you will need to run cables through an area of foot traffic. Depending on your tolerance for ugliness, you can run cables up over doorways, or through loops made from cable ties hung from the ceiling. If that is not an option, you may have to run cables across the floor. The best you can do is to minimize the number of cables by using snakes. Make sure the cables are tight and flat on the floor, using cable ties to bundle them together in a manner that won't cause a tripping hazard (see **Fig. 3**). You can also build a small ramp out of plywood that covers the cables, protecting them from damage.

Acoustic Approaches

To quote an oft-repeated aphorism, the single biggest sonic improvement to your tracks and mixes comes not from a new microphone or signal processor, but from tuning the room itself. Properly placed absorption, which particularly covers the low frequencies, is a critical aid in making mixing decisions that will translate well. Although complete coverage of room acoustics is beyond the scope of this article, there are a few things to keep in mind.

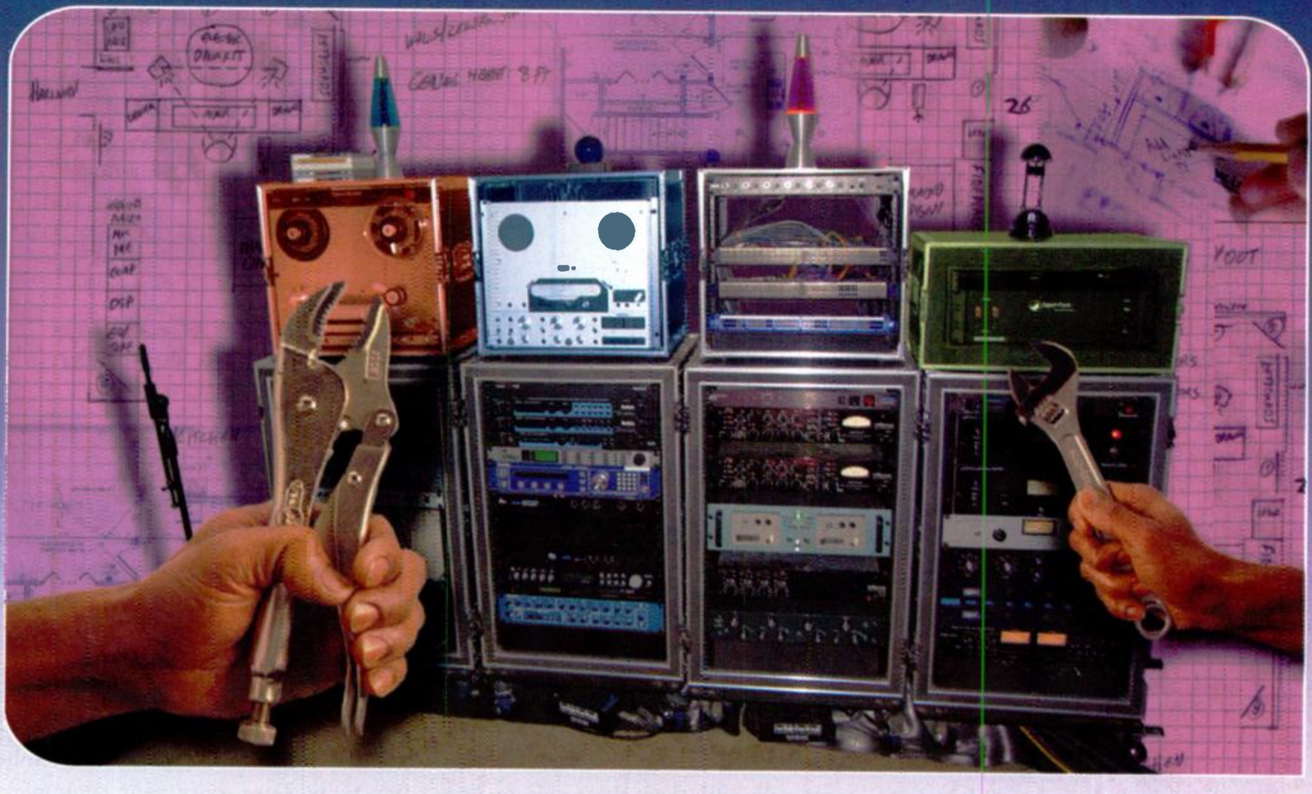
It is important to understand that surface treatment is not effective for soundproofing: it is used to balance frequencies and control reverberation time within the room. Surface treatments fall into the category of *absorbers*, which absorb particular frequency ranges of sound waves, and *diffusers*, which scatter sound waves.

Diffusers can be effective on the rear wall of a smaller space but are generally lower priority than absorbers for the typical project studio. Bass trapping and broadband absorption, on the other hand, are things that no studio should be without. You can start by placing bass traps across all corners of the space, because this is where low-frequency buildup predominates. Although each room has different needs, you should place absorption to the sides of the mix position, in the front of the room behind the monitors, across the rear wall, and on the ceiling, if possible, halfway between the mix position and the monitors.

Absorbers can be made from a variety of materials, but semirigid fiberglass batts and rock wool are the most common. The fiberglass batts are a solid, dense version of the pink stuff you have in your walls. These batts are typically encased in a wood or metal frame and covered with an acoustically transparent cloth, such as burlap. The thicker the batt, the deeper the frequencies it is effective against.

Bass traps are simply absorbers that are effective at absorbing bass frequencies. There are several ways to construct bass traps, including using thick batts with an additional air space between the batt and the wall.

Outfitting a recording space with effective, professional treatment can be a somewhat pricey proposition—on the order of several thousand dollars to do a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For example, the primary design considerations for my new studio were sonic excellence, analog solutions wherever practical, and portability. A key point was to be able to take the studio down in one day and set it up elsewhere in another. To that end, equipment is housed in touring racks or sets of drawers on casters whenever possible (see Fig. 1). The room acoustics are attached to stands, rather than being mounted to the walls. The wiring is clean and tight, and most of it can be quickly coiled together and attached with cable ties to the outside of the patch bay case. And when I move out of this space, it will look like a living room again with no revisions needed.

Once you have defined your goals, it's time to take a complete studio inventory. List every piece of gear you have, and with each one, ask yourself the following questions: Does this piece of equipment help advance my studio's goals, or is it a distraction? Does it interface well with my other gear? Is it redundant? If the item no longer serves a productive goal for you, put it into a pile of stuff to be sold. The proceeds can help finance other equipment that will better suit your current needs.

Next, figure out how much you are likely to make selling off the gear you no longer need, and add it to the money you've squirreled away for this little project. That budget will then set a cap on what you can buy.

Now comes the fun part: R&D. Make a list of problems that need solving, such as the need for acoustic conditioning, or a good mic for recording electric guitar. Then research your options by browsing the Web, asking your friends, and reading trade magazines.

Remember that this is not about finding the sexiest new toy on the block. Rather, it is about getting the best tool to solve one or more problems. For instance, if you have neither a good EQ nor a good compressor, you might consider finding a hardware or software channel strip that will do both jobs for you. In the end, the goal is to select the appropriate items that will integrate into a functional whole.

Once you've completed the R&D process, make a prioritized list of the things you'll need. Buy the highest-priority items until you've exhausted your budget, then slowly complete the list as funds become available.

The Physical Layout

When designers lay out a kitchen, they focus on creating a space that offers maximal productivity with minimal movement. To this end, they utilize the concept of the work triangle. The points of the triangle are the three most important kitchen areas: the refrigerator, the sink, and the stove. When preparing a meal, you interact continually with these three objects. As a result, they are laid out first, and the rest of the kitchen is designed around them.

You can take this same approach when planning the physical layout of a studio. Start by creating a bird's-eye-view layout of the room on graph paper (see Fig. 2). Be sure to note the positions of doorways, windows, power outlets, and any other features of the room that might impact the location of equipment.

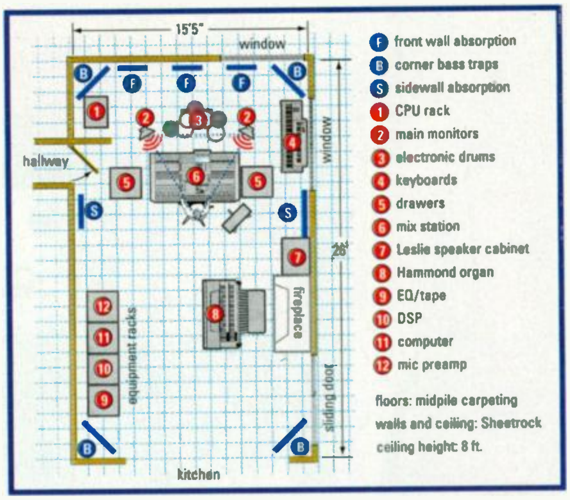
The central listening position and speaker placement are the first elements to determine. The optimal listening position is generally around the center of the room, or slightly forward of that location. Draw a picture of your head at that position on the graph paper. The speakers should be oriented to fire down the long axis of the room and placed equidistant from the sidewalls.

Though it may be tempting to place the speakers directly against the front wall, try to move them into the room at least a foot or two. That will help cut down on undesired bass-loading effects and will lower the amplitude of the early reflection off of that wall.

Now that you know ballpark locations of your listening position and the speakers, draw an equilateral triangle on the graph paper from your head to the speakers, adjusting speaker location until each line is equidistant. This will give you a starting point that can be fine-tuned later, once your equipment and room acoustics are in place. ("Truth or Consequences" in the November 2001 issue of EM shows you how to tune your control room to flatten its response; it's available online at www.emusician.com.)

With your speakers positioned, you can lay out the rest of the work space. The area directly in front of your listening position is the most valuable piece of studio real estate and should be occupied by whatever tools you use most frequently during mixdown. The obvious contenders for this spot are your mixer or control surface, as well as the controls for your recording device. If you spend a lot of time looking at the computer screen while mixing, its logical position is dead center between the two speakers, and close enough to allow you to read it without eyestrain. Be careful to avoid obstructing the sight lines to your speakers with your computer monitor, because it can create unwanted aural reflections.

In my case, I wanted to avoid the distraction of looking at a screen during mixdown, so I moved my computer monitors outside of the speaker area. My mixer is directly in front of the main listening position, and the remote control for my recording device (an iZ Radar hard-disk recorder) is on a roll-around stand within easy reach of my right hand. This decision has helped



CHUCK DAHMER

FIG. 2: In the initial design phase for my studio, I noted the locations of windows, doors, a hallway, and the kitchen, as well as where I wanted to place my CPU rack, monitors, instruments, and acoustic treatment.


NICK PECK

FIG. 1: A studio that is well organized and streamlined can be a great place to work.

This Pair Beats a Full House

Is it a gamble each time you set up your powered loudspeakers in a difficult environment? Mids don't cut...bass too thin...amps overheat...audio shuts down? If so, you're obviously not using Yamaha MSR400s. Behind their subtle, professional appearance is a speaker capable of overcoming the most adverse circumstances. Put a pair in front of a good sized crowd and see for yourself. We're talking loud and clean.

MSR400...your best bet for great sound, reliability, and value.

- 12" long-throw woofer / 1.75" compression driver
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me focus on what I'm hearing while mixing, rather than what I'm seeing.

Once these key decisions are made, the rest of your equipment can be added, with highest priority given to the objects you use most frequently. If your studio is primarily oriented toward a single composer-engineer creating electronic music, then special emphasis must be given to the position of your master MIDI keyboard controller. Placing this keyboard directly to your right as you face the speakers allows you to play melodies with your right hand while operating your DAW or recorder with your left. In studios that don't have a mixer or control surface, a MIDI keyboard controller can be placed directly in front of the listening position instead.

External signal processors, computers, and other ancillary equipment often work well in racks to the left of the mix position. If you have a lot of gear and ample room, consider getting a producer's credenza that sits behind you. That will give you plenty of rack space within easy reach, as well as a handy work surface. (For more information about studio furnishings, see **Web Clip 1**.)

Cables and Patch Bays

Though it may be considered a trifle mundane in some circles, the humble patch bay is the nerve center of your studio. Careful thought should be given to layout and your approach to wiring. When done right, patch bays are intuitive, reliable, and capable of withstanding changes in layout. When not planned out, they can become sprawling, inflexible monsters.

Patch bays come in various formats. My favorite type is a 1U bay that uses Tiny Telephone (TT) jacks, offering 96 points of balanced connectivity. Each pair of patch points can be individually removed from the patch bay for soldering at a workbench if required.

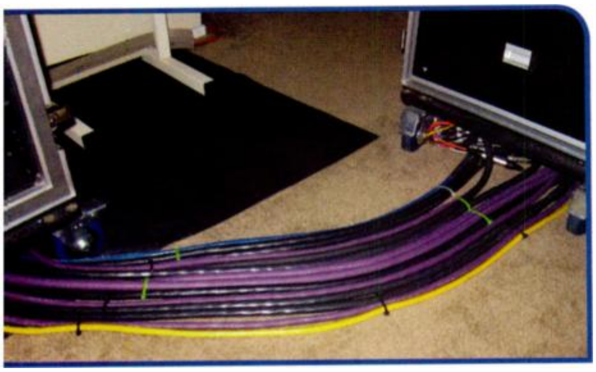
The rear of the bay comes in a number of configurations. Some terminate in solder lugs, which you would attach directly to your gear with cabling. Others terminate in EDAC, ELCO, or DB25 connections, which allow you to connect large, multipin snakes directly to certain mixing consoles and recorders. Still others terminate in solderless punch-down blocks, allowing you to connect individual audio cables without having to solder the ends to the patch points. Lastly, some bays separate the patch rows from the equipment connection terminals by using snakes to connect between them.

New patch bays are great, but there are bargains to be found in used ones. I bought my patch bays in an electronics surplus store for $20 each. You can also hunt for quality used bays on eBay or through Internet vendors such as Mr. Patchbay (http://home.flash.net/~motodata/patchbays/).

The key to a successful patch bay is planning everything out on paper before you touch a single wire. Make a list of every piece of equipment you have, noting the number and type of analog inputs and outputs. Group your items by function, utilizing normaling wherever appropriate to minimize the number of patch cables needed for your default work flow.

Once you've listed all your gear, make a diagram on graph paper or in a spreadsheet of your proposed patch bay layout. That will help you determine how many patch bays you need to do it right. Be sure to leave about 25 percent of the patch bays open for future expansion—you'll be glad you did when that next piece of kit comes along.

The next step is determining where your patch bays will be located. It is a critical decision because all your cable runs will be determined by this choice. I placed mine in a rack on top of all my outboard gear. This meant short runs to individual signal processors, but longer runs to my mixer and recorders. Snakes are perfect for multitrack and mixer connections, as they allow for a larger number of connections within a smaller package than individual cables.



NICK PECK

FIG. 3: Try to avoid running cables through areas of foot traffic whenever possible. When it's unavoidable, minimize the exposed cabling area and arrange the cables in flat, tightly bundled groups.

You will need to decide whether you will be opting for custom-length cables or not. I can't stand wading through pools of extra cabling, so I cut each cable to size. I had Redco Audio (www.redco.com) custom build DB25-to-XLR snakes for me, and I soldered the rest of the cables myself.

If you choose to solder cables yourself, be sure to use a quality soldering iron and cable jacks. Heat-shrink tubing, a good continuity tester, and a bit of patience are a must. The results will be reliable connections that will last a long time.

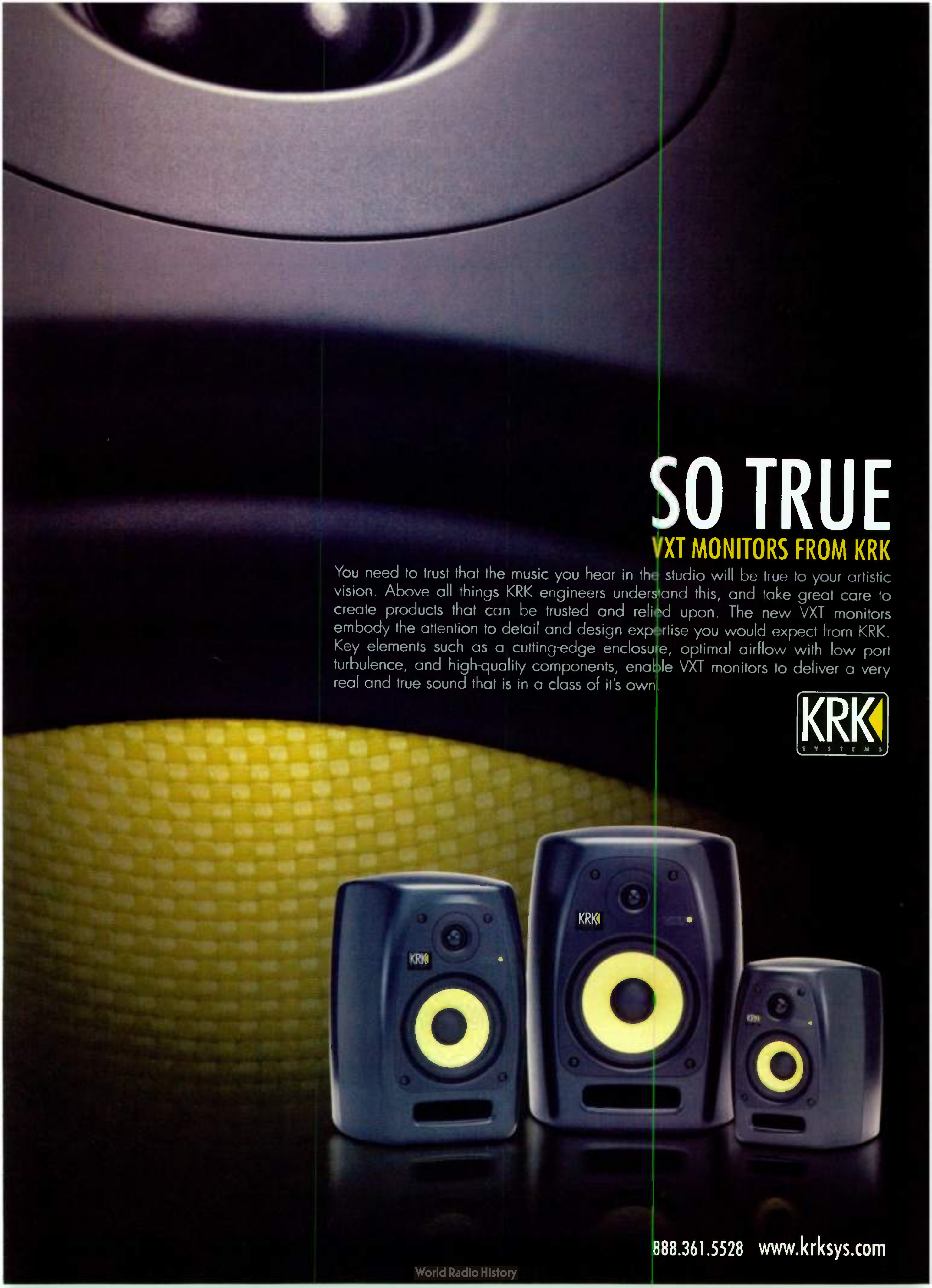
If you plan to make custom-length cables yourself, the next step is to take a full cable inventory. Lay them out in piles on the floor so you can easily see the ends and cable types. This is also a great opportunity to throw out those mounds of cheap, junky, or broken cables.

Measure the distance from a piece of gear to your patch bay, adding about 18 inches or so for slack. Try to find a cable that has the appropriate ends and is as close as possible to the length you need. Resist the temptation to stretch cables that are too short. For patch bay work, steer clear of cables that are thick, because they will take up too much room when clustered with dozens of others.

Once you find your cable, cut it to length and be sure to save any good-quality jacks that are on the cable end you are not planning to use. If you are soldering the cable to the patch point, remove that point from the rack and attach the cable at your workbench. Be sure to use heat-shrink tubing on the cable ends to minimize the possibility of short circuits between cables within such a tight space.

My equipment inventory and layout requirements resulted in a need for three 96-point patch bays. One is used for recorder inputs and effects sends and returns, while another is used for recorder outputs and mixer inputs, as well as group sends and miscellany. The third is dedicated to signal processing.

The patch bays live in a 10U rack of their own, in order to leave plenty of room for cabling, and each patch bay is separated by 1U of empty space. All the cabling routed to a single patch bay runs through loops of cable ties attached to the wall of the rack. This provides strain relief for the cable connections and keeps the cabling organized.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Cable positioning always seems to be a hassle. Rule number 1 is to avoid running cables through foot traffic areas. In the past, I've built simple cable troughs out of ABS drainpipe cut in half and attached to the walls. My previous studio had cable troughs built into the foundation under the floor.

Putting a cable run behind equipment and furniture is not a problem, but occasionally you will need to run cables through an area of foot traffic. Depending on your tolerance for ugliness, you can run cables up over doorways, or through loops made from cable ties hung from the ceiling. If that is not an option, you may have to run cables across the floor. The best you can do is to minimize the number of cables by using snakes. Make sure the cables are tight and flat on the floor, using cable ties to bundle them together in a manner that won't cause a tripping hazard (see Fig. 3). You can also build a small ramp out of plywood that covers the cables, protecting them from damage.

Acoustic Approaches

To quote an oft-repeated aphorism, the single biggest sonic improvement to your tracks and mixes comes not from a new microphone or signal processor, but from tuning the room itself. Properly placed absorption, which particularly covers the low frequencies, is a critical aid in making mixing decisions that will translate well. Although complete coverage of room acoustics is beyond the scope of this article, there are a few things to keep in mind.

It is important to understand that surface treatment is not effective for soundproofing: it is used to balance frequencies and control reverberation time within the room. Surface treatments fall into the category of *absorbers*, which absorb particular frequency ranges of sound waves, and *diffusers*, which scatter sound waves.

Diffusers can be effective on the rear wall of a smaller space but are generally lower priority than absorbers for the typical project studio. Bass trapping and broadband absorption, on the other hand, are things that no studio should be without. You can start by placing bass traps across all corners of the space, because this is where low-frequency buildup predominates. Although each room has different needs, you should place absorption to the sides of the mix position, in the front of the room behind the monitors, across the rear wall, and on the ceiling, if possible, halfway between the mix position and the monitors.

Absorbers can be made from a variety of materials, but semirigid fiberglass batts and rock wool are the most common. The fiberglass batts are a solid, dense version of the pink stuff you have in your walls. These batts are typically encased in a wood or metal frame and covered with an acoustically transparent cloth, such as burlap. The thicker the batt, the deeper the frequencies it is effective against.

Bass traps are simply absorbers that are effective at absorbing bass frequencies. There are several ways to construct bass traps, including using thick batts with an additional air space between the batt and the wall.

Outfitting a recording space with effective, professional treatment can be a somewhat pricey proposition—on the order of several thousand dollars to do a fairly large room correctly. However, an investment in these products ensures that you will receive expert guidance from the manufacturer and high quality.

You can also build your own absorbers and diffusers. Owens Corning makes a line of semirigid fiberglass batts—model numbers 703 and 705 are most frequently used for absorption. All the required materials are readily available. In fact, Ethan Winer of RealTraps has an article on the Web (www.ethanwiner.com/basstrap.html) that explains exactly how to build a bass trap this way. Before you get started, though, be ready for a bit of pain. the process of cutting fiberglass batts and gluing burlap to the surface is about as messy as it gets. Be sure you have a large work space, a ventilation mask, clothes you can throw away after the job, and a high tolerance for misery.

I built all the absorbers for my last studio myself, covering all the front and rear walls, the corners, and much of the ceiling space with 2-foot-by-4-foot panels. Because I was at the end of my budget, building my own was the right way to go, but it was an unpleasant and time-consuming process.

In my new space, I chose products from RealTraps (www.realtraps.com): four Mondotraps for the corners, three Minitraps for the front wall, two RFZ panels for the sidewalls, and a pair of Microtraps for the ceiling. Because the studio is in a rented space, I didn't want to mount the ceiling panels to the ceiling. Instead, I attached hooks to the joint where the walls meet the ceiling, and strung airplane wire across at high tension, laying the Microtraps on top of the wires. The RealTraps look far nicer than my homemade absorbers, and will travel with me throughout the rest of my career.

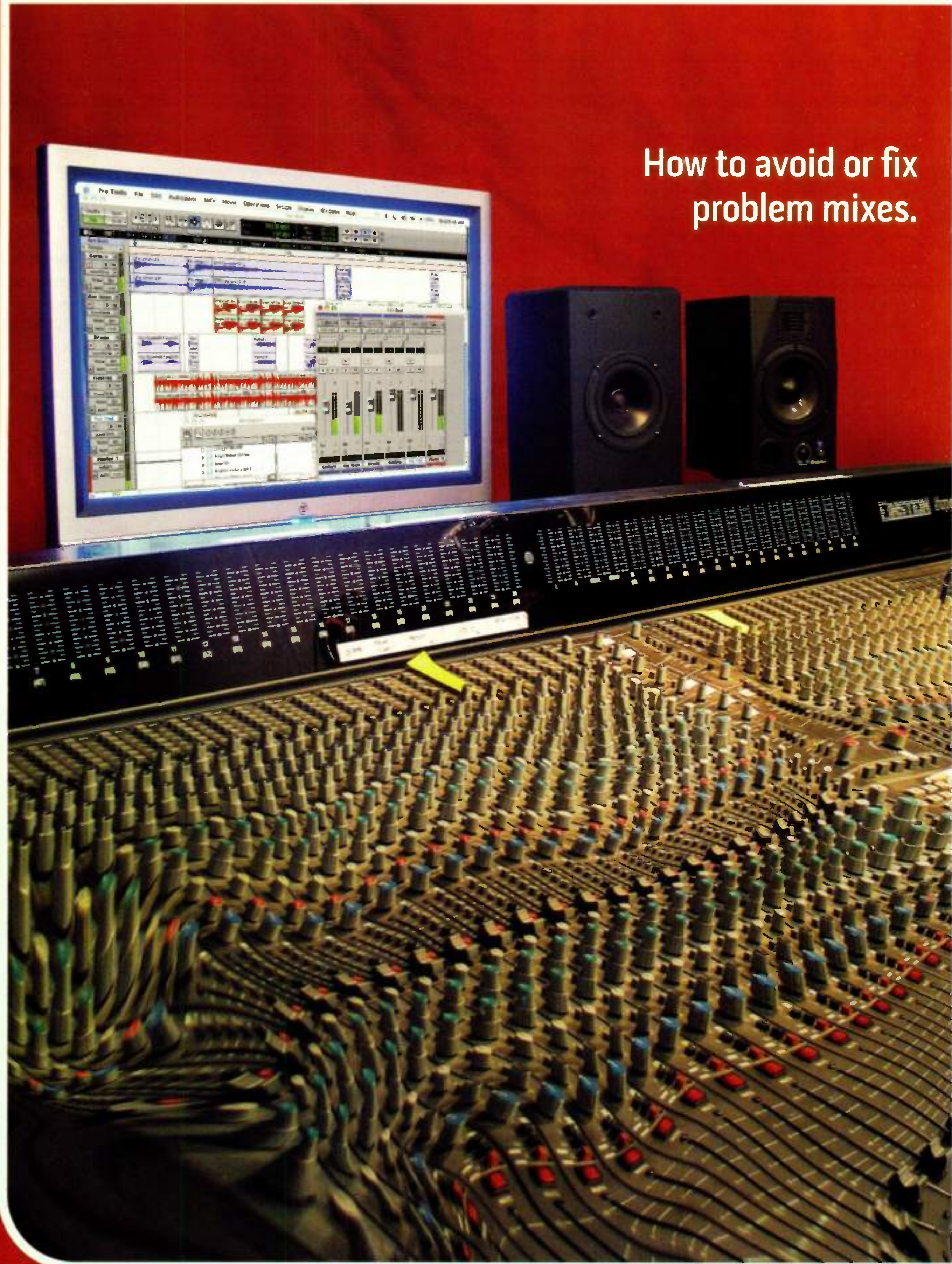
In addition, I purchased RealTraps stands rather than mounting my absorbers to the walls. That had three benefits: I didn't have to deface the walls, installation time was immediate, and, most important, the absorbers can double as gobos for tracking purposes. With this system, I have no need for a separate vocal booth: I simply move the panels into the middle of the work area, creating a temporary recording space. On a recent session, I placed a Leslie rotating speaker in the middle of my studio, put up some ribbon mics, and encased the area with four of the absorptive panels. The result was the best organ recording I've done yet.

Heavenly Sound

Your recording space can be a beautiful, intuitive, and ergonomically effective place to make music. Designing it takes nothing more than a little bit of planning, a little bit of money, and a little bit of elbow grease. Decluttering the area, laying things out for maximum efficiency, having clean, well-thought-out cabling, and making the investment in acoustic treatment will give you a personal studio that you'll never want to leave. **EM**

Nick Peck (www.underthebigtree.com) is a composer–keyboardist–sound designer–audio engineer in the San Francisco Bay Area.

How to avoid or fix problem mixes.



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Twelve Common 12 Mixing Mistakes

By Michael Cooper

As a mastering engineer, I hear a lot of mixes from other studios. Some are great, some are not. But what is striking to me is that the mixes that need help usually suffer from many of the same problems. The good news is that these shortcomings can all be avoided or corrected by using a few simple techniques.

In this article, I will describe 12 common problems with wayward mixes and discuss how to solve them. If your mixes are routinely restrained by a lack of punch, clarity, and detail; if your productions are held hostage by unruly dynamics and spectral imbalances; or if your results don't sound as wide and deep as the mondo tracks created by your competition, read on for some liberating pointers. I'll address each problem and its solution individually, beginning at rock bottom.

Boomy or Thin-Sounding Mix

The most common problem I hear with mixes is uneven levels throughout the range of bass frequencies. This can present itself as either a thin-sounding mix or a boomy one. Some mixes sound alternately thin and boomy in different sections of the song.

The main culprit behind a skewed bottom end is mixing in a room that has not been properly treated with acoustic products to help tighten up impulse response and attenuate room modes. (Room modes, aka standing waves, are narrow peaks and dips in frequency response; they are especially problematic in the bass range.) These acoustic problems might lead you to, for example, unnecessarily boost certain bass frequencies to compensate for a thin-sounding mix when, in fact, the mix already has a perfectly balanced bottom end, though the room's uneven bass response at the mix position is telling you otherwise.

In truth, even rooms that have had thousands of dollars poured into their acoustic makeovers usually have some persistent problems with uneven bass response (although the inaccuracies are usually dramatically reduced in number and severity with proper treatment). Typically, one or two prominent room modes remain at the mix position, making it difficult to properly assess the mix's bass content in these narrow bands.

In most control rooms, there is at least one spot where specific room modes and other bass-response anomalies that compromise monitoring at the mix position are much weaker or even completely tame. While this alternate bass-reference spot might have other problems and be less accurate overall compared with the mix position, it gives you another reference for bass balance in the narrow bands you can't hear properly at the mix position.

How can you tell where the alternate bass-reference spot is? First, assuming that you have more than one pair of reference monitors, play a respected, full-bandwidth mix (usually one that a prominent record label has had mastered and released) through the speakers that have the deepest bass response (include a subwoofer if you use one). Choose this reference mix carefully: it should be one that has always sounded great on the bottom end no matter what sound system you've played it on.

Walk around your control room while the reference mix plays, listening to how the sound of the bottom end changes as different acoustic influences come in and out of play. Note the spot where the bass response sounds the most even at the specific bass frequencies that are out of whack (too weak or too strong) at your mix position—that spot should become a second place you go to check the bottom end when making bass-EQ decisions on your mixes.

Unfortunately, the alternate bass-reference spot is often inconveniently located with respect to the studio's mixer or DAW controller. For example, the place where the 40 to 45 Hz band is most accurate in my control room is about 3.5 feet in front of the back wall.

There is an easy solution: remote control. I always use my Frontier Design Group TranzPort wireless DAW controller (see **Fig. 1**) when checking a mix's extreme bottom end at the back of my control room. Using the TranzPort to remotely start and stop playback allows me to set my control room's monitoring level high enough that I can really hear those subterranean frequencies without blasting my ears at close range. I listen, evaluate the bottom end, stop playback, make the relevant EQ adjustments at my mixer, and repeat the process until the bass sounds great at both the mix position and the alternate bass-reference spot.

FIG. 1: The Frontier Design Group TranzPort wireless DAW controller is excellent for working from an alternate position in the control room.

FIG. 2: Electric guitars sound awesome when processed with the PSP VintageWarmer 2 split-band compressor plug-in.

Edgy, Fatiguing Sound

Digital audio has a reputation for producing cold, brittle sound, but the problem often stems from poor engineering techniques. The most common factor contributing to an edgy, fatiguing mix is indiscriminate boosting of upper-midrange and high-frequency EQ on multiple tracks.

Here's a typical scenario: hours of mixing at high sound-pressure levels (SPLs) progressively compresses your ears' high-frequency sensitivity, and they become starved for the highs they're missing. To compensate, you boost the highs and upper mids to get back the detail and presence your tired ears can no longer hear clearly.

You check your mix the next morning after your hearing has recovered, and it's like fingernails on a blackboard. Rather than cut the offending frequencies, you opt to boost the bottom end to warm up the mix. Now you have phase shift (unless you've been consistently boosting using a linear-phase equalizer) and alternating peaks in response across virtually the entire spectrum, resulting in an overly edgy sound, not to mention decreased headroom.

The solution is to mix at lower SPLs and to cut offending frequencies whenever possible instead of boosting other frequencies to compensate. For instance, it usually sounds better to carve away bass frequencies than to hype the midrange EQ when trying to make a mix sound more present. As a general rule, using EQ to cut will sound better than using it to boost.

Other factors leading to a harsh-sounding mix include having too many midrange instruments in the

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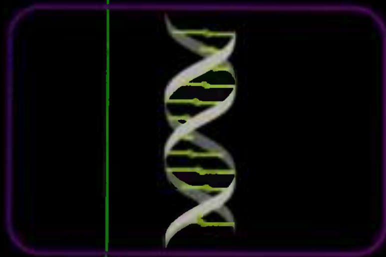

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arrangement or mixing them too up front with respect to the other elements. Know when to lower that bright organ pad to mellow things out a bit. Similarly, do you really need those 13 electric guitar overdubs? Consider muting some of the midrange elements that aren't essential and that only make the mix more fatiguing to listen to. Often the problem with a mix lies with the arrangement, and no amount of EQ will help.

No Sparkle and Bottom

Of course, sometimes EQ boost is needed to make a mix sound great. You can generally get away with boosting extreme bass and high frequencies more than you can boosting midrange frequencies. That's because the human ear is less sensitive to phase shift at the extremes of the audible spectrum. Even after boosting the bass and highs a bit, you may find that your mix still doesn't have the huge bottom end and sparkly highs you yearn to hear.

Again, the reason may be that you're listening at too loud a level while making EQ decisions. That's a problem because the human ear is subject to the Fletcher-Munson effect. In plain English, this means the ear is much less sensitive to bass and high frequencies when listening at low volumes than at high volumes. (Many consumer stereos have a Bass Loudness button to compensate for this reduced sensitivity to bass frequencies at low listening levels.) Ear fatigue aside, if you adjust EQ to taste while monitoring at loud levels, your mix might not sound sparkly and thunderous enough once the playback level is turned down.

Knowing this (and to preserve my hearing), I spend most of my mixdown time with my monitors set no louder than a spirited two-way conversation, and I'll often set them a lot lower. If I can get the mix to scintillate and thunder while listening at that low level, it is going to absolutely rock when it's cranked. Also working in my favor, my high-frequency sensitivity won't be trashed by sustained listening at loud levels, helping me retain an accurate perspective of spectral balance. That said, I will crank my control room monitors for about 20 seconds or so every hour when I'm mixing to confirm that the bottom and top ends still sound great and that I haven't taken any EQ boost too far.

One other point: if you compress tracks such as bass-guitar and cymbals post-EQ, the compression will at least partially negate the effects of any EQ boost on those tracks. Try placing the compressors before any EQ boost to get more sparkle and boom.

FIG. 3: The SPL Transient Designer can be used to increase the amplitude of the attack portion of drum tracks to create a punchier mix. The 2-channel TD2 is shown here.

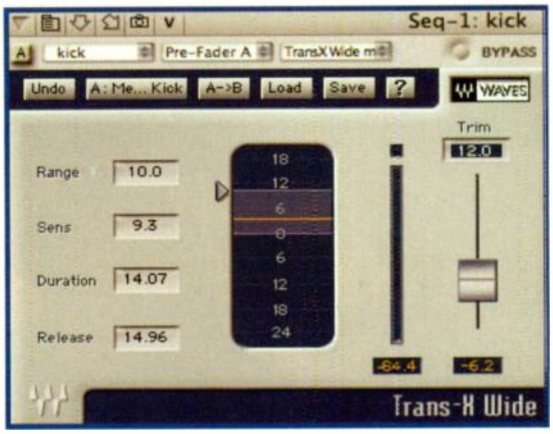

FIG. 4: The Waves TransX Wide plug-in, part of the company's Transform bundle, is set up here to deliver extra punch to a kick drum track.

Large Swings in Spectral Balance

Sometimes the timbre of specific elements of a mix (or of the whole enchilada) is a moving target. For example, the electric bass or acoustic guitar might sound boomy on some phrases yet be well balanced everywhere else in the song. The lead singer might have a shrill high register that bites your head off during the choruses, whereas the lower register sounds perfect during the verses. Or the entire mix might get edgy when, for instance, a bunch of midrange instruments pile on for one section of the song.

In these cases, static EQ settings won't sound good throughout the song. One worthy solution, albeit a time-consuming one, is to ride the EQ on individual tracks as needed. But a quicker and sometimes more elegant-sounding fix is to slap a split-band (aka multiband) compressor on the unruly tracks—or even on the entire mix. A split-band compressor divides the audio spectrum into multiple, adjustable frequency bands so that each can be compressed independently. Examples of outstanding split-band compressors include the Tube Tech SMC-2BM (a high-end analog unit) and the Waves C4 Multiband Parametric Processor, Waves Linear Phase Multiband, and PSP VintageWarmer 2 plug-ins (see Fig. 2).

Adjust the bandwidth of one or more of the split-band compressor's bands to include only the frequencies that exhibit large swings in level (for instance, bass frequencies that sometimes get too loud and make the mix boomy), and bypass the other bands. Then set each active band's threshold to be at or slightly below the level where the offending frequencies begin to annoy. Adjust each

active band's ratio, attack, and release controls to taste to limit how much (if at all) the unruly frequencies can bloom above the thresholds you've set. With the proper settings, a split-band compressor will automatically nip large swings in spectral balance in the bud. (For more in-depth information on how to use split-band compressors, see "Let's Split!" in the January 2004 issue of EM, available online at www.emusician.com.)

Insufficient Detail

When a mix is lacking in detail, boosting high-frequency EQ is often the wrong approach. When that just creates a glassy mix without solving the problem, try cutting the upper-bass and low-midrange frequencies instead. Too much energy in these bands can create a blanket of mud that obscures a mix's underlying transients, so try cutting between 200 and 500 Hz before boosting highs. Just be sure not to overdo it, or else you'll end up with a thin mix and too much detail.

Another thing to consider on a cloudy-sounding mix is whether sustained sounds such as string or synth pads are too loud. By simply lowering some or all of the tracks that exhibit minimal transients and loud average levels (sustain), percussive elements will more readily punch through. The end result will be a mix with plenty of detail that nevertheless retains its warmth because of minimal use of EQ.

Not Enough Punch

A mix lacking detail will also often lack punch, or transient elements married to tightly focused bass-frequency content. When a mix's spectral balance is already great, it can be a mistake to boost both bass and high frequencies to achieve more punch. The added highs might just

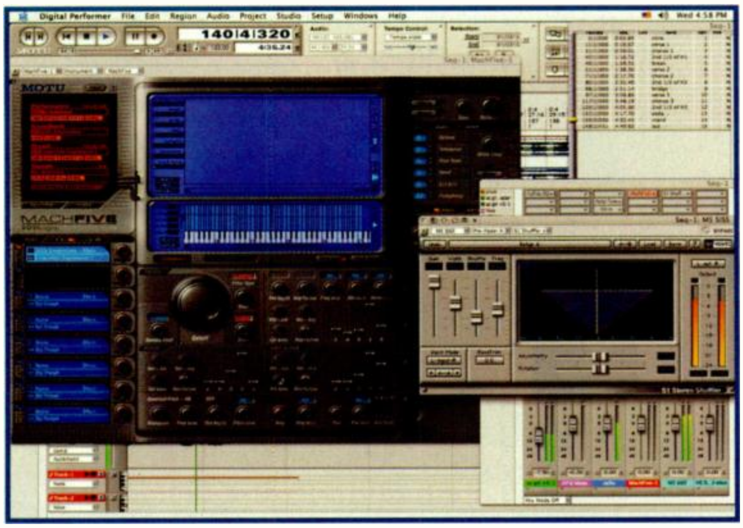

FIG. 5: A previously rendered track of a Sonic Implants Symphonic Strings ensemble section is patched through the Waves S1 Stereo Shuffler plug-in to increase its stereo width and create a dreamier sound.

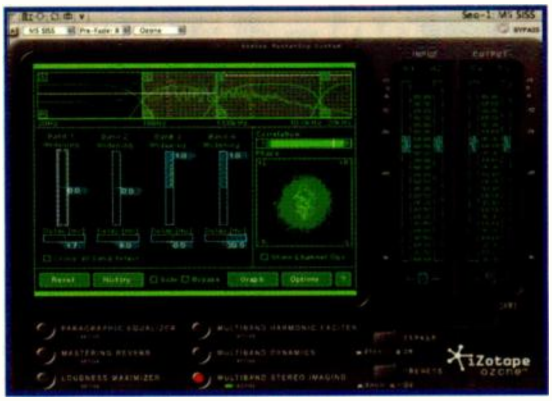

FIG. 6: The iZotope Ozone 3 plug-in bundle includes a Multiband Stereo Imaging component that can independently widen the stereo image of up to four frequency bands of a track.

make the mix sound glassy, whereas the extra bass boost could make it boomy.

Instead, use a dynamics processor to emphasize the attack portion of the low-frequency elements from which you want more punch (for example, trap drums and electric bass guitar). A solid-state, VCA-based compressor set to relatively slow attack and fast release times (start with 60 ms for each) will often do the trick.

The SPL Transient Designer, available in both 2- and 4-channel models, is an outstanding solution for increasing punch on individual tracks (see **Fig. 3**). This amazing analog processor uses an envelope follower to change the amplitude of the attack and release portions of an audio signal. With the twist of a single knob, the Transient Designer can greatly enhance the beater slap of a kick drum or the crack of a snare drum, and it can make a bass guitar track pop like balloons.

For reshaping transients inside the box, I often turn to the Waves TransX Wide plug-in, which is part of the Transform bundle (see **Fig. 4**). It offers much greater control over the attack portion of sounds than the Transient Designer but gives you no control over the release phase. TransX Wide is a surefire ticket to slammin' drum tracks.

Too Much Compression

These days, many mixes are so overcompressed that they become irritating and fatiguing to listen to after only one or two minutes. Overcompression is like a plague contaminating our industry. Make no mistake—I love stereo-bus compression, and I like my mixes loud, but there's a big difference between pumped-up, exciting dynamics and just plain annoying noise and distortion.

The old saw about using your ears when determining how far to push mix-bus compression is all well and good, but I have a more practical suggestion: watch the crest factor on your stereo-bus meters. The crest factor is essentially the difference between peak and average

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Vincent **di Pasquale**

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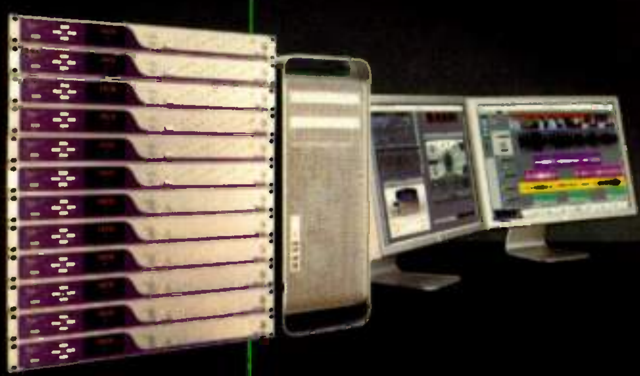

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levels, and keeping tabs on it is a good reality check against what ears addicted to volume might otherwise be pushing to accomplish.

Spend time listening to your favorite records—particularly those that have dynamics you'd like to emulate in your mixes—patched through the 2-track return of your mixing console or DAW, and keep a close eye on the meters. (Make sure that the meters are peak reading and set to prefader listen, and that all processing is disabled.) Note how much the meters rise above average levels during transient peaks throughout various sections of each song. Then shoot for roughly the same crest factor in your mixes. You can learn a lot by being a good meter reader.

The Chorus Doesn't Climax

You had high hopes for your new power-pop ballad, but something is holding it back. Your tracks were all captured with plenty of dynamic range, the performances were killer, and the arrangement positively soars during the hook. Yet for some reason, the chorus just doesn't deliver the big payoff it should in your mix. It's time to look at your mix-bus compressor settings again.

Sometimes an engineer will set up the mix-bus compressor for a big, in-your-face sound at the beginning of mixdown, when working on relatively quiet verses, and will just assume it's going to sound even bigger during the choruses and other climaxes. A compressor with too low of a threshold and too high of a ratio will suck the life out of the hook when it hits—sometimes the chorus will actually sound lower than the verses. Raise the compressor's threshold and lower its ratio to no more than 2:1 to give the hooks room to explode. You might also need to back off the compressor's attack time a bit.

Washy Sound with No Depth

Adding reverb to a mix is a great way to make it sound bigger. The larger the implied acoustic space, the more depth and width the production takes on. But running virtually everything through reverb in an attempt to make the mix sound huge is a common mistake of neophyte mix engineers.

Something can sound big only if something else sounds small. In part, it's the contrast between close-up and far away that gives a mix depth. (The nuance captured by superior mics and mic preamps is another contributing factor, but that's a discussion best left for another article.)

When many tracks are drowning in reverb, everything begins to sound indeterminately far away, and there is not enough of an anchor for the brain to get a picture of what is psychoacoustically up-close. Not only has depth gone out the window at that point, but the mix also takes on a washy character dominated by diffuse echoes that blanket any semblance of detail and punch.

One solution, of course, is to make some tracks very dry. You might even need to make a *lot* of tracks completely dry in order to attain the depth you desire. Instruments that produce inherently sustained or reverberant sounds, like cymbals and strummed acoustic guitars, often benefit by turning their reverb sends way down or completely off. That's especially true of dense arrangements that are prone to drown in ambient soup. The acoustic guitar already supplies built-in reverb from the resonating chamber that is its body. Piling on a bunch of additional reverb makes little sense, unless that instrument is being played in short, percussive bursts such as during a largely monophonic introduction or solo.

Despite the foregoing, there are instances where a healthy dose of time-based effects is needed to create the desired sonic landscape. In such cases, try adding predelay to some of your reverbs, or try substituting single echoes or multitap delays for reverb effects. These alternatives allow the dry signal to voice before the effect kicks in, giving a front-to-back effect in the soundstage that can really enhance perceived depth while preserving detail.

Another remedy for a washy mix is to eliminate one of the channels of a stereo track, thereby reducing that

FIG. 8: A lead vocal track is compressed by two Waves Renaissance Compressor plug-ins chained in series.

FIG. 7: The Avant Electronics Avantone MixCubes are outstanding passive monitors to reference how well lead vocals are sitting in a mix.

track to mono. Converting most of your stereo tracks to mono will help provide the pinpoint imaging that is a remedy for a washy mix. Conversely, using a lot of tracks that were recorded with spaced-pair stereo-miking is a recipe for mud soup. Each of those tracks is a rendering of an instrument playing in an acoustic space, and simply panning them differently to separate them won't necessarily lend focus and depth to your mix.

Panning a few stereo tracks across the stereo field is a common strategy. But if you pan one stereo track hard left and at ten o'clock (for left and right channels, respectively), another at ten and two o'clock, and a third at two o'clock and hard right, what have you accomplished? You now have three small rooms in a left-center-right arrangement superimposed over whatever other acoustic spaces are implied by added reverb on other tracks. No wonder the mix sounds washy!

In summary, to clean up a washy mix: Keep a number of your tracks mostly or completely dry. Mute one side of one or more stereo tracks. And use discrete delays and reverb predelays to create depth without sacrificing detail.

> Using a lot of tracks recorded with spaced-pair stereo-miking is a recipe for mud soup.

Collapsed Stereo Image

Suppose you've hard-panned a number of tracks, but your mix still doesn't sound as wide as you'd like. What's wrong with this psychoacoustic picture?

Your hard-panned tracks might have too much bottom end. Bass frequencies are inherently omnidirectional, meaning it's hard for the human ear to determine where they originate. That's because bass frequencies have long wavelengths, and easily wrap around the listener's head to either ear with minimal phase difference.

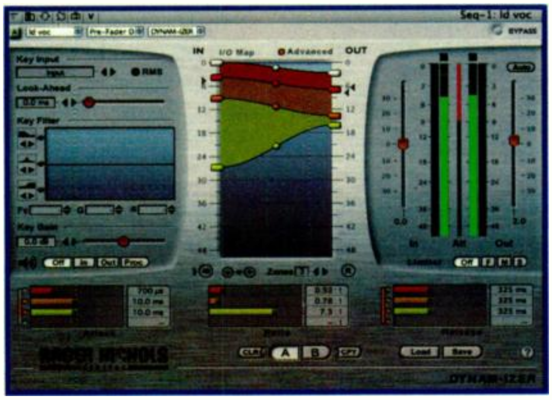

FIG. 9: Roger Nichols Digital's superb Dynam-izer plug-in divides a track's dynamic range into as many as four different zones for independent dynamics processing.

From a stereo-field perspective, tracks that are panned hard left and hard right are potentially the most directional elements of a mix, whereas center-panned tracks are the least directional. The more the prominent omnidirectional bass frequencies are in hard-panned tracks, the more the hard-panned tracks' perceived positions in the stereo field get pulled toward the center. Conversely, rolling off bass frequencies on hard-panned tracks will move them farther from the center.

There is no magic frequency at which omnidirectionality occurs. Sound becomes progressively more omnidirectional as its frequency gets lower. So the lower in frequency the bass content of a panned track, the more it will move toward the center (assuming that the high frequencies also present in the track don't compensate). Even hard-panned tracks with a lot of low-midrange frequency content will move slightly toward the center image.

To make a mix sound wider, try rolling off the bass and possibly some low-midrange frequencies on hard-panned tracks. Also, hard-pan tracks with lots of high-frequency content—such as cymbals, shaker, and piccolos—to gain more apparent width. If you still need more width in your production, running a single stereo track through a stereo-imaging plug-in such as Waves S1 Stereo Shuffler or iZotope Multiband Stereo Imaging (which is part of the Ozone 3 multicomponent plug-in bundle) will do the trick nicely. Be judicious, however; using this kind of processing on multiple tracks or on an entire mix can quickly make your production swim in a washy, diffuse soup (see **Figs. 5 and 6**).

Vocals Consistently Too Loud or Too Low

We've all been there. You thought you had the perfect mix, but then you hear it on a friend's stereo system, and the lead vocal suddenly sounds too loud, in front of and divorced from the backing music. Or it's buried underneath an onslaught of guitars, making your clever lyrics lost to all ears. What went wrong?

Setting the perfect vocal level can be difficult. The vocal's balance with respect to other tracks will always sound different on different monitors. What works for me is listening on bass-challenged monitors such as the Avant Electronics Avantone MixCubes (see **Fig. 7**) or the discontinued Yamaha NS-10M Studio. Without prominent bass frequencies masking the lead vocal, I can more accurately gauge how loud the money track is with respect to the other tracks.

If you have only one set of reference monitors and use a subwoofer, turn off the subwoofer when setting

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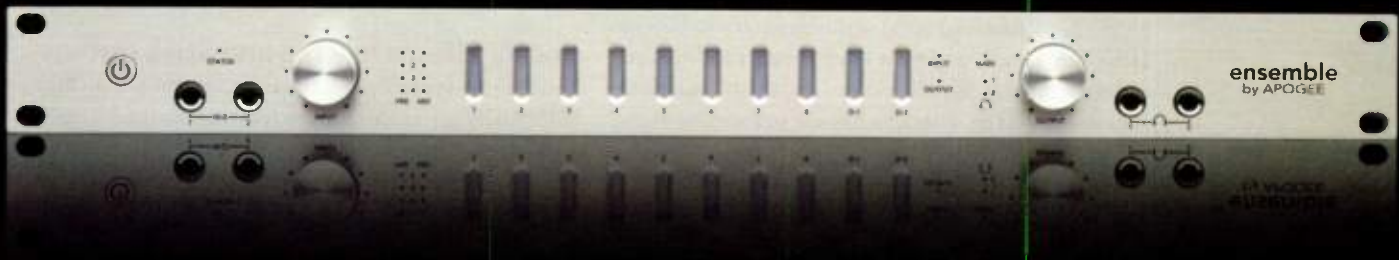

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the level of the lead vocal. Also, listen to the mix at very low volume to let the Fletcher-Munson effect decrease your perception of bass and high frequencies. That will leave you with an unobstructed window into the midrange, where the lead vocal primarily sits.

Slowly turning down your close-field monitors to the point of almost dead silence is another effective technique. If the lead vocal is the last track to become inaudible, you'll know it's loud enough to be easily heard on most if not all systems. If it's still relatively loud when all the instruments are practically mute, the lead vocal probably needs to be turned down.

Of course, some styles of music call for louder vocals than others. For example, the vocal should generally be mixed louder on a country song than on a rock production. But these guidelines should give you the needed perspective to make the right judgment call for your chosen format.

Vocals Alternately Dip and Stick Out

Lead vocals typically benefit from compression. That helps them sit at the proper level throughout a mix. Compression limits the dynamic range of the track so that it becomes neither too low nor too loud in the mix on any given phrase. But with a very dynamic vocal, it may be impossible to compress aggressively enough to accomplish this goal without completely squashing the track, ruining its timbre, and destroying any depth and nuance. If, after you push the compression as far as you dare, the vocal still dips too much on some phrases and sticks out too much on others, here are some alternatives.

Try chaining two or more compressors together in series, with each adjusted to more moderate control settings so that no single one is going to squash the track (see **Fig. 8**). For instance, the first compressor could have fast attack and release times and a high threshold setting so that it kicks in with its high compression ratio only during peaks. The second compressor might be set to a relatively low threshold and ratio and moderate attack and release times so that it is processing average levels pretty much all the time, but with kid gloves. Here, the second compressor isn't expected to clamp down on transient peaks, so it can be set for more moderate action on average levels that will preserve the track's timbre and nuance. Meanwhile, the first compressor needn't have its threshold set so low that it will rein in the average levels of the vocal track—that's the second compressor's job, and it will do it more gently.

Despite the time-tested procedure of chaining compressors together in series, Roger Nichols Digital offers a far more powerful and elegant solution to reining in extremely dynamic vocals. The company's groundbreaking Dynam-izer plug-in divides a track's unprocessed dynamic range into as many as four mutually exclusive and contiguous zones (see Fig. 9). It can then independently compress or upwardly expand the track across each zone using different ratio, attack, and release settings. The key point here is that each compressor or expander applies processing only across the input-level range to which it is assigned. You can, for example, optimize the zone settings to upwardly expand the quietest vocal phrases, gently compress moderately loud sections, and smash transient peaks forcefully.

After using the foregoing techniques, the lead vocal still might fluctuate too much in level on a few remaining phrases. Don't be afraid to ride the track's fader to even out those sections of the vocal, and record your fader moves with your DAW or mixer's automation. Also, some buried lyrics may be brought out more effectively by boosting upper-midrange or high frequencies rather than riding the fader (remember to undo the EQ boost immediately afterward). In some of my mixes, the lead vocal's track will have dozens of fader and EQ moves over the course of a three-minute song, depending on how even the singer's performance was. Don't be afraid to do whatever is necessary to make the vocal track perfect.

The Perfect Mix

None of the techniques discussed in this article will lead you to a great mix on their own. They must all be taken into consideration at once and balanced against one another. For instance, striving for too much detail and clarity can result in a thin, icy mix that will sound even more fatiguing if brickwall limiting is applied to achieve competitive loudness. And a mix with too wide of a stereo image and key tracks panned hard left and right might lose needed center focus and punch.

Keep your original vision for the song in mind while you mix, asking yourself along the way if any of these 12 problems are beginning to creep in. Note if any corrective tweaks you perform introduce their own problems, but be aware that effective mixing usually entails a series of smart trade-offs. Putting these compromises into perfect balance is the key to an outstanding mix. **EM**

EM contributing editor Michael Cooper is the owner of Michael Cooper Recording in Sisters, Oregon. You can hear some of his mixes at www.myspace.com/michaelcooperrecording.

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JANE RICHEY

Pushing the Right Buttons

By Paul Tingen

Bill Bottrell uses technology and psychology to produce award-winning music.

If you want an indicator of Bill Bottrell's skills as a producer, start by looking at the list of heavyweight artists he's produced: Rosanne Cash, Sheryl Crow, Thomas Dolby, Michael Jackson, Shelby Lynne, Madonna, Tom Petty, and the Traveling Wilburys, to name a few. Then there are his numerous Grammy nominations, his Grammy win (1994's Record of the Year for Sheryl Crow's *Tuesday Night Music Club*), and multiple millions in record sales.

But the ultimate proof of Bottrell's skills can be gleaned by listening to the albums he's engineered, produced, and in many cases also played on and cowritten material for. He has a knack for creating records that combine the refinement of a modern studio-recorded CD with a profound sense of rock 'n' roll vibrancy.

Tuesday Night Music Club is a relatively early example of Bottrell's gift. For more-recent demonstrations, listen to Rosanne Cash's *Black Cadillac* or Van Hunt's *On the Jungle Floor*, or check out the songs that are uploaded on Annie Stela's MySpace page (www.myspace.com/anniestela). Or, if you want to hear the ultimate triumph of feel over faultlessness—and get a sense of Bottrell's amazing songwriting, singing, and performing talents—listen to his DIY, lay-them-down-in-two-hours demos, a rotating selection of which can be found on his MySpace page (www.myspace.com/billbottrell).

Atmospherics

How is Bottrell able to marry sonic sophistication with such a strong sense of spontaneity? Part of the answer lies in his grasp of people skills. He doesn't simply turn on the red light and expect artists to perform at their peak. He believes in managing the psychological aspects of a session. That starts with providing a comfortable working environment, but it involves a lot more.

"The main thing that I do as a producer, for better or for worse," Bottrell says, "is to orchestrate emotions in a room toward something. I have always tried to begin with an interplay between musicians. I may put a little 808 beat down for the tempo and have the singer sing and play piano or guitar and have a couple of musicians play along. That would always form the basis from which I build the record. And to get a good performance, you need to set a mood—a vibe—and this has to do with the people and the environment, the room they're in.

"I carefully regulate who is in the room. I don't mean [that I] eliminate people—it can be a matter of consciously bringing people in to add to the vibe. I use musicians who can hang in a room and contribute to the energy, whose egos don't shut down the energies. I don't pick musicians for what they play, but for who they are and their ability to bring something to the room. Then I follow the energy of the group, wherever that leads. Group dynamics, the psychology of small groups of people, is a fascinating subject to me—the arcs of synergy, how that ebbs and flows. And the producer has to be integrator of all these energies, making sure they get focused in one direction.

COURTESY MIMI PARKER

FIG. 2: Bottrell and Mimi "Audia" Parker in the studio.

"I don't overly plan things. I like it loose, and I like it somewhat chaotic. I like to find every distraction that I can, short of taking people's minds off the music. Musicians have comfort zones and way too often play their most comfortable thing, as opposed to what's needed and what supports the energy of the whole [see **Web Clip 1**]. Getting them out of that is the whole challenge, and I may do things like put a thick book on the snare or throw things at the guitarist while he's doing a solo. The aim is to distract. The brain is pretty strong and people's thoughts are many, and distraction creates better performances."

JANE RICHEY

FIG. 1: Inside Bottrell's control room at School House Studio, with his Neve 8058 console in the background.

Strong Opinions

Before talking more about production, Bottrell discusses the state of the music business and music technology. "Recorded music has had its glory days, and it is on the wane now," he says. "The first indicator occurred during the '90s, the era of CD reissues, when I noticed that anything we put out had to compete with a rich library of 100 years of great recordings. This made it very difficult to create something that you felt would stand the test of time. Myself and others continued making records like the ones we had made in analog—full of nuance, fire, energy, and sometimes cacophony. But we found that on digital they sounded terrible. Only when producers and artists came up with ways of shifting focus, of saying things with minimal dynamics and arrangements, like in hip-hop, did things progress. The fact is that music morphs to fit the medium. And digital breeds Disney-fication."

In other words, the capabilities of digital workstations and sequencers to manipulate performance characteristics such as pitch and timing are taking the soul out of music. "We have lost something—the real value of music, the story, the humanity," he says.

"Music has become wallpaper. Everywhere you go there's a speaker playing pop songs, and people have become accustomed to pop music playing as a constant soundtrack to their lives. Pop music takes up 90 percent of the space in the media, leaving the dusty corners for the rest of us. Recorded music has become merely part of a franchise. It's Beyoncé who matters, not the record she makes. In addition, we have to look beyond the idea of recording and selling music, because music will be free. You can't control the copyrights, and you can't keep track of digital duplication."

Biting the Hand

Bottrell's diatribe seems to predict the end of the music industry as we know it, and when this is put to him, he responds, "Let's hope so!" In his view, what he calls "mediated music" will become purely a calling card for live performance—something he welcomes because he feels that "playing live is what musicians do."

The producer's words symbolize the love/hate relationship he has with the mainstream music machine. While on one hand Bottrell welcomes what he sees as the music industry's impending demise, on the other he has helped supply it with many of its biggest hits.

This ambivalence is nothing new—Bottrell has at various other stages in his career bitten the hand that fed him, and twice he dropped out entirely. Growing up in Los Angeles, the teenage Bottrell played in rock bands and in 1970 bought a Teac 4-track tape recorder. He developed his engineering skills through various jobs at commercial studios in the L.A. area. He went freelance in the early '80s, and by the end of the decade, Michael Jackson and Madonna had sought him out because of his unique combination of technical and musical skills.

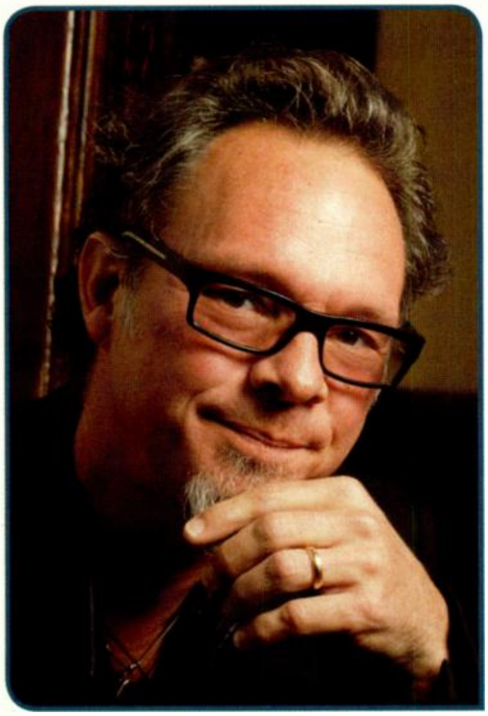
Despite his success, Bottrell felt empty and dissatisfied, and he dropped out in 1990. He built a studio in Pasadena called Toad Hall, and vowed to work only with what he called "marginalized artists." The problem with his plan was that some of those marginalized artists became famous through his Midas touch, one of them being Sheryl Crow. Bottrell nevertheless became known as the enfant terrible of the American music industry.

By 1996 he retreated to the Northern California town of Albion, where he spent time with his family and vowed to work and perform only locally. When he received a demo of Shelby Lynne in 1998, however, Bottrell couldn't resist his instincts. Using his Toad Hall equipment, he built a new studio, called it William's Place, and recorded, produced, and cowrote Lynne's *I Am Shelby Lynne*, which earned her a Grammy and Bottrell another Grammy nomination.

The Here and Now

Fast-forward to 2007, and the global music machine appears to be reciprocating Bottrell's ambivalence, increasingly rejecting his work. "In the past seven years," he says, "eight out of the ten full albums I have produced are either unreleased or have only been released as a contractual necessity."

While continuing to champion "marginalized artists," Bottrell appears to have overcome his antipathy to the big time; during 2006 he spent three months in Bahrain working on a new Michael Jackson album, but the singer never showed, so the project was shelved. Bottrell is now working with Seal on a new album. Part of the work for it was done in Bottrell's latest digs, School House Studio. William's Place was closed in 2004, but Bottrell recently moved his gear a few miles down the road and set it up in his home. The first thing to officially come out of the new setup was the theme song he did with Seal for the movie *The Pursuit of Happyness* (Columbia Pictures, 2006). In the context of Bottrell's tirade against digital and the music industry, and his idea that recorded music is going to die, what moves him to rebuild his studio and still record music?



JANE RICHEY

According to Bottrell, the most important part of his job is to "orchestrate the emotions" of the talent he's producing.

"A person wants to stay relevant," he says, "and I'm doing that through finding my own way in the digital medium while still applying certain standards of art and expression." From a technical perspective, how does Bottrell go about remaining relevant in the digital age? The omnipresence of analog and old-school equipment at School House hardly comes as a surprise, but has he changed his way of working or at least some of the equipment in his studio?

School House Studio's heart is a Neve 44-input 8058 console (see **Fig. 1**), which incorporates 28 31102 mic-pre/EQs and 32264a compressor/limiters.

In addition, he has an array of vintage mics, tube-mic pres, and classic signal processors. Naturally, there's a Studer A800 analog 16/24-track. Oh, and there's the small matter of a full-blown Digidesign Pro Tools HD 7.3 system, running on an Apple Power Mac G5.

But didn't Bottrell say that digital audio recording and editing programs like Pro Tools have helped take the humanity out of music? Has he sold his soul to the digital devil after all? "Well, I was a fan of analog, and I still am," he says. "It still excites me to work on analog, and if I have a project that's appropriate, I'll go straight to my Studer. The Rosanne Cash album was recorded at the Pass Studio in L.A. on a Studer 827 24-track. But all the other projects I've done in recent years were recorded to Pro Tools. Sonically, Pro Tools took a huge leap forward with HD. They really got the system over the hill, and high-definition digital definitely has more sonic detail [Bottrell routinely records at 24-bit, 96 kHz]. But the emotional details of a performance come through better on analog—it's just a different thing."

Of course, the million-dollar question still is, why do performances sound so gripping on analog? Countless digital applications try to answer this, with varying degrees of success; witness all sorts of Warmth, Harmonics, and Analog Mode buttons. I suggest to Bottrell that while recording on analog is like painting on a white canvas (that is, the background noise and tape compression), recording on digital is like working on a transparent canvas. "I think that's a very good analogy, and rather than what sounds best, you have to ask what sounds most appropriate for the music," he says.

"I've never attempted to make things sound as realistic as possible, but I still go for the illusion of a bunch of people playing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a room; that's the basis," explains Bottrell. "You don't work it too much. You go for early takes and you go for immediacy, as opposed to any sort of perfection.

"I never believed in perfection, and the temptation to endlessly rework things in Pro Tools is not irresistible for me. I've done many things with Pro Tools as just a recorder, using the same recording techniques as I did with analog. I'll use the Pro Tools arrangement and editing facilities, but I really dislike the sound of an all-digital mix, so I'll always split it out and put it through my Neve console. I think that's a fundamental difference. Anyone who uses Pro Tools or a computer should save up $1,500 and go out and get an analog board from the late '80s or early '90s and mix through that. Leave the EQ flat and do what you want to do in Pro Tools, or whatever system you're using, but get the sound out by running it through a console."

Perhaps unexpectedly, the way Bottrell applies his Pro Tools rig today goes much further than just as an analog tape recorder with advanced editing facilities. Although he has a Pro Tools engineer, Mimi "Audia" Parker (see **Fig. 2**), the producer now very much likes to get his hands digitally dirty (see **Web Clip 2** for studio techniques from Bottrell and Parker). "The sound of my Neve console is lovely, so I use the EQ and often instinctively walk over and twist the knobs. I'll also use the Neve mic preamps and the Pro Tools A/D converters. I take plug-ins on a case-by-case basis [see **Web Clip 3** for plug-in settings from Bottrell and Parker]. I can't remember the names—Mimi sets all these things up for me—but there's this equalizer that's green [McDSP Filterbank F2], and I can put that thing over everything that I record. I love that thing. It is flexible and brilliant—all I need from an EQ. I also use Pro Tools for creating samples, depending on the music. I have an open mind about Pro Tools now and will try anything, but I'll also know when it's crap. The Auto-Tune thing is a blasphemous thing to do to a vocalist, absolutely evil. We can talk about changing media and the death of the music industry, but part of the blame for the state of music has to go to crap music made by people with poor taste."

BILL BOTTRELL: A SELECTED DISCOGRAPHY

Rosanne Cash, *Black Cadillac* (Capitol, 2006); producer, engineer, mixer

Van Hunt, *On the Jungle Floor* (Capitol, 2006); producer, engineer, mixer

Annie Stela, *Annie Stela* (Capitol Records, 2006); producer, engineer, mixer

Five for Fighting, *The Battle for Everything* (Sony Music, 2004); producer, engineer, mixer, cowriter

Kim Richey, *Rise* (Mercury/Lost Highway, 2002); producer, cowriter, musician

The Stokemen, *Class of Dude* (This Records, 2002); producer, cowriter, musician

Alisha's Attic, *The House We Built* (Mercury UK, 2000); producer, engineer, mixer

Elton John, *Songs from the West Coast* (Mercury UK, 2000); mixer

Shelby Lynne, *I Am Shelby Lynne* (Island, 2000); producer, mixer, cowriter

Tom Petty and the Heartbreakers, "Surrender" from *Anthology: Through the Years* (MCA Records, 2000); producer, engineer

Rusted Root, *Welcome to My Party* (Island Def Jam, 2000); producer

Sheryl Crow, *Sheryl Crow* (A&M Records, 1996); cowriter

Michael Jackson, "Earth Song" from *History: Past, Present and Future—Book 1* (Epic Records, 1995); producer, engineer

Linda Perry, *In Flight* (Interscope Records, 1995); producer, mixer, cowriter

Rusted Root, *When I Woke* (Mercury Records, 1994); producer, engineer, mixer

Sheryl Crow, *Tuesday Night Music Club* (A&M, 1993); producer, engineer, mixer, cowriter

Michael Jackson, *Dangerous* (Sony Music, 1991); coproducer, engineer, mixer, cowriter

Madonna, *Truth or Dare* (Miramax Films, 1991); producer (soundtrack)

Madonna, *Dick Tracy* (Warner Brothers, 1990); coproducer, engineer, mixer (movie songs)

Madonna, *Like a Prayer* (Warner Brothers, 1989); engineer, mixer

Tom Petty, *Full Moon Fever* (MCA Records, 1989); engineer, mixer

Traveling Wilburys, *Traveling Wilburys*, vol. 1 (Warner Bros., 1988); engineer, mixer

Thomas Dolby, *Aliens Ate My Buick* (EMI, 1987); coproducer, engineer, mixer

Free and Clear

So what advice does Bottrell have for those recording in personal studios? "Technique has to be secondary to what you want to express and hear. Whatever tool you have, you have to master it and make it transparent for you, and then move on to the art. I probably cursed Pro Tools for two years. At first it subdued me, and then I subdued it, and now it's transparent for me.

"You start with something vibrant, and if you are the artist recording yourself, be the artist. Technique should be put out of your mind. I suppose that's what I do on my solo work. I record quickly and spontaneously, and whatever comes, I leave it, even as I may later go back to it and edit or otherwise change it. But it's very important to not let anybody or any concept stop you because you think it is not supposed to be done in a certain way. I like to start over with every project that I do and see where that leads. There are a thousand ways to make a record, and we have to find the one that works." EM

EM WEB CLIPS

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

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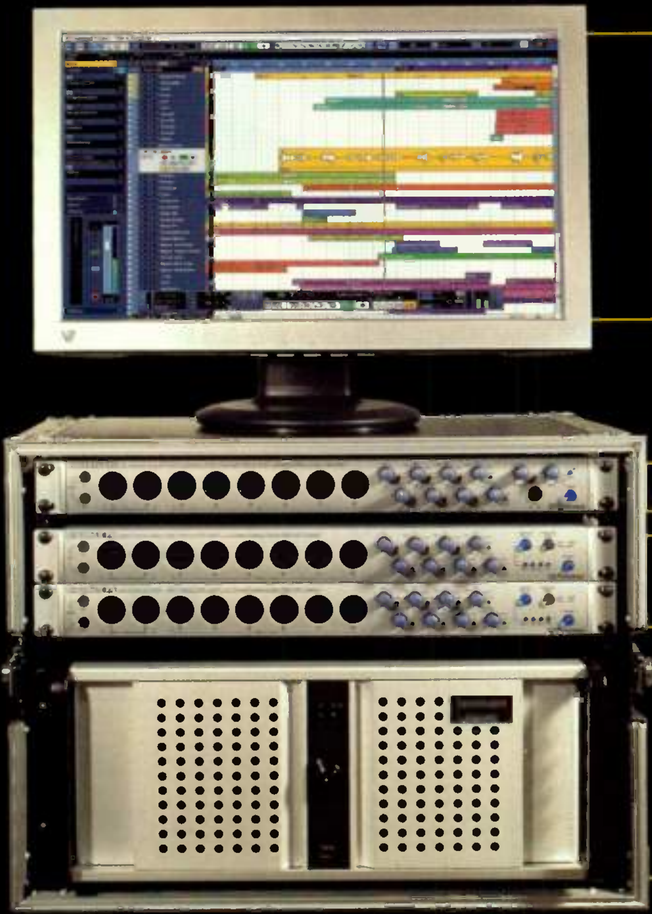

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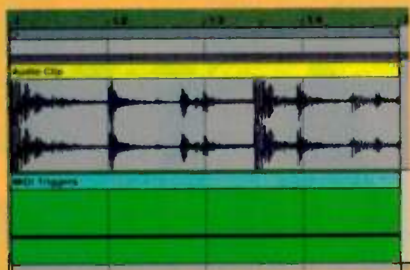
RECORDING

STEP-BY-STEP INSTRUCTIONS

1



Set Live's tempo to match the audio clip's. Insert a MIDI clip on an adjacent track with a note at C3 extending the full length of the clip.

2

Insert locators in the arrangement at the desired audio slice points.

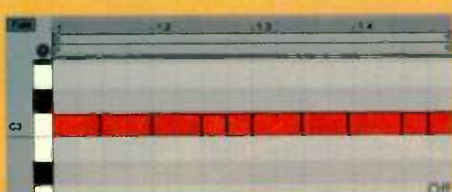
3

Split the audio and MIDI clips at each locator. Do not consolidate the slices.

4

Drag each audio slice to its own Impulse slot.

5



Consolidate each MIDI slice. Nudge the note in each slice (except the first) twice to the right and once to the left to separate the notes.

6

Consolidate the MIDI slices into a single MIDI clip and rearrange their pitches to taste.

To do that, move the trigger notes in the MIDI clip. Moving a note vertically changes the slice it plays; moving it horizontally changes the slice's timing. You can use the latter to tighten the feel or to destroy it altogether by quantizing (see **Web Clips 3 and 4**).

You can make subtle or radical changes by replacing some or all of the samples in the Impulse slots. Beyond that, you can route any Impulse slot to its own audio track for separate effects processing, and that even works with several Impulses embedded in an Instrument Rack (see **Web Clips 5 and 6**). EM

EMWEB CLIPS

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

Making the Least of It

By Vijith Assar

What lossless file compression is and why to use it.

If you've ever tried to conserve disk space by archiving old recording projects, you probably know that standard compression schemes like ZIP and RAR don't work well on audio data. These schemes shrink file sizes by minimizing data redundancy and work best with the types of repeated patterns common in file formats such as Microsoft Word's DOC format. Although audio signals often consist of repeating waveforms, the repetition is represented in a manner that is completely alien to standard data-archival programs. At best, you can expect percentage savings that are in only the single digits.

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Enter the *lossless codec*. Lossless codecs are functionally equivalent to compression schemes used to create ZIP and RAR files but use data-coding methods optimized for audio. *Linear predictive coding* (LPC), for example, saves bits by "predicting" what future sample values will be by looking at previous samples. *Run-length encoding* (RLE) involves using the same number to represent repeated values (such as those of a long passage of silence). The resulting files take up significantly less disk space than the uncompressed versions—achieving a savings of 40 to 60 percent is quite normal—and the compressed files sound identical to the uncompressed one. You can even convert a compressed file back into a WAV or an AIFF file that is bit-for-bit identical to the original because all the original data is intact.

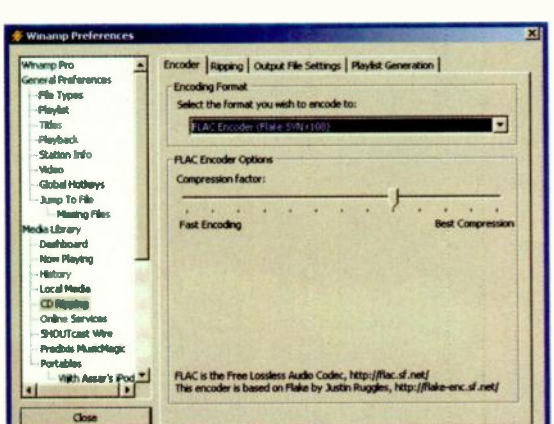


FIG. 1: The most important part of configuring a FLAC encoder is choosing the compression level.

Just as television has moved from black-and-white to color to HD, lossless compression is rising in the audio world. All industry pros should therefore know how to package their mixes into a high-quality portable file.

Meet the LACers

By all accounts, the Shorten format, an open-source format developed most recently by Wayne Steilau and Jason Jordan (for complete developer contact information, see **Web Clip 1**), was the first method of lossless compression to make it to market. Available as early as 1993, it's still used widely for live-concert trading among jam-band fans, most notably at the concert trading hub of etree.org. For most other uses, however, it has fallen behind newer counterparts such as Josh Coalson's Free Lossless Audio Codec (FLAC), which was released in 2003 and has since garnered the most widespread hardware and software support of any lossless codec. With few exceptions, any platform that incorporates any sort of lossless compression supports FLAC. The freely available source code makes it easy for interested developers to integrate FLAC into their products, so it's by far the most future-proof of the available formats.

Although FLAC is a command-line program, graphical front ends such as Kevin Athey's MacFLAC are available for most operating systems. FLAC provides a tremendous number of intimidating text commands, but the only one worth worrying about is the compression level, which is specified by a number between 0 and 8. The resulting sound quality is the same for all levels, but higher values give you smaller files. You are usually able to choose this setting with a slider that's on the Preferences page of your encoder's front end (see **Fig. 1**). The highest compression levels take much longer to encode yet give only minimal returns regarding file size (see **Fig. 2**). Further, because they require more processing power to decode, some portable playback devices can be picky about the compression levels they'll support. In general, the best values to use are between 4 and 6.

Impulsive Decisions

By Len Sasso

Using Ableton Live's Impulse with sliced loops.

Ableton Live's Impulse drum sampler makes an excellent player for sliced audio files because of its ability to apply separate processing and output routing to each of its slots. In this month's "Making Tracks" column (see p. 60), I show how to slice audio files directly in Live and combine several Impulses in a Live Instrument Rack to reach beyond Impulse's 8-slot limit. Here I'll give some tips for making the most of that rack.

I have based my examples on a rack that contains four Impulses triggered by the first eight notes in the C chromatic scales (C through G) starting on C3, C4, C5, and C6, respectively (MIDI Note Numbers 60–67, 72–79, 84–91, and 96–103). **Web Clip 1** contains such a rack. The most straightforward way to use that rack is to arrange the trigger notes for the resulting 32 slots manually, which is what you'd do with slices that have been sequentially carved out of a single audio file.

A Better Alternative

You can also fill each Impulse with alternative slices from similar loops. Many MIDI loop collections come with several variations on the same basic loop. The timing of the slices in these variations is usually close enough for you to use the same trigger file for any variation. In that case, you can redirect the trigger notes on the fly to mix the variations, and Live's Random MIDI effect is perfect for that.

The Random effect shifts incoming notes by some interval, and four parameters (Chance, Scale, Choice, and Sign) determine that interval. Chance sets the probability of a shift occurring. Scale sets the smallest possible shift interval, and Choice specifies possible multiples of that shift interval. Sign determines the direction of the shift (Add for up, Sub for down, and Bi for either [see **Fig. 1**].

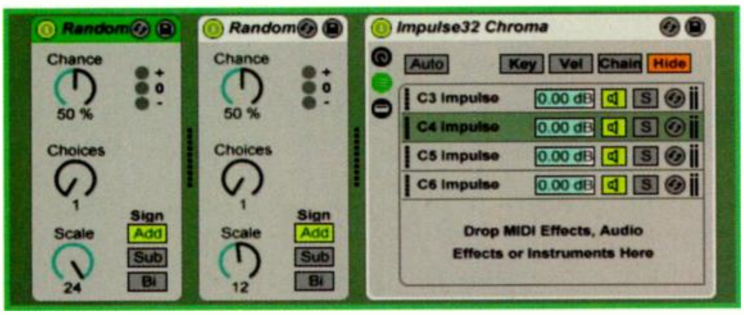


FIG. 1: The Live Instrument Rack (right) contains four Impulses in parallel. The Random MIDI effects (left and middle) ensure that the Impulses are selected equally.

To choose between the four Impulses, use two Random effects in series, set both Sign buttons to Add, both Choices to 1, and their Scale knobs to 24 and 12. The first Random effect shifts the focus between the first and second pair of Impulses in the rack. The second Random effect shifts the focus within the chosen pair. Together, the two Chance knobs set the probability of each Impulse receiving the incoming note, which should always be between C3 and G3. For example, with both Chances set to zero percent, the top Impulse will always be used. With both Chances set to 50 percent, each Impulse is equally likely to receive the note.

Mutations

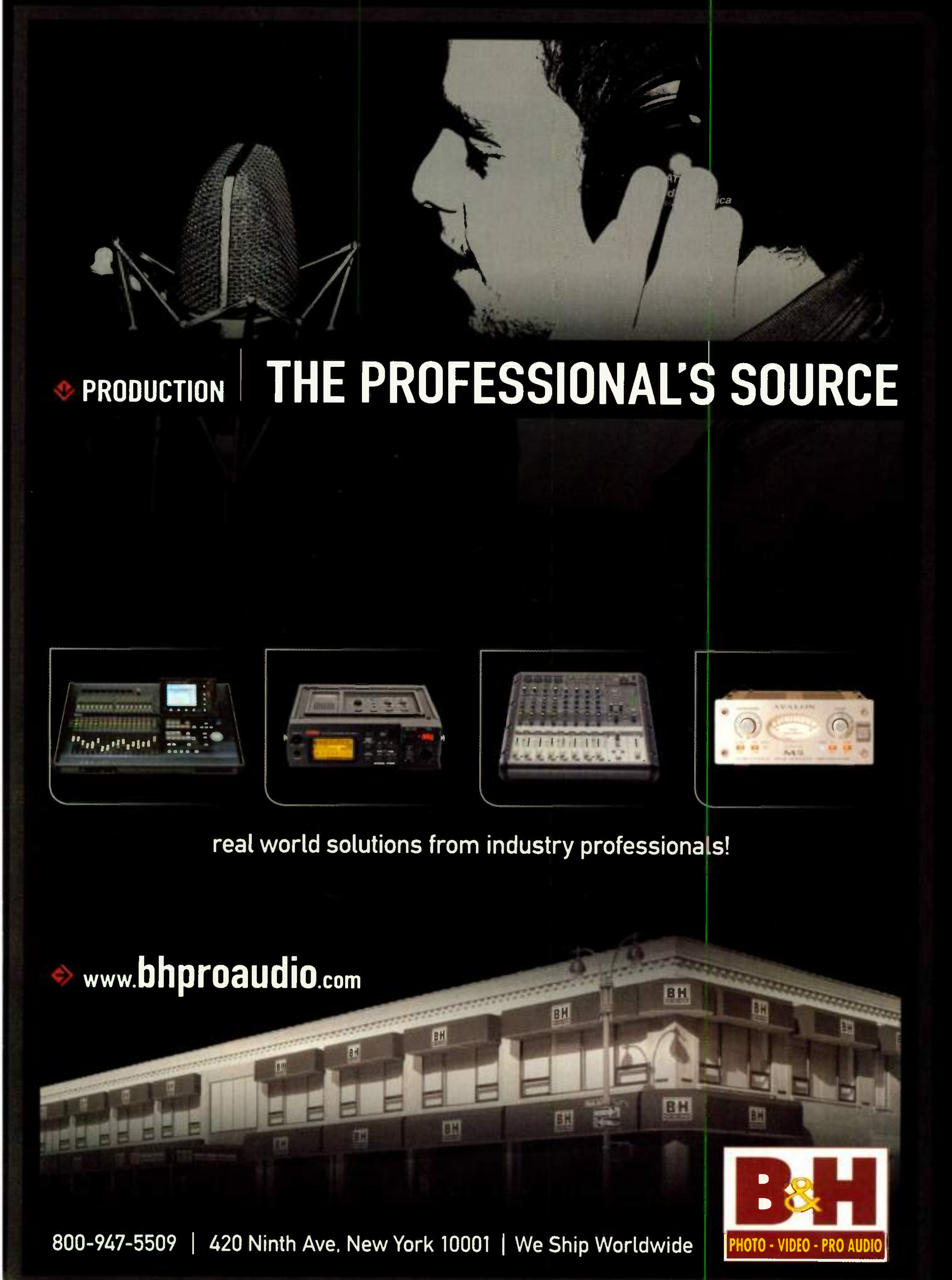
You can use Impulse's individual slot parameters to add variation to your slices; Transpose and Stretch work especially well. Use them with the Mode switch to control the amount of pitch-, timbre-, and time-shifting. Using a notch filter with a high Random setting is another good option.

You can apply effects plug-ins in various ways. You can insert plug-ins after the Impulses in the rack to apply them to all slots, but I prefer to create additional Live tracks, route a whole Impulse or individual Impulse slots to those tracks, and apply insert or send effects on the added tracks.

Beat Repeat is one of my favorite Live effects to use for processing percussion. If the slices are evenly spaced in time, setting the Interval knob to the slice length will cause each slice to be processed with the probability set by the Chance knob. The effect is determined by the remaining controls. Try midsize Variation settings in Auto mode and experiment with the filter and Pitch Decay settings. If Beat Repeat is a send effect, use its Gate mode; otherwise, use Mix or Ins mode, respectively, for a more- or a less-dense effect.

Grain Delay and Auto Filter are good to use when applied to individual slots. Their effects can be radical, but with each Impulse getting only 25 percent of the action, the effects are used sparingly. Don't overlook each effect's *x-y* Controller; that's a great way to adjust the effects in real time. With Auto Filter, use the Phase, Offset, and Spin controls to affect the filtering relationship between the stereo channels. Also note that the Quantize Beat setting determines how often the modulation updates and is independent of the modulation rate, even when that is set in note increments (see **Web Clip 2**). EM

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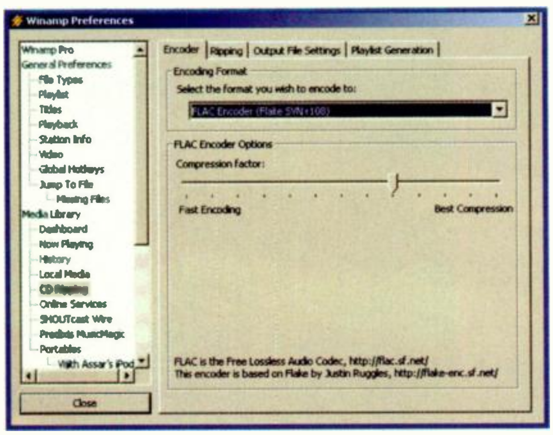

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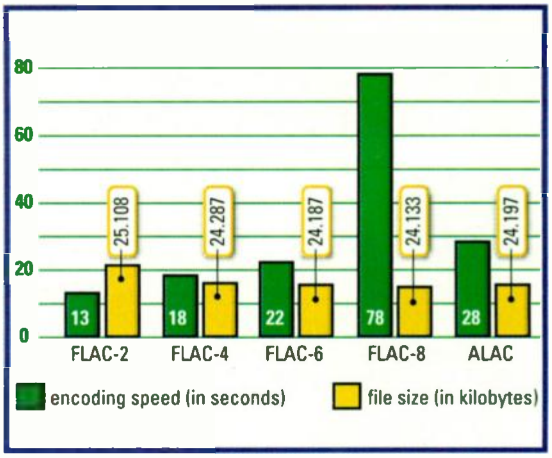
Even though FLAC has the most widespread support, Apple's Lossless Audio Codec (ALAC), a proprietary format introduced in a 2004 update to iTunes, has the greatest market share. Support in popular consumer packages such as iTunes, QuickTime, and the iPod has propelled ALAC to the forefront of lossless digital audio. Indeed, to put iTunes into high-definition overdrive, just choose Apple Lossless as the audio format in the program preferences; there are no compression options as there are with FLAC. Note that Apple lossless files share the .m4a file extension with AAC, a lossy compression format also favored by Apple.

Who, What, Where

Like most modern media formats, lossless codecs usually include robust support for *tagging*, a method of storing additional information about the audio (*metadata*) inside the file. The most useful metadata fields are the basics (song name, album title, release date, and so on), but the tag can also hold more-esoteric elements such as URLs, pictures, and even more audio. In fact, because most lossless codecs were developed well after MP3, they often correct some of the flaws of the metadata implementations used by MP3. For example, newer formats often allow custom field names and multiple values for a single field. Storing large audio archives in metadata-rich formats makes organization a breeze. Instead of agonizing about whether to put a TR-808 bass-drum sample into the Drums folder or into the Hip-Hop folder, you can just enter both terms into the metadata and have the file show up in a search for either.

Using lossless files can be a savvy alternative to MP3 files, especially if you're trying to promote your studio's high production values or pristine sound quality. ALAC is particularly useful for these purposes, because it will play back seamlessly on any computer that has QuickTime or iTunes. Downloading files may take longer than with MP3, but for your listeners to get the full experience of hearing your high-resolution mixes, it's important to give them the option of listening to lossless files. ALAC's increasing popularity is perhaps best demonstrated by Apple's AirTunes, a popular home-audio streaming product that wirelessly connects your computer to your home stereo using an AirPort Express router. AirTunes will automatically transmit all audio in ALAC format to avoid bandwidth hiccups.

Similar bandwidth savings could be realized in long-distance collaborative projects carried out over the Web. But before sending a large batch of files to a production buddy, make sure that he or she has a way to access or decompress them. At the moment, lossless audio compression formats are not well supported by your average DAW. Apple Logic supports saving as ALAC from the File Export menu, and the open-source program Audacity, originally developed by Dominic Mazzoni and Roger Dannenberg, does the same with FLAC. More importantly, Ableton Live has supported FLAC import since version 5. Considering the impressive metadata capabilities of that format, it's safe to say that a carefully tagged sample library in FLAC format would be a tremendous addition to any Live-based studio environment.



CHUCK DAHMER

FIG. 2: Very high compression levels take much longer to encode but produce only a minimal decrease in file size.

The best support, however, comes from the fledgling sequencer Cockos Incorporated Reaper, which can import, export, and even record to FLAC. Users of other software will want to keep their session audio in their program's native format but should consider storing exports, mixes, and final products using lossless compression because of the metadata management possibilities. All the necessary conversions can be handled by power tools like iTunes, illustrate's dBpowerAmp, and Peter Pawlowski's foobar2000.

Transmission Complete

MP3 format will be with us for some time to come; even the multitude of technically superior competing lossy formats hasn't managed to unseat it. But for audiophiles and professionals alike, lossless is the only sensible option at this point. Large hard drives are cheap enough to allow storage of lossless audio in ways that even five years ago weren't financially sensible. What's more, lossless codecs offer complete format portability: because no data is ever discarded, there's nothing to stop you from converting to new formats as many times as you want, and you can always go back to uncompressed WAV or AIFF if needed.

As media technology expands, new technologies are developed and new formats emerge. But in the audio world, the past ten years have brought about a significant decrease in quality. For those surfing the bleeding-edge of audio, lossless audio compression can not only be a powerful tool but it also puts us one step closer to restoring widespread use of high-resolution audio. EM

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Vijith Assar works at the Music Resource Center in Charlottesville, Virginia.

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Q&A: Mark Bright & Stephen J. Finfer

By Michael Cooper

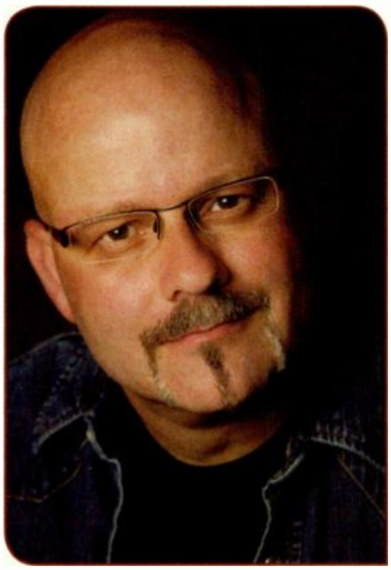
How to pitch your songs to industry insiders.

For many songwriters, the business side of music publishing is a curse; they would rather focus on the creative aspects of their jobs and leave the marketing of their songs to the "suits." But for others, pitching their own songs is empowering. After all, nobody is going to believe in your music and look after your interests as much as you are. So why not take control?

Perhaps the biggest challenge for songwriters trying to gain a foothold in the music-publishing business is obtaining contacts with the industry's decision makers (including record producers and labels' artist and repertoire staff) and learning how to most effectively pitch their songs. To find out how it's done, I interviewed two industry veterans who are in different markets and on opposite sides of the table: one pitches songs, and the other solicits them.

Mark Bright (see **Fig. 1**) is both a famed record producer and a publishing-company executive. The Nashville-based country-music icon has produced hits for Cowboy Crush, Sara Evans, Rascal Flatts, Lonestar, Jo Dee Messina, Jamie O'Neal, Carrie Underwood, and many others. Formerly the vice president of EMI Music Publishing, Bright went on to cofound Teracel Music (since acquired by Dimensional Music Publishing). He currently owns Mark Bright Productions and My Good Girl Music, the latter of which is a publishing coventure with Sony ATV Music.

FIG. 1: Mark Bright is one of the most sought-after producers in country music today.



TONY PHIPPS

Stephen J. Finfer (see **Fig. 2**) is co-owner and chief operating officer of Hollywood-based Arthouse Entertainment (www.arthouseent.com), a parent company to the Art for Art's Sake (ASCAP) and Art in the Fodder (BMI) music-publishing companies. Arthouse also includes production and management operations. In terms of publishing market share for the pop charts, Arthouse Entertainment ranked in the Top 10 throughout 2006. Finfer previously held prominent positions at TVT Music Publishing, Famous Music Publishing, and Universal Music Publishing.

In the music business, success comes only after making the right connections. How does an unknown songwriter get his or her foot in the door in order to pitch songs to a producer or an A&R person?

Bright: It's really important to focus on affiliations. Our business is built on relationships and networking. It's always going to be more difficult, and in some cases almost impossible, to send an unsolicited song through the mail and have somebody hear it [if for no other reason than] the legal repercussions that can come from that.

Finfer: That stack [of unsolicited songs] builds and builds until they get a truck and throw it out—because that's the lawsuit bin. It's one in a million [great songs] out of that bin, but five out of that million will sue you [on fraudulent copyright-infringement charges]. So it's not a bin, you really want to go into. What you need to do is find a filter, someone who's going to say, "Listen to this guy." Those people are lawyers, managers, agents, and friends of A&R people. When it comes from a lawyer or agent, I'm going to listen to it.

Are referrals from writer- or publisher-relations staffers at the performing-rights societies (ASCAP, BMI, and SESAC) helpful?

Finfer: Yes! Any music-industry professional works.

Establishing initial contacts is only half the battle. Considering that most producers and A&R people are too busy to answer many of their phone calls and emails, how do you find out who's looking for songs to record?

Finfer: All major publishers prepare an internal "who's looking" list. If you have a friend who's signed to a major publisher, ask them for a copy of it.

Bright: The information on those sheets is gleaned from personal conversations with A&R people at the top of the pecking order at record labels.

Finfer: There are also many services that you can pay for that will give you briefs [tip sheets] on a weekly or monthly basis. Many new writers use them.

REVIEWS

its toll on the FireWire bus. SSL recommends dedicating a FireWire bus to the Duende, but I say it's a necessity. Note that many computers have multiple built-in FireWire ports that often are all on the same bus, so it is likely you will need to add a FireWire card, which is not expensive but does take up one of the slots in your computer. In certain situations, there can also be some pesky interactions between the Duende and your other computer hardware (more on this later).

A Fine Pair

The Duende comes with only two plug-ins: Channel and Bus Compressor. SSL plans to develop and release additional plug-ins in the future, the first being the DrumStrip drum processor ($299), which should be available by the time you read this.

Channel re-creates the classic SSL 4000 channel strip, with a 4-band equalizer, highpass and lowpass filters, a compressor, and an expander/gate (see **Fig. 1**). (SSL says the EQ and dynamics are taken from the XL 9000 K-series console, but it sure looks like the 4000 to me.) Bus Compressor is modeled after the 4000-series master bus compressor, renowned as one of those audio devices that just make things sound better coming out than they sounded going in (see **Fig. 2**).

Both the EQ and the dynamics in the Channel plug-in have a few tricks up their sleeve, nearly all of which are based on the original channel strip. The EQ has two parametric bands and two bands of shelving EQ, but the Bell button on the shelving bands makes them into peaking filters. The 4000 consoles had two primary incarnations: the E- and the G-series. One difference between them was that the E-series EQ had no overshoot, making it more surgical, whereas the G-series had overshoot and undershoot that made it more musical, if less precise. The E button in the Duende plug-in's interface allows you to switch between these two response types—a very nice touch.

The control paths, or sidechains, of Channel's compressor/limiter and expander/gate are in parallel, with the two control signals applied to a single gain element, which presents some interesting processing possibilities. The compressor/limiter has switch-selectable peak or RMS detection, and both parts of the dynamics section have autosensing attack times.

FIG. 1: Channel's basic settings are very easy to grasp, and a variety of internal routing options are available.

Channel offers an almost complete matrix of routing options: filters before or after EQ, dynamics before or after EQ, filters and/or EQ in the dynamics sidechain, and so forth. Input and output level controls and an input polarity reverse switch complete the module.

The plug-in's interface replicates the 4000's layout and look, which unfortunately is as constraining as it is understandable. For example, Channel reproduces the 4000's 6-LED ladder meters, which were always just a bit better than nothing at all, rather than taking advantage of the computer's graphics capability to show more-informative meters.

FIG. 2: Bus Compressor is modeled on the 4000 console's master bus compressor, famed for its ability to pull together mix elements.

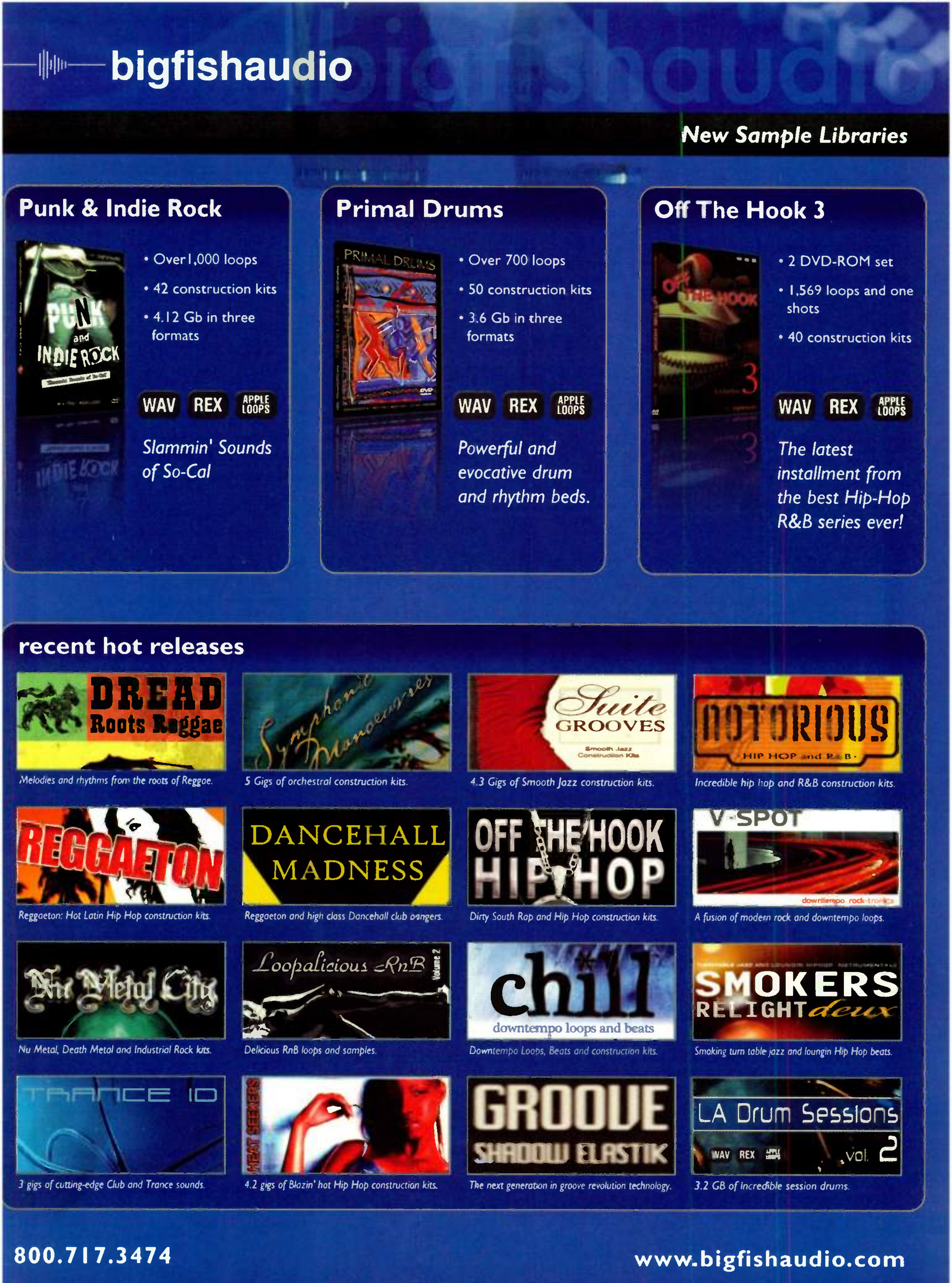
Bus Compressor is a totally straightforward compressor, although with stepped attack, release, and ratio controls, and with only three ratios to choose from (2:1, 4:1, and 20:1). It also has an interface that replicates the original, providing a familiar look that is of dubious ergonomics in the digital realm. In the case of Bus Compressor, the issue is the stepped switches, which are changed by clicking-and-dragging—an unnatural and imprecise method.

Both Channel and Bus Compressor can be instantiated as mono or stereo plug-ins. The Duende supports up to 32 instantiations.

Speed Bumps

Getting the Duende operational was not as easy as I would have liked, though I couldn't call it a nightmare. When I received the device, Intel Mac support was just being released. This required me to update both the firmware for the processing module and the software that runs inside MOTU Digital Performer (DP), my digital audio sequencer. The updates had to be performed in the right order, but at first the Duende seemed reluctant to execute them properly. After a few attempts, I finally got both updates done (firmware 1.15 and driver version 1.24) and started to work with the Duende inside DP.

I quickly realized I could not keep my FireWire drives running while the Duende was online, because I do not have a FireWire card and my 3 GHz quad-core Mac Pro Intel Xeon machine has only one FireWire bus for the four ports on it. For a while, I kept the Duende window open all the time as I explored the product's features.



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Tip sheets can sometimes be woefully short on details, and artists sometimes change their musical direction from one album to the next. How do you know if the songs you're pitching are on target?

Bright: As a publisher, we got a couple of songs on Faith Hill's *Breathe* album [Warner Bros., 1999], and the way we did it was by doing homework and trying to look at what she was saying in the press and on fan sites. In interviews that Faith was doing toward the end of the project, she was saying, "I'm going to be more contemporary this time out." We publishers and songwriters have to be better at the marketing side of being creative. The more-successful songwriters are those that tend to be a lot more savvy at figuring that stuff out. Go to the artist's fan site and blogs and see what they're saying right in that moment.

Finfer: You've got to get the tip sheet [so you aren't] shooting in the dark. Artists these days are changing their colors from record to record. Someone who made a rock record last time could now be making a country record. Was Kelly Clarkson going to make an urban record? She made a rock record. If you can't find anything else, you guess that you need to make the new version of the last record, the contemporized version of what was on the last record.

NATALIE AVAKIAN

FIG. 2: Stephen J. Finfer manages the day-to-day operations of Arthouse Entertainment, a busy pop publishing company.

So if details on the new project's direction aren't available, you should stick closely to the previous album's direction?

Bright: Yes. Don't try to become an A&R person yourself and think [that] maybe they're going to go off in this wild direction. Stick closely to what they've done before. You need to listen to the last album and try to cast the song as best you can. That's extremely important.

MySpace is a great research tool in that way. Even major-label artists post some of their recent songs there.

Bright: It's made it a lot easier, I agree.

Okay, suppose that you pitch your song and the producer or A&R rep passes on it. How many times can you pitch that same song to them, but for a different artist or project, before it wears out its welcome?

Bright: There isn't a statute of limitations. If you've gotten [an] indication that an A&R person or record producer likes that song, and then a new artist comes up and you feel like it's the right pitch, you pitch it. It gets down to casting it well for the next artist and plugging it again. "Bless the Broken Road" megahit for Rascal Flatts] is a case in point. I had pitched that song to eight or nine acts along the way. It took me until Rascal Flatts's third album to finally get that to work.

Finfer: It's really if the A&R person likes it or they don't, not how many times you've pitched it. If they like it, I've had people hold on to songs for years waiting for the right artist. They think it's a hit, but the artist didn't like it or it wasn't the right artist for the song.

But you'd better be sure they liked the song in the first place before pitching it to them again.

Finfer: If you keep coming back to an A&R guy with a song they don't like, you're going to lose your credibility with that person and it's going to overflow to the other things you pitch. You have to not waste their time. You've got to find what they like so you know they're not going, "Uh! Here it is again." You also don't want to be the guy who's sending the same song all the time. You want to always be coming with something new and fresh.

Any final thoughts?

Finfer: If you get that opportunity to have somebody listen, you better have your production together, not just your song. We're in a world now where everyone has the best gear you can have for very little money. It's a lot to ask people to hear just songs. In my world, people are more likely to react to great productions than to songs these days. You can have a great song and play it on a guitar-vocal or piano-vocal demo—that's going to be tough. If you've got a reactive, dynamic production that screams out at you, you'd be surprised by how things can find their way.

Bright: I don't really care what writer writes the song. I don't ask first who wrote the song, ever. And you'll find that most record producers in this town really don't care who wrote the song. I just came off of a big hit with Carrie Underwood: "Before He Cheats." It was written by two writers who had never had a hit before. EM

EM contributing editor Michael Cooper is the owner of Michael Cooper Music. Some of his songs are posted at www.myspace.com/michaelcooperRecording.

REVIEWS

SOLID STATE LOGIC Duende (Mac/Win)

Hardware-accelerated plug-ins with the SSL sound.

By Larry the O

Although founded to make solid-state control systems for pipe organs, Solid State Logic (SSL) became widely known for its 4000-series large-format mixing consoles in the late '70s. As the first consoles to include dynamics processing on every channel, and boasting a pioneering automation system, the 4000-series became a studio standard, familiar both sonically and operationally to legions of engineers (myself included).

A great many SSL products have been built on the legacy of the 4000-series. The Duende DSP host is one of these products, though its technology was actually lifted from the company's C200 digital production console.

As the market for large-format music mixing consoles has waned, SSL has branched out into broadcast and postproduction consoles and, most recently, the project studio market. The Duende is intended to bring the famous sound and features of an SSL 4000-series channel strip within reach of the project studio market, and it does so quite well.

The SSL Duende is a DSP plug-in host that connects to your computer via FireWire.

Thinking Outside the Box

In order to run the C-series algorithms, the Duende appears in a digital audio sequencer as if it were a software plug-in, following the recent trend of hardware DSP hosts such as the Waves APA, the TC Electronic PowerCore, and the Universal Audio UAD-1 card. The Duende is a 1U device that communicates with a computer over a FireWire 400 port. It supports VST, AU, and RTAS plug-in formats, at sampling rates up to 96 kHz. The front panel has only an AC power switch, while the back panel has two FireWire ports and an input for DC power from the wall-wart power supply.

Because the Duende takes care of the 40-bit floating-point digital signal processing, the minimum system requirements for the host system are not steep: a 1 GHz Pentium 4 or AMD computer running Windows XP SP2, or a 933 MHz Macintosh G4 running Mac OS X 10.4.8. The Duende needs 80 MB of hard-drive space.

Although the device goes easy on your CPU, it takes

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

its toll on the FireWire bus. SSL recommends dedicating a FireWire bus to the Duende, but I say it's a necessity. Note that many computers have multiple built-in FireWire ports that often are all on the same bus, so it is likely you will need to add a FireWire card, which is not expensive but does take up one of the slots in your computer. In certain situations, there can also be some pesky interactions between the Duende and your other computer hardware (more on this later).

A Fine Pair

The Duende comes with only two plug-ins: Channel and Bus Compressor. SSL plans to develop and release additional plug-ins in the future, the first being the DrumStrip drum processor ($299), which should be available by the time you read this.

Channel re-creates the classic SSL 4000 channel strip, with a 4-band equalizer, highpass and lowpass filters, a compressor, and an expander/gate (see Fig. 1). (SSL says the EQ and dynamics are taken from the XL 9000 K-series console, but it sure looks like the 4000 to me.) Bus Compressor is modeled after the 4000-series master bus compressor, renowned as one of those audio devices that just make things sound better coming out than they sounded going in (see Fig. 2).

Both the EQ and the dynamics in the Channel plug-in have a few tricks up their sleeve, nearly all of which are based on the original channel strip. The EQ has two parametric bands and two bands of shelving EQ, but the Bell button on the shelving bands makes them into peaking filters. The 4000 consoles had two primary incarnations: the E- and the G-series. One difference between them was that the E-series EQ had no overshoot, making it more surgical, whereas the G-series had overshoot and undershoot that made it more musical, if less precise. The E button in the Duende plug-in's interface allows you to switch between these two response types—a very nice touch.

The control paths, or sidechains, of Channel's compressor/limiter and expander/gate are in parallel, with the two control signals applied to a single gain element, which presents some interesting processing possibilities. The compressor/limiter has switch-selectable peak or RMS detection, and both parts of the dynamics section have autosensing attack times.

Channel offers an almost complete matrix of routing options: filters before or after EQ, dynamics before or after EQ, filters and/or EQ in the dynamics sidechain, and so forth. Input and output level controls and an input polarity reverse switch complete the module.

The plug-in's interface replicates the 4000's layout and look, which unfortunately is as constraining as it is understandable. For example, Channel reproduces the 4000's 6-LED ladder meters, which were always just a bit better than nothing at all, rather than taking advantage of the computer's graphics capability to show more-informative meters.

FIG. 2: Bus Compressor is modeled on the 4000 console's master bus compressor, famed for its ability to pull together mix elements.

FIG. 1: Channel's basic settings are very easy to grasp, and a variety of internal routing options are available.

Bus Compressor is a totally straightforward compressor, although with stepped attack, release, and ratio controls, and with only three ratios to choose from (2:1, 4:1, and 20:1). It also has an interface that replicates the original, providing a familiar look that is of dubious ergonomics in the digital realm. In the case of Bus Compressor, the issue is the stepped switches, which are changed by clicking-and-dragging—an unnatural and imprecise method.

Both Channel and Bus Compressor can be instantiated as mono or stereo plug-ins. The Duende supports up to 32 instantiations.

Speed Bumps

Getting the Duende operational was not as easy as I would have liked, though I couldn't call it a nightmare. When I received the device, Intel Mac support was just being released. This required me to update both the firmware for the processing module and the software that runs inside MOTU Digital Performer (DP), my digital audio sequencer. The updates had to be performed in the right order, but at first the Duende seemed reluctant to execute them properly. After a few attempts, I finally got both updates done (firmware 1.15 and driver version 1.24) and started to work with the Duende inside DP.

I quickly realized I could not keep my FireWire drives running while the Duende was online, because I do not have a FireWire card and my 3 GHz quad-core Mac Pro Intel Xeon machine has only one FireWire bus for the four ports on it. For a while, I kept the Duende window open all the time as I explored the product's features.

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When I finally closed the Duende window, DP started stuttering horribly. This was an entirely consistent problem. It took a few exchanges with SSL technical support (which was quite prompt, pleasant, and professional in our email exchanges) before I understood that the problem could be remedied by selecting Always Run In Real Time from the contextual menu in the plug-in window. I'd never seen this command in DP before. It is designed to allow prerendering for DSP-intensive plug-ins, which isn't possible with external DSP hardware.

Selecting that command fixed the problem. But every time I instantiated a Duende plug-in, I had to remember to select Always Run In Real Time or else I'd suffer the consequences. SSL says that this problem is unique to DP and does not occur with any other host.

Even after I made the proper menu selection, several times DP went into an odd mode where tracks played back out of sync. Deinstantiating a Duende plug-in seemed to fix the problem; I would then reinstantiate the plug-in and everything would play okay. I suspect (but can't prove) that this is related to communication glitches between DP and the Duende.

The Sweet Sound of Success

Having more or less tamed all the beasts, I got down to working with the Duende and enjoyed myself quite a bit. While I'd always found the 4000 E-series EQ serviceable (I've had little experience with the G-series EQ), it had never flipped my wig. I found the Duende's G-series EQ much more engaging than the E-series, and I started using it extensively (**see Web Clips 1 through 3**).

I found myself reaching for Channel often, even when I already had other plug-ins on a track. It didn't meet every need—sometimes I needed more bands of EQ or a different compressor characteristic, for instance—but Channel frequently made me very happy. I liked the EQ, especially in the low end, and the compressor usually

> PRODUCT SUMMARY
>
> **SOLID STATE LOGIC Duende**
>
> DSP host and plug-ins
> $1,995
>
> | | |
> |---|---|
> | FEATURES | 3 |
> | EASE OF USE | 3 |
> | AUDIO QUALITY | 4 |
> | VALUE | 4 |
>
> RATING PRODUCTS FROM 1 TO 5
>
> **PROS:** Famous SSL sound, including both E- and G-series EQ. Up to 32 instantiations. Plug-in setup is straightforward.
>
> **CONS:** Only two plug-ins. Needs dedicated FireWire bus. Some problematic interactions when Digital Performer is host. No factory presets.
>
> **MANUFACTURER**
> Solid State Logic
> www.solid-state-logic.com

worked well to tame dynamics problems. It was also very nice to be able to instantiate Channel in stereo, which you could not do on the 4000 consoles.

Bus Compressor, on the other hand, is magic. It's not a gain maximizer, it's not quite a leveler, and it's not really transparent. In fact, it combines a bit of each of these features to create an effect that pulls together all of the mix elements and makes them sit together well (see **Web Clips 4 and 5**).

The great part of having it as a plug-in was that I didn't just use it across the stereo bus; I often used it on individual tracks. It did a beautiful job of combining several background vocals into one creamy layer. Bus Compressor also made a crunchy electric guitar track feel like the instrument had been really loud when it was recorded. On bass, it just smoothed out the peaky notes. There are many different kinds of compressor sounds, and no compressor works for everything, but Bus Compressor is a very valuable tool to have for a variety of applications.

I often use more than one compressor on a track that really needs dynamics management, and I was interested to note that both Channel and Bus Compressor worked excellently in that situation. Again, you can't throw any two compressors on a track and get that well-managed sound without lots of pumping, but the Duende's compressors seemed to play well with others nearly all the time.

Although you can save your plug-in settings as presets, no presets are included with the Duende. Granted, the EQ and dynamics processing are pretty straightforward and not too hard to set up, but it would be helpful to have some of the classic sounds as jumping-off points.

SSL Authenticity

At $1,995, the Duende is at the high end of DSP hosts, especially when you consider that it includes only two plug-ins. On the other hand, Channel has a lot of functionality, both plug-ins sound great, and 32 instantiations is about what you would have gotten on a 4000 console, and more than you can get out of some other plug-in hosts.

Although the price of entry may be daunting to some, for those who really want the sound of an SSL console's dynamics and EQ, the Duende is the most affordable way to get it. Keep in mind that you will need to add a FireWire bus (unless you have a machine that actually has more than one bus and can afford to dedicate it to the Duende). But the result will be a classic flavor with which to season your music. EM WEB CLIPS

Larry the O is finally ready to throw away his 8-inch floppy disks of SSL automation sessions.

FIG. 1: The PreSonus DigiMax FS provides front-panel access to its input connectors, gain controls, and sync options.

PRESONUS DigiMax FS

A multichannel mic pre and Lightpipe master clock.

By Rusty Cutchin

See Product Specs @emusician.com

PreSonus has channeled its knack for designing preamps and handling digital conversion into the DigiMax FS, an 8-channel mic preamp with some interesting sync capabilities. It doesn't interface with a computer directly—there are no USB or FireWire ports—but if you have an interface with ADAT Lightpipe connections, such as Digidesign's 002 digital mixer, the DigiMax FS can serve as a high-quality expander for more mic inputs.

On top of that, the DigiMax FS has extensive analog insert and output options, making it very useful as an independent mic pre that can feed an audio interface in a budget home studio. For the semipro or pro studio with unused Lightpipe-equipped devices, the DigiMax FS can serve as a high-quality clock source.

Combo Deal

The DigiMax FS provides eight Class A preamps, which you access through Neutrik combo connectors on the front panel (see **Fig. 1**). Although some users may not care for the idea of a 1U mic pre with all its XLR connections on the front panel, the multitude of rear-panel jacks makes it obvious why this design was necessary. Users who install the DigiMax FS in a rack will probably want to keep at least a 1U space free above the unit for mic cables entering the rack from the rear. (There are no controls directly above or below the inputs, so cables don't get in the way when you're operating the unit.)

Inputs 1 and 2 are configured for instrument level on the connector's TRS jack. The rest of the ¼-inch inputs are set for line level. (You can use the rear-panel insert jacks to feed line-level signals to inputs 1 and 2.) To the left of the input connectors are two rear-illuminated buttons that let you assign phantom power to inputs 1 to 4 and inputs 5 to 8.

To the right of the input connectors are gain pots for the eight inputs. The pots have markings for both mic and line gain ranges (0 to 60 dB and –20 to +20 dB, respectively). The controls are staggered, which makes them easier to grasp and operate. LED clipping indicators sit to the side of each pot.

At the right edge of the front panel are a power button and two buttons for setting up the unit's sync configuration. An Internal Clock button cycles between 44.1, 48, 88.2, and 96 kHz settings. An External Sync button changes color when pressed: red for ADAT sync and blue for synchronizing via the unit's rear-panel BNC connectors. Sync is internal when the button isn't illuminated.

To the Rear and Out

PreSonus has provided a wealth of analog outputs on the DigiMax FS's rear panel (see **Fig. 2**). For starters, there are eight TRS outputs marked DAC that are fed directly from the Lightpipe inputs. Because all connections are active at all times, you could use these to, for example, process Lightpipe tracks (say, from an archived ADAT

tape) through analog effects devices while also passing the original digital versions to a separate DAW.

FIG. 2: A multitude of analog outputs, including preconverter insert points, makes the DigiMax FS a serious front end for studios with ADAT Lightpipe connections.

Another section of the DigiMax FS's rear panel supplies eight direct outputs from the unit's input channels. A set of eight insert points (ring equals send) completes the analog output section. This is the perfect way to add compression or other effects to signals from mics or instruments before the signals are digitized.

On the left of the rear panel are BNC in and out connectors with a 75Ω termination switch, which can be engaged on the final DigiMax FS in a chain. You can add as many of the units as you like to create additional input and conversion capabilities.

Alongside the BNC section is the 96 kHz ADAT-SMUX section, which provides four Toslink connectors, two each for input and output. Standard ADAT devices (up to 24-bit, 48 kHz resolution) use the first connector in each section for all eight channels of input or output, whereas units capable of higher sampling rates and with SMUX implementation use the first connector for channels 1 through 4 and the second for channels 5 through 8 in each section.

Sync City

What should make the DigiMax FS attractive to studio owners with older Lightpipe-equipped devices is PreSonus's new JetPLL technology (*Jet* stands for Jitter Elimination Technology). JetPLL was developed by TC Applied Technologies (a division of TC Electronic), and the technology is part of the DICE II chip, which is used in TC Electronic's Konnekt 24D and PreSonus's new FireStudio audio interfaces.

The promise of JetPLL is that any incoming digital source will be adjusted to the same jitter spec as the device with JetPLL, no matter which unit is set to be the master clock. This should also provide better performance from the unit's converters, along with faster locking and rock-solid synchronization. Owners of units like Digidesign's 002 and 002 Rack, Yamaha's DM-series digital mixers, and RME's recording interfaces, as well as of older devices with early Lightpipe implementations, will benefit from reduced jitter.

In Session

I have been a fan of PreSonus's economical yet high-quality mic preamps for years but had not tried out one of its multichannel units. It's been a long time since my studio used any Lightpipe-equipped devices, but I have been doing some old-school analog processing and dumping tracks into MOTU Digital Performer 5 on my dual-processor Power Mac G5. I connected the direct outs of the DigiMax FS to my interface, a FireWire-equipped Mackie Onyx 1620 mixer. I used the aux ins of the 1620 to avoid problems with gain staging that could have occurred had I plugged the analog outs into the Mackie preamps.

The DigiMax FS preamps sounded very clean, especially with large-diaphragm condenser mics, and they gave me plenty of headroom on all channels. The instrument inputs worked well with several guitars I plugged into them, and the unit's insert points were able to drive a couple of different guitar processors without introducing audible distortion. Without even testing the unit's DAC or sync capabilities, I was very impressed with the DigiMax FS's audio quality and PreSonus's consistently good approach to preamp design.

For further testing, I took the DigiMax FS to a colleague's professional studio, where he had occasionally used a Digi 002 rack unit in a composing room for writing and recording projects when his extensive Pro Tools HD rig wasn't needed. We set up the DigiMax FS as the master clock for a session feeding instrument mics to a Pro Tools LE project. The DigiMax FS found sync faster than any external unit I'd seen set to master clock and driving a digital mixer. Best of all, there were no surprise noises or pops as we experimented with different bit and sampling rates. The DigiMax FS worked like a charm in this situation, and its converters sounded very, very good at high sampling rates with quality mics on instruments like acoustic guitar and solo violin.

PRODUCT SUMMARY

PRESONUS DigiMax FS

multichannel mic preamp
$799

FEATURES	4
EASE OF USE	4
AUDIO QUALITY	4
VALUE	4

RATING PRODUCTS FROM 1 TO 5

PROS: Very good mic preamps. Multiple analog outputs. Preconverter insert points. Exceptional clock-sync technology.

CONS: Front-panel mic connectors.

MANUFACTURER
PreSonus
www.presonus.com

To the Max

Although the average home recordist will probably be more interested in one of PreSonus's multichannel audio interfaces for all-in-one convenience, the studio owner who wants to make use of unused Lightpipe connections to add extra channels to a DAW (or even a full analog) recording setup should give a listen to the DigiMax FS. Its mic preamps alone make it a solid, midpriced front end for such systems, and its converters will probably outperform all but the highest-quality vintage Lightpipe units (like the ones that recorded all those ADAT tapes you haven't transferred yet). If Lightpipe is still your plumbing of choice, the DigiMax FS is an excellent way to beef up the digital signals flowing through the tubes.

Rusty Cutchin is a former editor of EM and a producer, engineer, and music journalist in the New York City area.

FIG. 1: Play your effects (or your soft synths) with the KP3's 8 × 8 light-up display. A trail of LEDs follows your finger to show the last value. The four sample pads trigger one-shots and unmute loops.

KORG KP3 Kaoss Pad

Light up your music.

By David Battino

Variety is what makes life—and music—interesting. In the third version of its Kaoss Pad, Korg has made it easier than ever to warp your music in ear-pleasing ways. And not only is the KP3 Kaoss Pad (KP3) a hands-on effects processor, it's also a surprisingly deep MIDI controller.

The KP3 is an eye pleaser too, thanks to its illuminated touch pad and buttons (see **Fig. 1**). Whereas the KP2 changed color based on the position of your finger, the KP3 turns the touch pad itself into a 64-pixel display. Stroke the pad, and a trail of fading red lights follows your fingertip like fairy dust. Hit the Hold button, and the LED under the last point you touched remains lit so you can see where you left off. Change programs, and the name of the new effect scrolls by like a message on a bank clock. Switch to External Controller mode, and the eight columns of LEDs can emulate eight faders. It's both slick and useful.

Korg upgraded the sampling as well, adding two more sample-playback pads (for a total of four), resampling, an SD memory card slot, and USB sample and parameter transfer. And because effects are the heart of the Kaoss Pad series, the KP3 got a fresh crop of unusual new ones.

See Product Specs @emusician.com

8 × 8 State

After reviewing a parade of plastic gear over the last few years, I was pleasantly surprised when I pulled the KP3 from its box. It's bigger than it looks in photos—with cables connected, the KP3 is about the size of a copy of EM. Looking like Darth Vader's track pad, it sports a sturdy metal case with enough heft (almost 3 pounds) to keep it from sliding around. Unfortunately, the knobs are attached directly to the circuit board rather than being bolted to the case, exposing the ones on the front to potential damage. The headphone level knob scraped against the housing until I gently bent it back.

The star of the front panel is the *x*-*y* touch pad, which is about 3 inches wide by 3.75 inches tall and has a lovely, smooth feel. It senses the lightest touch of your finger, unlike most other track pads, which make you grind down.

Although the LED grid beneath the pad divides it into eight zones vertically and horizontally, the pad actually transmits 128 values in each direction over MIDI, and at least that many to the internal effects parameters. Pad sensitivity varies with the program that's loaded. Some of the synthesizer programs span an octave from left to right; others cover several octaves (see **Web Clip 1**). With the drum groove programs, sliding your finger

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For years, engineers have been drawn to the sound of SSL. There's certain magic in the circuitry that adds a sonic character that could only be found in SSL consoles — until now. With AlphaChannel and Duende, the classic SSL sound that has graced popular music for 40 years is now available to the legions of DAW users worldwide.

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Alpha Channel

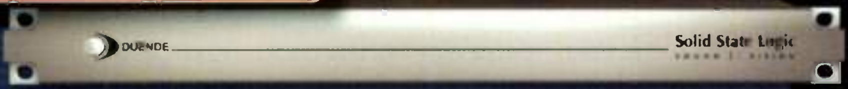
A great recording starts with a great front end, and the AlphaChannel is the perfect lead-in for your DAW. The studio-grade mic preamp boasts "Variable Harmonic Drive" along with SSL parametric EQ and limiting for the ultimate fidelity for DAW-based studios. It's literally like you pried a channel from an SSL board, minus the fader.

Solid State Logic
SOUND || VISION

SSL Alpha Channel at a Glance:

- SSL analog processing for your DAW
- SSL mic preamp with 3-band EQ, Filter, and Limiter
- Variable Harmonic Drive circuit
- Analog or digital outputs direct to your DAW
- Intelligent auto compression for maximum headroom

Duende



After you've laid down tracks with the AlphaChannel, do your processing with SSL as well — with no load on your CPU. Duende conjures its own brand of magic with EQ and dynamics ripped straight from the C200 console. Plus, you get the world-famous Stereo Bus Compression that engineers have been using for years to "Stick a mix like audio glue!"

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SSL Duende at a Glance:

- Classic SSL sound in a plug-in
- DSP-accelerated processing lightens load on CPU
- EQ, dynamics, and stereo bus compression
- FireWire connectivity
- VST/AU (RTAS available with included wrapper)

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from left to right switches among eight patterns; with the filter programs, you get a smooth sweep.

In addition to sensing fingertip position, the pad also detects taps, which you can assign to MIDI events such as Note On commands. It doesn't, however, detect Velocity or Pressure.

Knobs and Buttons

Below the touch pad are four Sample Bank buttons, which light up in green when assigned to loops and red when assigned to one-shots. When a loop is playing, its button turns orange (more on sampling in a moment). In fact, almost every button on the KP3 lights up, blinking when necessary to guide you through the occasional multistep sequence of pushes. That visual feedback makes the KP3 especially easy to navigate.

Above the touch pad are eight illuminated Program Memory buttons. As on the KP2, these store shortcuts to your eight favorite programs, but on the KP3 they also perform a time-slicing of sorts, letting you skip over portions of each sample to create interesting rhythmic variations (see **Web Clips 2a and 2b**). Press the nearby Shift button, and the Program Memory buttons become enabled to transfer samples to and from the memory card, adjust MIDI and display settings, and put the KP3 into USB Mass Storage mode so you can access an SD card (not included) from a computer.

Other control improvements include dedicated buttons for Mute (which makes it easier to perform gating effects), Pad Motion (which records a few seconds of your movements on the pad), and audio tempo detection. You can now save Pad Motion performances in the Program Memories as well and play them backward. Unfortunately, Pad Motions always run at the original speed rather than adjusting to the sequence length, which makes them awkward to use with loops.

The fader is another KP3 innovation. Normally it controls the level of all four samples in parallel. Hold the Shift button, and the fader instead controls the feedback amount for an eighth-note delay on the effect signal. The delay kicks in when you lift your finger from the touch pad. Korg calls this feature FX Release; it's designed to smooth the transition between processed and dry sounds, and it does.

You can also program the fader, FX Depth knob, and Hold button to send MIDI data. The FX Depth knob

FIG. 2: The KP3's back panel includes a power jack (with no cord lock), MIDI I/O, USB, a Direct/Send switch for in-line or aux-send hookups, and RCA audio I/O. A headphone output, a ¼-inch mono mic input, and an SD card slot are on the front.

FIG. 3: With the straightforward editor-librarian, you can transfer samples to and from the KP3, make extensive controller assignments, and even change the scrolling message.

doubles as an output level control for the drum groove and synthesizer programs, but it would have been helpful to have a true master output knob.

On the KP3's right side are the tempo controls, including the detented Program Change/BPM Adjust knob. (Pressing it toggles functions.) The Tap/Range button lets you tap the tempo or set a detection range for analyzing incoming audio. I found that automatic tempo detection worked quite well on music with a prominent beat, and I liked the way it continually readjusted to tempo changes. Pressing Shift and the Tap/Range button simultaneously invokes the Align function, which restarts all looping samples on the downbeat. You can tap it repeatedly to create stuttering effects.

Jacks in the Box

Unlike the KP2, the KP3 has no turntable inputs, just RCA line inputs and a ¼-inch mono mic input (see Fig. 2). A handy lever switches between them. Some people complain that the Kaoss Pads use RCA jacks for line-level I/O; Korg contends that they provide a more secure fit than ¼-inch jacks and that adapters are readily available. I feel the lack of an XLR mic input is a bigger inconvenience. Then again, you're probably going to mangle the mic signal with effects, so a cheapo mic should suffice. Another option would be to put the KP3 in Send (100 percent wet) mode and run it off the effects loop in a mixer. In that case, you could use any I/O your mixer provides.

You also get MIDI In and Out as well as USB. The USB jack lets you transfer samples and parameters to and from the software editor-librarian, send and receive MIDI data between the KP3 and other software, and access an SD memory card from a computer. Unfortunately, you can't address the hardware MIDI ports over USB, which would have enabled the KP3 to be a MIDI interface.

Rounding out the back panel are a power button and a jack for the 12V wall wart. I would have liked a cord lock for the latter and a Kensington security slot. Korg says those additions would have raised the price significantly due to the metal case.

Effects Appeal

With 128 effects and sound generators in 14 categories, I found it maddening that the only documentation on them was a single page listing their names and *x-y* control parameters. The manual in general is shallow. Be prepared to do a lot of experimentation. I recommend starting with the video walk-through on Korg's site, which was uploaded just before we went to press.

Fortunately, exploring the KP3 is fun. The majority of effects sync to tempo, and those that don't can pick up the synchronized echo from the FX Release feature.

The filters don't sound as juicy as I'd hoped, but they do provide intriguing parameters such as noise level and shape (see **Web Clip 3**). With the 8-band graphic EQ, you can draw the frequency curve on the touch pad.

Fans of abrasive textures will enjoy the new Decimator effect, which reduces the sampling rate and bit depth (see **Web Clip 4**). Alas, the reverbs are gritty too.

LOOP TRIGGER HACK

The most confusing thing about the KP3 is the way it handles looped samples. Rather than triggering loops from the beginning when you press a sample pad, it runs the loops continuously in the background and then unmutes them. (One-shot samples behave as expected, starting from the beginning.) If you have four loops of equal length that all start on a downbeat, that design ensures they'll always be synchronized with each other. But it also means there's no way to know which section of the loop you'll hear when you press a pad.

To get around the problem, use this hack I discovered: to trigger one or more loops from the beginning, hold down the Mute button, start the loop(s), hold down the Shift button, and then simultaneously release the Mute button and press the Align button.

LFOs account for a fifth of the effects, and they're all beat-synced. Generally, the *x*-axis controls speed, ranging from 8-bar sweeps to frantic burbles. I particularly liked the infinite highpass filter and flanger effects, which apply a sonic barber pole to your sound (see **Web Clip 5**).

There are a number of interpolating delays, which change pitch instead of glitching as you change the delay

REVIEW

time—great for dub effects (see **Web Clip 6**).

Grain shifter is a new effect reminiscent of the Grain Delay in Ableton Live. It seems to grab short slices of the sound periodically and loop them for a buzzy, stuttering effect (see **Web Clip 7**). (Here's where a descriptive manual would be welcome.) For longer, rhythmic looping—with wonderfully grungy pitch transposition—there are 20 looping and sampling programs (see **Web Clip 8**).

Four crossfade programs let you fade between the four sample pads on the *x*-axis while applying another effect on the *y*-axis. Because that design never lets you hear more than two samples at once, I would also have liked a mode that put each sample at a corner of the touch pad and let you fade between them Wavestation-style.

The four drumbeats are useful, if bland; I resampled them to drive several of the Web Clips. The pad performance layout is really cool, though. In two programs, the *x*-axis selects among eight patterns and the *y*-axis varies delay depth. In others, the axes control tone, decay, and delay. The two remaining drum programs are finger drums, which trigger different sounds based on where you tap.

I loved playing the ten synth programs, which ranged from a brassy sawtooth to screeching digital textures, all fed through rock star echo. It's hard to hit specific notes when playing from the pad, but it's simple to scrape out wild, expressive swoops and blips (see **Web Clip 1**).

The final effect category is vocoder. Although these vocoders are supposedly derived from the vaunted Korg Radias, I found them disappointing. The sound is rough and gargly, and you can't specify the pitch with MIDI notes. You just slide your finger around and hope the pitch lines up with the song (see **Web Clip 9**). That makes it tough to produce traditional "robot singing," but I did find myself using the vocoders to munge loops into background textures.

PRODUCT SUMMARY

KORG KP3 Kaoss Pad

effects processor/loop sampler
$460

FEATURES	4
EASE OF USE	3
AUDIO QUALITY	3
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Smooth, informative touch pad/display. Tempo-synced, playable effects. Straightforward USB editor-librarian. Hefty metal housing. Surprisingly deep MIDI controller capabilities.

CONS: MIDI sync drifts. No XLR mic input. No time-stretch on samples. No power cord restraint. No MIDI note input on vocoder. No master output knob. Skimpy manual.

MANUFACTURER
Korg
www.korg.com

Sample This

The KP3 offers a unique take on sampling (see the sidebar "Loop Trigger Hack"): everything is referenced to the current tempo. When you hit the Sampling button, you're given the choice of recording 1, 2, 4, 8, or 16 beats. Hit the sample pad to which you want to record, and the eight Program Memory buttons start counting down the remaining time. (A tricolor LED indicates level.) If you terminate sampling before the countdown ends, you get a one-shot; otherwise, the audio you recorded starts looping. The process is extremely fast.

If you subsequently change tempo, loops will change speed and pitch to match (there's no time-stretching, alas). One-shots always play back at the original pitch and tempo. The only way to figure out a loop's original tempo is to transmit it to the computer editor-librarian (see **Fig. 3**).

There's rudimentary sample editing for loops; you can adjust the start time plus or minus one beat in 32nd-note increments. (One-shots can't be trimmed.) When I missed catching a downbeat, I found it easier to resample the loop to a new pad (see **Web Clips 10a, 10b, and 10c**).

When resampling, you can run the sound through the effects. You can also change the level of individual pads and, with loops, mute individual slices in real time with the eight Program Memory buttons. I had fun chopping up a politician's speech that way (see Web Clip 2).

You can also import 8-, 16-, and 24-bit mono or stereo WAV or AIFF files at 44.1 or 48 kHz; the KP3 converts the file to 48 kHz and truncates the length to 16 beats. Samples can be saved to and loaded from the SD card as well, although they must be named with 2-digit numbers, not text.

Computer-MIDI Kaoss

I installed the KP3 MIDI driver and editor-librarian on both Mac OS X and Windows XP. Transferring samples and settings back and forth was simple, although the PDF manual was very sketchy on MIDI control functions. I had to hook up a MIDI monitor to figure out what was going on.

Pressing Shift-8 on the KP3 puts it in a special controller mode in which it stops processing audio and assigns MIDI commands to the eight Program Memory buttons, the four Sample Bank buttons, the *x-y* pad, the FX Depth knob, the fader, and the Hold button. What's more, in this mode, the *x-y* pad can send up to eight different MIDI Control Change messages based on position and one more for taps (see **Web Clip 11**). I easily mapped buttons 6, 7, and 8 to Start, Stop, and Continue on my sequencer.

All is not well in MIDI-land, though. After reading several online reports of MIDI sync problems, I created a drum pattern in my keyboard sequencer, sampled it into the KP3 with the latter in External Sync mode, and then played both patterns simultaneously. They drifted apart in less than 30 seconds. Driving the keyboard from the KP3's clock did keep the notes in sync, although that would be a hassle live, because you'd have to enter the tempo manually on the KP3 for each song. Interestingly, the KP3 did a better job syncing to the keyboard when I set it to derive tempo from the audio signal. I also crashed the KP3 once. Korg recently released a firmware update (version 1.03) that addresses the crashing and "some MIDI clock sync issues." It is still investigating the loop-drift reports.

See the Light

The KP3's DJ pedigree gives it a refreshing directness. There are no layers of confusing menus here, just immediate control. Being able to play in three dimensions at once (vertical, horizontal, and tapping) on the luxurious-feeling touch pad is unspeakably cool. I tried mapping two knobs on an external MIDI controller to the KP3's *x*- and *y*-axis parameters, and it felt about as fluid as an Etch A Sketch.

If you're looking for a groove box or studio effects processor, you may be disappointed. There are only four sample pads and they don't support time-stretching, the MIDI sync currently drifts, several effects (particularly reverb) sound rather gritty, and the I/O is designed for DJ mixers, not studio ones. For a street price of $399 I can't complain too much, though. On the other hand, the KP3 makes a stunning controller for other effects and synthesizers, and it's so tactile and fun to use, you'll be tempted to sprinkle its chaotic spice on everything. EMWEB CLIPS

David Battino (www.batmosphere.com) is the coauthor of The Art of Digital Music *(Backbeat Books, 2004) and the audio editor of the O'Reilly Digital Media Web site (http://digitalmedia.oreilly.com).*

FIG. 1: The Spectralis is more than a synth module with sample-playback and virtual analog sound engines. It's also a sophisticated drum machine, step and pattern sequencer, and 10-band filter bank.

RADIKAL TECHNOLOGIES Spectralis

Powerful synthesis, sampling, sequencing, and filtering. By Geary Yelton

I first became aware of the Spectralis more than three years ago, just as the German manufacturer Radikal Technologies was pinning down the final details of its design. It sounded fantastic on paper: a digital synthesizer with analog filters, a fixed filter bank, user multisampling, and independent step and pattern sequencers, all in a performance-oriented tabletop module with loads of hands-on control. It promised to be the mother of all groove boxes, but was it too ambitious to be true? During subsequent NAMM shows I was able to see it, touch it, and hear it in action. I liked what I heard, and it appeared to deliver on its promises. At the very least, I knew that Radikal was on the right track.

The Spectralis is the brainchild of Jörg Schaaf, who was instrumental in developing synths for Quasimidi and has recorded and performed with electronic-music superstar Klaus Schulze. The Spectralis is a self-contained production workstation with features that make it attractive for live performance as well as studio use. Like most workstations, it can be used by itself, connected to other MIDI instruments, or installed as part of a computer-based system.

Although the Spectralis has now been shipping since February 2005, its operating system is a work in progress, with new features arriving in dribs and drabs. I downloaded two minor updates while I was writing this review. Since August 2006, the only official operating system has been update 097. Currently I'm working with update 097L. Radikal says that 098 is right around the corner, but you shouldn't expect every announced function to work until version 1.0.

Uncharted Territory

One look at the Spectralis Analog Synthesizer/Beat Matrix/Filter Bank should tell you it's a serious piece of audio hardware (see **Fig. 1**). It's as solid as a brick, and it performs virtually all the functions you might expect from a groove box, and more. A 2 × 40–character yellow-green LCD and wood side panels give the Spectralis a distinctively retro appeal. A total of 37 knobs and 52

illuminated buttons are strewn across the front panel. You can press on all the knobs as if they were buttons. Every knob is an encoder that rotates infinitely—even the dedicated Master Volume knob. Because that knob's value isn't displayed in the LCD and it doesn't stop turning when it reaches maximum, you have to guess.

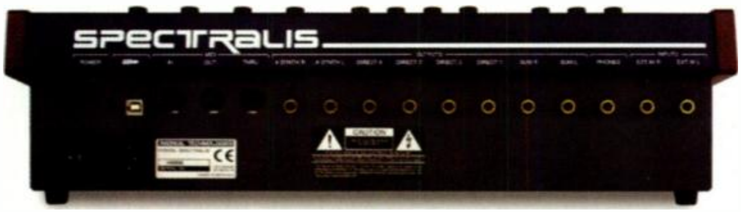
At first glance, the control panel layout may be overwhelming. The panel is divided into 15 discrete sections. At the bottom, a row of 16 numbered buttons lets you perform functions such as programming sequences and selecting parts, patterns, grooves, and motifs. You can also use the buttons to enter note data and to transpose patterns. Just above the buttons are 16 encoder knobs for controlling parameters such as part levels and step-sequencer values.

Above the 16 knobs is another row of 16 buttons, divided into 5 sections. The leftmost section is Num Button Assignment; pressing any of its 5 buttons determines the function of the 16 numbered buttons. To its right is Sequence Edit, which contains 3 buttons that open sequence-editing menus in the main display; pressing the buttons also assigns the 16 knobs and 16 numbered buttons to sequencer functions.

Next is the Groove Edit section, which contains just two buttons. Pressing Step/Accent lets you program drum sequences using the 16 numbered buttons to select steps, the first 12 knobs to select drums, and a menu in the display to specify parameters such as pitch and Velocity. Pressing Function opens another menu, but its functions haven't been implemented in the current OS (more about that later). The Filterbank section also has just two buttons, Level and Pan. To its right is the Mixing Desk section, which supplies buttons that determine whether the 16 numbered knobs will control volume, panning, or one of two effects sends for each of the 16 parts.

In the front panel's upper left, the Part section's 4 buttons—Select, Mute, Solo, and Play—also determine what the 16 numbered buttons do. When you press Select, you can use them to select parts for editing. If a pattern is playing, pressing Mute and selecting a part removes it from the sequence, and pressing Solo removes all the other instruments. Pressing Play allows you to play notes or beats on the currently selected instrument using the first 13 numbered buttons. In Play mode, you can use the final three buttons to shift octaves and to hold a note or chord.

FIG. 2: In addition to MIDI and USB ports, the Spectralis has nine analog audio outputs and a pair of audio inputs for processing external sounds through its filter bank.



Sections to access the synthesis parameters take up most of the front panel's top half. Three knobs control the LFOs, and three control the VCAs (the amplifiers, despite the name, are not voltage controlled). The VCO section (again, not voltage controlled) has four knobs, as well as two buttons for transposing a maximum three octaves up or down. The VCF section has cutoff and resonance knobs for each of the two filters, four buttons to select the filter slope, and one nonfunctioning button labeled Set Routing. Pressing on the knobs opens filter menus in the display.

The Spectralis's LCD is in the upper right, and below that are four knobs for controlling whatever parameters it displays. Below the knobs, the Advanced Editing section has buttons for scrolling displayed pages up and down, a Shift button for accessing alternate front-panel functions, and Enter, Exit, and Save buttons. Just below Advanced Editing is the Transport section, which includes a Tempo knob.

Making Connections

Other than USB, an IEC power receptacle, and MIDI In, Out, and Thru ports, all rear-panel connections are on unbalanced ¼-inch audio jacks (see **Fig. 2**). Two inputs allow you to process external sounds through the Spectralis's filter bank and synth engines. Sum R and Sum L are the main outputs, and Direct 1 through 4 are assignable outputs for individual parts. The synthesizer section has its own pair of outputs, and a headphone jack is also on the rear.

The Spectralis's USB port is exclusively for transferring software updates, samples, and other data files to and from your computer. It does not serve as an audio or MIDI conduit, and there's neither an editor-librarian nor a plug-in for controlling the Spectralis. When USB is connected, the Spectralis's internal 64 MB of flash memory appears on your computer's desktop as if it were a removable hard disk. If you view its contents, you'll see song folders and dozens of files that only the Spectralis recognizes. You can expand the onboard memory by plugging SmartMedia cards into two slots on the front panel, or you can store files on your computer's hard disk and drag them to the Spectralis's flash memory as needed.

Protosynthesis

At the heart of the Spectralis's Hybrid Synth are four analog-modeling digital oscillators that generate continuously variable waveforms. That means you can morph from one waveshape to another by applying a modulation source. You can also modulate the spectrum of any waveform in a fashion similar to pulse-width modulation using a technique that Radikal calls Time Linearity Modulation. If you want extra grit, you can reduce the bit depth of individual oscillators—a capability that's unusual on hardware instruments. Each oscillator has its own amplitude and pitch envelopes, and you can

individually adjust their glide times—very nice. Because the audio oscillators can function as modulation sources, you can combine them in various ways for frequency-, ring-, and phase-modulation synthesis.

The Hybrid Synth (which the manual also calls the analog synth) has two real analog filters that give the Spectralis lots of character. One is a lowpass filter with a 24 dB-per-octave slope, and the other is a 12 dB-per-octave filter with lowpass, highpass, bandpass, and notch outputs. Both can resonate up to self-oscillation.

Remarkably, the Hybrid Synth supplies 10 LFOs and 14 envelope generators. The three LFO knobs let you control rate, depth, and waveform, which is continuously variable from sine to sawtooth to random. Four of the LFOs offer additional parameters such as individual envelopes and definable start points. The Spectralis's envelope generators are of the ADSR variety, but with an additional hold stage. They are hardwired to 14 logical destinations; 8 control volume, 4 control pitch, and 2 control the lowpass and multimode filters. Despite the Spectralis's abundance of front-panel controls, it has no dedicated envelope knobs, but assigning the four knobs beneath the display is only a knob press away.

You might expect an instrument with so many knobs and buttons to make most parameters available on the surface, but the Spectralis goes much deeper than most synths. When you press on a knob in the VCO section, you can scroll through 26 oscillator pages, each containing 4 hidden parameters ranging from routing individual oscillators to the filter bank to specifying Time Linearity Modulation settings.

Surprisingly, the manual provides only limited detail about VCO parameters. Fortunately, you can find information hidden in the Spectralis's onboard help system. Just hold the Shift button, press one of the four encoders beneath the display, and a description will appear in the LCD. Pressing the Page Up and Page Down buttons allows you to read any additional text.

PRODUCT SUMMARY

RADIKAL TECHNOLOGIES Spectralis

synthesizer workstation module
$2,500

FEATURES	4
EASE OF USE	2
QUALITY OF SOUNDS	4
VALUE	3

RATING PRODUCTS FROM 1 TO 5

PROS: Warm, fat, and punchy. Impressive versatility. Solid construction. Loads of oscillators, LFOs, and EGs. Analog filters. Imports user samples.

CONS: Unfinished features. No user-programmable effects. Limited USB functionality. No digital I/O. Poor documentation. Difficult to learn. No sample-file translator for the Mac.

MANUFACTURER
Radikal Technologies
www.radikaltechnologies.com

Alternate Engine

Along with the Hybrid Synth, the Spectralis has 3 DSP Synth parts with a combined polyphony of 32 voices. The DSP Synth engine is most useful for playing user samples and drum and percussion sounds. It has its own stereo digital filter with resonance and a 12 dB-per-octave slope; it operates as a lowpass, highpass, or bandpass filter. Two ADSR envelopes modulate the filter and amplifier, and Velocity can modulate them as well.

The Spectralis comes with a collection of multisamples that loads automatically. You can import your own samples by converting 16-bit, 44.1 kHz WAV or multisampled SoundFont 2.0 files into the Spectralis format using the included SmplImp.exe application, but only in Windows (a Mac version is planned). A simple drag-and-drop procedure quickly converts files.

On the Fly

The Pattern Sequencer lets you create songs by arranging as many as 32 patterns. Each pattern plays 16 individual parts and may be as long as 16 measures. Parts 1 through 11 and part 16 are dedicated to drum and percussion sounds, and parts 12 through 15 are dedicated to synth sounds. Within each pattern are motifs that play each part; a motif is a melody, drumbeat, or chord pattern. The Spectralis comes with a large library of motifs, and you can create your own by either real-time or step-time recording. Every motif can have a different length and time signature.

The Spectralis gives you complete, real-time control over the arrangement and mix of each pattern and song (see **Web Clip 1**). You can mute and solo parts, adjust their levels and panning, and replace any sound with any other available to that part. You can even exchange motifs and grooves while the sequencer is playing, and jump back and forth from pattern to song mode.

The Step Sequencer has 32 tracks, each with a maximum of 192 steps. You can route tracks to control various parameters that include note value, amplitude, filter cutoff, and so on. You can determine duration, Velocity, and even probability—the likelihood that an event will be triggered when a sequence line repeats. Steps can be played, muted, or skipped, and they can be any length from a 192nd note to 12 measures. In addition, the Step Sequencer can transmit MIDI data to control external hardware and even software instruments.

With the 097L update, the Spectralis has gained not one but six arpeggiators. A complete set of user parameters includes direction, resolution (from 192nd to whole notes), octave range, gate time, and more. A nice twist is that the arpeggiators can respond to notes played by the Step Sequencer. (For information about the Spectralis filter bank, see **Web Clip 2**.)

To Be Continued . . .

The Spectralis has something for everyone. For keyboardists who want variety, it's a versatile instrument offering some very good factory sounds and the ability to import your own samples. For sound designers who want to tweak parameters they've never seen before, the Spectralis goes as deep as you want to dig. And for musicians who just want to reach out and make some music, its preprogrammed motifs and patterns serve up speedy gratification. The Spectralis's virtual analog synthesis sounds warm and fat, and its versatile sequencing encompasses the instrument's strongest features.

The Spectralis's routing flexibility is very impressive. You can direct sounds from almost any point in the signal path to virtually anywhere. You can process drum grooves or samples from the DSP Synth with the Hybrid Synth's analog filter, and you can process sounds from any source with the filter bank.

Although you can do an awful lot with the Spectralis, its development is ongoing, and several rather basic functions have not yet been implemented. Most notable is that only one of the two effects processors works at all; its only effect is delay, and it has no user parameters. It's not unusual to maneuver your way to a parameter in the LCD, only to discover that the associated knob is nonfunctional, as indicated in the display by brackets around the parameter name. On the other hand, the Spectralis's designers have left plenty of room for future expansion. You can always add more features through software updates, but adding more knobs and buttons would be impossible.

Radikal describes the Spectralis as being open ended, and I can see some sense in that approach. Instead of having to replace your synth when new and desirable features become available, you could just update the one you have, much as you can with the Korg OASYS or Access Virus. However, the Spectralis's degree of non-completion is somewhat unusual for a hardware instrument, and more akin to what you might expect from beta software. Somewhere over the horizon are new features such as dynamic effects, Control Change automation, event editing, and support for REX files.

In the meantime, the Spectralis is a very versatile synthesis workstation that suffers somewhat from a rather complex and confusing user interface. It makes some operations (such as copying motifs from one pattern to another) easier than most groove boxes do, but many are more difficult. Even after spending many hours learning my way around, I frequently had trouble finding exactly the parameter I wanted to adjust. If you were to devote yourself entirely to mastering the Spectralis, though, it would offer you a satisfying platform for creating all kinds of electronic music.

EM WEB CLIPS

EM associate editor Geary Yelton has been playing, programming, and writing about synthesizers for more than 30 years.

TRUE SYSTEMS

P-Solo

By Myles Boisen

The True Systems P-Solo ($695) is a unique single-channel, solid-state mic preamp/DI with a small footprint (3 inches by 6 inches) that makes it ideal for desktop studios with limited space. Despite its diminutive size, the transformerless P-Solo offers a variety of pro features and boasts exemplary specs, including ultralow distortion and an astonishing 1.5 Hz to 500 kHz (+0/–3 dB) bandwidth.

The attractive aluminum faceplate is dominated by an oversize gain knob. At its fully counterclockwise position, the control offers a low-gain mode (+6 dB at the mic input and –14 dB for the DI). When rotated clockwise, the knob clicks audibly before switching into a normal variable-gain mode (+16 to +64 dB for the mic input and –4 to +44 dB for the DI).

A ¼-inch DI input is on the front panel, as are switches for +48V phantom power and an 80 Hz low-cut filter. An array of four LEDs provides output gain metering: signal present (green), +4 and +12 (yellow), and overload (red). Regrettably, the P-Solo lacks a phase-reverse switch.

On the rear panel is a balanced XLR mic input, balanced +4 dBu XLR and ¼-inch TRS outputs, a standard IEC connector for the internal power supply, and a power switch. In mid-2007 True Systems will offer a 4U rackmount adapter tray that holds five Solo units—any combination of P-Solo preamps and the upcoming C-Solo compressor.

The True Systems P-Solo offers a low-gain mode and a front-panel DI that sounds great on electric guitar.

Flying Solo

I had numerous occasions to test the P-Solo in sessions at my Guerrilla Recording studio in Oakland, California. On a percussion session, recording to a modular digital multitrack, the P-Solo paired well with a Neumann TLM 49 solid-state condenser mic. When layering congas, shaker, tambourine, and cowbell, the mic and preamp combination captured tight, organic sounds that were well suited to the roots reggae tracks.

On this date I noted that the P-Solo delivered good transients and was never overly bright. It also worked well with an AEA R84 ribbon mic to smooth out a trombonist's often raspy tone.

I auditioned the P-Solo on two different vocal sessions and in both cases found it lacking in some respects. On a rock session with a female singer, I paired it with a Lawson tube mic. The resulting track was warm but sometimes too dark sounding, making vocal intelligibility a challenge amidst the drums and guitars. Switching to another preamp instantly restored much-needed high-end clarity to the track.

With a mellow male folksinger on the Neumann TLM 49, the P-Solo didn't work out at all. The preamp emphasized an unpleasant midrange and sibilance that I had never heard from this mic or singer before, producing a surprisingly unflattering result. Substituting a Grace Designs Model 101 preamp in the vocal chain restored a fuller and more usable timbre to this track.

Overall, the P-Solo has a clean and very warm solid-state sound with minimal coloration. In comparison to other solid-state preamps in my rack, the True Systems design tends to sound a bit dry or flat, sometimes lacking openness or sparkle in the high end. Of course this quality can be a plus or a minus, depending on one's taste, the interaction with a given microphone, and other factors.

Going Direct

The P-Solo's DI circuit compared favorably to that in the Model 101 during tests with a mid-'70s Fender Stratocaster. The Model 101, one of the few preamps that sound really good on both bass and guitar, provided a thumpy bass and high-end clarity. But the P-Solo had a thicker tone and a more pleasing midrange. With a timbre closer to what one might get out of a high-tech guitar amp, the P-Solo was a winner on guitar.

I found the P-Solo to be a bit thin sounding on electric bass when tested against the Model 101 and a Millennia Media TD-1. Although the low end was not quite as full, the P-Solo DI issued a nice tone with good midrange detail.

Based on how it sounds with vocals, and considering the price, the P-Solo might not be the best choice for someone looking for their first general-purpose mic preamp. However, the P-Solo's warm tone, fine DI circuit, unique gain-attenuation feature, and portability make it worthy of consideration for those adding to their preamp collections.

Value (1 through 5): 3
True Systems
www.true-systems.com

WALDORF

The Waldorf Edition 1.2.4

(Mac/Win)

By Geary Yelton

Plug-ins have come a long way since 1998, when German synthesizer pioneer Waldorf introduced the virtual filter D-Pole, followed by the soft synth PPG Wave 2.V in 2000 and the virtual percussion module Attack in 2001. Now all three are available in a suite called the Waldorf Edition ($149.95). They haven't changed much since their initial release, except that they now run on modern computers such as Intel-based Macs and cost no more than the original price of just one. They're now compliant with AU and VST 2.4, and you can control them using Novation's Remote SL and other MIDI controllers.

Installing the Waldorf Edition on my dual-processor 2.3 GHz Power Mac G5 with Mac OS X 10.4.9 couldn't have been easier.

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DrumCore Demo and DrummerPack example tracks available at submersiblemusic.com.

I ran the installer, opened Apple Logic Pro 7.1.1 and Steinberg Cubase SX 3, and immediately began using the plug-ins.

Filter Flavor

D-Pole is more than a simple filter; it's also a ring modulation, distortion, envelope follower, and delay plug-in. In addition to lowpass, highpass, bandpass, and notch, it has an allpass filter type called Resonator. All five are resonant to self-oscillation, and slope is switchable from 12 dB to 24 dB. D-Pole displays frequency in hertz and resonance as a percentage—very nice. You can increase overdrive to 52 dB and prevent overload by adjusting the volume.

D-Pole's maximum delay is two seconds, and delay's feedback can loop infinitely. Damping lets you dial down the high-frequency content. Delay automatically syncs to tempo, and clicking in the Time display reveals a menu for selecting rhythmic values. You can apply the plug-in's LFO to modulate cutoff, panning, or oscillator frequency.

Ring modulation provides an oscillator with adjustable frequency and three waveforms. You can even dial down D-Pole's sampling rate to as low as 210 Hz for bizarre distortion effects caused by skipping samples. An envelope follower can modulate filter frequency in a positive or negative direction. Thirty-one included presets show off how much fun D-Pole can be (see **Web Clip 1**).

The New Wave

Wave 2.V is an 8-part multitimbral soft synth that resurrects the sounds and front-panel layout of the PPG Wave 2.3, an 8-bit wavetable synthesizer. Wave 2.V furnishes 9 banks of 128 sounds, including the original factory programs. Many Wave sounds have a digital bite that's difficult or impossible for other synths to reproduce (see **Web Clip 2**).

When you turn a knob, its corresponding parameter is shown in a tiny display. Clicking on a row of buttons opens windows to select from 31 wavetables, program the arpeggiator, reroute modulation, graphically edit envelopes, and more. Copy and Paste buttons let you move programs from one bank to another.

Wave 2.V offers numerous ways to impart timbral complexity. The Basis knob spreads individual voices across the stereo spectrum. Three keyboard modes stack voices so that two, four, or eight sound when you play a single note; tuning the voices relative to one another by semitones lets you play one-finger chords and complex timbres. You can use Aftertouch or the LFO to scan the wavetable. And if you enable True PPG, the plug-in exhibits the original instrument's limitations, including aliasing and other signs of instability.

Attack Is Back

Attack is a multitimbral plug-in I've long admired because it offers capabilities no virtual drum machine has matched. Its analog-modeling engine has two 9-waveform oscillators, frequency and ring modulation, a 6-mode filter, and two graphic envelope generators with an unusual Shape parameter. The LFO and the sophisticated dual-delay section can sync to tempo. To simulate handclaps, a unique Crack Modulator generates a sawtooth LFO that's applied to amplitude.

Attack comes with 28 kits you won't find elsewhere, many focusing on unusual sounds (see **Web Clip 3**). When you load a kit, the names of its 24 parts appear on bars on Attack's left side; clicking on a bar plays its corresponding sound and allows you to edit its parameters. In addition to assigning the lower two octaves of MIDI notes to play individual drum and percussion sounds and effects, each kit has 12 sounds that include bass and pads you can play melodically and polyphonically on 12 MIDI channels.

The Waldorf Edition bundles and updates three oldies but goodies—D-Pole, Wave 2.V, and Attack—with dozens of presets.

A Tasty Salad

Wave 2.V and Attack were respectable bargains when they were sold separately, and they're outstanding values now. My only complaint about D-Pole and Attack is that the plug-in windows are so small that their text can be difficult to read. With a street price well under $100, however, any one of the three plug-ins is reason enough to buy the Waldorf Edition.

EM WEB CLIPS

Value (1 through 5): 5
Waldorf/QTec Designs (distributor)
www.qtecdesigns.com

DIGIDESIGN

Mbox 2 Mini

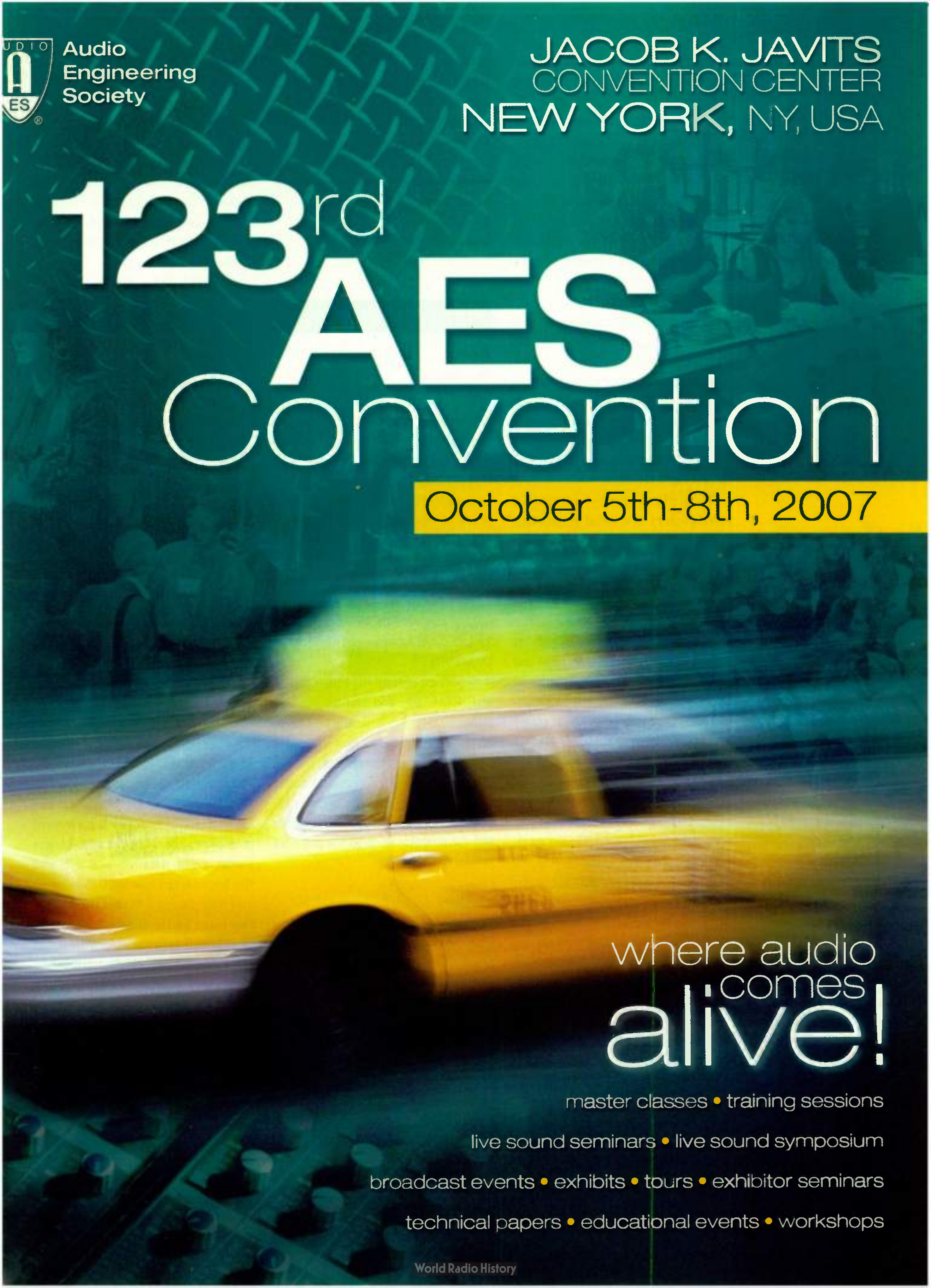
By Gino Robair

With stereo I/O that includes a single XLR mic input, it's safe to say that Digidesign created the Mbox 2 Mini ($329) with singer-songwriters in mind. The USB 1.1 bus-powered Mbox 2 Mini (M2M) is less than half the width of the Mbox 2 Pro, and a third as heavy. And because Pro Tools interfaces act as a dongle—you cannot use the program without one—the M2M's small form factor makes it useful for the traveling Pro Tools user who might not have the space for a larger Mbox but needs only basic I/O for listening and editing.

The M2M also provides the cheapest way into a Digidesign-branded Pro Tools system. (Although you can get into an M-Audio Pro Tools M-Powered system for around the same list price, that system won't let you use the DV Toolkit option for postproduction work.)

Bare Necessities

To reduce the interface to this size, Digidesign includes a minimum of analog I/O, completely eliminating MIDI and S/PDIF ports. The rear panel holds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a single XLR mic input, a pair of unbalanced ¼-inch line/DI inputs, a pair of unbalanced ¼-inch line outputs, a USB port, and four buttons—a pad switch for each channel, a 48V phantom power switch, and a Mic/DI switch for channel 1. This last button lets you select between the XLR and ¼-inch input, so you can keep cables plugged into both jacks.

The front panel has a ¼-inch headphone jack and level control, a mute switch for the monitor outputs (which doesn't affect the headphone output), an input level control for each channel, and a control that mixes between the direct input signal, for latency-free monitoring, and the playback signal coming from the computer. A green LED indicates when the M2M is receiving power from your computer. The package also includes the *Pro Tools Method One* instructional DVD for new Pro Tools users.

The interface's housing feels solid but is reminiscent of a metal project box. Four rubber nubs on the bottom of the case keep it from sliding around the desktop.

The M2M's specs are similar to those of the Mbox 2 Pro. The main differences are less than 50 dB of gain, as well as unbalanced line-level jacks and a few decibels less dynamic range on the M2M's I/O. However, the two interfaces sounded nearly identical to me in daily use with my 1.33 GHz Mac G4 laptop running Mac OS X 10.4.8. When I wanted to edit my Pro Tools sessions while touring, the M2M was the winner hands down, not only because of its size but also because it doesn't require external power. If I had a wish list, it would be to have balanced I/O and the buttons relocated to the front panel.

The Mbox 2 Mini offers a simplified feature set in order to maximize portability.

On the Go

The Mbox 2 Mini is great for Pro Tools LE users who don't need more than one mic at a time. It's also a boon for those of us who like to get some work done while in planes, trains, and automobiles.

Value (1 through 5): 3
Digidesign
www.digidesign.com

VIR2 INSTRUMENTS

Acoustic Legends HD

(Mac/Win)

By Marty Cutler

Not everyone owns a collection of high-end acoustic guitars. For that reason, Vir2 Instruments offers Acoustic Legends HD ($299.95), which provides a diverse collection of guitars with distinct tonal characteristics, so you can choose an appropriate instrument for your music. As a bonus, other fretted instruments are included, such as mandolin, acoustic bass guitar, and banjo. All told, the library amounts to 19 GB of 24-bit, 96 kHz stereo samples distributed over three DVDs.

Acoustic Legends HD comes in AU, VST, RTAS, DX, and standalone versions. Installation is simple and relies on Native Instruments' new NI Service Center feature. It's convenient, but you will need Internet access and a fast connection. I put Acoustic Legends HD through its paces on my dual-processor Mac G4 with Mac OS X 10.4.8 and 2 GB of RAM.

KP Duty

Acoustic Legends HD uses Native Instruments Kontakt Player 2 (KP2), which eliminates the need for multiple, dedicated standalone applications: libraries using KP2 can launch within a single, multitimbral instantiation. If you need to edit the sounds in greater detail, you can load the library into Kontakt 2. The KP2 interface bears a strong resemblance to its full-featured Kontakt 2 sibling, albeit without user access to the latter's powerful script-creation and sampling features.

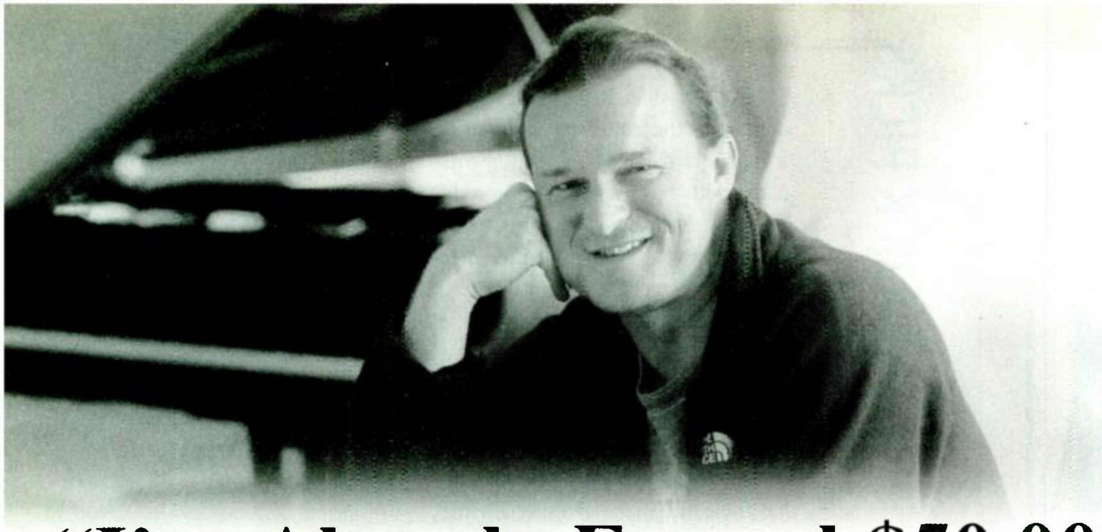
Nonetheless, Acoustic Legends HD gives you a performance-oriented layout with a number of important controls at your fingertips. The parameters you can tweak vary by instrument, but the most common ones belong to reverb, EQ, and stereo width. Most of the conventional guitar patches give you control over the release samples and level of fret noise, whereas the chord banks do not offer decay controls.

One of the most challenging guitar techniques to reproduce with a keyboard is a convincing strum. With that in mind, Acoustic Legends HD offers a bank of strummed chords, including muted strokes for rhythmic variation. Keyswitches let you access different chord inversions and a relatively generous set of chord qualities, such as major, minor, major seventh, and half-diminished. If you need to change keys, a built-in capo function limits the range of low notes and inversion choices, just as a real capo would when placed on a guitar.

The Xtra Special bank is for players who are less insistent on authenticity. The bank contains guitars mutated into pads, tempo-synchronized arpeggios, and just plain weird and rude sounds. I appreciate some of the rhythm-oriented, tempo-sync patches, but many of the other patches strike me as filler.

High Strung

Appropriately, the conventional instruments sound like their sources: the Gibsons are brassy with a prominent midrange; the Martins and Taylors sound bright, detailed, and full; and the fingerpicked Lowden 025 is sweet and mellow (see **Web Clip 1**). The two main guitar folders break down into picked (presumably with a flat pick) and fingerpicked instruments. However, I question a couple of the instrument choices therein. For example, the Gibson J200 is a natural for fingerpicking, yet there are no fingerpicked Gibsons at all in the collection. I would have preferred the Martin D-35 as a picked, rather than fingerpicked, instrument. In all fairness, these are reflections of my own taste rather than rules set in stone.



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Unlike some of the other conventions I've attended, the panelists at the Rally are friendly and accessible. I've never been anywhere that gives you so much great information, and so many chances to meet people who can help your career.

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Although steel-string guitars predominate, the collection does offer a couple of nylon-string instruments played with a pick and fingers. In addition, the Xtra bank contains some decent banjos, ukuleles, and a nice mandolin tremolo patch—fortunately, with a uniform tremolo speed across the keyboard. Nice-sounding variations are available throughout the library, including keyswitched instruments with slides mapped to alternate keys or keyswitched chord inversions. The bank of harmonics has some nice material, especially the J200 harmonic chords patch (see **Web Clip 2**).

A typical problem with sampled guitars is the unnatural uniformity of timbre. In real life, an upstroke on a guitar often sounds different than a downstroke, and a strum may not occur in the same location over the instrument's sound hole, resulting in significant timbral variation. Although Acoustic Legends HD uses sampled up- and downstrokes, I noticed an apparent randomizing of samples that added authenticity to the performance.

There are a few holes in the library's documentation. Although the booklet has a list of all the instruments and model numbers, and it explains controller assignments, patch types, and nomenclature, in many cases I needed to investigate with my keyboard what the keyswitching assignments actually changed, because they were not specifically documented.

Vir2 Instruments Acoustic Legends HD uses Native Instruments Kontakt Player 2, which can host multiple instruments and sound libraries.

Pickin' and Grinnin'

Overall, Acoustic Legends HD is a solid and varied assortment of eminently playable, high-resolution virtual guitars and fretted string instruments. If you are looking to add sampled acoustic guitars to your desktop studio, be sure to check out this collection.

Value (1 through 5): 3
Vir2 Instruments/Big Fish Audio (distributor)
www.vir2.com

ABBEY ROAD STUDIOS/ CHANDLER LIMITED

EMI TG12413 Limiter Pack

(Mac/Win)
By Eli Crews

The TG12413 Limiter Pack plug-in ($450; $335 for the LE version; $560 for the TDM version) is the first foray into the digital realm for Abbey Road Studios and Chandler Limited wunderkind Wade Goeke. It's modeled after the first solid-state limiter made by the technicians at EMI in England in the 1960s, the TG12413, which was designed to replace the venerable Fairchild 660 and 670 tube limiters and EMI's own RS124 tube compressor. It's quite possible that some of your favorite albums were recorded through the TG series of equipment, if you happen to like, you know, small-potatoes bands like the Beatles and Pink Floyd.

Time Machine

The TG12413 Limiter Pack plug-in comes in two flavors: 1969 and 2005. The 1969 version has the controls of the original TG limiter used on 1969's *Abbey Road*: Comp/Limit, Recovery (release time), Hold (which adjusts the amount of compression and was an abbreviation for *threshold*), and Output. The 2005 version is based on the Chandler TG1 and exchanges an Input knob for the Hold control and adds an extra 12 dB of input gain for more-extreme limiting capabilities. The two versions share retro EMI TG-series styling, from the red, gray, and black chicken-head knobs to the unique vertical VU meters, in which the needle bounces up from the bottom to reflect the amount of gain reduction induced by the effect. The VU meter also doubles as an indicator of the Hold level on the 1969 version when signal is not passing through the effect.

Installation is easy, and the authorization lives on an iLok dongle. The plug-in is cross-platform, and both TDM and RTAS versions are available (AU and VST versions are in beta development). Also, the plug-in works very well with the Mackie Control Universal surface.

In Action

The plug-in sounded great on vocals, guitar, bass, and even on the stereo bus (I actually mastered a couple of tracks through it with great success). But TG12413 really shows its colors on drums, whether on individual mics, the room mics, or the entire drum bus. I went back and forth between the 1969 and 2005 versions a lot, and they're both stellar, but for different things. In addition, switching between Comp and Limit modes on each version yielded widely varying results, because not only does the ratio change from 2:1 (Comp) to 8:1 (Limit), but the attack times and recovery settings also change between the two modes.

I found the 1969 in compressor mode at low gain-reduction levels and a slow recovery to be the perfect thing on a jazz drummer's bus. The kit just jelled in that way that makes it sound like a single unit rather than a bunch of separate instruments. On a rock kit, I slammed the room mics through the 2005 with a fast recovery and got the most musical pumping and breathing I have ever achieved. A/B'ing this against my beloved UAD-1 Fairchild plug-in, I could hear distinctive character traits in each, and both sounded fantastic, but I definitely liked TG12413 better for that particular application. The Abbey Road/Chandler plug-in tightened up the low end in a way that was both punchy and warm, while getting rid of some of the boominess on the raw tracks.

Abbey Road Studios and Chandler Limited's exquisite software version of the classic EMI limiter sets a new standard for hardware modeling. The plug-in comes in two different versions: 1969 (left) and 2005 (right).

Model Student

I also had the distinct pleasure of A/B'ing the plug-in against the original hardware retooling of the Abbey Road classic—the Chandler Limited TG1 Limiter, which Wade Goeke was kind enough to lend me for this review. Although there were subtle differences here and there in the way that the hardware unit and software plug affected the sound of the material, it was shockingly close, and I'd be hard-pressed to say which one I liked better. The TG1 is an excellent piece of gear, but it retails for $4,000 (for two channels). The plug-in, on the other hand, has virtually unlimited instances in Pro Tools LE 7.3.1 on a dual 2 GHz Apple G5 with 4 GB of RAM, so the fact that it sounds so similar to the hardware unit is mighty impressive.

In a nutshell, I think the TG12413 plug-in sounds amazing. If $450 for a compressor plug-in with only four controls on it seems expensive to you, consider this: not only are there two versions, but each version has two settings that sound completely different, so in a sense it's four compressors in one. This kind of value is possible only in the digital age, and I commend Chandler Limited for making its awesome sounds available to the budget-limited DAW community.

Speaking of which, keep an eye out for Chandler/EMI/TG EQ plug-ins, which are in beta release as of this writing. If they sound anywhere near as good as the limiter plug-in, you'll definitely want to pick those up as well. EM

Value (1 through 5): 5
Abbey Road Studios/Chandler Limited
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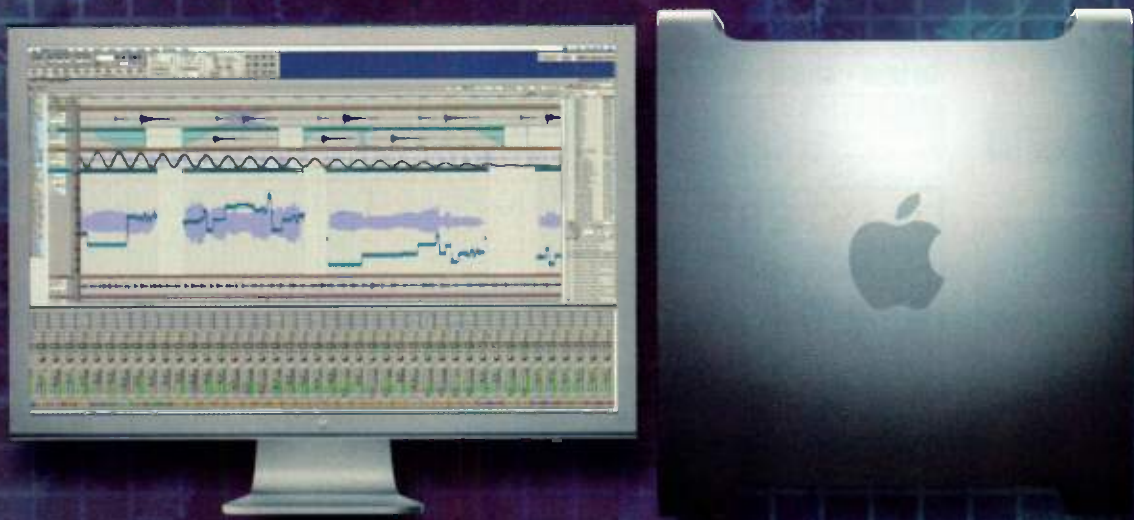

8-Core Intel Processing

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.

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Waves native processing

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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!

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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.

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Montreal Pop

By Diane Gershuny

Synthetic and acoustic collide on WhiteRoom's CD.

At the core of the band WhiteRoom is the duo of Alex Dray and Eddy Silva, who started playing together during high school in Montreal, Canada. The pair drifted in and out of bands on the Montreal scene, trying out various musical directions and influences. Five years later, they "gave up natural light and hunkered down in a claustrophobic basement bunker," says Dray. "To call it a studio at that time would have been delusional, since it was essentially a bare room with a computer running Steinberg Cubase 5, a sound card, and two pairs of $20 headphones."

Dray and Silva invited in musicians, including horn and string players and a classical guitarist. They blended those instruments and a range of inventive synthetic textures and sound design into more rock-oriented rhythm tracks. The result? A sound that they call "neoromantic pop," which is described on the band's Web site as indie rock that drifts "fluidly between Latin jazz, urban beats, haunting trip-hop, and romantic cinema soundscapes."

The band's debut CD, *WhiteRoom* (Freeworld Records, 2006), was recorded on a Dell Pentium III computer with an Event Gina 16-bit sound card running Cubase 5. Dray and Silva consider themselves "children of the digital domain" and therefore had no inclination to track to analog tape. "We disagree fundamentally with audiophiles who swear that tape-based systems, tube compressors, and vinyl recordings have a 'soul' that is lacking in digital setups," Dray says. "Of course, we have some beautiful pieces of analog gear: a Roland Juno 60, a Roland JX-3P, and a Korg MS-20 are prized possessions. We used the Roland Space Echo RE-201 lavishly and with great affection. For sampling we used an Akai S6000, which has its own particular D/A A/D coloration and warmth."

EMILY GAN

***WhiteRoom*/Dray and Silva (left and right, respectively) in their studio.**

RIFFS

WhiteRoom

Home base: Montreal Canada
Sequencer of choice: Steinberg Cubase 5
Primary vocal mic: M-Audio Luna
Web site: www.whiteroom.ca

All the instruments were tracked with a stereo pair of Shure SM81s and an AKG 414 BULS—in a room with low ceilings and very little acoustic treatment. "We had to close-mic everything to reduce the effect of flutter tones, flanging and phasing, and just about every other nasty room noise that you can imagine," Dray says. Vocals were captured primarily with an M-Audio Luna mic.

Once the songs entered the production phase, they often went in unforeseen directions. "On the song 'Ender,' Shen Qi was invited to improvise on erhu [a 2-stringed Chinese instrument that's bowed like a violin]. We were left with about an hour of material from which we created a one-minute solo. The solo itself contains over 100 different samples from the original improvisation, edited together to make a fluid musical line," Dray says.

Live drums were generally tracked last. Dray and Silva upgraded to the 8-input, 20-bit Layla card and borrowed some mics, including an AKG D112, a few more SM81s, and some SM57s. "Almost every song was recorded and produced initially to sequenced drums," Dray says. "Although it made performance difficult and adjustments were [often] made after tracking was finished (but with no Beat Detective–type editing, thank you very much), there are many instances when we chose to double-layer the original sequencing with live drums over the top."

Dray sums it up: "In retrospect, we'd like to think that that was our goal for the whole album—to move casually in and out of electronic, synthetic, and acoustic instrumentation without ever making statements like 'This is the electronica bit!' and 'This is the acoustic bit!'" EM

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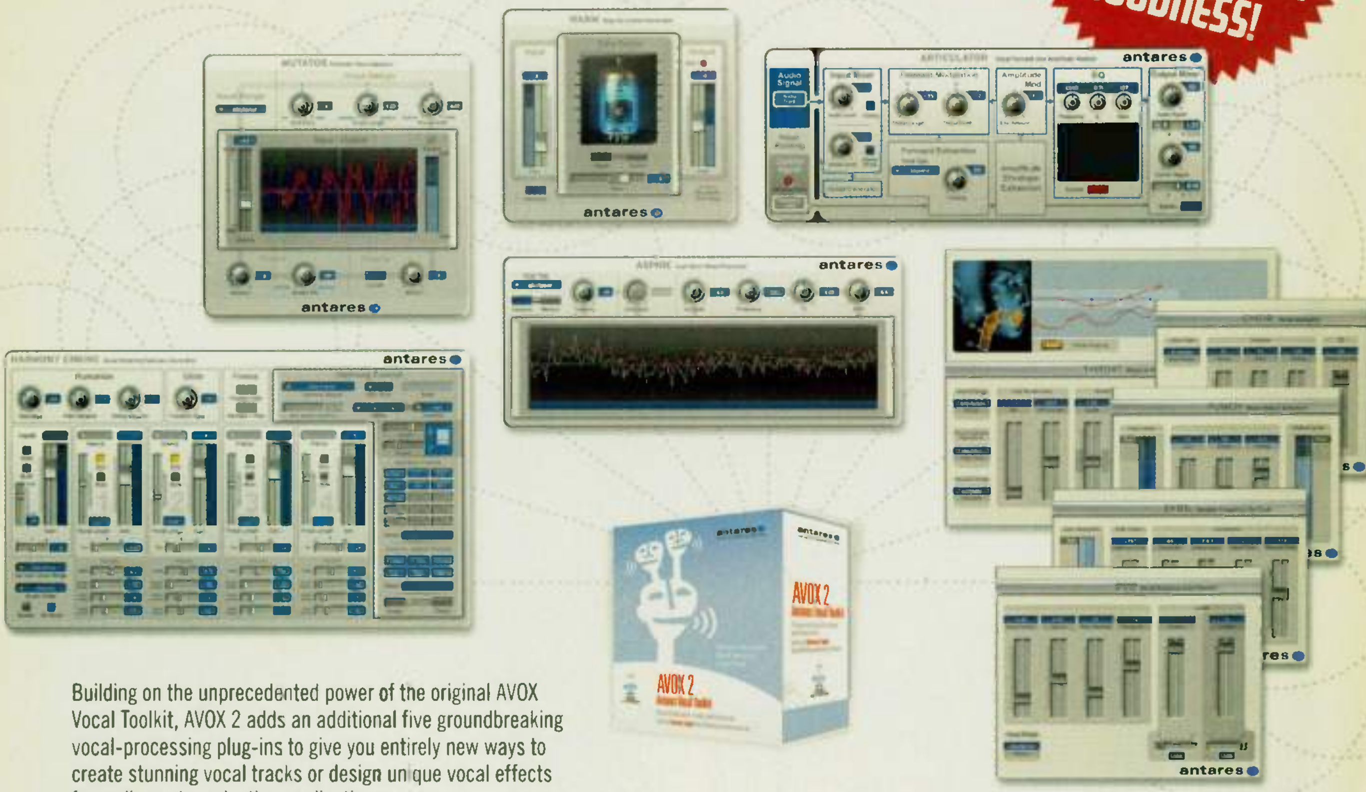

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