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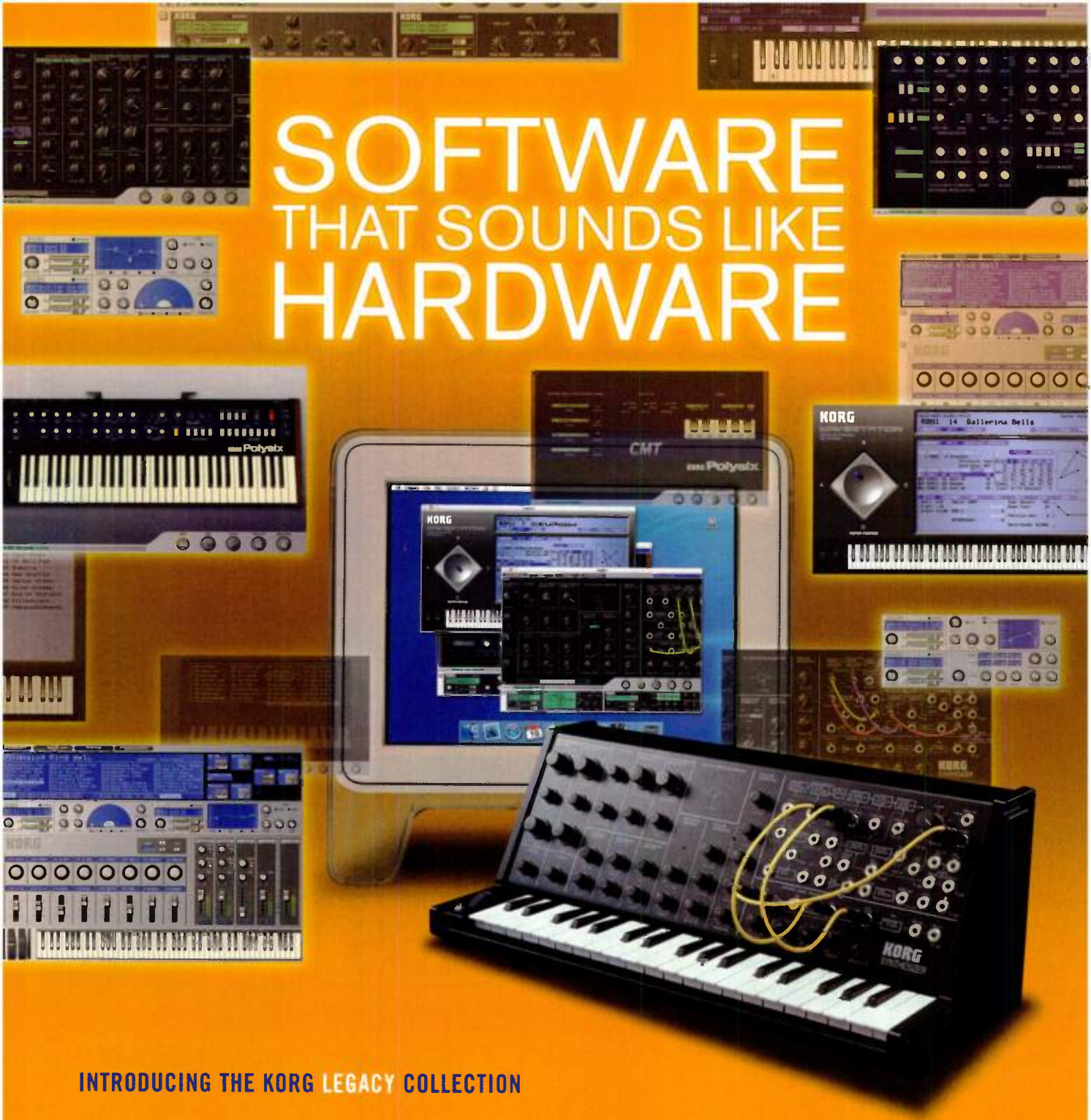

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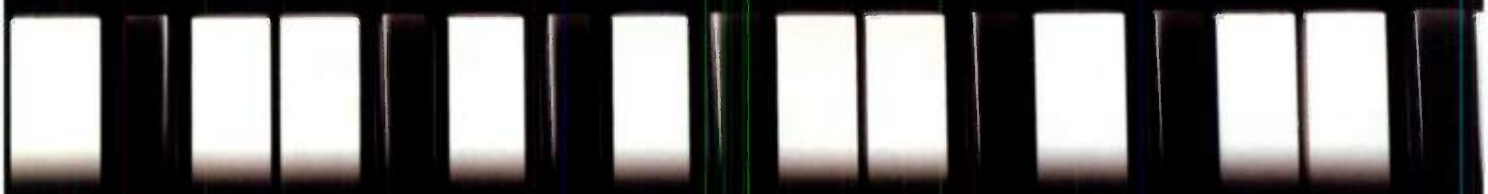

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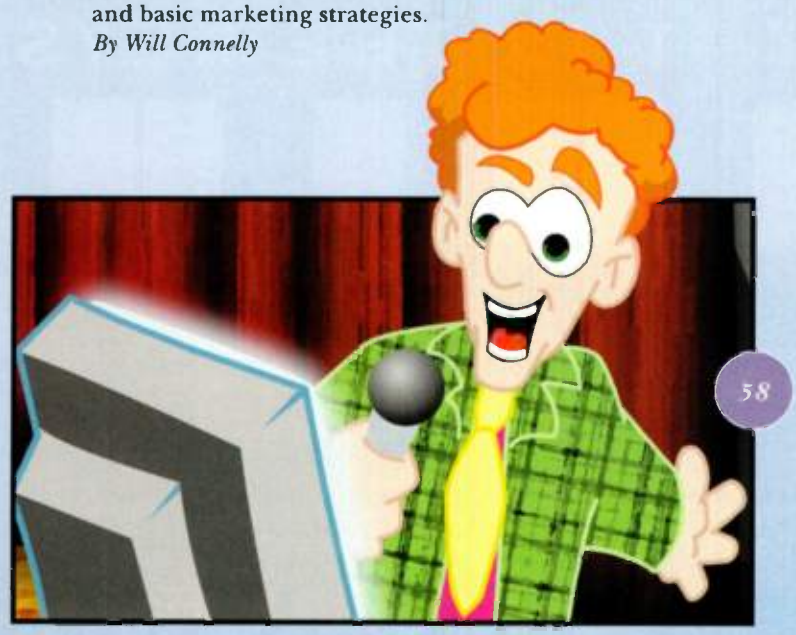

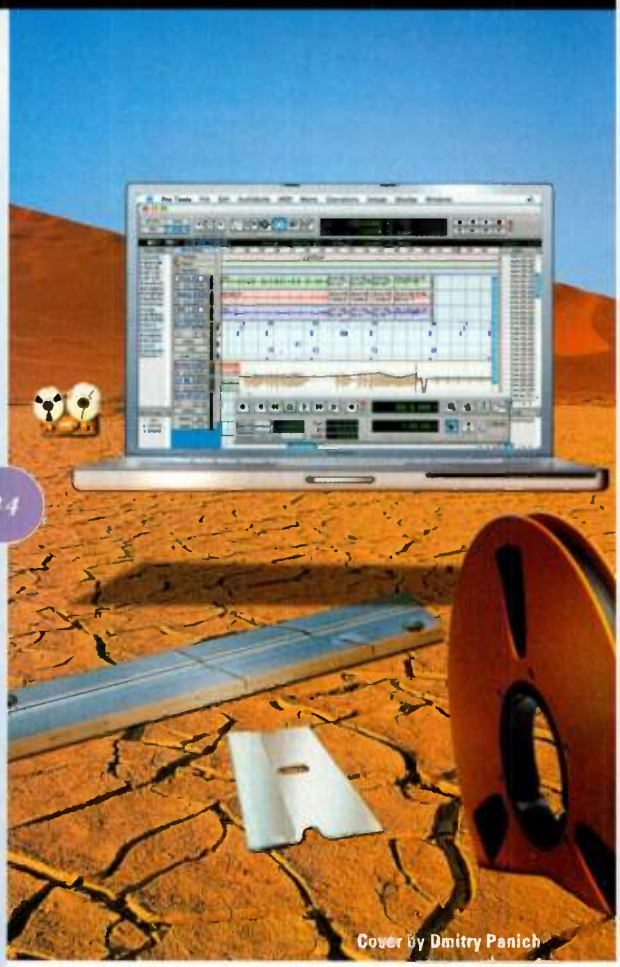

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Electronic Musician® (ISSN 0884-4720) is published monthly except semimonthly in July by PRIMEDIA Business Magazines & Media Inc., 9800 Metcalf Ave., Overland Park, KS 66212 (www.primediabusiness.com). This is Volume 20, Issue 7, June 2004. One-year (13 issues) subscription is $40; outside of the U.S. it's $75. POSTMASTER: Send address changes to *Electronic Musician*, P.O. Box 1929, Marion, OH 43306. Periodicals postage paid at Shawnee Mission, KS, and additional mailing offices. Canadian GST #129597951. Canada Post International Publications Mail Product (Canadian Distribution) Sales Agreement No. 40597023. Canada return address: DP Global Mail, 4960-2 Walker Road, Windsor, ON N9A 6J3.

Electronic Musician
JUNE 2004 VOL. 20, NO. 7
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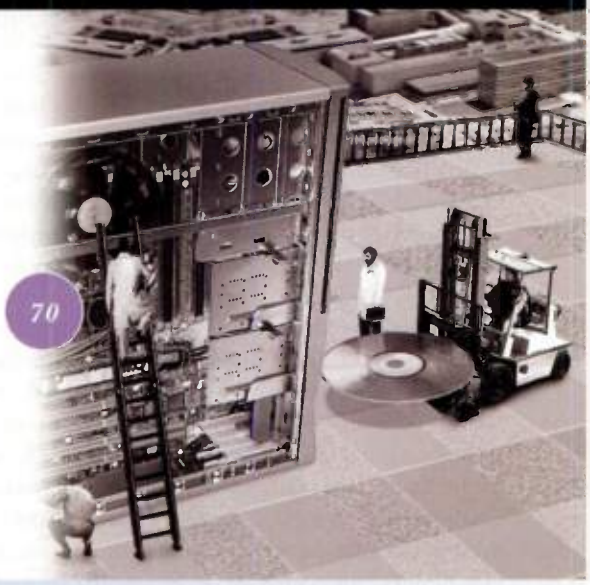

REVIEWS

Producing Good Music for Bad Singers


ANTHONY PIDGEON

One reason that choosing topics for EM stories is such a challenge is that our readers have such a diverse range of production experience and musical taste. In order to help as many electronic musicians as possible, we usually focus on genre-neutral, bread-and-butter subjects. For instance, in this issue we have Nick Peck's "Making the Cut" about audio-editing techniques; Maureen Droney's "Sweet and Low," in which top producers discuss how they record electric bass; Brian Knave's "Recording Musician" on mixing drums; and Kevin Smith's "Desktop Musician," which explains how to get your computer ready for service as a DAW.

Once in a while, though, we like to offer a story you probably didn't anticipate. We have a good one this month with "CD+G=$," in which author Will Connelly gives us the inside scoop on producing graphics-enhanced compact discs for karaoke.

Admittedly, karaoke production is an unusual field for many musicians. Most of us think of ourselves as original and creative artists, and karaoke is all about covering the hits. Besides, many karaoke singers are so bad that we'd rather listen to a catfight, and some musicians have a hard time accepting the idea of producing music for such a purpose. But a lot of people love to sing karaoke, and if you can make the right business connections and deliver the goods, producing the needed CD+G discs can be another modest revenue stream for your studio. So check it out; I assure you, reading the story is painless, and you will not be required to listen to bad singing!

• • • • • • • • • •

In the "they come, they go, and the beat goes on" department, long-time associate editor David Rubin has left our staff and returned to the freelance life. David has been a linchpin of our team for the past seven years, and we hated to lose him. Ironically, you may see his byline in EM more than in the past because he now has more time to write articles.

Fortunately, we got lucky and were able to add not one but two outstanding new associate editors: Len Sasso and Rusty Cutchin. Len Sasso needs little introduction in these pages; his byline has appeared in more than 80 EM articles, mostly about products and techniques for desktop music production. It should be no surprise, then, that Len has taken over David Rubin's "Desktop Musician" column.

Rusty Cutchin is also a familiar name to many in the music industry because he is a former editor of *Home Recording* magazine. Rusty is a fine writer and researcher who has a strong audio-engineering background and is equally at home recording with hardware and software.

I could not be more pleased with our two new "bandmates," and I am confident that our readers will be just as delighted with the results.

Steve Oppenheimer
Editor in Chief

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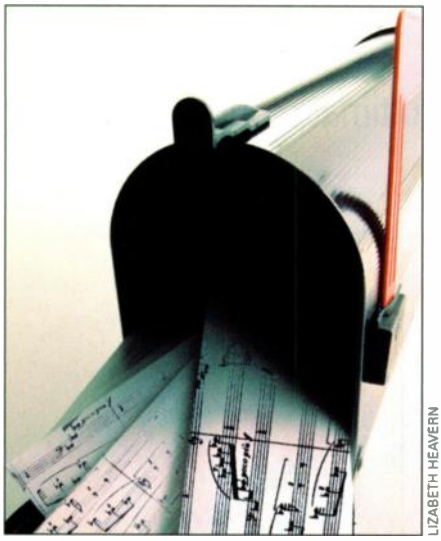

LIZABETH HEAVERN

MIX WELL

Kudos to author Maureen Droney and the editors of **EM**! Unlike the elusive article about mastering engineers ("Masters on Mastering," September 2003), Ms. Droney's article ("Mixing Strategies of the Pros," April 2004) offered some great insights. When it comes to mixing, it's so easy to get tunnel vision while thinking, "This is how it should be done."

I would like to thank the three recording engineers whom Ms. Droney interviewed for sharing their thoughts. I hope to fill their shoes someday!

Cary King
Creative Sound Images
Fort Wayne, Indiana

GREAT PERFORMANCES

Sean Carberry's "Tracking the Elusive Vocal" (April 2004) perfectly summed up what it takes to get a great vocal performance: making the vocalist feel confident, setting up a variety of mics and switching between them as the vocalist warms up, and helping the vocalist keep the vibe.

Many of my clients gripe about engineers who have no patience or diplomacy and leave no room for experimentation. One guy evidently tells his clients, "It's your money, dude!"

I wish I had had your article years ago. It would have saved me a lot of breath in explaining to new engineers how to "track the elusive vocal."

Joe Danger
Joe Danger Records

ACOUSTIC SOLUTIONS

Ethan Winer's piece on small-room acoustics ("A New Approach to Personal-Studio Acoustics," April 2004) nailed the problem. When Mr. Winer talked about the hideous problems in bass perception caused by a small room, I recognized my own bedroom studio and the last dozen mixes I've done there.

However, Mr. Winer did not nail the solution. The acoustics in my bedroom studio suck at 80 Hz. What do I do about it? Mr. Winer's only concrete suggestion is to check out stereo mixes in my car. Okay, but I can't mix in my car, so this is nothing like a complete solution.

I understand the problem, but what next? I suggest that this should be the subject of a detailed follow-up piece.

Richard Hunter
via e-mail

I generally agree with Ethan Winer's points in his article "A New Approach to Personal-Studio Acoustics" in the April 2004 issue of **EM**. However, I must challenge his old-school thinking with respect to the use of subwoofers and equalization.

You can't tune a world-class monitoring system to a smooth frequency response using only wideband absorption panels, as Winer suggests. Proper room treatment includes *some* wideband absorption and diffusion. In some cases, calculating the room's length, width, or height for axial-mode trapping will treat the primary mode and its octaves. This is important for designing any control room.

You can achieve a full-frequency and accurate monitoring system with the following steps. First, carefully position the subwoofers and speakers to attenuate the first- and third-width axial modes in the center of the room where the engineer will sit. Second, choose highpass and lowpass filter frequencies to fill in dips where appropriate. Third, "polish" the system with a high-quality parametric equalizer using a 1/24-octave analyzer. Finally, make sure the system is correctly phased using 360-degree phase control. Mixes from this environment should travel extremely well from playback system to playback system.

I suggest that Mr. Winer switch from an RTA measuring method to a time-frequency domain FFT method, such as the Meyer SIM system or the SIA SMAART system. Measure a monitor system at the sweet spot in a control room before treatment and then after treatment to see the limitations of using only wideband absorption panels. A good dual-sub system and parametric EQ can show how wonderful the marriage of all the elements of a control-room monitoring system can be.

Carl Tatz
Carl Tatz Design

I was wondering if you could clarify something for me. On p. 34 of the April 2004 issue, in the sidebar entitled "Acoustical Physics," the formula to determine the resonant frequency between two walls reads as follows:

$$\text{Frequency} = \frac{1{,}130}{\text{Feet} \times 2}$$

I'm wondering what the mathematical

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function is between [1,130] and [Frequency]. Wouldn't the resonant frequency also depend on the material the wall was made of? Thanks!

Jamie Harding
via e-mail

Author Ethan Winer replies: The most effective solution for low-frequency response errors in small rooms like Richard Hunter's is to add a sufficient number of broadband bass traps. Bass traps reduce reflections that skew the low-frequency response, and they reduce low-frequency modal ringing. Modal ringing is not unlike reverb; it contributes to a muddy low end and makes it difficult to distinguish which notes are being played by bass instruments.

Carl Tatz refers to world-class monitoring, but my article addressed getting good results in the small-sized rooms commonly used for personal studios. I wanted to explain why small rooms have unique problems that can't be solved using the traditional methods.

A large control room with a high ceiling would be a great place to build custom bass traps and line the entire rear wall with QRD diffusors. But in small rooms, the walls are generally too close to the mix position to benefit from diffusion. Moreover, when every wall is nearby, the room modes are less important than the comb filtering that occurs at all low frequencies. Reflections from close walls are also very strong, and response nulls of 30 dB or more are typical. You simply can't correct such large aberrations with an equalizer. That's why broadband absorption makes more sense (for small rooms) than trapping that targets specific frequencies.

I agree completely with Mr. Tatz regarding room-measurement software. I use Acoustisoft .com's ETF program, which is excellent and affordable. However, my goal was to show readers how they can get a fully accurate measurement of their room's low-frequency response using tools that they already own.

The formula Jamie Harding inquires about is very simple, but its formatting was scrambled. It should have read:

$$Frequency = \frac{1,130}{Feet \times 2}$$

As I explained in my article, the natural resonant frequency between two parallel surfaces is related to the distance between them. Reflectivity is also an important consideration. In order to foster resonance, the surfaces must be sufficiently rigid to reflect the waves. Cardboard "walls," for example, would allow low frequencies to pass right through. In truth, all room boundaries reflect, absorb, and pass sound waves—all at the same time—in varying amounts depending on the frequency.

PLAY YER MIDI GUITAR

In the March 2004 "Square One" column, "Tricks for Tracks," author Jim Aikin writes that there is usually no reason to use sequencer tracks that can transmit data over multiple MIDI channels. I must point out that such a feature is a common editing tool for MIDI guitarists and other musicians who are not oriented to using keyboards.

Because each string of a MIDI guitar controller can transmit on its own MIDI channel, a multichannel MIDI track preserves discrete assignments of events from the controller's MIDI data stream. That's especially handy for Pitch Bend messages issuing from individual strings. The multichannel track retains the controller's guitarlike performance.

Tracking of stringed-instrument controllers has vastly improved, bit there are still occasional glitches. Consequently, the ability to view and edit MIDI guitar events in a single track-edit window is a huge time-saver.

Marty Cutler
via e-mail

INSPIRED REFLECTIONS

Larry the O's informative and insightful "Final Mix" column is always a great read after absorbing the rest of *Electronic Musician*. His March 2004 column ("Where Do You Want to Go in This Life?") ranks with his best.

I, too, have been drawn to music. Coming from a musically inclined family, I was inspired to follow music, as well. But other distractions and vocational attractions have pulled me in diverse directions throughout the years, and only now do I realize that that is okay. Not everyone has an immediate calling to his or her eventual profession.

I've come full circle and discovered, as so eloquently stated in Larry's article, that I now have a deeper knowledge and appreciation for all that I do in work and life. I was unable to clarify and distill my thoughts on this subject, and now Larry has put into words what I could not.

Bravo! Thank you, Larry, for shedding light on a murky subject: sometimes a jack-of-all-trades can find his or her calling after all, and ultimately bring more to the table because of his or her experiences. You have made a difference in my life through this observation, and for that you have my gratitude.

Paul Quinn
Pipedreams Recording Studio
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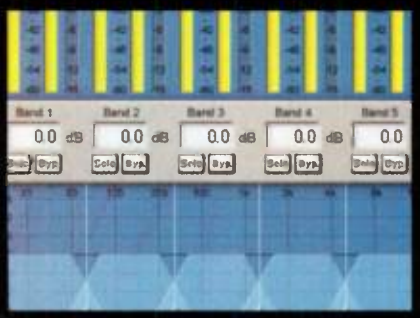

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Roland

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WHAT'S NEW

By Geary Yelton

VIRSYN CANTOR

VirSyn Cantor (Mac/Win, $349) is a new soft synth that emulates human singing. Just enter the lyrics in plain English and then play the melody on a MIDI instrument, add vibrato and breath noise for expression, and even change gender in real time. Go offline to alter the base spectra of vowels and consonants, enter note and performance data, or define the formant structure and noise characteristics of even the tiniest of phonemes.

Cantor provides 8-part vocal synthesis and supports microtuning. It runs either standalone or as a VST, RTAS, or Audio Units plug-in. To use Cantor on the Mac, you'll need at least a G4/400 MHz and Mac OS X. PC users will need a minimum Athlon XP or Pentium III/600 MHz and Windows XP. Both platforms require 256 MB or more of RAM. VirSyn Software Synthesizer; tel. 49-7240-202-956; e-mail info@virsyn.com; Web www.virsyn.com.

BEHRINGER B-CONTROL

Nothing beats real buttons, knobs, and faders for controlling sequencers, effects plug-ins, virtual instruments, and MIDI hardware. To that end, Behringer has announced its new B-Control series of USB/MIDI control surfaces. The BCF2000 ($299.99) is a tabletop device that provides eight motorized 100 mm faders. The BCR2000 ($199.99) is a similar unit with 24 rotary encoders, each with a 15-element LED ring. Both units also have eight infinitely variable push encoders (also with LED rings) that you can turn and press to send various MIDI data, as well as 20 illuminated buttons that can send MIDI on/off messages. In addition, the BCR2000 has two footswitch inputs and the BCF2000 has a footswitch input and a footpedal input.

You can easily assign every control either manually or by using the B-Control's MIDI Learn mode. Write-in fields allow you to label custom control functions, and a 4-digit LED displays abbreviated parameter names and values. The BCF2000 and the BCR2000 store 32 user presets, each with four controller groups. Behringer promises that a dedicated editor-librarian and drivers for popular software will be available soon.

Both B-Control models also operate as 32-channel MIDI interfaces with one MIDI In and two MIDI Out ports. A MIDI merge function enables you to cascade multiple B-Control units for even greater control capabilities. Behringer USA; tel. (425) 672-0816; e-mail sales@behringer.com; Web www.behringer.com.

ROLAND FANTOM-X SERIES

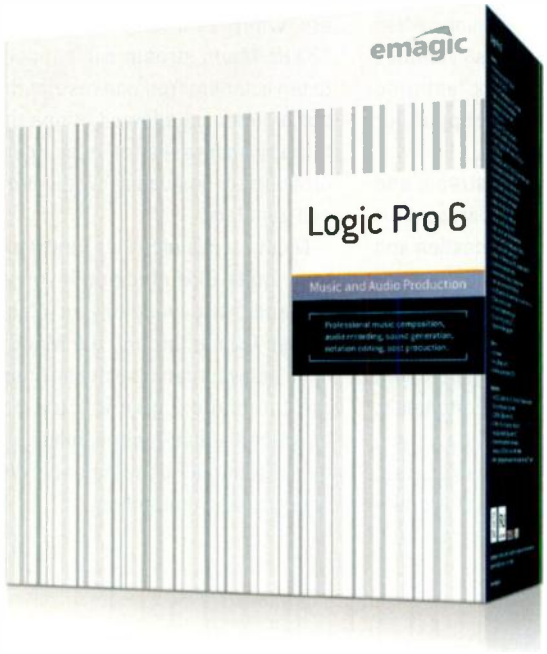
Roland has begun shipping its Fantom-X instruments. The series includes three keyboard workstations—the Fantom-X6 ($2,395), the Fantom-X7 ($2,895), and the Fantom-X8 ($3,395)—and one single-rackspace module, the Fantom-XR ($1,595). The X6 and X7 have 61- and 76-note synth-action keyboards, respectively, and the X8 has an 88-note progressive hammer-action keyboard. All four instruments offer 128-note polyphony, 128 MB of onboard waveform ROM, and significant expansion capabilities. In addition to master reverb and chorus, every Fantom-X has three multi-effects processors with COSM guitar-amp modeling, tempo-synced delay, multiband compression, and more.

The most remarkable feature of the three keyboard models is a large color LCD. Each also has 16 Aftertouch- and Velocity-sensitive gel pads, a D Beam controller, a V-Link interface, and resampling capabilities. A high-resolution sequencer lets you record and edit audio and MIDI data. The keyboards also have four SRX expansion slots and DIMM slots that accommodate a maximum of 544 MB of sample RAM. The Fantom-XR has slots for six SRX boards and tops out at 528 MB of sample RAM, but it has no sequencer, no color LCD, and fewer real-time controllers. On all models, a PC Card expansion slot gives you up to 1 GB of user sample storage.

Every Fantom-X sports a USB port for exchanging audio and MIDI data with computer software. Computer-based patch-editing software (Mac/Win) is included. Audio jacks include four ¼-inch outputs, two ¼-inch inputs, and S/PDIF I/O. Roland Corporation U.S.; tel. (323) 890-3700; Web www.rolandus.com.

emagic

All in one box.



The most widely used program of its kind for professional music creation and audio production is now more attractive than ever before. The new Logic Pro 6 contains all Emagic's superb plug-ins and software instruments, making it not only the best-equipped music production software available, but also the one with the most unbelievable price/performance ratio. Emagic's considerable expertise and experience in music composition, audio recording, sound generation, notation editing and publishing, post production, and film scoring is now all in one box – in Logic Pro 6.

www.emagic.de or www.apple.com/software/pro

DOWNLOAD OF THE MONTH

ROGUE AMOEBA NICECAST 1.5

Want to get your music on the air? Nicecast (Mac, $40), from Rogue Amoeba software, allows any computer running Mac OS X to stream audio directly to the Internet. Nicecast includes a broadcaster and a server. The broadcaster captures audio from any CoreAudio input device or from any audio application currently running on your computer and broadcasts it to the server of your choice. You can use Nicecast's built-in server (the simplest choice) to stream the audio to the Internet, or you can set Nicecast up to broadcast to any external server of your choosing. Once you start streaming, you can choose to have Rogue Amoeba add your stream to its MacStreams.com Web site. You can then e-mail the link to your friends and link to it on your own Web page.

In the simplest case, setting up Nicecast is straightforward. Launch Nicecast, choose an audio source, and click the Start Broadcast button. You might need to take care of some additional settings if you connect to the Internet from behind a firewall or through a router such as an Apple AirPort Base Station. If so, step-by-step documentation can help you out, and the process is really quite simple. I use an AirPort Base Station from behind a firewall, and I was on the air in about ten minutes.

If you choose to add your stream to MacStreams.com, you can provide information, including an icon graphic, which will appear with the stream listing. If your source software supports it (as iTunes does), that information can include the name of the currently playing track. In fact, using iTunes Shuffle and Repeat All options, you can set up a continuous broadcast of any iTunes Playlist, although iTunes hung several times when I tried to do that. Nicecast also includes an effects-processing matrix featuring 15 built-in effects as well as support for VST and Audio Units plug-ins.

The quality of your music stream and the number of listeners you can support depends on your Internet connection and the speed of your computer. NiceCast encodes audio in MP3 format on the fly before streaming. That is necessary to make the stream bandwidth manageable, but it is a CPU-intensive process. Fortunately, an encoding-quality control allows you to reduce the CPU drain. You can also set the streaming data rate to accommodate your connection speed. The rate you

choose is per listener; choosing 128 Kbps, 44 kHz Stereo with a DSL or cable-modem connection will allow one to three listeners, whereas a lower-quality 24 Kbps, 22 kHz Mono stream will support three to ten listeners. You can restrict the number of listeners allowed at one time and thereby ensure that no one experiences dropouts, or you can allow the process to self-regulate.

Nicecast is a nice idea, and it turns out to be quite CPU efficient. If you want to add audio streaming to your Web site or just go fishing through the MacStreams .com and your fan club doesn't number in the thousands, give it a try. You can download a time-limited, fully functional demo from the Rogue Amoeba Web site, www.rogueamoeba.com.

—Len Sasso

YAMAHA i88X

Yamaha recently introduced the i88X ($1,295), a 1U rackmount mLAN-based audio and MIDI interface. The i88X provides 18 simultaneous channels: 2 balanced XLR inputs (with ¼-inch inserts), 6 balanced ¼-inch inputs, 18 balanced ¼-inch outputs, stereo S/PDIF I/O, and 8-channel ADAT Lightpipe I/O. The first input can handle high-impedance instruments in addition to mic- and line-level signals, and peak LEDs are provided for the first two inputs. Two mic preamps, which were designed by the engineers who developed the DM2000 and 02R96 mixing consoles, supply switchable 48V phantom power. The i88X's master clock lets you select an internal source or an external ADAT or mLAN source. As with other mLAN interfaces, all digital audio and MIDI data travels over FireWire.

The i88X can be used either standalone or as an expansion unit for an existing mLAN system. The A/D converters accommodate 16- or 24-bit audio at 44.1, 48, 88, and 96 kHz sampling rates. The i88X includes drivers for Windows XP, Mac OS 9, and Mac OS X. Yamaha Corporation of America; tel. (714) 522-9011; e-mail infostation@yamaha.com; Web www .yamaha.com.

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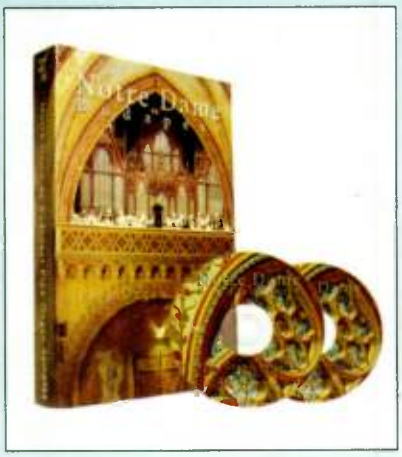
SOUND ADVICE▲▲▲▲



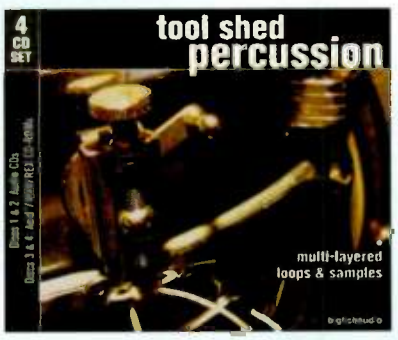
▲ SHIROKUMA

If you've always wanted an immense pipe organ but couldn't afford it, the new *Notre Dame de Budapest Pipe Organ Samples* ($499.95) is for you. At nearly 9 GB, this 2-DVD GigaStudio library is undoubtedly the most massive compilation of symphonic-organ samples ever.

The collection features the symphonic organs from two Hungarian cathedrals: the 99-stop Rieger-Kloss organ in Notre Dame de Buda and the 38-stop Varga organ in Notre Dame de Kipest, both in Budapest. All the most useful combinations of stops were recorded at 32 bits and 96 kHz and then downsampled to 16-bit, 44.1 kHz recordings. Each sample is at least 30 seconds long, and pedal notes are a full minute. The cathedrals' natural reverb was captured and stored in the release samples. A MIDI expression pedal controls the swell-box effect and crescendo, and MIDI Breath Control affects tremolo. A user's guide provides detailed registration information. Shirokuma Ltd./EastWest (distributor); tel. (800) 833-8339; e-mail sales@eastwestsounds.com; Web www.soundsonline.com.

▼ BIG FISH AUDIO

Big Fish Audio's new *Tool Shed Percussion* ($99.95) isn't your average sample library. It's a unique collection of drum-kit loops that feature fake-leather luggage, metal buckets, circular-saw blades, a propane tank, and dozens of other found instruments, in addition to traditional drums and cymbals. Odd sound sources include a metal bolt in a water-filled mixing bowl and a leaf rake scraping across a wood floor. Even some of the less radical instruments, such as African goat hoofs, bowed cymbals, and a marching bass drum played with brushes, are a little unusual. The documentation is quite complete and sometimes rather amusing; for example, an instrument called Gut Drums is described as "hand slaps on a large, round human belly."



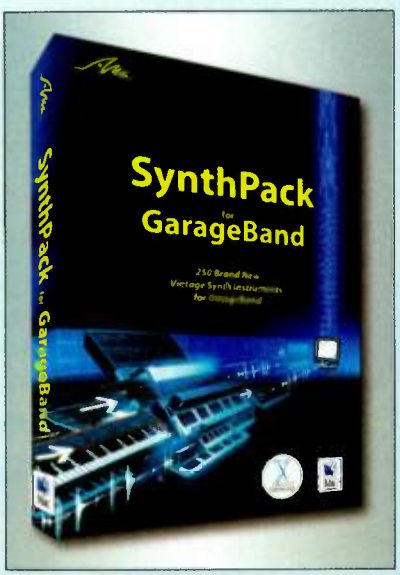
Tool Shed Percussion comes on four discs: two audio CDs and two CD-ROMs containing Acidized WAV and REX2 files. Each pair of discs contains 49 construction kits, each with a percussion-loop mix and its individual loop elements, for a total of more than 600 loops. Disc 2 also supplies a large assortment of individual hits ranging from kick drums and cowbells to pitchforks, tape measures, and T-squares. Big Fish Audio; tel. (800) 717-FISH or (818) 768-6115; e-mail info@bigfishaudio.com; Web www.bigfishaudio.com.

▼ AMG

Thanks to a large potential user base, several developers have begun offering soundware for Apple Soundtrack and GarageBand users. British sound designer AMG is one company supplying GarageBand instruments as well as Apple Loops. *SynthPack for GarageBand* (£30) adds more than 150 vintage synth sounds to GarageBand's arsenal of software instruments. Sampled analog classics range from the Memorymoog and ARP 2600 to the Roland MC-202 and Oberheim Matrix-12.

At least seven AMG sample discs are available in Apple Loops format. *2Step Ahead* (£60), produced by Leo Cavello, is an assortment of dance-oriented samples categorized as Groove Kits, Processed Beats, Keys & Synths, Vocals, Fx, and One Shots. Each Groove Kit supplies a folder of loops you can organize into a complete mix. Another Apple Loops disc is Luke Cage's *Ultramagnetic Beats* (£60). Its 58 hip-hop grooves are also broken down into their constituent loops. Every folder contains loops of the bass, the drums, the melody, and the full mix, and some supply additional or alternative parts.

The sample discs are available in other formats, too. Prices are in British pounds only and include worldwide shipping. You can also order AMG samples from Apple's online store at the current exchange rate. AMG; tel. 44-125-271-7333; e-mail matt@amguk.co.uk; Web www.samples4.com.



IR-1 Convolution Reverb

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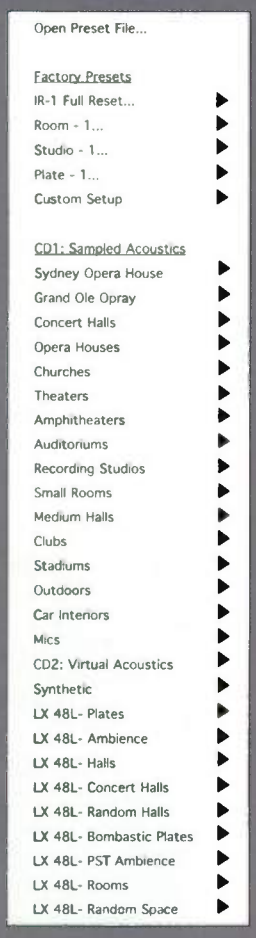

IR-1 Offers the best features of Sampling and Traditional Reverbs without the compromises. IR-1 is the first Convolution Reverb with "classic parameters" that affect the convolution, not just "filter the result".

IR-1 comes with an extensive library of over 60 carefully sampled impulse response files that recreate both the acoustics of real spaces and the sounds created by classic electronic devices. Waves worked with leading acoustician professor Angelo Farina to develop new and innovative techniques to create a library that establishes new benchmarks for clarity and accuracy in sampling reverbs.

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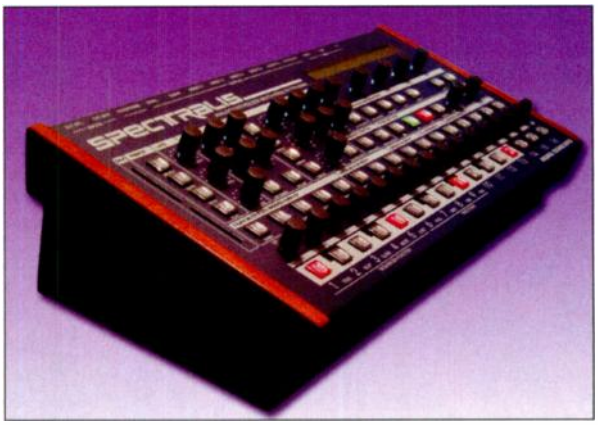
RADIKAL TECHNOLOGIES SPECTRALIS

German manufacturer Radikal Technologies has announced the Spectralis ($2,500), a 5U rackmountable or tabletop groove box that combines 48-note DSP synthesis and sampling, monophonic analog-modeling synthesis, real analog filtering, 32-track polyphonic step sequencing, and matrix-style drum sequencing. The analog-modeling section has four digital oscillators with ring modulation, FM, and continuously variable waveshaping, as well as a 4-pole lowpass and a 2-pole multimode analog filter. A programmable analog fixed filter bank provides one lowpass, one highpass, and eight bandpass filters; 20 VCAs handle complex routing between the filters. The analog electronics can process external audio signals and use them as modulation sources.



The Spectralis can resample synthesizer voices and patterns, allowing you to play analog-modeled sounds polyphonically. In addition to sampling, the DSP section will accommodate new synthesis algorithms as they become available, according to the manufacturer. Drum and synth samples are stored in 32 MB of onboard ROM, and 128 MB of sample RAM comes standard. The Spectralis provides two Smart Media card slots, and a USB 2.0 port allows it to exchange data with a computer. Additional I/O includes two audio inputs, two analog-synth outputs, six DSP-synth outputs, and three MIDI ports. Radikal Technologies; tel. (201) 836-5116; e-mail info@radikaltechnologies.com; Web www.radikaltechnologies.com.

NATIVE INSTRUMENTS GUITAR RIG

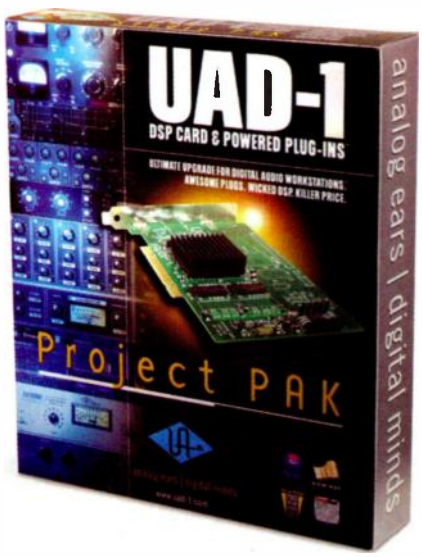
Native Instruments' Guitar Rig (Mac/Win, $499) is a virtual modular studio for electric guitarists. Guitar Rig's onscreen rack contains assorted computer-modeled analog electronics—classic and modern guitar amps, speaker cabinets, mics, stompboxes, and tape decks—that you can combine in any order. Guitar Rig comes with the Rig Kontrol, a hardware unit that supplies an integrated DI box, four footswitches, and a footpedal for real-time parameter control.

Guitar Rig's tube-modeling technology emulates the Mesa/Boogie Rectifier, the Fender Twin Reverb, and the Marshall 50W Plexi. Virtual mics and cabinets supply 5 mic types, 4 recording positions, 15 speaker types, and as many as 8 speakers at a time, each with its own mic configuration. You can split the signal into two paths, process them independently, and then mix or crossfade them. Virtual Tape Deck modules are on hand for recording, overdubbing, and learning songs. There's also a tuner, a metronome, and a library of audio loops.

Guitar Rig operates standalone or as a VST, RTAS, DX, or Audio Unit plug-in. On the PC you'll need a Pentium/700 MHz or Athlon/1 GHz, Windows XP, and 128 MB of RAM. On the Mac you'll need a G4/733 MHz, Mac OS X 10.2.6, and 128 MB of RAM. Native Instruments USA; tel. (866) 556-6488; e-mail info@native-instruments.com; Web www.native-instruments.com.

UNIVERSAL AUDIO UAD-1 PROJECT PAK

Universal Audio has announced the UAD-1 Project Pak (Mac/Win, $499), an entry-level bundle comprising the cross-platform UAD-1 DSP expansion card and 15 version 3.5 plug-ins. The software supplies users with many of UA's most popular dynamics and effects processor emulations, including 1176SE, Pultec EQP-1A, RealVerb Pro, CS-1, and Nigel. The CS-1 suite models a complete channel strip consisting of the EX-1 EQ and compressor, DM-1 delay modulator, and RS-1 reflection engine. Nigel, UA's guitar-processing modeler, furnishes amp and cabinet simulations, echo and delay effects, a modulation filter, a phase-shifter, and more. You can expand the UAD-1 Project Pak by purchasing additional software plug-ins, such as LA-2A, 1176LN, Fairchild 670, Dreamverb, or Cambridge EQ.



The UAD-1 card is compatible with PCI and PCI-X expansion buses. It can run alongside other DSP cards, and you can install as many as four UAD-1s in a single computer. UAD-1 plug-ins support VST, DX, Audio Units, and MAS hosts in Mac OS 9, Mac OS X, and Windows 98SE, 2000, ME, and XP. For Pro Tools users, Project Pak also includes a UAD-1-specific version of FXpansion's VST-RTAS Adapter. Universal Audio; tel. (866) 823-1176 or (831) 466-3737; Web www.uaudio.com.

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ADOBE AUDITION 1.5

Approximately one year ago, Adobe Systems acquired Cool Edit Pro 2.1 from Syntrillium Software. Now Adobe has announced a new edition of the popular audio editor. Along with numerous other enhancements, Audition 1.5 (Win, $299) adds support for ReWire and VST plug-ins to its already extensive list of editing, effects-processing, mixing, and mastering features. The program's post-production capabilities now include

video thumbnails, which make it easier to sync sound to picture, and support for native DV, MPEG, and Windows Media Video (WMV). Now you can correct pitch and eliminate pops and clicks more quickly and easily. The new Center Channel Extractor can accentuate or attenuate certain instrument or vocal parts. Spectral View lets you isolate a sound by its frequency and location in time so you can apply edits.

Audition 1.5 comes with 500 new 32-bit music loops in various genres, bringing its bundled library to more than 5,000 loops. Additional new features include integrated CD burning, clip time-stretching, user-definable keyboard-shortcut sets, and 20 factory-programmed sample sessions that you can customize to suit your needs.

Upgrades from Audition 1.0 or Cool Edit Pro 2 are $69; upgrades from previous versions are $169. Audition 1.5 requires a minimum Pentium/400 MHz, Windows 2000 or XP, 64 MB of RAM, 55 MB of disk space, and DirectX 9.0 for video import. Adobe Systems; Web www.adobe.com.

TASCAM GIGASTUDIO 3

Tascam has announced a complete rewrite of its sampler program for Windows, GigaStudio 3 (GS3). Available in three versions—Orchestra ($599), Ensemble ($369), and Solo ($199)—GS3 sports an updated GUI and supports VST effects plug-ins; 24-bit, 96 kHz samples; and ReWire connectivity. GS3 is a true multichannel sampler that can simultaneously record as many as 32 live audio inputs. An expanded 128-channel mixer section offers compression and 4-band EQ on each channel. Quick Edit allows you to modify envelopes, filters, and other parameters in real time. GigaStudio's MIDI capabilities have been expanded, and you can load all the instruments you want on a single MIDI channel.

In addition to a new sampled piano called GigaPiano 2, GS3 includes GigaPulse SP, a convolution engine that models physical environments, instrument resonances, and even microphones. GS3 Orchestra also includes GigaPulse Pro for importing impulse responses, as well as Sample Translator for importing foreign formats. GS3 Solo and Ensemble are 96- and 160-note polyphonic, respectively; GS3 Orchestra's polyphony is limited only by hard-drive and processor speed. Windows XP is required to run GigaStudio 3. Upgrades range from $50 to $375. Tascam; tel. (323) 726-0303;Web www.tascam.com.

PRIMERA TECHNOLOGY BRAVO II

For anyone publishing recordable CDs or DVDs in quantity, Primera Technology has announced the Bravo II Disc Publisher (Mac/Win). Two models are available: the Bravo II CD Publisher ($2,195), with a 52× CD-R recorder, and the Bravo II DVD Publisher ($2,695), with a Pioneer A07 DVD±R/CD-R recorder that burns DVDs at 8× and CDs at 24×. Like the original Bravo, the Bravo II offers hands-free, automated duplication and printing of as many as 25 CDs or DVDs in succession, and its Kiosk mode doubles that capacity. A robotic arm transports discs to the built-in disc recorder and then to the integrated printer, which prints in full color directly on their surfaces. Primera says that compared to the original model, the Bravo II has a 10 percent faster disc-production rate and a 15 percent faster printing speed while doubling print resolution to 4,800 dpi.

The Bravo II includes Charismac Engineering's Discribe 5.0 duplication software and Magic Mouse Production's Discus printing software for the Mac. For the PC, it includes Sonic Solutions' PrimoDVD 2.0 and the SureThing CD Labeler Primera Edition. Windows-based networking software is available as an option. Bravo II is compatible with any PC running Windows 2000 or XP that has a USB 2.0 port, and with any Mac running Mac OS X. Primera Technology; tel. (800) 797-2772 or (763) 475-6676; e-mail sales@primera.com; Web www.primera.com. ❦

BM Series

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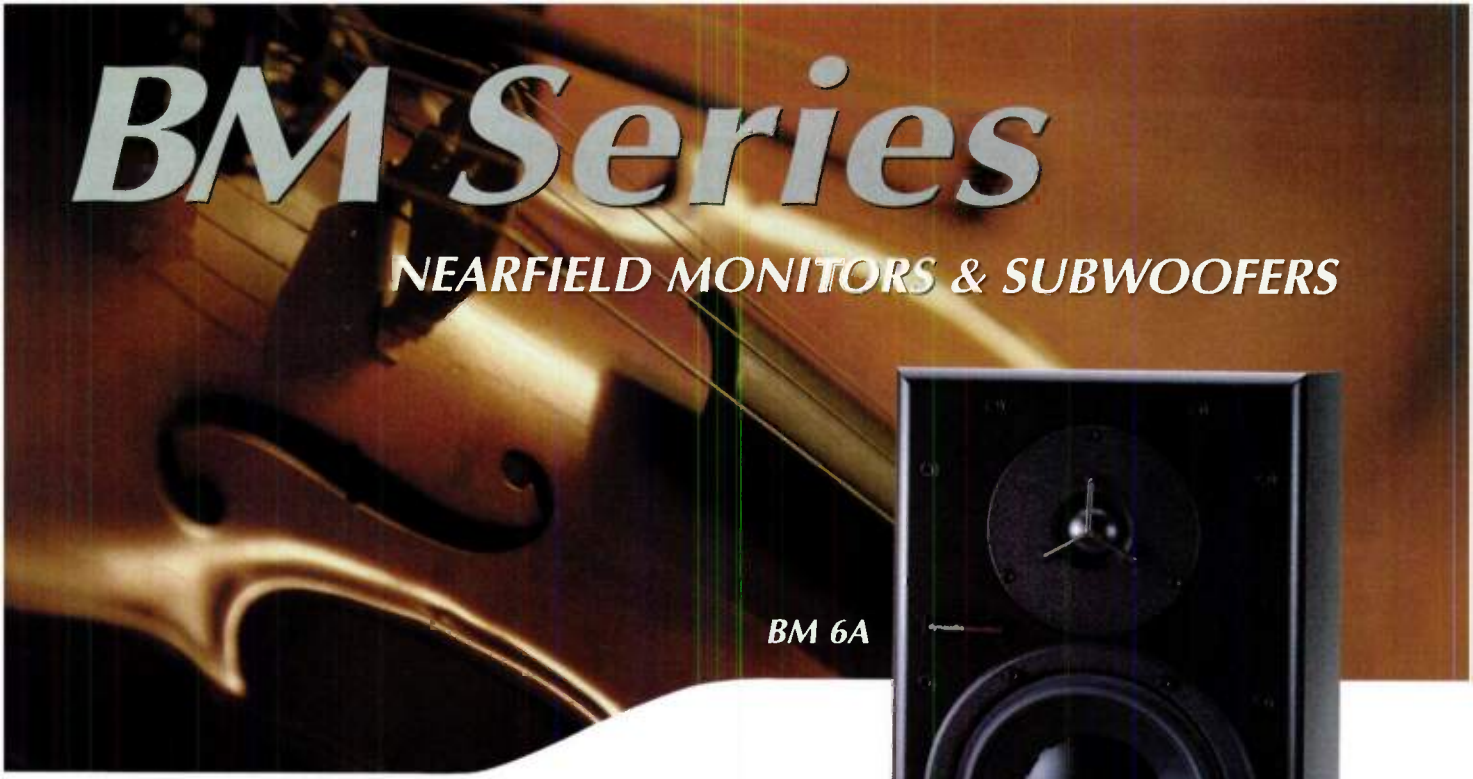

POWERFUL AND EXCELLENT MONITORS

The BM Series monitors - both passive and active - are clean, powerful and accurate. The transparent sound ensures the most realistic listening conditions for a wide variety of monitoring environments. Comprised of renowned Dynaudio driver technology, BM monitors have become the standard when it comes to high performance. In combination with the new BM10s and BM12s subwoofers, BM Series nearfields are also perfectly suited for multichannel applications.

Main Features for the BM series

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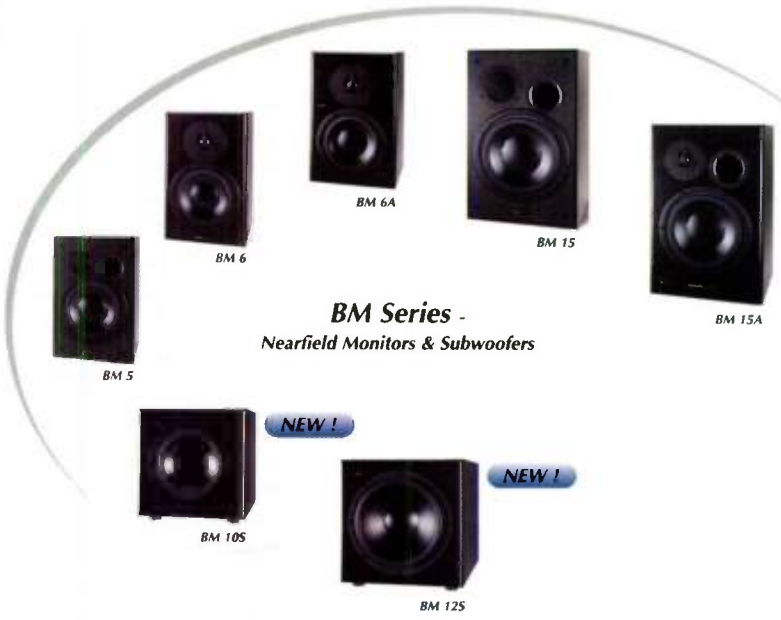

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Sweet & Low

LAURA WILLIAMS

Advice from three top pros for recording great-sounding electric-bass tracks.

By Maureen Droney

In an ideal world, recording master-quality electric-bass sounds in your studio would be a piece of cake. You'd simply connect the bass to a DI and maybe to a preamp, adjust gain to avoid overload, slap on a compressor, press the record button, and off you'd go. But reality is more complicated. You have to take into account the quality and sonic signature of the outboard gear, the condition of the instrument, the skill of the player, the musical style, the acoustics of the control room, whether the sound of an amp will be added into the equation (and if so, what mics will be used), and so forth.

To help sort through these variables and others that are integral to bass recording, I consulted a panel of experts: three veteran engineer-producer-musicians. All have labored in the trenches of both home and pro recording and have plenty of useful advice to offer on capturing great bass sounds.

MEET THE PANEL

Quadruple-threat Dusty Wakeman (dusty@maddogstudio.com) is a bass player, an engineer, a producer, and the owner of Mad Dog Studios in Burbank, California. A Texas native who migrated to Los Angeles, Wakeman worked in recording-system sales and studio design at West L.A. Music before scoring his first big break—engineering for producer Pete Anderson on the cow-punk classic *A Town South of Bakersfield*, a compilation that featured Dwight Yoakam, among other notables.

Wakeman went on to engineer 12 albums for Yoakam and to produce and engineer records for artists such as Lucinda Williams, Jim Lauderdale, Rosie Flores, Me'Shell Ndege'Ocello, The Bonedaddys, Reacharound, and the Lonesome Strangers, all the while continuing to tour and record as a bassist with artists such as Jackson Browne and Michelle Shocked.

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Sweet & Low

Carmen Grillo (carmen@carmengrillo.com) is a singer, songwriter, and guitarist who teaches audio recording as a mentor for Recording Connections in Los Angeles. Rita Coolidge and Bill Champlin are just two of the artists he's toured and recorded with; he also spent ten years as a member of the legendary funk group Tower of Power. During his tenure with TOP, Grillo taught himself engineering and began recording the band's rehearsals and album demos. Word got around, a home studio resulted, and soon he was as much in demand for his skills as a producer and engineer as for his musicianship. The studio is now called Big Surprise Music, and it's where Grillo did most of the work on *Everybody on the Bus*, the solo album he produced for TOP's legendary bassist Francis "Rocco" Prestia. That project reunited the members of the band's mid-1970s rhythm section: Prestia, guitarist Bruce Conte, keyboardist (and longtime Santana band member) Chester Thompson, and drummer David Garibaldi.

Producer-engineer Ken Kessie has amassed plenty of bass-recording experience working on platinum albums in major studios with R&B artists such as Tony! Toni! Toné!, En Vogue, and Bel Biv DeVoe, and with rockers like Santana, Blue Oyster Cult, and Ronnie Montrose. Like most pro engineers these days, he also has a home studio. That's where he recorded an indie CD for Los Angeles rockers Sevensoft and two well-received, Billboard-charting major-label jazz albums, *Hidden Agenda* and *Midnight Morning*, for trumpeter Greg Adams (who, coincidentally, was the original horn arranger for Tower of Power).

KEN KESSIE

Although Dusty Wakeman generally records his bass through high-end compressors like the UREI 1176 or the Empirical Labs Distressor, he sometimes uses a Boss CS-3 compressor pedal for its abundant sustain and particular tonal quality.

ROOM OF DOOM

The panelists agreed that one of the biggest problems facing project-studio owners trying to record proper bass tracks is bad-sounding control rooms. Although correcting a misleading listening situation is beyond the scope of this article, an abundance of helpful information is available from many sources. (For instance, see "A New Approach to Personal-Studio Acoustics" in the April 2004 issue of **EM**, online at www.emusician.com.)

The important thing to realize is that to get a great bass sound, you need an honest and trustworthy reference from your monitors and from your control room. If you don't have that, you may think the instrument or your recording technique is causing the uneven bass response, when room sound adding or subtracting frequencies is the real culprit. With any sound, but especially with bass, effective room treatments will make more difference than your preamps or EQ in your final product.

Even if you have an accurate-sounding room—and especially if you don't—it pays to use as many kinds of monitors as possible. A couple of different pairs of bookshelf speakers, a boom box, your car system, computer speakers, headphones—take advantage of them all. But do have a primary set of quality monitors that you know well.

And at all stages in the recording process, don't forget to use reference CDs of your own and others' music. Not only do reference CDs provide inspiration and create challenges by giving you a sound to aspire to, they will also keep you from going too far astray.

HIRING THE BEST

"I'm going to let you in on a secret," says Kessie with a laugh. "One of the main reasons the big-name engineers and producers sound so good is that they use the best musicians." So if you're not playing the bass parts yourself, make sure to get the best player you can. Don't hire a jazzer to handle an aggro drop-tuned rock track. Don't get a country guy to play funk, or vice versa.

Of course, there will be many situations in which the choice of bassist isn't yours to make. But any time it is, get the most qualified player you can. This will save you hours of corrective work. Not only do great players have good time and tone, but their instruments are usually in fine condition.

TOP-NOTCH AXES

That brings up another important point: you should record the best possible

instrument you (or whoever the bass player is for the session) can get your hands on. Don't waste time on a bass with fret rattles, buzzes, or hums. Get new strings and break them in a day or two ahead of time. Make sure the instrument is set up and intonated properly. Good bass tuning is essential in maintaining clarity in the low end. Have an effective and easily accessible tuning system. (You might want to assign a bus on the console to your tuner and feed it with a clean, preeffects split from your direct box so you're always set up to check tuning.) On the other hand, don't rely completely on your tuner. Sometimes shifting the pitch a cent up or down can make a track glow and shimmer.

String choice, of course, can make a huge difference. "For rock, roundwounds are my usual choice," says Wakeman. "They're bright, and when you're recording rock, that brightness helps the bass compete with the guitars and bass drum. Having edge on the top also helps if you're going to add any kind of distortion. For country, blues, reggae, or even a McCartney kind of sound, flatwounds give you a tone that's darker and warmer and rounder."

THE DIRECT RESULT

These days, the direct box is almost always the first part of the signal chain for bass recording. Even if you're miking up an amp, you're still likely to be using a direct box to split your signal so that you can also send a direct feed to the multitrack (often passing through preamps and compressors on the way). Direct boxes run the gamut from expensive tube and solid-state models to battery-powered "stage" boxes like those made by Countryman and Whirlwind, but your basic choice is between active and passive. Active boxes are powered by either an internal battery or phantom power; passive boxes simply transform your instrument's signal to mic level without being powered. Active DIs generally have user controls such as volume, tone, and impedance selection; they also tend to have more headroom.

Wakeman alternates between two favorite DI boxes, using an Avalon U5 for a clear sound and the popular Demeter STDB-1 tube model when he wants more grit. Kessie's favorite DI is an '80s-era active box by Audio Design. "It's the only one I have," he admits. "It has no controls on it at all, but I like its fat tone and the fact that it's bulletproof—I never have problems with it." Other DIs that Kessie likes and deems "affordable" are the JDI Mk3 Passive and JDV Mk3 Active Class A from Radial Engineering and the Avalon U5.

For Prestia's bass on *Everybody on the Bus*, Grillo used a simple signal chain (see **Fig. 1**) that included a DI he constructed himself. "It's a passive direct box," he says, "and it has almost nothing in it—just a transformer, a ground switch, a pad, and a jack. The best way I can describe it is to say it sounds 'old.' Its simplicity gives a strong, clear signal."

PREAMPS

All three panelists generally avoid using their consoles when recording bass. Remember, one of the keys to getting a good sound is recording with the shortest signal path possible; avoid even the patch bay if you can.

There are plenty of preamps to choose from: you could use one of the popular vintage models by Neve and API that many recordists swear by or one of the sleek new designs from Martech, Avalon, Manley, or Focusrite—to name just a few. While professionals readily acknowledge that a great preamp is worth its high price, real pros also make the most of whatever tool they have available. If that's the console preamp on a Mackie board, so be it. Use your ears and try to maximize the gain structure of your signal path. Every piece of gear has its "sweet spot," a designed-in optimal setting that you should not ignore. Your manufacturer may provide information on what the most favorable setting is. If not, a three-quarters-on setting is a good place to start.

Unless you're going for distortion, what you generally want on bass is as big, clean, and full-frequency a sound as you can get. If you're looking for a little rasp or even some flat-out blazing distortion on your track, keep in mind that you might be better off getting it at a later stage. Distortion locks you into a sound that is difficult to revise. It's preferable to get a nice, clean, fat sound and mess with it in the mix.

Also keep in mind that a preamp is not always necessary. For example, on Prestia's bass for *Everybody on the Bus*, Grillo found that his home-built DI (into a dbx 160X compressor-limiter)

KEN KESSIE

Carmen Grillo recommends a medium attack setting on the compressor when recording bass. If settings are too fast, you can lose the sound of the finger or pick plucking the strings.

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provided sufficient gain. "I just used the output control from the compressor to get the gain that I needed into the ADAT we were recording to," he explains.

These days, Grillo is fond of using an Avalon VT-737SP preamp, a 3-in-1 combo unit that combines a dedicated instrument input with an optical compressor and equalizer. He likes the sound of the Avalon and the convenience of having everything in one box. "The compressor sounds good," he notes, "and it's a very versatile equalizer. It has two bands of EQ with sweepable shelves on the high and low end. I use it a lot for bass to dial out a little bit of 400 cycles. That seems to lift the bass in the track and keep it from getting muddy."

Kessie's first-choice bass preamp is a classic Neve 1073 preamp-EQ module, about which he says, "It's fat and tough sounding. I guess because of my R&B roots, I like some grit in my bottom end, and Neves tend to provide that. It's not that they're dirty, but they add coloration. They're warm, but something about the transformers always puts the sound right in your face. I can't explain it—I'm not sure anybody but Rupert Neve can—but that's why these units are so popular and in demand."

Obviously, there aren't enough 1073s around for everyone to own one even if everyone could afford one. On the other end of the price scale, Kessie suggests the RNP8380, "an excellent and very affordable preamp made by FMR Audio. Some other reasonably priced good pieces are M-Audio's Tampa, the Eureka by PreSonus, and Summit's 2BA-221."

Wakeman loves classic gear and reaches for Neve or API preamps whenever possible, but he also praises several more modern and less expensive options, including the Summit 2BA-221 and the Bellari MP110 tube model. "Good transient response is one of the most important things to look for in a preamp," he says, "But it's a very personal choice. I highly recommend developing a relationship with a music store and a salesperson there. If they know you will do repeat business, they'll help you out. You can get to try out a number of them until you find the one you like."

DYNAMIC CHOICES

Compressors are essential in modern recording, but they're also overused. Each panelist had favorites, both cheap and expensive, but all cautioned about the dangers of abuse during recording. Once recorded, compression is difficult if not impossible to undo, so use it sparingly. Grillo, Kessie, and Wakeman recommend light compression to tape, and they advocate saving final compression choices for the mix. That said, here are their favorites.

▼

You should shoot for 3 to 4 dB of gain reduction.

Grillo frequently stays "in the box" with his Avalon VT-737SP for compression. "I tend to use a medium attack, not quite half way up," he says, "because I want to make sure I keep the sound of the finger or the pick, whichever is plucking the strings. I don't want that getting lost with an attack that's too fast. And I generally use a fast release. Especially with someone like Rocco, who plays a lot of notes, I want the compressor set to give up the note quickly enough so that it can start working on the next note."

Grillo also likes the dbx 163X on bass but prefers to use it on mixdown rather than during tracking. "It's a great compressor because it's idiot-proof. There's just one slider, and it does what it needs to do: it makes the bass stick out in the track evenly. Notes that were hit softer come up, those hit harder come down, and overall it still sounds good."

Grillo's wild-card compressor choice is the 1980s-era solid-state Valley People Dynamite compressor-limiter. "For some bass players you might need a little more impact, a more aggressive sound, and the Dynamite is great for that," he says. "It only has two controls:

MAUREEN DRONEY

Ken Kessie uses EQ only as a last resort when tracking bass parts. He suggests first trying to get the tone you want from your instrument, amp, or other components.

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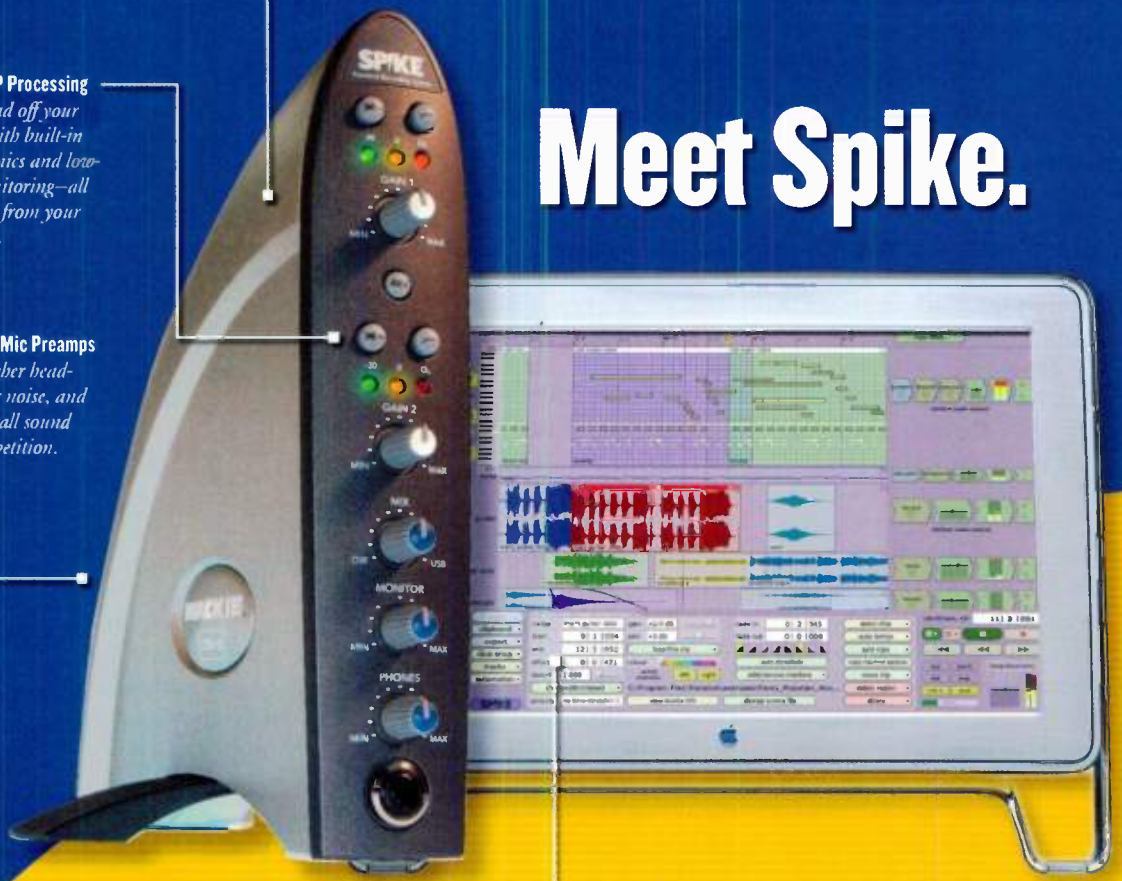

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input and release. The only way to determine the amount of compression is with the input control. It's a strange box, but it's got a sound."

As the owner of a commercial studio, Wakeman can take his choice of a wide selection of compressors; UREI 1176s, Summit tube boxes, and Empirical Labs Distressors are his top picks. "If I could have only one compressor in my studio, I'd make it a Distressor," he says. He also has several suggestions for more affordable compressors. "The Summit half-rackspace [TLA-50] is quite good. Dbx 163s are good on bass; I used to sell a lot of those. And the dbx 161, the consumer version of the 163, is a good choice if you can find a used one. It's the same as the 163 but doesn't have balanced inputs. It's perfect for home recording and good for bass.

"Ultimately, I'm always recording with the mix in mind," he says. "I want the bass to sit evenly instead of having it coming and going. If I'm lucky and the player is consistent, I won't have to compress a lot. Sometimes, if I want to compress something a lot to control the level but I don't want the sound of compression, I'll use a slow attack and release so that you don't hear it working."

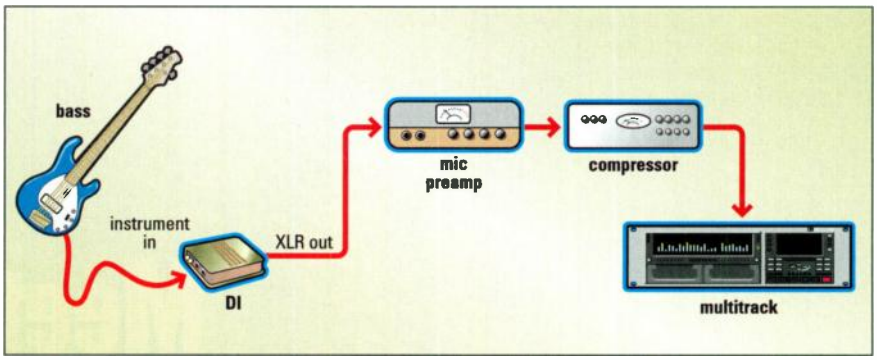

FIG. 2: Here's an effective signal chain for recording bass direct. Like the other setups shown in this article, this signal chain bypasses the console to maintain signal purity.

By contrast, Wakeman at times uses compressors to add color. "Especially with a UREI 1176," he points out, "you can play around with the attack and release. Particularly on bass, it can sound almost like you're using EQ.

"Of course," he continues, "the amount of compression I use also depends on the style of the music I'm working with. Sometimes, especially on rock stuff, I'll want to get something a little different, so I'll play through a little blue Boss compressor pedal—the CS-3. I'll record to tape with that. I'm totally locked into the sound, but I'm prepared to live with it. Running through the Boss will affect how I play: I'll use a different touch because it gives me sustain, and a particular kind of suck on the attack. That's something important to remember: how you treat the sound can definitely affect how the bassist plays. It all works together to get the right sound for the right part."

Kessie often uses a Teletronix LA-2A to add color; its tube sound provides what he calls "tone and glue." Other, more budget-friendly compressors he recommends include FMR Audio's RNC1773. "It's an incredible box for the money, and it has attack and release controls," he says, "perfect for dialing in a rhythmic bass part. I also own an ART Pro VLA, which is very transparent. And I've heard from other engineers that the PreSonus APC-88 and Summit TLA-50 are great values and easy to use."

ALL THINGS BEING EQUAL

When it comes to EQ, the panelists are adamant: they avoid using it during tracking whenever possible, preferring to get the sound right at the source. "I used to EQ to death," Kessie laughs, "but I've learned that with less EQ, sounds translate better on different speakers."

Wakeman says, "If I absolutely need to EQ going in, I will. But I always try first to get the sound right with the instrument and amp."

Grillo agrees, saying, "It just sounds more true in general if I don't EQ during recording. And when you do, and then you EQ again in the mix, it builds up and you lose the natural sound of the live instrument."

Speaking more broadly about EQ, Kessie says, "Good hardware EQ is expensive, and EQ is an area where plugins, in general, really shine. If you want the good hardware EQ for your major projects, consider renting instead of buying. API 560 graphics sound great

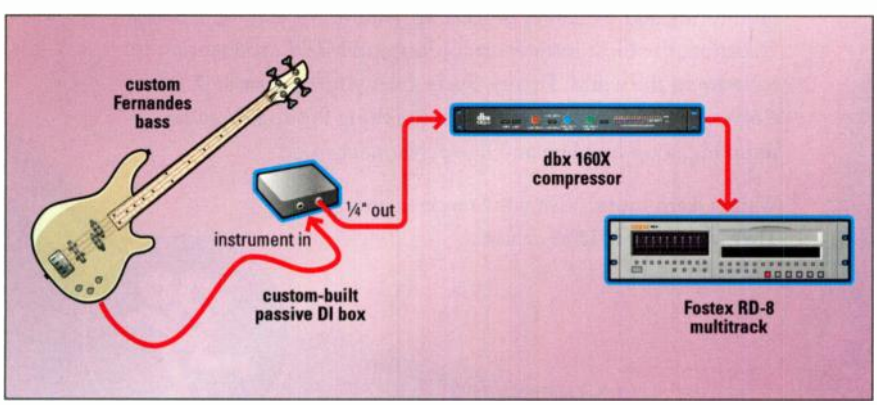

FIG. 1: When Carmen Grillo recorded Tower of Power bassist Rocco Prestia's solo CD, he got good results with this pared-down signal chain.

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and are very easy to use. I also recommend Audio Arts parametrics. I bought a reasonably priced used pair some time back, and they are very nice sounding.

IN THE BOX

Both Grillo and Wakeman are Pro Tools owners and operators, and both have used numerous kinds of plug-ins on bass. "My favorites for bass and other instruments are by McDSP," says Grillo. "They make great compressors, and they also have excellent British-sounding EQ. It has filters on the top and the bottom and two bell curves in the middle that help you dial-in what you're looking for. I like Waves' MaxxBass, which adds a kind of resonant frequency that you can mix in with your original sound. It's great for bass, kick drum, or any kind of low-end material. I also like the Bomb Factory Joemeek compressor [now owned by Digidesign], and their EQ, the Meequalizer, works great on bass. It's very string friendly.

"I've used Digidesign's Line 6 Amp Farm for distortion on bass, and it works very well. And IK Multimedia Amplitube is good. It can give you distortion and EQ that really sounds tubey. It totally goes to 11! *[Laughs]* And it is great for emulation. If you have a bass player who normally plays smooth jazz, but he's the only guy you can find for a rock 'n' roll date, you might want to use Amplitube on his bass to put him in the band."

While Wakeman loves his hardware 1176s and LA-2As, he also raves about Universal Audio's plug-in versions of the same units. "The thing is," he explains, "once you get into the digital domain, you want to stay there. Because every time you convert your sound back and forth you lose something. I prefer to stay digital rather than go through converters over and over again. So, once I'm digital, I'm happy to have plug-in versions of the hardware I like."

Wakeman also gives props to the compressor and EQ plug-ins by JoeMeek, and he loves Bomb Factory Sans Amp PSA-1 and Amp Farm. "But," he says, "I'm still waiting for the bass versions."

KEN KESSIE

FIG. 4: One of Wakeman's preferred cabinet-miking configurations consists of a Sennheiser MD421 aimed at the center of the speaker, and a Royer R-121 ribbon mic a short distance back and off to the side.

MAKING CONNECTIONS

The panelists offered specific setup suggestions for various bass-recording needs.

Going direct. Using high-quality cables, connect the bass to the DI, the XLR output of the DI to the mic input on your preamp, the line out of the preamp to your compressor, and the line out of the compressor to the multitrack (see **Fig. 2**).

Adding an amp. If an amp is important to your sound, try this setup: after connecting the bass to the DI, run a ¼-inch line from your DI out to your amp head. (Hopefully, it's readily accessible in the control room with your bass player.) Next, run a line from the head to the speaker cabinet. Record to two tracks, one with the miked bass sound and one with the direct feed from the DI (see **Fig. 3**).

As for what microphone to use, the panel had a number of suggestions: the AKG D 112, the Sennheiser MD421, the Electro-Voice RE20, the Royer R-121, or the Soundelux IFET7. Wakeman likes using two mics. "I usually put the 421 or the RE20 pretty much right in the center of the speaker," he says, "and then I put the Royer a little off the center and back a bit" (see **Fig. 4**).

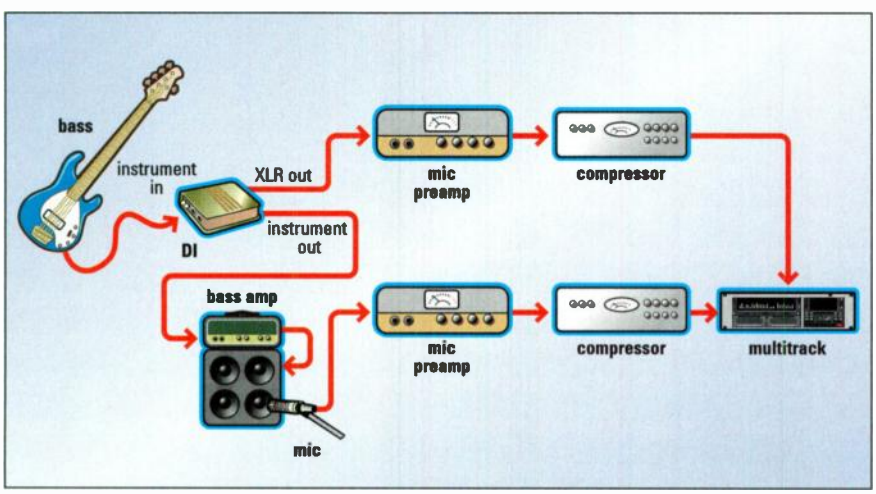

FIG. 3: The panelists recommend this basic signal chain for miking a bass amp.

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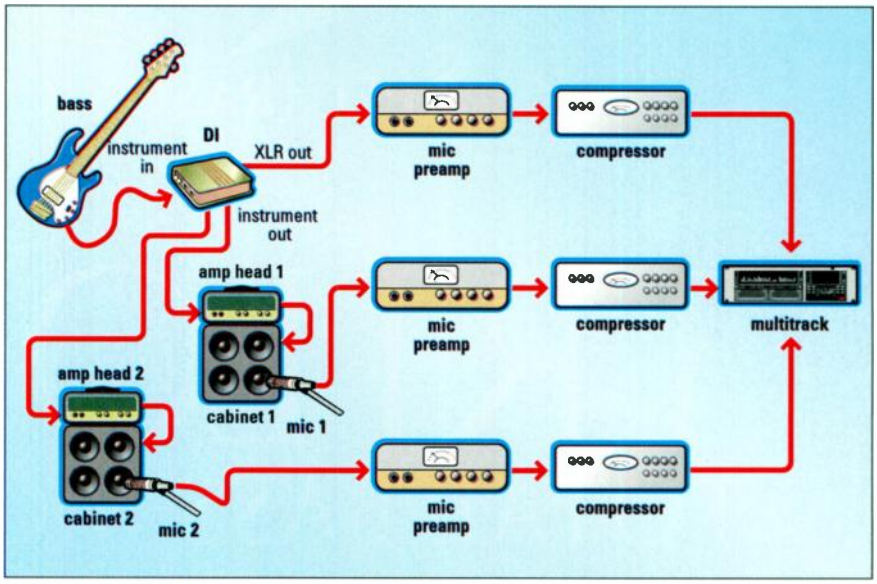

FIG. 5: For ultimate control of your bass sound, Kessie suggests using this quasi-biamping configuration, which involves two amps—one set for extra treble and one set for extra bass—and a direct feed, all going to separate tracks.

It's important to keep your signal chain clear and clean, because the bass cabinet will probably add some dirt to your sound. Of course, you'll want to record on two tracks, one from the mic and one from the DI, and you can experiment with how much of each to use when you're mixing. You'll probably require more of the amped sound for heavy rock tracks and less for lighter rock and country. Reference CDs will be helpful in figuring out the best proportion.

If you're recording on a DAW, time-aligning your direct and amplified tracks will improve your sound. (Time alignment is necessary because the amped sound will take a few milliseconds to reach the microphone, while the direct sound is virtually instantaneous. Shifting the amplified track forward until it lines up with the direct track usually produces a "fatter" sound.) Remember that if you're not happy with the amp sound, you can always reamp the track on which you recorded the direct signal. Or you can add plug-ins like those mentioned earlier for grit and attitude. Line 6 Bass Pods are also popular amp substitutes.

For even more control, try recording the bass using two cabinets and two heads to achieve a bi-amping effect. Get great bottom out of one amp and some cutting treble out of the other. Record to separate tracks, and use each in the mix as needed. This is a great way to compete with sampled kicks and detuned guitars (see **Fig. 5**).

GETTING SET

Whether you're going direct or using a combination of direct and amped signals, it's helpful to get your levels maximized before you record.

Start off with the bass's volume at 10 and the compressor set to 0 dB with the ratio at 1:1. Have the bass player strum some loud three-note chords, and set the preamp gain so that you're just overloading your recorder. That way, when the bassist plays normal lines, you will have plenty of headroom. If your bass player is planning to play loud, strummed chords in the song, turn the preamp down far enough to let those loud phrases pass through it without overloading the recorder.

Next, have the bass player play along with the track (or the band, if you're recording live). Make sure the drums—especially the kick—are loud and clear in the player's mix. Turn down anything that won't help him or her lock tight to the groove.

If at that point you feel that the bass is uneven, kick in the compressor. Try a 2:1 ratio to start. Keep the compressor attack slow enough to let you hear the attack of each note. Keep the release fast enough so that each note is not affected by the note before it. Be careful: if the release is too fast, the compressor will "chatter" and distort as long notes sustain. (If you're using two compressors as part of a DI-and-amp setup, start by setting the compressors similarly, then fine-tune them to taste.)

You should shoot for 3 to 4 dB of gain reduction. Remember, you can always compress more at mix time. You

TONAL TWEAKING

If the character of your bass's tone isn't what you want for a particular song, you can alter it using some of the following suggestions.

To sound brighter:

- Switch strings to roundwounds.
- Switch to a brighter bass.
- Use a pick, or pick closer to bridge.
- Switch to a brighter preamp or compressor (solid-state units are usually brighter than tube).
- As a last resort, patch in a high-quality EQ between the compressor and the recorder. Try removing sub harmonic frequencies below 30 Hz, pulling out 2 or 3 dB around 400, or adding at bit at 1.8 kHz.

To sound darker:

- Switch strings to flatwounds.
- Switch to a darker bass.
- Use fingers, or pick closer to the neck.
- Switch to a darker preamp or compressor. (Tube models are usually darker than solid state.)
- Live with it. You'll probably appreciate the high end when the overdubs pile up.

will probably have to increase the output gain of the compressor slightly to compensate for gain reductions

BASS-IC INSTINCTS

The central message that Wakeman, Grillo, and Kessie communicated to me is this: your goal when recording bass should be to get as clean and full-frequency a sound to tape or disk as you can. You'll achieve that more easily by working in an acoustically treated control room, or at least using monitors that you're familiar with. You want to use the best-quality instruments, DI boxes, preamps, compressors, cables, and mics that you can get your hands on, and keep the signal path as short as possible.

If you can, try to get your tone from the instrument and the components you're using rather than with EQ. You're better off saving effects and heavy compression for the mix, where you can mess around with the track to your heart's desire. That way, you have the option of going back to your original sound if you want to.

▼

String choice can make a huge difference.

On the subject of the mix, here's a suggestion from Kessie to leave you with. If you're mixing from a DAW, don't move every note of the bass track to line up with a kick or a grid. You'll probably waste hours and just end up making everything sound smaller. Of course you want to clean up gross errors, but getting too precise can kill a groove. And a bass part without its groove—no matter how well recorded it is—is useless.

Maureen Droney *is Los Angeles editor for* Mix *magazine. As an engineer, some of her own notable bass recording credits include Rocco Prestia, Alphonso Johnson, Randy Jackson, Pete Sears, and Larry Graham.*

Making the Cut

By Nick Peck

Editing has been a part of music production for 50 years, but the advent of ubiquitous nonlinear hard-disk editing has changed the face of music, for better and for worse. Systems like Nuendo, Pro Tools, SAW, and every other DAW give us unprecedented speed, accuracy, and flexibility in editing audio. We can fix mistakes, adjust timing, clean up recording flaws, experiment with rearranging the sections of our songs, and generally fly things around and chop them up in unexpected ways.

Enhanced efficiency is perhaps digital audio's greatest boon. One of the most important efficiencies, a feature used so commonly that we take it for granted, is the ability to make sophisticated edits quickly and to a level of precision that analog just can't touch. Even among some die-hard analog-heads, it is now common practice to lay parts down to 2-inch, 16- or 24-track tape, then fly the tracks into a digital audio workstation for editing.

But we can take editing to obsessive extremes, robbing tracks of their feel as we try to attain a mythical perfection (see the sidebar "Overdoing Perfection"). Music producers routinely use editing as a crutch for subpar musicianship, fixing shaky timing and replacing wrong notes, cobbling together solos and vocal performances from dozens of takes, and disguising a band's sloppiness by painstakingly lining up and retiming the various tracks after recording.

In this article, I'll examine a host of editing techniques that you can use to change and improve your music production. I will also focus on some tricks of the trade to increase your editing efficiency. You can take a quick look at how the track-editing process developed in the sidebar "A Brief History of Editing."

Advanced editing tips for fine-tuning your audio.

DMITRY PANICH

Making the Cut

FADES, FADES, FADES

The simplest form of edit involves placing two regions next to each other in time so that one region abruptly stops as the other begins. This type of edit, called a *butt splice*, is frequently all you need to get the job done. Butt splices have two disadvantages, though. If the splice does not take place at zero-crossings of the waveforms, a tick or pop is often heard (see **Fig. 1** and **Web Clips 1** and **2**). In addition, this type of immediate, direct transition can often sound jarring or rough. This is where fades enter the picture.

There are three types of fades: *fade-ins*, in which the signal grows in amplitude from silence to its nominal level over a period of time; *fade-outs*, in which the signal gradually drops to silence; and *crossfades*, which are a combination of the first two. Crossfades smooth over edit points by fading out the old material and fading in the new simultaneously. Crossfades can last from a few milliseconds to several seconds and require sufficient material on either side of the edit point to work with. Crossfades can be used to segue smoothly from one song to the next, to fix mistakes in musical performances by allowing notes to be replaced seamlessly, and to create strange new sounds by grafting different elements into a unified whole.

Fades and crossfades have amplitude curves, which specify the rate of attenuation over the length of the fade. The simplest example is a linear curve (which is actually a straight line), in which the rate of attenuation stays constant over the length of the fade. *Linear* fades are perfectly useful, but they are far from being the only option. *Logarithmic* fades start quickly, then slowly taper off toward the end. *Exponential* fades start slowly, then move quickly toward the end. *S-type* fades start quickly, slow down toward the middle, then speed up again toward the end. Each of these fades has a subtly different effect and is useful in various situations.

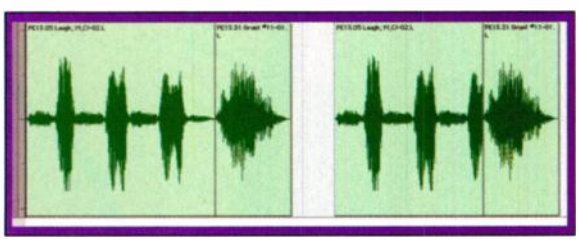

FIG. 1: Butt splices align two segments of audio end to end. The butt splice on the left occurs at a zero-crossing, so no pop or click should occur as it will with the edit on the right.

Many DAWs allow you to create custom fades by adjusting these basic fade shapes or by drawing your own freehand fades. When creating crossfades, you can specify the shape of the fade-out curve and fade-in curve separately, although symmetrical shapes are commonly used. Linear (or equal-gain) crossfade curves are often specified as the default crossfade type. In my experience, though, equal-gain crossfades often leave the impression of a hole or dip in amplitude at the edit seam. Logarithmic (or equal-power) crossfade curves don't exhibit that characteristic, because the in and out regions are both being played at higher amplitudes around the edit (see **Fig. 2**). This is the curve I use as my default curve.

Many songs end by fading out over a vamp or repeating chorus. Pay special attention to these fades, because they play an important role in the listener's final impression of the song. Start by thinking about where you want the fade to end. I tend to like fades that end at a musical boundary, such as the end of a chord progression. Another effective technique is to fade gradually over the length of a chorus, then end the fade after the beginning of the next chorus. This creates the impression that the song is continuing on and on.

When creating song fade-outs, use a mouse or control surface to create fader automation; don't create a computer-generated fade region. The manual fade has a more human, musical character, making the fade feel like part of the performance.

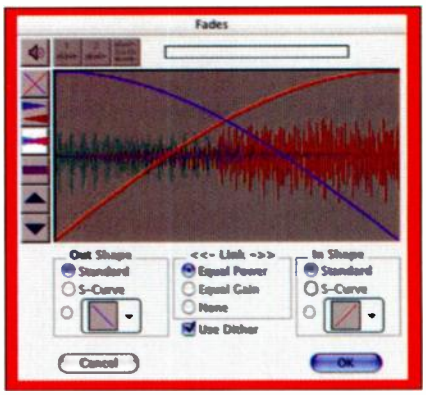

FIG. 2: Equal-power crossfades frequently work best for seamless musical transitions.

EDITING MULTITRACK MUSIC

Some vocalists are lucky enough and talented enough to be able to deliver a flawless performance of a song in a single run-through. Those of you in that category can skip this section; for the rest of us, the art of vocal comping (or compositing) comes into play. Comping involves recording several takes of the tune against the backing tracks, then selecting the strongest elements and combining them into a composite performance.

The lead vocal of a song is almost always its primary focal point, so it's very important to assemble the most in-tune, mistake-free performance you can. However, musical perfection must always be balanced against spontaneity and emotional delivery, both of which tend to get lost after too many takes. Having too much material to choose from can also slow the creative process to a crawl. So when you're preparing for a vocal-comping session, consider limiting yourself to a maximum of four source tracks. If the vocalist delivers more than four takes, choose the best four to work with and throw the rest away.

Next, line the tracks up in your DAW. Listen through the takes a verse at a time. Find the one you like the best

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and use it as your primary track, replacing elements with materials from other takes as necessary. The amount of detail you apply to the vocal edit is largely a matter of personal taste. Musical genre, skill level of the vocalist, and the producer's working style all factor into your decision-making process here. Some people like to choose whole sections of music, some work on the level of lines and phrases, some get to the level of words, and some people even edit syllables of words from different takes. How you work is up to you, but remember that pitch perfection can come at the expense of musical flow.

When the red light comes on, even the most advanced musicians make mistakes. This is especially true in improvised parts and solos. I feel that the most important thing to capture is the musician's energy in a spirited performance, so I don't let the occasional flub worry me too much. However, there are times when a beautiful solo is marred by a particularly egregious honker. If a note is a bit sharp or flat (within a semitone or so of the desired pitch), you can use pitch shifting to correct it (see **Fig. 3**). The problem is that significant pitch shifting disrupts the delicate attack transients that are so important to a sound's definition and also changes the formant relationship of the various overtones that define the instrument's timbre. Therefore, it's a good idea to try replacing a bum note with another note from elsewhere in the track. This treasure hunt does not always pay off—you need to find a replacement note with appropriate pitch, duration, and amplitude. Longer notes can be used to replace bad notes if you trim their duration carefully.

When you find a good substitute note, paste it into an empty track directly above the bad note. Zoom in to the beginning of the waveform and line up the attack portions of the notes as closely as possible. Drop the replacement note onto the old note's track and create crossfades before the attacks of the new note and the note that follows.

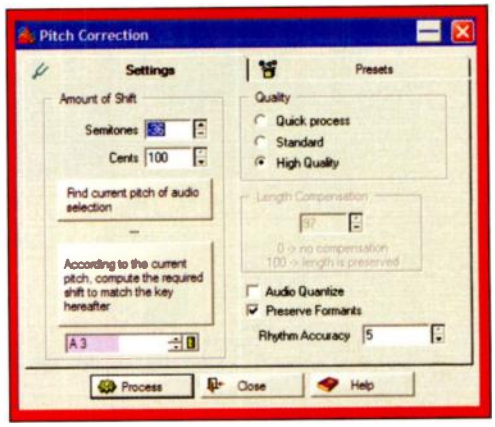

FIG. 3: Like other audio editors, Steinberg Wavelab offers pitch correction. You can shift a note up or down as many as 36 semitones in intervals of one cent (one hundredth of a semitone).

DOING DRUMS

Drums are one of the most difficult instruments to record well. A series of distinct sound sources emanate from different locations in proximity; multiple mics and tracks are often used to allow greater mixing control. Every mic picks up sound from all the sound sources in the kit—the snare drum bleeds into the kick-drum mic, for example. Thus, any EQ or mix decisions applied to the kick will be reflected to a lesser degree on the snare drum on that same track. The snare information in the kick-drum mic will also be slightly out of phase with the dedicated snare track, smearing the all-important attack transient. Multiply this effect by 7 to 12 mics, and a multitrack drum kit can quickly become a dull, thuddy, boomy mess.

This problem can be ameliorated somewhat by carefully editing out material on tracks for drums that are not being struck at a given moment. In practice, this usually works best with tom-toms, since they are struck less frequently than the kick and snare. Listen through the drum tracks for the various tom fills. Delete everything on the track except for the fills, fading the regions you are keeping after the tom

OVERDOING PERFECTION

In theory, digital editing allows us to produce letter-perfect performances. All the techniques discussed in this article are used every day to create audio illusions: releases that are metrically perfect and far beyond the musical capabilities of the band or musicians that purportedly recorded them. But endlessly worrying a musical track to try to perfect it can lead to sterility and a lack of perspective. And all the editing in the world won't make a beginning guitar player sound like Steve Vai. The question is no longer whether we have the capabilities of altering reality in this way, but whether we should do so. The reaction against false perfection has led to a resurgence in the standard recording technique of yesteryear: recording whole bands in a room together with minimal overdubs. This approach is finding mainstream success in the records of such artists as Norah Jones.

At the 2003 AES convention in New York, legendary producer Arif Mardin discussed this phenomenon in his keynote address. In his opinion, digital tools were all too frequently overused to manufacture performances. In effect, this amounts to the "dumbing down" of commercial music in an effort to pander to a supposedly unsophisticated public. Mardin felt that the public is far more perceptive than the commercial-music industries give them credit for, and that Norah Jones's success shows that honest, heartfelt performance is what ultimately counts.

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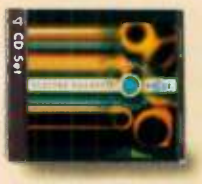

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fills are completed. Although this may sound unnatural when you listen to the tom tracks by themselves, the stereo-overhead tracks glue the whole sound together, creating a smooth, clean-sounding drum track in which the toms jump right out of the speakers. Gates are frequently used for this purpose as well, but editing by hand avoids false triggers.

The multitrack nature of drum recording makes other types of edits a challenge as well. When moving drums around, you must remember to keep all the drum tracks grouped and edit them simultaneously. A drum edit's biggest giveaway is the unnatural decay of the cymbals at the edit point. You can sometimes disguise this by extending the crossfade region for the overhead tracks a second or so past the edit point. You can also fly in a new cymbal crash right at the edit weld if that makes musical sense.

KEEPING TIME

Digital-editing technology gives users the ability to retime acoustic-drum performances, moving the individual drum hits around in time. You can use this technology to tighten up an unsteady performance or even to change the tempo, within reason. In the past, music producers have used traditional time-compression or time-expansion algorithms to change a drum part's tempo, but this can't correct for a sloppy groove. And this type of digital manipulation can have deleterious smearing effects on nice, crisp drum recordings.

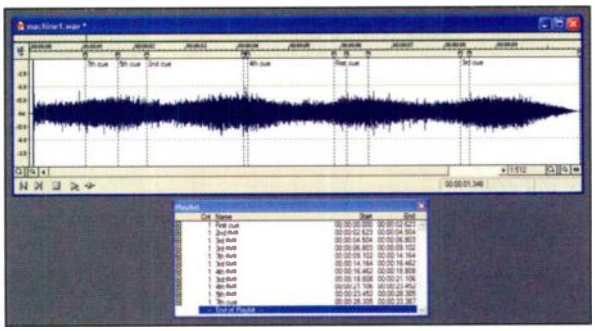

FIG. 5: Playlist editing can bring fresh perspectives by allowing you to quickly rearrange the order of musical sections. In Sony Sound Forge, the Playlist can be converted to a new file.

An alternative approach is to chop the drum tracks into a series of small regions, using functions such as Pro Tools' Beat Detective. Beat Detective works by analyzing a drum performance's transients in relation to the tempo of the track, breaking it down into user-specifiable subdivisions of bars and beats. Once the settings are dialed in, Beat Detective separates the performance into dozens or hundreds of individual regions (see **Fig. 4**). These regions can then be quantized against Pro Tools' metrical grid just like a MIDI drum track would; even groove-template and swing parameters can be specified. Once everything is lined up, Beat Detective applies a series of short crossfades to smooth over the edit points.

If your workstation of choice doesn't offer a feature like Beat Detective, you can chop your track by hand, although the procedure is a little more tedious. Start by using a strip-silence command to break the drum tracks into component regions. You will have to tweak the threshold and region-length settings on a per-track basis for optimal success. Be sure to leave enough time before the threshold trigger so the very beginning of an attack isn't chopped off. You can then use a quantize-region feature to snap those parts in place and a global-crossfade feature to weld the regions together again. Try a crossfade time of 10 milliseconds to start with. Listen through the tracks carefully, and you'll find that you'll have to fix a few things by hand (I end up having to do this when using Beat Detective as well).

To be honest, I've had mixed results retiming drum tracks in this fashion. Sometimes it works beautifully, and other times it makes the parts sound weird and robotic; it varies with the individual performance and musical genre. If the performance feels good but is drifting against the click, there is an alternative to consider. Try breaking

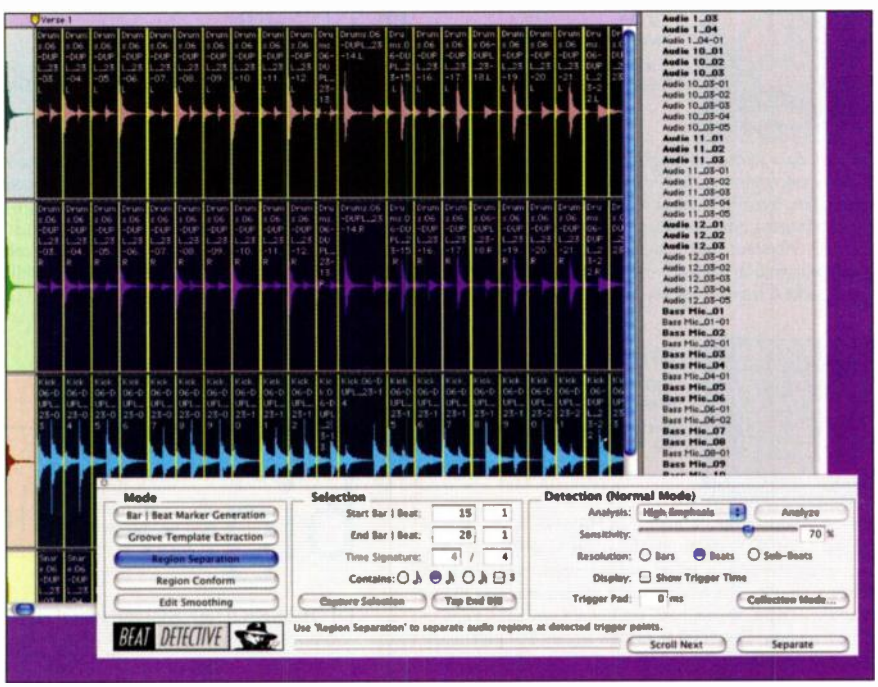

FIG. 4: Retiming a drum track can be time-consuming, but it is the best way to tighten up a shaky performance. Tools such as Pro Tools' Beat Detective (shown here) make the job easier.

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the performance into one-, two-, or four-bar pieces. Quantize these larger pieces against the grid. This works because it lines up the downbeat with the click, but the internal structure of the groove is better preserved, creating a more natural-sounding edited track. Resyncing the drums every bar or two maintains the track's rhythmic integrity while retaining everything we like about human feel.

EDITING STEREO MIXES

A half-century ago, iconoclastic piano virtuoso Glenn Gould created controversy throughout the classical-music world by heavily editing his recorded piano performances. His finished recordings were composited from multiple takes, which sparked debate over the nature of the recording process itself. Now, structural editing of 2-track mixdowns is common across all musical genres.

Finished tunes are routinely edited for length; entire sections are removed

A BRIEF HISTORY OF EDITING

Analog audio editing has been around as long as analog tape itself and is still in widespread use today. The process is fairly straightforward but demands precision for good results. To start with, locate the edit points by shuttling the tape to the desired location. Then disengage the playback motor and manually rock the reels back and forth until you can hear the edit point line up exactly with the playback head. At this point, place a mark on the back side of the tape with a grease pencil and then find the other edit point, repeating the process. Next, slide the tape into an aluminum tape guide (known as an editing block) attached to the recorder. There are narrow grooves in the block at 90- and 45-degree angles. Line up the marked edit point with one of those narrow grooves (typically the 45-degree angle groove, because this creates more of a crossfade effect), and carefully slice the tape with a razor blade.

Spool off the tape you don't want until you come to the next edit point; one more slice there, and the edited material is gone. The two ends of tape that remain are then carefully lined up, abutting each other in the editing block, and are connected with a piece of cellophane tape known as splicing tape. If the edit point is the beginning or end of a song, the tape is not spliced to more recording tape, but rather to white tape known as leader tape (see Fig. A). Leader tape is nonmagnetic and thus creates no sound when passed over the playback head. Its contrasting color makes it easy to visually pick out the beginning and ending of the song on the reel.

This laborious process has been common practice for 50 years and has been used creatively in countless situations. When recording the Beatles' "Strawberry Fields Forever," John Lennon liked the first half of one take and the second half of another take; he asked producer George Martin to splice them together. Unfortunately, the takes were recorded at different tempos and a semitone apart in key. Martin accomplished this nearly impossible feat by gradually slowing down the tape as it approached the edit point. That edit happens at exactly one minute into the song.

A significant limitation of analog editing is that it is an all-tracks-at-once proposition; it is impossible using standard splicing techniques to move one track in time relative to another on the same tape. So if a bass note is late relative to the kick drum, the best option is to punch in and rerecord it. Necessity is, however, the mother of invention. The need to fix tracks after the musicians were not available led to a high-wire proposition of last resort: the infamous "window edit." This technique involves slicing a "window" of tape that corresponds to the location of a physical track lengthwise down the reel of tape for the duration of the edit, then slipping the window in time relative to the rest of the multitrack tape, then taping it back into its new position

NICK PECK

FIG. A: Engineers have been editing audio since the early days of music production. When an edit point is at the beginning or end of a song, white tape (known as leader tape) is used.

with splicing tape. Performing this task in the digital world is trivial by comparison.

Using technical innovation to invent new genres of art is nothing new. In the '50s, pioneering composers such as Pierre Schaefer, Pierre Henry, Karlheinz Stockhausen, and Edgard Varese sliced up recordings of everyday sounds: door slams, church bells, children playing, and so forth. They then recontextualized those sounds by gluing them back together and presenting them as unified compositions. This early branch of electronic music, known as *musique concrete*, was the first form of music that could not have existed without audio editing. The impact of musique concrete was utterly profound—echoes of this early work have resonated through virtually all electronic music to the present day.

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to make the songs more radio friendly. Sections can also be moved around or removed after the fact to tighten a song up or improve its flow. Artists are often too close to a song during primary production to look at large-scale structural issues. Once the mix is completed and some time has passed, the song can be looked at with a new perspective, at which point editing can come into play to improve pacing or momentum.

There are a number of 2-track audio-editing packages such as Bias Peak and Sony Sound Forge that offer a nondestructive playlist-editing feature. This allows you to define musical regions in your mixdown that can then be rearranged quickly within the playlist window (see **Fig. 5**). Some playlist systems provide crossfades to help smooth out the edit points. If the playlist is not sufficiently sophisticated for a project's needs, you can still rough in your arrangement ideas in a playlist, then implement them in greater detail within your DAW software.

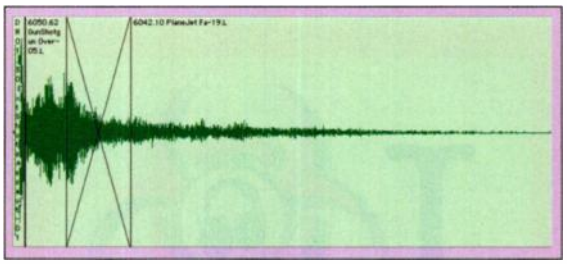

FIG. 6: Mixing segments of different sounds is a great method for designing unique effects.

GOT LIVE

I love recording live performances; there is often an energy, an improvisational spirit, and a lack of inhibition that can't be duplicated when the red light goes on in the studio. But live performances can also ramble: improvisatory expositions that are captivating onstage can feel long-winded when played back in the living room. Aggressive editing can be used to turn 20-minute jams into 8-minute gems.

Start by listening to the raw performance several times. Draw out a rough road map, identifying energy levels, spontaneous arrangements, and soloing instruments. Identify natural areas of tension, build, and resolution. Next, look for obvious places to tighten: meandering sections where no one is really playing a lead, areas of solos that feel aimless or repetitive, and sections where the band is not grooving well. Remove those sections, looking to leave logical in and out points to

join the remaining material. Try to sculpt the piece to create a sense of forward motion—of building toward and reaching a climax, then building again.

Making musical edits sound convincing can be a laborious process. Sometimes sections will fuse together naturally, but often they won't—it's the luck of the draw. If an edit isn't convincing, be honest with yourself about it and look for alternative entry and exit points. This process can force you to discard your original notions but can lead down interesting new paths. Sometimes the effect can be spellbinding. With a little luck and some skillful editing, group free improvisations can end up sounding like carefully composed works.

IN THE MOOD

Film and television make heavy use of preexisting musical works, fitting them to picture as needed. In fact, music editing is a particular specialty job within the post-production world. Successful music editors have an encyclopedic knowledge of a wide variety of musical genres, as well as large music libraries available at their fingertips. They can identify the mood for a scene and cut appropriate music to picture, often using multiple pieces from a variety of sources within a single scene. This skill is frequently put to the highest test in movie trailers, where the music editor has 30 seconds to knock the audience out of their seats.

There are times when the edit transition between two disparate pieces or sections feels awkward or forced. A music editor can soften the transition by composing bridge material such as a musical flourish or string pad or by adding sound-effect whooshes or hits at the edit. I often smooth over unconvincing orchestral edits by applying a bit of reverb to the tail of the outgoing piece. This creates a subtle bridging wash for a second or two under the incoming piece.

SOUND DESIGN AND EFFECTS

When you're creating sound for picture, editing is your most important tool—they don't call them sound editors for nothing. Existing sound effects are constantly trimmed, sliced, and diced to fit the duration of events onscreen, in films, television, or video games. But in addition to simply trimming sounds for length, editing can be used as a creative tool for sound design. Different sounds are frequently spliced together to create a new hybrid sound, using, say, the attack portion of an M-16, the body of a shotgun, and the tail of a jet fly by (see **Fig. 6** and **Web Clip 3**). Using this mix-and-match approach, you can generate a large variety of gunshots, explosions, and body falls from a limited set of materials.

According to psychoacousticians, a sound's attack is the most critical component in defining its timbre, so that's where I put a great deal of thought when designing sounds. This approach is particularly appropriate for trying to get punchy weapon fire, explosion,

or percussive sounds. Layering sounds is a very useful tool, but there is only so much energy you can build into a single attack transient—digital 0 (that is, all bits on) is still 0 no matter what you do. So when I want to make a really cool, impactful sound, I often take the one-two-punch approach, grafting in a low-frequency thud just before the sharper high-frequency report of the weapon (see **Fig. 7**). Listeners' ears will fuse this together into a single, seamless sound that is big and mean, without my having to squash the dynamic range through overcompression.

This technique can be used to create new musical instruments for your sampler as well. Years ago, the venerable Roland D-50 created a new battery of sounds by offering sampled attack transients and synthesized body waveforms. It was great fun to graft a flute attack onto a string body or a string pluck onto a horn sounds. You can do the same thing now: try replacing the first 250 milliseconds of a violin sample with the first 250 milliseconds of a piano sample of the same pitch. Use crossfading to blend, and presto, you've created a pianolin (see **Web Clip 4**).

FIG. 7: Grafting a new attack onto an existing sound can add punch to your audio.

TIPS AND TRICKS

In editing, speed is power. You can greatly increase your editing speed by memorizing your DAW's edit-keystroke commands. In Pro Tools, turn on Single-Keystroke Command mode (click on the little A-Z button in the upper left-hand corner of the edit window), and keep your left hand on the middle row of your computer keyboard. A and S trim the top and tail of a region, D and G create fade-ins and -outs, and F creates a crossfade in the selected area. Also try turning on Tab to Transient mode, which moves the selection bar from attack to attack with each stroke of the Tab key. Weaning yourself from the mouse wherever

possible promotes maximum speed and efficiency.

You can use pre- and postroll to test your edits in context without having to move the selection bar away from the splice point. Setting a 2- or 3-second preroll allows you to concentrate on fine-tuning the edit without losing your place. When selecting large regions of music to be edited, it is all too easy to lose track of your edit-in point while hunting for your edit-out. Dropping a marker at the edit-in point solves this problem handily. Once you've identified your edit-out point, you can jump right back to the marked in point to quickly and accurately create your edit region.

Many digital audio workstations support a global-fade function, allowing you to create multiple crossfades over a large series of regions. For a starting point, try specifying an equal-power curve with a 10-millisecond crossfade length, then listen through for any fades that don't work and tweak them by hand.

In video-game and multimedia work, engineers are often called upon to edit and batch-process hundreds of files at a time. If you don't have time to check each file by hand, you can safeguard against non-zero-crossing-induced ticks and pops at the tops and tails of the files by specifying a 2- to 5-millisecond fade-in and fade-out within your batch processor. AudioEase's BarbaBatch is an example of batch-processing software that supports this very useful feature.

People have been slicing, dicing, and shredding their music for decades, shaping their work in interesting and useful ways. The art of editing is an acquired skill and a critical tool in any audio engineer's bag of tricks. Try some of these techniques in your next musical project and see how editing can clean up, tighten up, and open up new sonic vistas. EM WEB CLIPS

Composer, keyboardist, sound designer, and engineer **Nick Peck** *lives and works in the Bay Area. He still likes editing with razor blades on an old Ampex 440-B ¼-inch reel to reel. Find him at www.underthebigtree.com.*

CD+G=$

Karaoke disc production in the personal studio.

By Will Connelly

Karaoke is a relatively small but rapidly growing segment of the music-production business. According to one software manufacturer, the market for karaoke products (hardware, software, and recordings) grew from roughly $1 billion in 2000 to $15 billion in 2003. If you have a personal or a project recording studio, catering to that market could improve your studio's profitability. Producing karaoke CDs involves either turning out one-offs for artists and other specialty users or creating masters for volume replication.

Virtually every form of music with vocals has found its way onto a karaoke disc. Everyone has heard the discs being used in karaoke bars, where the music roars from the speakers, song lyrics appear in sync on a giant TV screen, and wannabe singers dreaming of stardom warble into a microphone. The bad news for musicians is that a KJ (karaoke jockey), like a DJ, costs a bar owner less than a live band. The good news is that studio owners can still profit from this relatively new frontier. You can find work arranging and performing the music, mixing and editing the tracks, and creating master CDs with the special software (currently available only for Windows) and hardware that the format requires.

If you use standard studio gear and editing tools to process and mix tracks that fit karaoke's vocal requirements, you can use CD+G (compact disc plus graphics) authoring software to record karaoke discs. You can sell your services to others or use them to turn your own music into reproducible, marketable karaoke discs.

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PLAYING THE MARKET

The market for karaoke CDs is generally limited to KJs and to home enthusiasts who have karaoke parties or use the discs to teach and entertain children. Despite relatively few customers, however, making the discs is still a potential source of income. For small record companies, selling hundreds or a few thousand copies of a disc with low production costs can be profitable. Karaoke discs typically offer fewer songs at higher-than-average prices. Discs containing from five to eight songs, each between three and five minutes long, are most common. The CDs are often packaged in multidisc sets at prices approaching $100.

Although sales volume is building rapidly, the karaoke market is not yet big enough to be a part of the traditional record-distribution system. Karaoke CDs are sold through specialty distributors and dealers that also sell pro- and home-karaoke equipment. Manufacturers also sell discs direct to users. You can find such outlets by searching under "karaoke records" on the Web. If you choose to create and duplicate karaoke CDs, those dealers and manufacturers are your potential wholesale customers. You can sell direct if you have your own Web page, and you can find additional customers on Internet newsgroups like alt.karaoke.music.

The market appears to be growing. Major retailers like Target, K-Mart, and Radio Shack carry low-cost karaoke players along with a few companion (mostly children's and party) CDs. A strong indicator that the market will expand is illustrated by a record distribution deal struck in early 2004. Warner Brothers has agreed to distribute karaoke CDs produced by a company called The Singing Machine, the largest U.S. maker of karaoke equipment and discs.

Although karaoke-CD production and marketing may become more formal and dominated by high-volume manufacturers, three factors make karaoke production an option for studio owners to consider. First, a small producer of discs can still deal directly with buyers at specialty stores. Second, a producer can take a demo (and, if necessary, someone to sing along with it) into a karaoke bar, and the KJ will probably be delighted to audition the work and gauge the audience's reaction. Third, for a while at least, any studio that is equipped to burn karaoke CDs will have a competitive service advantage over one that isn't.

TECHNICAL AND LEGAL

Invented by Philips NV in Holland and developed primarily by Phillips, Sony, Pioneer, and JVC, the technical standard that makes karaoke possible is an extension of the Red Book specification for burning audio CDs. The extension governs the creation of compact discs in CD+G format, which enables simultaneous recording and playback of music (WAV or MP3), text (TXT), and graphics (BMP or JPEG) files.

Karaoke discs contain music and images, but a karaoke CD+G disc is neither an audio CD nor a standard CD-ROM. Preparing the music file for a karaoke CD involves the same production tools used to make any other recording. The musical arrangements are written to support individual singers, and the final stereo mix is recorded with no lead vocals.

Karaoke disc producers sometimes use vocal-suppression processing on the original hit recording. In theory, this process leaves bass, highs, and overtones intact so that a shell of the original instrumental chart remains. (The two software programs that I'll describe later include vocal removers.) Vocal suppression is not always possible or completely effective, however. The degree of success depends upon factors such as the level of the vocal in the original mix, the frequency range of the singer's voice, and whether the source material was originally in stereo or in mono.

Although most karaoke productions are remakes of songs that are already pop hits, a karaoke arrangement must not violate copyright law. In the United States, anyone who wants to record any published song can legally do so with or without the approval of the copyright owner by obtaining a compulsory license (usually through the

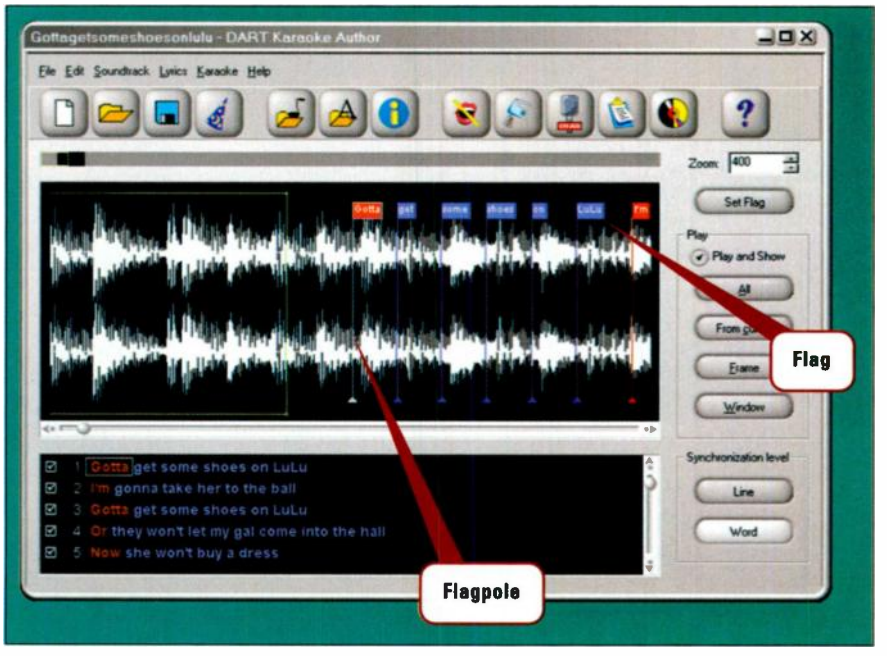

FIG. 1: After you have synchronized the words and music, DART Karaoke Studio's Authoring screen lets you make timing corrections by dragging flagpoles and flags.

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Harry Fox Agency) and paying the statutory royalties set by the congressionally mandated Copyright Tribunal. But compulsory licenses do not cover the charts, and most owners of the arrangements of original hits will not sell or license them and cannot be compelled to do so. Legitimate karaoke producers write or commission their own arrangements (which are sound-alikes but not outright copies) to avoid copyright infringement. Consequently, vocal suppression isn't always an ideal solution.

When the karaoke producer owns all the intellectual property rights to a song, its lyrics, and the charts, however, legal and ethical questions are no longer issues. Of course, singing new original material might not appeal to the average nonprofessional singer in a karaoke bar, but a CD+G disc of original music might come in handy for a performer doing a track date at a dance club or another venue equipped with a professional karaoke player.

THE PRICE IS RIGHT

Less than five years ago, software to create CD+G karaoke discs sold for $15,000 or more. Two software suites that currently sell for less than $200 are the only commercially produced and supported programs I have found. They appear to be among the music industry's best-kept secrets.

Karaoke technology was invented and developed by major consumer-electronics firms, but none of those giant corporations openly market the software necessary to create CD+G masters. In the United States, that job has been left to two relatively small companies—Dartech Inc. (DART) and Micro Technologies Unlimited (MTU). Both produce programs that assemble music, lyrics, and graphics into a composite file. Downloadable demo versions of the software are available on each company's Web site. I tried out the most recent versions on a PC running Windows 98SE, equipped with an Athlon/1.6 GHz and 512 MB of RAM.

KARAOKE STUDIO CD+G

You can download DART Karaoke Studio CD+G (Win, $199.95; currently version 4.1.6) from Dartech's Web site, or you can order a copy on CD-ROM for $10 more. You can also download a fully functional trial-period demo package. DART Karaoke Studio CD+G contains several interacting components, including a basic version of the company's standalone disc-burning software, CD Recorder 4.1.

Karaoke Studio CD+G runs on Windows 98 or later and requires a 350 MHz CPU, 16 MB of RAM, and 1.2 GB of temporary hard-drive space to store the composite file for each 74-minute (maximum) CD. A sound card and a suitable CD-RW drive (more on this later) are also required.

The program's Wizard imports WAV or MP3 files for music, and TXT files for lyrics. In addition, it collects title, writer credit, and miscellaneous notes such as copyright information. (The graphics files are added later.) Karaoke Studio CD+G's Wizard is intuitive and easy to learn and use, but you can bypass it using optional file-opening and -importing features.

To import lyrics, you prepare them as a standard text file with any word processor. Capitalization and punctuation should be used to match the vocal and musical phrasing. Although the program will wrap long lines, the best practice is to divide phrases in such a way that lines contain no more than six or eight words (depending on word length) to make it easier for singers to read from the video display.

You can manually synchronize words and music in the Karaoke Author window that comes up when you exit the Wizard screen, creating a DAX file containing those newly coordinated elements (see **Fig. 1**). The object of synchronization is to enable successive lines of lyrics to be displayed on screen by either scrolling or repainting the screen, and to enable either individual words or whole lines of lyrics to change color just before they are to be sung.

As the music WAV or MP3 file is imported, its stereo waveform is displayed. Standard waveform selection and zooming tools are available. The imported lyrics appear in a window below the waveforms. You synchronize music and lyrics by playing the music and watching the lyric display. At every point in the music that corresponds to a word, you tap the computer keyboard Spacebar and a vertical "flagpole" marks the waveforms in both channels.

At the top and attached to each flagpole, the lyric word or line appears as a horizontal banner, or "flag." Poles and flags can be manually adjusted to correct for any mistiming of the Spacebar keystroke and then locked in position. Because the waveform can be magnified to any desired level of detail, you can move and position the flags precisely. As you're placing the flags using the Spacebar keystrokes, they are positioned on the waveform at a spot just

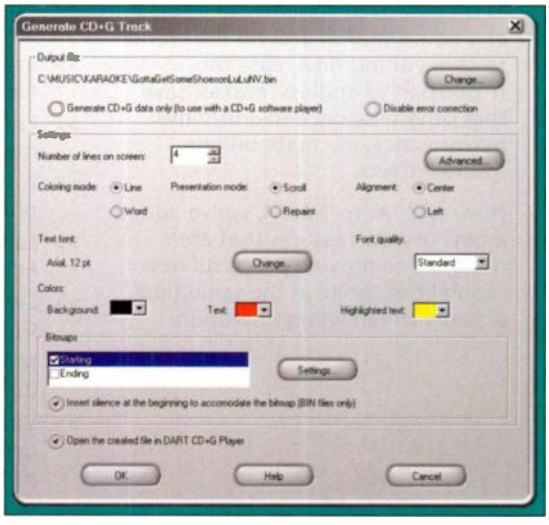

FIG. 2: Manage the creation of recordable binary (BIN) files in Karaoke Studio's Generate CD+G Track window. You can choose from file and display options and add graphics to the opening and closing credits.

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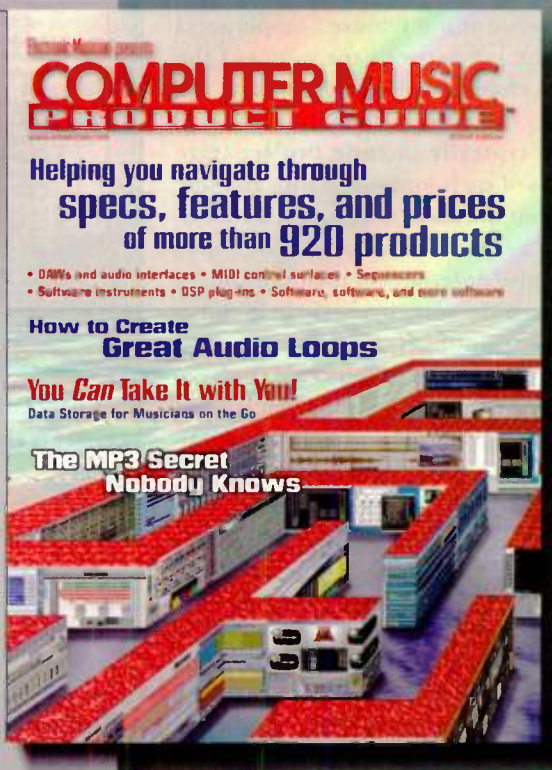

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before the word is to be sung. The exact position is variable in milliseconds; the factory default is 100 ms. This pre-positioning feature compensates for the lag between the moment the word changes color and the time the singer can react. Commercial karaoke discs typically include two separate tracks of each song—one with and one without the lead vocal audio. Both display the lyrics when played. The object is to give entertainers the opportunity to learn the song by singing along before attempting it solo.

For $14.95, you can upgrade the basic DART CD Recorder program to provide vocal removal and some rudimentary audio-editing capabilities. It includes tools for creating WAV files from MIDI or sound-card input, applying DirectX effects, equalizing, fading in and out, and compressing WAV files to WMA format. The bulk of the file massaging will usually have been done earlier, before the music and lyrics are assembled into a DKX file, but these tools may well prove handy.

A NEW GENERATION

Although Karaoke Studio CD+G will work on compressed MP3 files, WAV files are preferred. Obviously, the WAV or MP3 file imported into the Authoring window should be a finished mix. Any audio editing must be done to the source WAV file; the final binary file from which a CD is burned cannot be edited. When you exit the Authoring window, the program automatically creates an intermediate DKX file containing the composite words and music and saves it to your computer's hard drive.

FIG. 3: Karaoke Studio's CD+G Player can play BIN files containing the fully assembled words, music, and graphics of each song.

FIG. 4: DART CD Recorder's playlist lets you drag and drop tracks to set the order in which they will be burned to disc.

Selecting the Generate CD Track command accesses the Generate CD+G Track window, in which final assembly of all music, text, and graphic images takes place (see **Fig. 2**). If this window is accessed just after a song has been synced, that song will be brought up for processing. If a previous project is being reopened, the DKX file for that project can be called up instead.

Once the appropriate DKX file is in place, you can specify what kind of file you want the program to generate. The choices are to create either BIN files, which contain the fully assembled music, graphics, and text that are to be recorded as a karaoke CD, or CD+G files, which contain all the graphics data and assembly code associated with the project, but no music.

If CD+G files have the same name as MP3 or WAV files and both sets of files are on the same CD or in the same folder, then they can be played in sync on any computer equipped with CD+G player software (but not on karaoke machines). The BIN files are fully playable with word display and music on professional and home karaoke machines and on computers with the DART program installed.

Other options in the Generate CD+G Track window include selecting how many lines of lyrics will appear on the performance television screen at the same time. Another option is to choose whether the onscreen words will be scrolled or repainted as the song progresses. You can select whether lines will be justified left, center, or right. You can also choose the color of words or lines.

The Generate CD+G Track window screen also allows for the insertion and preview of introduction and ending BMG or JPEG graphic images such as the logo of a record company. In addition, you can insert and preview textual material such as song titles, key signatures, songwriters' names, and copyright notices. There's also a provision to

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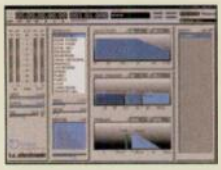
SoundStage & FX:reverb



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* The VS8F-3 Expansion Board is compatible with Roland's VS-2000CD/2400CD/2480 series recorders. Support for the VS-1680/1880/1824 is planned for summer 2004. Five Roland plug-ins are included with the VS8F-3 (Stereo Reverb, Tempo Mapping, Pre-Amp Modeling, Mastering Toolkit, Vocal Channel Strip). All other plug-ins, VGA output, and monitor are sold separately.

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insert silence so that the music doesn't play until after the brief introductory logo display.

Finally, you can specify that as soon as the final composite BIN file is created, it will be played back by the CD+G Player program (see **Fig. 3**). At that point, the OK button starts the assembly and file-making process.

AUDITION AND BURN

The CD+G Player screen has the same transport and track-selection features as an ordinary computer CD player. It also displays a simulated television screen on which the lyrics and graphics are reproduced, just as they would be on a performance karaoke machine. Despite its name, the Player subprogram cannot read commercially available karaoke discs or discs created by the DART software.

If the playback performance is satisfactory, your next step is to burn the karaoke CD. If the playback doesn't meet your needs, then you can go back and revise the original source files and repeat the steps in the generation process.

At this point, your computer's hard drive will contain as many as seven files for each song: the music; the lyric text; the initial composite DKX file; the intro and ending logo files; a CD+G file; and the final, fully assembled binary (BIN) file. The binary file is the one that will be burned to disc, but standard CD-recording software such as Ahead Software Nero or Roxio CD Creator cannot correctly burn a karaoke disc; it must be burned with the CD Recorder program included in the DART package (see **Fig. 4**). When this program is started or called from a menu button, a page will appear in which all the songs to be recorded on a new CD can be imported from their resident folders and placed in the desired order on a drag-and-drop list.

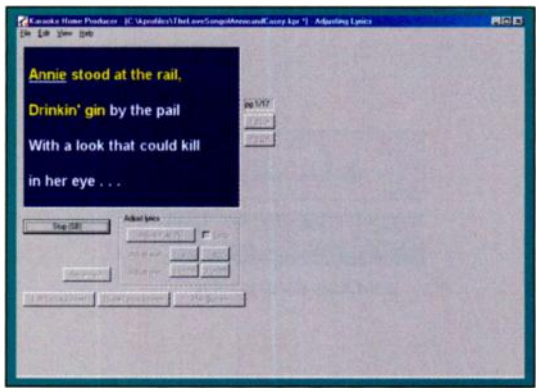

FIG. 6: KHP's Adjust Lyrics screen allows you to correct synchronization errors precisely.

When all of the desired song processing has been completed and the track list has been compiled, the final step before starting the burn is to set the recording speed. As I'll discuss later in more detail, karaoke disc burning is a tricky process that is more likely to succeed at the slowest speed available (typically 4×).

When the resulting karaoke disc is examined with Windows Explorer or other disk-management software, the file names will have CDA suffixes like other audio CDs. That transformation is necessary because no CD-ROM drives or players, including karaoke players, recognize BIN files as audio files. It is technically possible to record the binary files that DART software creates to a CD using Roxio or Nero software, and the BIN suffixes will show up on a Windows Explorer examination, but the discs will not play in either conventional audio-CD or karaoke players.

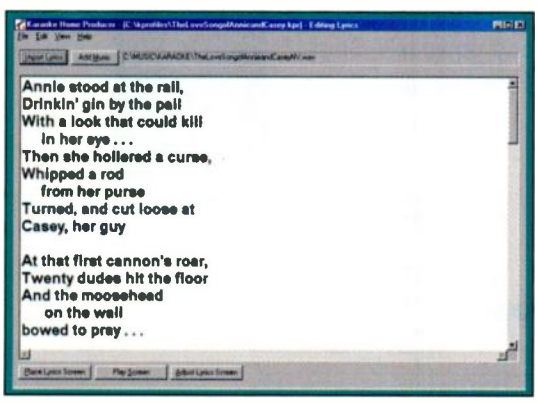

FIG. 5: After music and text files are loaded in MTU's KHP, tapping the Spacebar in time with the music sets the color-change points in the lyrics.

MTU KARAOKE SUITE 4

Micro Technology Unlimited (MTU) offers a four-piece software suite called Karaoke Suite 4 (Win; $138.95 for download; $150.95 on CD-ROM). As of this writing, Karaoke Home Producer 4.103 creates CD+G files; Vogone 2.210 eliminates vocals; Keyrite 1.400 manages pitch change; and Microstudio 2.517 drives the karaoke CD burner. The vocal removal and pitch-change components are standalone products, whereas the creation and burning software are interdependent.

Karaoke Home Producer (KHP) anticipates that you will have all the necessary files in final form before embarking on a CD-recording project. The program doesn't offer detailed audio and image editing. Lyric text files should be prepared with singer phrasing, font size, and visible text lines in mind. A provision for creating simple text and graphics for the opening and closing credits is included, but fancier creations will need external programs.

When you start KHP, a plain vanilla screen with a Windows Explorer–style frame appears for selecting the TXT-file lyrics (which will fill the blank screen immediately) and the WAV audio file (see **Fig. 5**). When both files are loaded, a single Place Lyrics button will appear. Pressing it will access the Adjust Lyrics screen and display the lyrics in reversed type (see **Fig. 6**). The first word in the lyric will be underlined, and when you press the now-visible Start button or the Spacebar, music playback will begin.

When playback starts, tapping the Spacebar in time with the music moves the underscore from word to word and fixes the point at which a color change will occur. There is no option for line-by-line rather than word-by-word color change.

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For correcting errors, you can tweak the timing of individual words in the Adjust Lyrics window. You can set a playback loop for any number of words before and after the one to be corrected, and four buttons will move the starting point and duration of the color sweep for the word. The color-shift effect will be seen as the music loops. This arrangement lets you select and play individual words and sections for precise timing.

Lyric font selection and graphics management are handled on a clear, easy-to-understand Song Settings page that you access from KHP's Edit menu (see **Fig. 7**). You can import images for the song's intro and closing credits in a variety of formats, including BMP, JPEG, PNG, and TIF, and you can set how long they will appear on screen.

Closing the KHP program creates a KPR file that can be reopened and edited. The KPR file contains the lyrics and the coded instructions for assembling the music, the text, and the graphics. You can play this file only from within the program.

When the synchronization process is finished, a single command in the File menu creates and exports a fully assembled CD+G file of the song, which you can play in an onscreen replication of a karaoke machine. The CD+G file is also the one from which CDs are burned using Microstudio.

FIG. 7: You can adjust font attributes and graphics in KHP's Song Settings page.

Microstudio is a standalone program that you open by clicking on its desktop icon (see **Fig. 8**). The opening menu provides access to the controls for drive speed (6× is the maximum recommended), caching, and jitter. Toolbar buttons call up other screens for importing the CD+G files (either from the hard drive or a CD), devocalizing, pitch change, and creating a savable playlist that puts the tracks in the desired order for recording. When the playlist is ready, pushing the Write button burns the CD. The write process begins with no warning, and if it is interrupted the disc is spoiled. The finished disc is automatically ejected, and the CD+G tracks will have CDA file name extensions.

BURN, BABY, BURN

Most CD-RW drives are incapable of burning CD+G discs because they do not have internal firmware that recognizes files containing the graphics extensions to the Red Book standard. Some older drives capable of writing CD+G files with earlier software are listed on the DART Web site, but they might not work with the new software described here. Drive manufacturers traded off the CD+G burn feature in later drives to accommodate the firmware requirements of new CD-R/RW and DVD+R/-R/+RW/-RW multifunction drives.

Specifications for several new drives list CD+G file compatibility but do not always reveal whether a drive is limited to read-only capabilities. Old CD-RW drives might not work with DART or MTU software, and buyers need to be careful in selecting a new drive. Be sure the specs state that the drive reads and writes CD+G and BIN files. All CD-ROM drives will play the audio from karaoke discs, but very few will play the graphics files, even when the computer is equipped with demodulating software.

DART advises that the only drives now in production that work with its

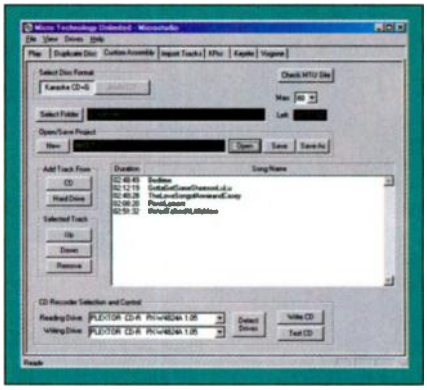

FIG. 8: MTU's Microstudio manages the selection of recorder burn settings, creation of a playlist, and the final assembly of music, text, and graphics.

current software are those offered by Plextor America, which traditionally has made drives for professional applications. According to Plextor, the company's drives have black trays to reduce optical jitter effects and provisions for future firmware updates via download. I used a Plextor PlexWriter 48/24/48A while testing the software and media discussed in this article. Most but not all Plextor drives support CD+G read and write; check the specifications before you buy a drive. Although Plextor drives contain the technology to prevent buffer underrun, as well as a PowerRec feature that automatically selects the optimum recording speed for conventional CD burns, DART strongly recommends a manual burn setting of 4×.

MTU recommends a rate no higher than 6×. It also recommends only its own karaoke-certified CD-ROM drives. The company's software worked with the PlexWriter, however, and I did not test an MTU drive.

Not all discs are created equal, and not even the Plextor drive can successfully burn all brands. You'll need to test discs from each batch you buy, as formulations are constantly changing. Plextor recommends Verbatim discs; DART has tested and recommends Kodak and Fuji. MTU recommends only its own branded discs. Both companies caution against using "audio" or rewritable discs, because they simply don't work.

CONTACT INFORMATION

Dartech Inc. tel. (800) 799-1692 or (952) 844-0217; e-mail info@dartech.com; Web www.dartpro.com

Micro Technologies Unlimited (MTU) tel. (919) 870-0344; e-mail sales@mtu.com; Web www.mtu.com

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FREEDOM OF CHOICE

I used software programs from both companies that were the most recent versions in January 2004. Each one exhibited minor bugs that were corrected during the course of writing this article. DART had by far the better tech support.

Karaoke Studio CD+G and Karaoke Suite 4 both use copy-protection schemes designed to prevent sharing. Every MTU program installation (even from the CD) demands a direct or indirect (via floppy disk) Internet connection, requires personal information, and compels acceptance of a secrecy agreement. It also limits the number of times the software can be installed and on how many machines. I found this mechanism to be irritating and intrusive.

A discrepancy exists between the two products in how the final files are named. DART calls the file containing the composite music, text, and graphics a BIN file; MTU calls it a CD+G file. Apparently, there's no industry standard.

The DART and MTU software accomplish the core job of creating discs that can be played in home and commercial karaoke machines. Both programs are complex and can lead to some frustration and spoiled discs before you have mastered the entire production process. The two products demand some manual dexterity for the word-by-word music synchronization (DART's line-by-line option is a plus). Assuming that the basic music, text, and graphics files are ready to go, with practice you should be able to process one song to prepare it for recording in a half hour or less.

Even assuming that you have to buy a new CD-ROM burner, either program will give you a foot in the door of a fast-growing business for far less than the cost of one good microphone. And the great part is that you can keep fussy, off-pitch vocalists out of your studio and in the bars where they belong.

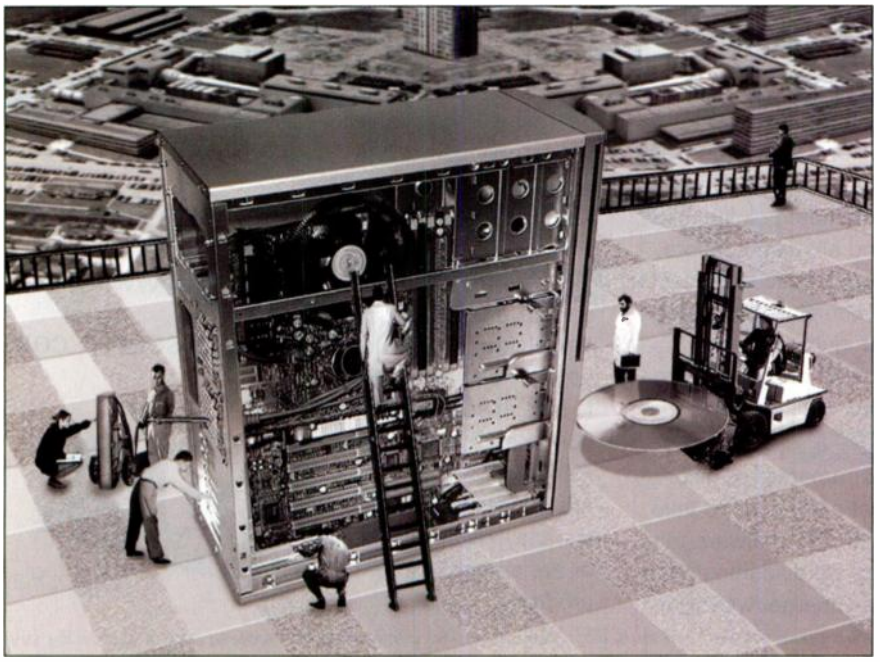
Will Connelly *leads the River Liffey Saloon Jazz Band in Fort Lauderdale, Florida, and has produced records, concerts, and television. Special thanks to Nancy Franks, a South Florida KJ who helped test CDs created with DART and MTU software.*

Conquering the DAW

Preparing yourself and your computer for a digital audio workstation.

By Kevin Smith

Your computer purrs next to you as you read the advertisements touting unlimited tracks, real-time digital effects, virtual-instrument hosting, onscreen mixing, full automation, and on and on. You think, "I've gotta have that!" But before you march down to your local dealer or surf to your favorite online shop, make sure you take the steps necessary to ensure that you need what you get and get what you need. In this article, I'll walk you through the crucial steps in choosing digital audio workstation (DAW) software that matches your computer, your needs, and your budget.



DMITRY PANICH

Although the term *DAW* is often reserved for high-end systems that include both hardware and software, most of what I'll cover here applies equally well to any software designed to arrange and play back audio and MIDI files. The charts for "Computer-Based Digital Audio Workstations" and "Sequencers" in the **EM** *2004 Personal Studio Buyer's Guide*, available online at http://psbg.emusician.com, list all programs of that type.

The first question you must ask in evaluating a DAW is whether your computer is up to the task of running it. To answer that, you need to know what you have (or, if you'll be buying a computer, what you're getting). Take the time to research the main components of your system. What type of the central processing unit (CPU) does it have? How fast is the CPU? How much random access memory (RAM) is installed? Can the memory be expanded? How big is the hard drive? How much space is still available? What types of expansion ports do you have? How many of each? What size is your monitor? What

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resolutions does it support? Can you add a second one? Finally, what version of the operating system (OS) do you have? Will your computer support an OS upgrade if necessary? You shouldn't overlook any of those questions; your computer's System Properties (PC) or System Profiler (Mac) window can provide many of the answers (see **Fig. 1**). It's a little like finding out whether your car has antilock brakes and air bags; you don't want to just assume it does.

Although PCs running Linux are beginning to appear on the music scene, for all practical purposes your decision is between a PC running Microsoft Windows and a Macintosh running Apple OS X. I am often asked which is better. Historically, the Mac has been the computer of choice for music and graphical design, but the PC has now bridged the gap. PCs usually require a little more research; Macs tend to be more expensive. Not all digital audio software runs on both platforms, so your chosen DAW may make the decision for you. Compatibility with your collaborators is another important consideration. It ultimately comes down to your personal preferences and the requirements imposed by the software you use and the environment in which you work. (For some helpful tips, see the sidebar "Computer Checklist.")

SPEED IS KING

It's true that DAWs that include digital signal processing (DSP) hardware, as Digidesign's Pro Tools does, tax your computer's resources less than *native* systems, which rely on your CPU for all their processing. But a fast processor, lots of RAM, and a big, speedy hard drive are always better. Native systems are less expensive, of course; however, when you add the cost of the extra computing power necessary to achieve equivalent performance, the savings you anticipated may vanish.

All of that means you need to be realistic about your computer's capabilities. Whereas your die-hard Pentium III/466 MHz lets you surf the Internet like nobody's business and never crashes, it will probably not be able to handle 24 tracks of 24-bit, 96 kHz digital audio. Add DSP plug-ins, real-time automation, virtual instruments, and so on, and you're going to need a considerably more powerful machine to get any work done. That your computer meets the DAW's minimum system requirements simply means the DAW will launch and successfully record and play back a track or two. You can set yourself up for success by investing in a machine that is designed to handle the workload you intend to give it.

You can save money and headaches by evaluating your needs carefully. How many tracks do you need to be able to play back simultaneously? Will you be recording live at all? If so, how many tracks at a time—do you plan to record your whole band at one go, or to track them one instrument at a time? Are you going to be doing post-production mixing and finalizing, or will you simply be recording and editing tracks? Do you plan to use a MIDI control surface for automation and a MIDI keyboard for playing virtual instruments? Once you've defined your primary goals, you can find the product that meets your needs, offers potential for growth, and fits your budget.

Unless you are on the cutting edge of technology, there's a good chance that other people are using the exact system you're contemplating. Find out about their experiences. Don't just read the manufacturer's brochures and visit the dealer's Web site; look for real-world

COMPUTER CHECKLIST

With so much riding on the decision, choosing the right computer is a critical part of setting up your studio. Here is a checklist of some of the more important factors you should consider in order to make the right choice.

✓ **The computer's age.** You can update the processor, RAM, and hard drive in most computers, but that's not the full story. A number of equally important features—bus speed, number and type of expansion ports, number of hard-drive slots, and maximum RAM capacity—usually cannot be upgraded. A current processor may make it easier to surf the Internet, but if your computer is showing its age, consider a new machine for audio instead of an upgrade.

✓ **Processor speed.** You should usually get the fastest processor you can afford, especially if your DAW is going to rely on it for DSP. A faster processor allows for more tracks, more plug-ins, and increased stability. However, you don't necessarily need to buy the latest model to get a fast processor. Frequently, excellent used machines are offered at bargain prices by users who *do* insist on the latest thing. Keeping your eyes open for such an opportunity can save you lots of money. Furthermore, if your DAW doesn't take advantage of new features such as dual processors or 64-bit busing, you're wasting your money buying a machine that has them.

✓ **RAM.** Have at least 256 MB more RAM than the minimum stated requirements for your DAW. More is always better, and 1 GB is now common on music systems.

✓ **Hard drive.** Have at least a 100 GB internal hard drive. Better still, have two internal drives and dedicate one to recording and playing back audio files. Also consider getting a separate hard drive (internal or external) for backing up your files.

✓ **Expandability.** Expandability extends the life of a computer by allowing your system to grow along with your needs. It is an area in which desktop and laptop machines differ significantly. Whereas laptops give you a variety of expansion ports, including FireWire 400 and 800, USB 1.1 and 2.0, Ethernet, and PC card slots, desktop models typically give you those as well as multiple PCI slots, multiple peripheral bays, and greater RAM capacity. Think about your expansion needs and buy a machine that can meet them.

✓ **Operating system.** The latest operating system is not always the best choice, especially if it represents a recent and major upgrade. Software and peripheral manufacturers typically take some time to iron out compatibility issues, and if you've jumped on the bandwagon too early, you can find yourself with a temperamental (or worse, inoperable) system. Before you upgrade or buy a new machine, make sure the new version of the OS is supported by all your critical software and by the drivers for your audio card and other peripherals.

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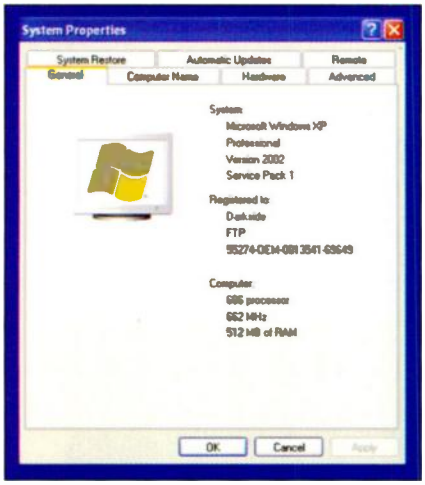

FIG. 1: The Windows System Properties window (shown here) is the best starting point for gathering information about a PC; on a Mac, it's the System Profile window.

accounts. Use your favorite Web search engine to compare products. Just search on the word *review* or *compare* along with the name of the product you are considering. Yahoo Groups are another great resource. You can use them to connect with other users and learn about potential pitfalls before you commit to a product. Many companies host user groups or product forums on their Web sites. There, customers discuss issues and exchange ideas with the manufacturer. Some companies also provide downloadable manuals. Reading the manual is a great way to get a better idea of the product's features and requirements and to get a feel for how it works before you buy.

ROLL UP YOUR SLEEVES

Once you have settled on a program, you are ready to prepare your computer for digital audio. Here again, thinking ahead can save you a lot of grief. You may not have anticipated the extent to which a DAW will take over your computer. I strongly recommend dedicating a computer to your DAW. Keeping your favorite games and graphical-design software and their associated drivers on the same machine as your DAW will increase your system's susceptibility to conflicts. Space is not the issue—the days of small hard drives and limited RAM are over. The problems a separate computer lets you avoid are program and driver conflicts that lead to crashes and loss of data.

If you're not in a position to dedicate an entire computer to digital audio, the next best option is to partition your hard drive and install an OS on each partition, placing your DAW on one partition and any software unrelated to audio on the other. Because you have to restart your computer when you want to switch systems, it's not exactly like having two computers, but the added stability is worth the inconvenience. Macs and PCs come with utilities for formatting and partitioning hard drives (see **Fig. 2**). Remember that partitioning a drive usually requires you to erase all of its contents, so be sure to back everything up first.

Whether you're using two computers or two operating systems on the same computer, I suggest running your DAW on a fresh copy of the operating system. That way, the only drivers that reside in that system are those installed by the DAW and its ancillary hardware and software.

File management is another crucial aspect of digital audio, and you need to know how your computer works with files and folders. DAWs typically rely on the operating system for file management. You need to know how to save, retrieve, find, move, copy and delete files on your system. Audio files are just like other types of data and can be handled in the same way. The difference is that you'll eventually have hundreds or even thousands of audio files to keep track of, and one misplaced file can be a showstopper.

I also recommend that you start with a clean hard drive—one that has been freshly partitioned or at least recently defragmented—for your audio projects, and that you use that drive solely for your audio-project files, not for your software or operating system. Create a folder hierarchy that makes sense and

TIPS FOR SUCCESSFUL TECH SUPPORT

The success of your tech-support phone call is largely in your hands. Even well-trained, patient tech-support representatives are only as effective as the information they are given. I asked the highly regarded tech-support specialists at Digidesign to tell me the five most important actions callers can take to make a tech-support call successful; here's what they said.

1. Be prepared. Having as much specific information available as possible will greatly improve your chances of solving your problem. Know as much as you can about the components of your system. Search for the answer in the manual and online before you call. Carefully note the steps that led to the problem and the steps you've taken to try to solve it.

2. Have patience. The hold time for tech support for popular products is usually long. Starting out with a bad attitude because you've been hanging on the line for 20 minutes, however understandable your frustration, is guaranteed to get the conversation off on the wrong foot. Remember that in all likelihood, the tech-support representative just got off the phone with someone who was as anxious as you are to get things up and running.

3. Have realistic expectations. The immediate solution to your problem may be a work-around—a makeshift and possibly awkward way of achieving the desired result. Although you should continue to push for a long-term solution (that's what you paid for), bear in mind that tech support's primary objective is to keep you productive.

4. Listen. In many cases the solution will not be a simple, one-step fix. A tech-support representative typically needs to make educated guesses to get to the root of the problem and solve it. That means you need to listen, follow directions carefully, and be willing to repeat something you may have already done.

5. Be courteous. Tech support may well be your only path to a solution, and courtesy can make all the difference in getting results. Keep in mind that many tech-support representatives are working musicians and can understand your position. And although they're trained to take the heat for problems with the product, a personal attack will lead most representatives to terminate the call.

is easy to follow. For example, you might set up a folder for each client; within those folders, you could create a folder for each client project, and within the project folders you could have separate folders for DAW project files, audio files used in the project, and other data files related to the project. You might also have a folder for audio resource material and, within that, separate folders for different types of audio files such as sound effects, instrument samples, percussion loops, and so forth.

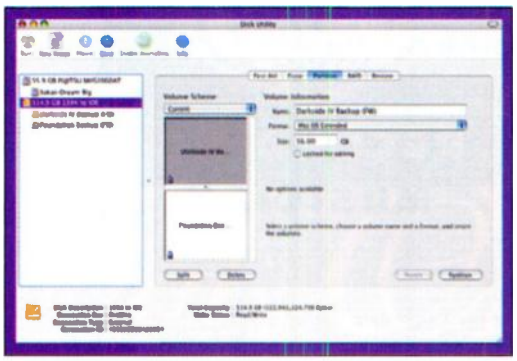

FIG. 2: Apple's Disk Utility has been greatly enhanced for OS X. You can use it to format, partition, check, and repair your hard drives.

It's vital to know where your files are being saved by your DAW and to have instant access to all your data. Diligence in naming and dating your files is also a must. Invariably you'll wind up with two versions of the same song (perhaps the guitar player took a version home to lay down some tracks) and want to keep only one. If everything is clearly named and dated, you'll have a much easier time sorting it out.

TAKE YOUR TIME

A rushed job is usually a botched job. Not only is it important to take your time when installing a new system, but it's crucial to do so only when you are not facing an impending project deadline. It takes just one missing driver or flaky piece of hardware to put you out of action.

Whenever possible, I suggest doing a new installation or an upgrade in stages. If you are relatively new to the process, try to work a couple of hours a day for several days rather than doing the installation in one marathon session. Remember, it's exacting work, and when you're tired, you are more likely make mistakes.

It's also important to take your time learning to use the system and your software. You may not like reading manuals, but doing so is well worth your while. Also, don't neglect any tutorials that came with the software. Understanding your software will save you time and money you might otherwise spend on tech support. Hanging on a long distance phone call for 20 minutes only to have the technician point out that your question is answered on page three of the manual is avoidable.

On the other hand, don't be afraid to seek assistance from tech support if all other avenues fail. To get the most out of your tech-support call, be prepared. Have your hardware, operating system, product version, and registration information at hand before you call. Do your best to formulate your question clearly and omit unnecessary details. Remember that the person you're talking to is not looking over your shoulder and has just started working on your problem. The more information you can provide succinctly, the better. See the sidebar "Tips for Successful Tech Support" for more suggestions on calling tech support.

As the excitement of getting started with your DAW takes hold, keep the points I've discussed here in mind. Advance preparation is invaluable, and making informed decisions will save you time and money. Once you decide on a system, take the extra time to prepare your computer, because it is the hub of your studio. There is no such thing as downtime insurance, so avoiding a crash, or at least its most dire consequences, is your best hope. (Aren't you glad you made that backup?)

Kevin Smith *is a San Francisco–based writer, consultant, and musician. Visit his Web site, www.familytreeproductions.com, to see what he's up to.*

It's helpful to start a mix with a clear vision of the song, a mental final mix of how you think it should sound. Only when you have some direction for the song can you properly assess the role of the drums. It's all about context.

THINK 3-D

Every great stereo mix also sounds three-dimensional—tall, deep, and wide. *Tall* refers to frequency range (how well all the frequencies are represented), *wide* to lateral positioning (where the instruments "appear" left to right across the soundstage), and *deep* to depth of soundstage (how close up or far away each instrument sounds in relation to the "front" of the soundstage).

Therefore, think in 3-D terms, for not only the whole mix but also for the drums. A drum kit, after all, comprises a bunch of parts spread out all over the place. Reflecting a (believable) sense of that spread-outness will deepen the dimensionality of the whole mix.

One of my personal mix tricks is to take the three dimensions one at a time: width first, height second, and depth third. That helps me focus on maximizing each dimension. Here's the strategy, laid out in that order.

START AS IS

Once you have the vision for the song, see how close you can get to realizing it by manipulating volume levels and pan positions only. Starting that way keeps you focused on the song as a whole and helps avoid the easy distraction of soloing and processing individual tracks, only to find that they don't work once reintroduced to the mix.

FIG. 1: The Waves Renaissance Compressor is a cross-platform plug-in that can emulate both VCA-based and optical compressors.

Don't worry—in the course of shaping the mix using only the tracks as they are, you will soon become aware of what's missing, what's clashing, what needs tweaking or muting, and so on.

SEE THE KIT

When panning drums, envision an area on the soundstage where the kit is set up. Note that you can mix as if looking at the drums from behind or in front of the kit—there's no right or wrong here.

What's important is to choose one perspective and stick with it. To that end, make sure the panning of the overhead mics corresponds with the panned positions of the toms (assuming they are individually close-miked). Simply solo the panned overheads (during a tom-fill passage), note which direction the tom fill goes, make a mental note of where each tom hits, and then pan the individual tom tracks accordingly. This will tighten the drum mix and will help clarify the location and punch of each tom in the stereo field.

The usual goal when panning is to create an uncluttered soundstage in which each instrument can clearly be heard and its position identified. Assuming you're going for a natural drum perspective (the best place to start, usually), keep the overall "width" of the drum image consistent with the number of instruments on the soundstage. In the case of an 18-piece big band, for example, the drum kit would likely be panned into a small area (between, say, 11 and 12 o'clock), so it won't "step on" the other instruments. A power-trio mix, on the other hand, would allow for wider drum pans.

LISTEN, DON'T LOOK

When setting pans, don't rely visually on the location of pan pots. Pan pots used in budget consoles are not very exacting, with the result that their positions on the mixer don't necessarily correspond to the position of the instrument in the stereo field. The solution is to make panning decisions based only on the sound. (I often close my eyes while panning, to better focus on what I'm hearing.)

A very useful technique is to finalize drum-pan tweaks with the mix summed to mono. That might not seem to make sense, but it works. For example, to find the best location for the hi-hats, sum to mono, turn the hi-hat pan knob slowly back and forth (in the general area where you had assigned it), and listen for a spot where the hats seem to step forward a bit or come into clearer focus. What you're doing, basically, is combing the mono sound field in search of a slice of free space.

The mono-panning technique is helpful for positioning kick and snare, too. Though it's customary to pan kick and snare drums on top of one another, typically dead center, try panning them ever so slightly apart, again while in mono, and listen for a spot where you get a sense of separation (the same as you get when listening to a live drum kit). By putting a slight bit of space between the two drums, you unmask the signals from one another, which helps clarify each in the mix.

CHECK PHASE

If the tracks you're mixing include stereo overheads, make sure to check the phase relationships between the two mic signals—poor overhead-mic placement is not uncommon, and its results can ruin a mix. Sum the overhead tracks to mono and listen closely from top to bottom. If the sound collapses, loses frequency content, or in any way sounds weird, you likely have a phase problem.

In that case, there are a couple of solutions. One is to go into a DAW and manually align the stereo sound waves. If that's not feasible, the easy fix is to eliminate (mute) one of the overhead channels. You can always create a faux-stereo image later by panning an effect opposite the mono track, delaying it a bit, and equalizing it differently.

It also pays to check phase relationships between the overhead mics and any close-miked drums. Sometimes the overheads are positioned at such a

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FIG. 2: If silence is golden—at least between hits—the Drawmer MX40 Punch Gate has the Midas touch. Sophisticated yet affordable, this 4-channel noise gate features frequency-dependent gating and a special circuit that adds dynamic punch to drum hits.

height above the kit that the sound waves are practically phase-inverted when they reach the mics (in relation to the sound that's captured by the close mics).

In that case, you might find that reversing the polarity ("flipping the phase," as it's called) on the overhead-mic signals improves the drum sound—for example, by making the kick and snare sound fuller. But before making a decision which way to go, be sure to solo *all* the drum-kit tracks and listen closely, in mono, throughout the song. There are often trade-offs you must make when changing phase relationships, and you should take those into account.

COMPRESS NEXT

After levels, pans, and phase checks, the logical next step is dynamics processing. Compressors tend to alter tonality a bit, so it makes sense to compress before getting into EQ and other processing.

Typically, the most important drums in a mix are kick and snare, so it's wise to reserve your better compressors for those tracks. Unfortunately, most inexpensive compressors don't have what it takes to get awesome kick and snare sounds. This is one of those instances in which the gear can make a big difference. (This explains why so many pro mixers favor UREI 1176LN peak limiters for kick and snare—they sound marvelous.)

In my experience, the best affordable pick for kick and snare drums is a full-featured, VCA-based compressor. One midpriced unit I've had great results with, especially on kick drum, is the Aphex 661 Expressor.

For overheads, you can usually get by with a cheaper compressor (again, VCA based). Lower ratios—1.5:1 to 3:1—are good here, and only slight to moderate gain reduction. The goal is to bring up the low-level stuff enough to make subtleties audible, but without crushing the life out of the performance.

If you're working in a DAW-based studio, you also have the option of using one of the many plug-in compressors on the market. (This holds true for any of the compression applications discussed here.) Though in many cases I still prefer the sonic character of hardware compressors, especially when I'm going for sounds that are more radical, there are many engineers who swear by their plug-ins. The Waves Renaissance Compressor is an example of a sonically flexible and relatively inexpensive compressor plug-in. It sounds good, is easy to use, runs on Mac and PC in a variety of formats, and can be set to emulate either

FIG. 3: Don't be fooled by the diminutive size of the Joemeek MC2 stereo compressor. This elfin box has gigantic powers. Featuring photo-optical control and an auto-variable slope (ratio), the versatile MC2 is perhaps the best-sounding inexpensive hardware-based compressor available—that is, if it's attitude you're after. It works especially well on drum subgroups and room mics.

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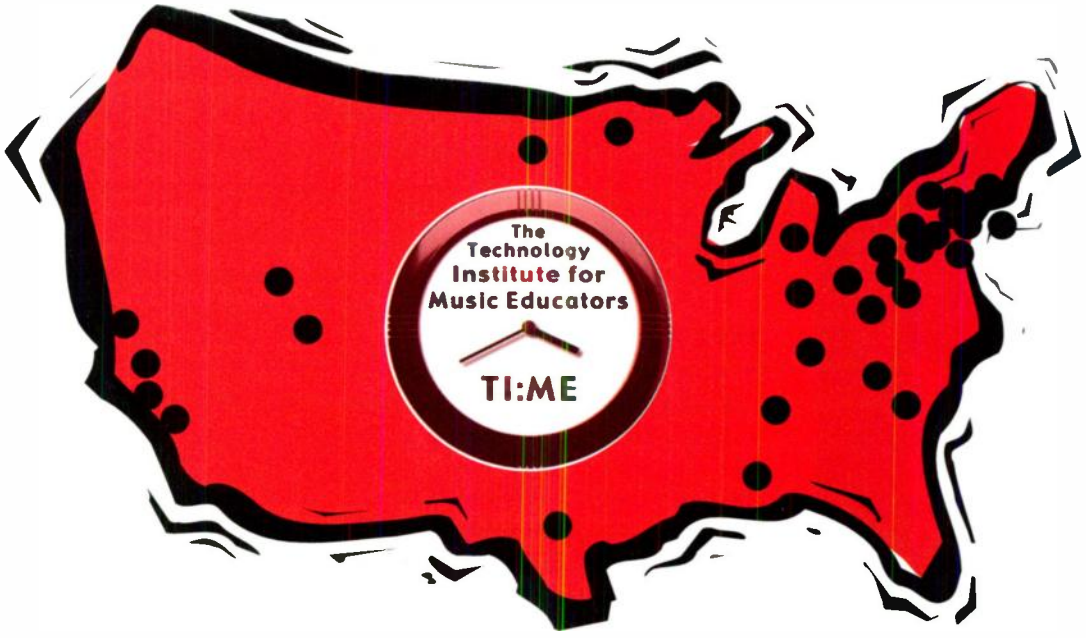

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a VCA-based or optical-controlled compressor (see **Fig. 1**).

If the toms are individually miked, some form of noise gating is almost essential. In the analog realm, my favorite affordable gates are the 4-channel Drawmer MX40 Punch Gate (see **Fig. 2**) and those found on the Drawmer MX30 Gated/Compressor/Limiter (an excellent all-around 2-channel dynamics processor). Of course, if you're working on an automated console or a DAW (or both), you can cut or reduce noise any number of ways—with automated mutes, deletes and crossfades, noise-gate plug-ins, and so forth.

TAKE A BUS

To help make the drums sound huge and extra solid without sounding overly compressed, try compressing all (or most) of the drum tracks a second time through a stereo compressor. Either bus the drums to a subgroup with a stereo compressor on the inserts or return the compressor outputs to a pair of faders, to allow for equalizing. Hit the compressor fairly hard and bring up the compressed channels just beneath the other drum tracks. Mix to taste.

I prefer photo-optical-controlled compressors for this duty. A very affordable opto unit that works remarkably well is the Joemeek MC2 (see **Fig. 3**). The MC2 is the least expensive unit I've found that gets close to a high-end compression sound on snare drum. Opto-controlled compressors are also great for processing room mics. Another low-cost analog compressor capable of giving near-high-end results is the FMR Audio RNC1773, better known as the Really Nice Compressor. A VCA-based stereo unit, the versatile 1773 is useful in many drum applications, including stereo overheads, kick drum, snare, and subgroups.

THE GREAT EQUALIZER

Back to the vision thing; it helps to know *why* you are equalizing a signal, or to at least know what you're going for. There are two main kinds of reasons to equalize, and both describe types of EQ: corrective and creative.

Trying to make an instrument sound clearer, better defined, better balanced, or just more natural all fall into the corrective camp. So does carving out space by juggling frequencies between competing instruments (kick and bass, for example). Creative EQ, on the other hand, moves beyond mere correction of frequency imbalances. Here, you try to make the sound into something different—bigger than life, as if recorded through a telephone, or whatever.

When applying corrective EQ, work to strike a balance between boosting and cutting; too much of either will typically make the instrument sound unnatural. The idea is to keep the envelope as linear and consistent as possible. If you must favor one or the other, favor cutting—cutting never adds noise, but boosting often does.

Bring up the entire rhythm section when dialing-in drum-kit EQ. It's fine to solo here and there from time to time in order to focus in on a particular instrument. For the most part, though, keep the bass and other rhythm instruments in there with the drums. Remember, the concept of equalization pertains to *a relationship among elements*. You need to balance the frequency content of the whole foundation—not of just one element. The goal is to strike a musical balance between all the elements, leaving each more audible in the mix by ensuring that it isn't fighting other elements for the same sonic territory.

Note that effects can be used to help

FIG. 4: The T.C. Electronic M300 Dual Engine Processor is a real jewel for drum processing. Surprisingly affordable given its top-notch audio quality, the M300 has a smart, easy-to-grok interface that lets you quickly dial in two different effects and then blend them with the turn of a knob.

equalize a signal. For example, if the overheads sound too bright, processing the tracks with a dark room rather than a bright one can help take some of the edge off. Better yet, route the effects returns back through spare fader channels so that you can EQ the effect.

DRUM 'N' BASS

In terms of juggling frequencies, pay particular attention to how the kick drum and bass guitar work together. Solo the two and listen closely. Which produces the predominantly lower note and which the higher?

Try accentuating this low/high relationship: carve out some space by equalizing so that like frequencies don't overlap. For example, if you boost the kick at 60 Hz, cut the bass at 60 Hz. If you cut the kick at 400 Hz (usually not a bad idea), try boosting the bass a bit in the same range. Again, the goal is to find a musical balance between the elements.

KNOW YOUR PATCHES

It pays to be well acquainted with your effects processors, too, as they usually play a big part in a drum mix. Keep a list of favorite drum patches handy, or save your favorite presets for your plug-ins, so that you know you can always dial up something that will work.

Drums tend to allow for more creative processing than do other instruments, however, so it's good to keep your ears (and mind) open. Sometimes the unorthodox—a lead-guitar patch on a snare drum, for instance—can be just the ticket.

Personal-studio operators are blessed these days with a plethora of first-rate effects at low prices, be they plug-ins for DAW systems or inexpensive outboard multi-effects units. My current favorite multi-effects units for drum processing are the Lexicon MPX 110 and the T.C. Electronic M300 (see **Fig. 4**).

TWEAK THE TIMING

Time-domain effects, such as reverb, and delay are where it's at for making a drum kit come alive in the mix. There are many ways to use such effects to create depth in a mix, but they all boil down to the same idea: positioning some instruments in the foreground, others in the background, and others in between.

Some engineers are quite finicky about this process, to the point that they calculate minutely different early-reflection (predelay) times for different ends (channels) of the same instrument. Other engineers, including *moi*, do it more by feel.

With multimiked drums, you might need to factor in ambient sound captured by overhead and room mics. Even leakage between close-miked drums can affect drum ambience. The trick is integrating the natural ambiences with any artificial ones you dial in.

One thing I like to play with that can enhance drum-kit dimensionality is the *amount* of effect in each channel. After all, the wetter the signal, the farther back it tends to sound. Even using only a single effect, you can help dimensionalize a kit simply by varying the amount of effect returning to the different channels. Try, for example, putting no effect on the kick, just a bit on the hats, a moderate amount on the overheads, even more on the toms, and the most on the snare.

GET A REFERENCE

Any tricks of the trade that help with mixing in general will help with mixing drums. Those tricks include honoring Fletcher and Munson by monitoring at different levels, employing multiple monitors (including a cheapie system such as a boom box), and remembering to check things in mono from time to time.

I'll close with the most important trick for those who are serious about improving their drum-mixing chops: compare your mixes with those of the masters. When your drum mixes can hold a candle to those of seasoned pros such as Elliot Scheiner, Frank Filipetti, and Bob Clearmountain (to name just a few), you can put the reference discs away. Until then, keep 'em spinning. They will keep you honest.

Brian Knave *is a former senior associate editor at* **EM**. *He lives in Salvador, Bahia, Brazil, where the drums never stop.*

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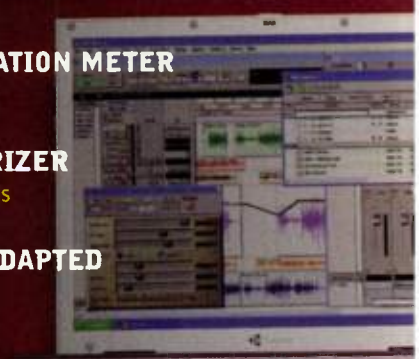

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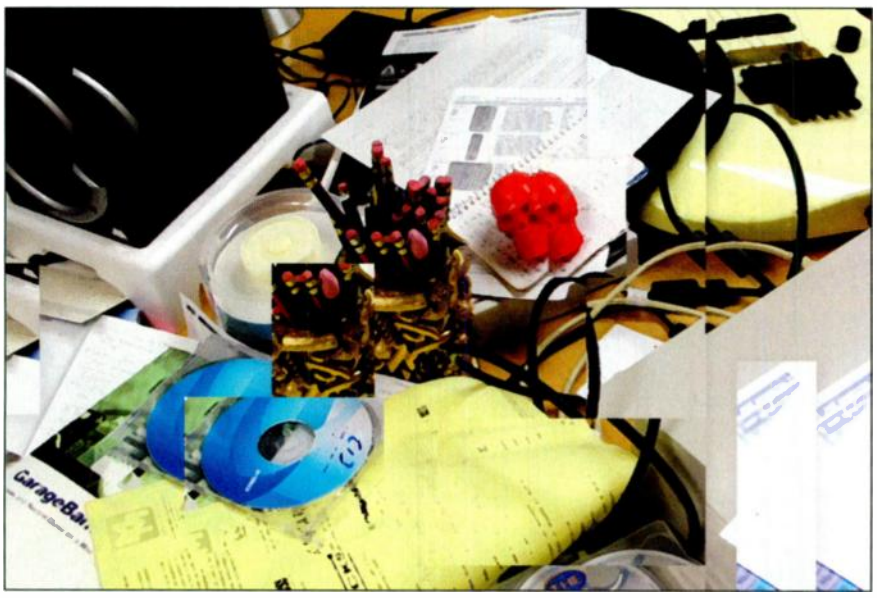
Taking Care of Business

Create billing and booking policies to keep your studio on track.

By Michael Cooper

Running a successful commercial recording studio takes more than great technical and creative chops. It also requires good business sense. Maintaining a healthy cash flow—a steady revenue stream that exceeds your operating expenses—is critical to your studio's success. One aspect of this is controlling and budgeting for both capital expenditures (new equipment acquisitions) and recurring expenses such as phone and utility bills, advertising costs, and inventory replenishment (CD-R and DVD-R blanks, recording tape, and such). It's equally important, however, to protect your business's revenue stream, and that means, in part, making sure that studio bookings don't dissolve and that you get paid for the sessions you execute for your clients.



MIKE CRUZ

Every time a flaky customer cancels on the eve of a session or, worse yet, fails to pay your invoice for completed sessions, you lose precious time and energy that could have been spent serving a paying client. These problems have a direct impact on your cash flow, top-line (revenue) growth, and bottom-line (net income) growth. Over the past 20 years, I've developed my own business guidelines to help preclude these kinds of setbacks.

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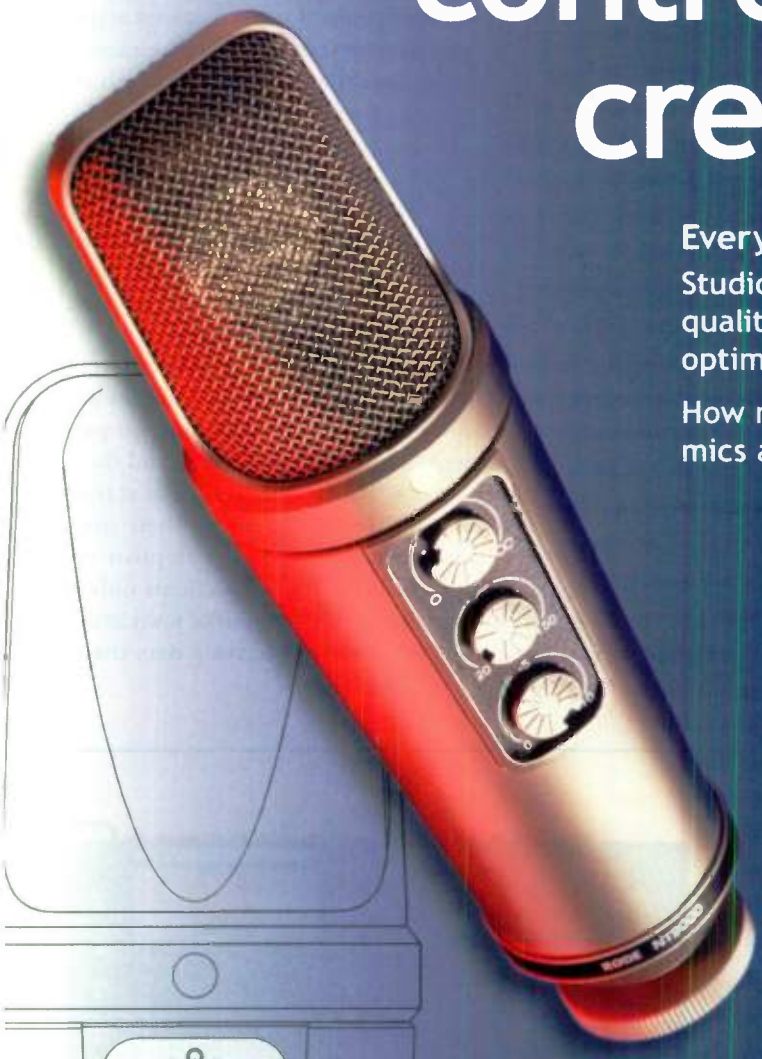

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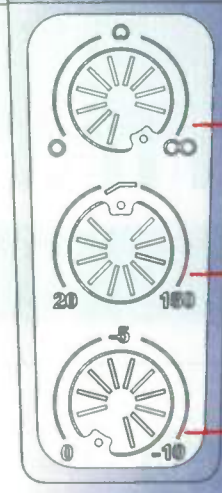

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A MATTER OF POLICY

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- Costs for archival and recording media
- Costs for renting ancillary equipment, if available
- Deposit requirements
- Cancellation policy
- Billing for late arrivals and no-shows
- When payments are due
- Penalties for bounced checks
- Qualifying for flexible payment terms with your studio
- Responsibilities of clients who damage studio equipment

session. Once you're holding their money, they share a common interest with you to make sure the relationship proceeds smoothly and equitably.

You'll need to decide for yourself how big a deposit to require from your clients. For new clients and those who have established a perfect prior payment history, I feel that demanding full payment in advance is not reasonable. Instead, I ask for a deposit equal to one-third of my full-day rate (one-third of my typical pay for a full day's work) for each day a client wants to reserve a studio session.

You'll also need to decide how hardcore you want to be about receiving a deposit for every single session. Some studios require a deposit in all instances from each of their clients, but I've decided to place limited trust in established clients who have a long and perfect payment history with me. For this group, I relax the deposit requirement for reserving a single session and insist on receiving a deposit only if they want to book multiple dates, either in one contiguous block or scattered across the calendar (because in those cases a major cancellation would really hurt me). If a formerly reliable client cancels a session with insufficient notice for me to book with someone else, I require a deposit from him or her thereafter for any and all sessions they wish to reserve. Of course, I always make an exception if the cancellation was necessary because of illness or a family emergency.

I almost always require a deposit from first-time clients. First-time clients are an unknown entity: they have no payment history with your studio, and there is no practical way to know ahead of time whether they usually keep their commitments. I'll waive the deposit requirement for first-time clients only if they contact me from out of town at the eleventh hour to reserve a date that I

have open and that's not otherwise likely to be booked (that is, when there is insufficient time before the session for their deposit to arrive by mail or FedEx).

For clients who have cancelled sessions repeatedly in the past, even with plenty of notice, I always require full payment in advance to reserve any session dates, because these people have demonstrated that they are not serious about their commitments. The same policy applies to clients who have had trouble paying their studio bill in the past, whether or not they've ever cancelled any dates.

CANCELLATIONS

If a session goes as scheduled, I apply the client's deposit as a credit toward their bill. If, on the other hand, the client cancels a scheduled session with less than 48 hours notice before the session begins, *and* I can't book the cancelled date with an alternative client who previously requested the same date, I will politely insist that the client who cancelled forfeit their deposit. I won't keep the deposit if I manage to book the cancelled date with another client. Nor will I keep the deposit if nobody else was initially interested in the cancelled date. I keep the deposit only if I turned down another client for the same date, subsequently suffered a cancellation, and lost business income because I couldn't rebook with the client I previously denied.

I always forgive a last-minute cancellation if my client falls ill or has a family emergency. That's only fair. However, clients who conveniently become sick or suffer family "emergencies" time and time again on the eve of reserved sessions will eventually challenge my good nature and be subjected to stiffer deposit requirements and penalties for cancellations. The only way to handle such ambiguous situations is to follow your gut instinct and do what you feel is equitable. You might even decide you no longer wish to work with such unreliable people. No matter how you decide to handle such clients, however, it's important that you warn them, in advance of their next infraction, the exact penalty they'll incur if it deviates from your standard policies. Nobody, not even a flaky client, deserves surprise treatment.

In all cases, it pays to have a waiting list of alternative clients who have expressed an interest in booking dates that are already reserved for someone else. In the event of a cancellation, a simple phone call or two to people on that list may be all it takes to book the cancelled session.

TIME IS MONEY

Is it just my imagination, or are musicians more time-challenged than other folk? Many of the clients I've worked with could never seem to arrive at my studio at the scheduled time for their sessions. For this reason, I've instituted a policy wherein the "clock" starts rolling within 20 minutes of the scheduled start time. That is, if a client has still not arrived 20 minutes after the session's scheduled start time, I begin billing them at the full rate from that moment on. If you feel that 20 minutes is too long to sit around without getting paid, set a shorter time limit. I prefer

to be a little more relaxed about it while still covering my butt.

If a client is more than 20 minutes late for a session, I call them to see if they're on their way. In fact, it's always a good idea to confirm a session's date and start time with a client the day before a session is scheduled. What if the hours pass, you are unable to contact your client, and they still have not shown up for the session? Once enough billable time has elapsed for the full amount of the client's deposit for the day's session to accrue, I cancel the remainder of the session and move on to doing self-sufficient work, such as performing studio maintenance or installing new software. That policy keeps me from waiting around all day for a client's arrival when only one-third of the day's potential pay (in the form of the deposit) is in my pocket.

GETTING PAID

I require the majority of my clients to pay any outstanding balance, minus the deposit amount, at the end of each single session or, in the case of multiple bookings, at the end of a contiguous block of sessions. The latter arrangement gives clients a convenience incentive to book back-to-back dates, thereby reducing the amount of time I spend setting up and breaking down mics and repatching gear for alternating clients.

I often give select clients who represent large accounts and who have a long and perfect payment history greater flexibility in paying their studio bill. For example, they might be permitted to pay a portion of their balance within 30 days, instead of being required to pay their balance in full immediately after each session or block of sessions. (Because I operate a small studio, the exact concessions I'm able to offer vary according to my cash flow requirements at the time.) A flexible payment plan encourages my very best clients to rack up extra studio hours while maintaining their positive cash flow.

Occasionally, clients forget to bring their wallets or checkbooks to the studio and will have no way of paying the bill at the end of a session. This is usually an innocent oversight, but it can also be a ploy used by dishonest people who don't ever intend to pay the bill. In any case, I always insist that nonpaying clients sign a statement on their bill before they leave the studio that acknowledges that they are responsible for paying the bill and agree to pay the invoiced amount. At least in the state of Oregon, failure to secure such signed documentation leaves you without any power to collect payment, either through the courts or a collection agency. It's a good idea to find out how your state's laws handle such matters.

Beyond maintaining their personal integrity, what inducement does a client have to pay their bill? That's simple: no payment, no masters. I never release finished product—even to clients who have a perfect credit history—before receiving

full payment for work performed to date. I have no problem with releasing rough mixes upon receiving a signed promise to pay any balance due, but final mixes and mastered product are never released until I get paid in full for my work.

SPELL IT OUT

Most clients will accept all of the above policies if you spell them out ahead of time in written form. Prepare a policy sheet (see the sidebar "A Matter of Policy") that details your studio's rates and how your business handles such matters as deposit requirements, cancellations, and scheduling of payments. You should also note any fees for bounced checks and damaged studio equipment on this form. Regarding the latter, inform all prospective clients in writing that they will be held responsible for paying for the repair of all damage to studio equipment arising from their abuse, negligence, or misuse.

As a routine procedure, hand out copies of your studio-policy sheet to all people who tour your studio, or mail copies to those who book sessions over the phone. If a client later takes issue with the details of their bill, you can remind them that such matters were fully disclosed in the studio policy sheet they received before their first session. If you really want to cover your butt, have your clients sign a statement acknowledging receipt and acceptance of the terms disclosed in your policy sheet.

The business policies I've outlined here may seem too corporate or hard-nosed to the studio owner who just wants to make great music and form creative bonds with clients. But you will only come off as being hard-nosed if you present and enforce these policies without empathy and respect for your clients. Your delivery is what determines whether your clients perceive you as reasonable or pushy. Be polite, but take care of your legitimate financial interests. It's all part of running a successful business.

EM *contributing editor* **Michael Cooper** *is the owner of Michael Cooper Recording, located in beautiful Sisters, Oregon. Cooper's studio offers recording, mixing, and mastering services.*

REVIEWS

ANTARES

FILTER 1.0 (MAC/WIN)

An effects plug-in that delivers more than its name implies.

By Len Sasso

What's in a name? In the case of Antares Filter, not nearly enough to thoroughly describe the variety of effects this DSP plug-in can deliver. From the control panel alone it's clear that a lot is going on here (see Fig. 1). However, the panel's complexity belies the ease with which Filter can be learned and used. For sound shaping from the simple to the bizarre, Filter is well worth investigating.

Filter is a four-stage processor; as its name implies, each stage contains a filter. Each stage also has a tempo-synced feedback-delay line you can place before or after the filter. Furthermore, Filter's four stages can be configured in any of six arrangements: parallel, serial, and four combination routings. Add four LFOs, four envelope generators, two 48-step Rhythm Generators, a full modulation matrix, and an envelope follower, and you have plenty to play with.

FIG. 1: Filter's dense control panel includes sections for four filter and delay stages (top), a modulation matrix (right), a graphic response-curve display (center), two Rhythm Generators, and four Function Generators (bottom).

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The plug-and-play DAW controller with 2.5 motorized faders per finger.

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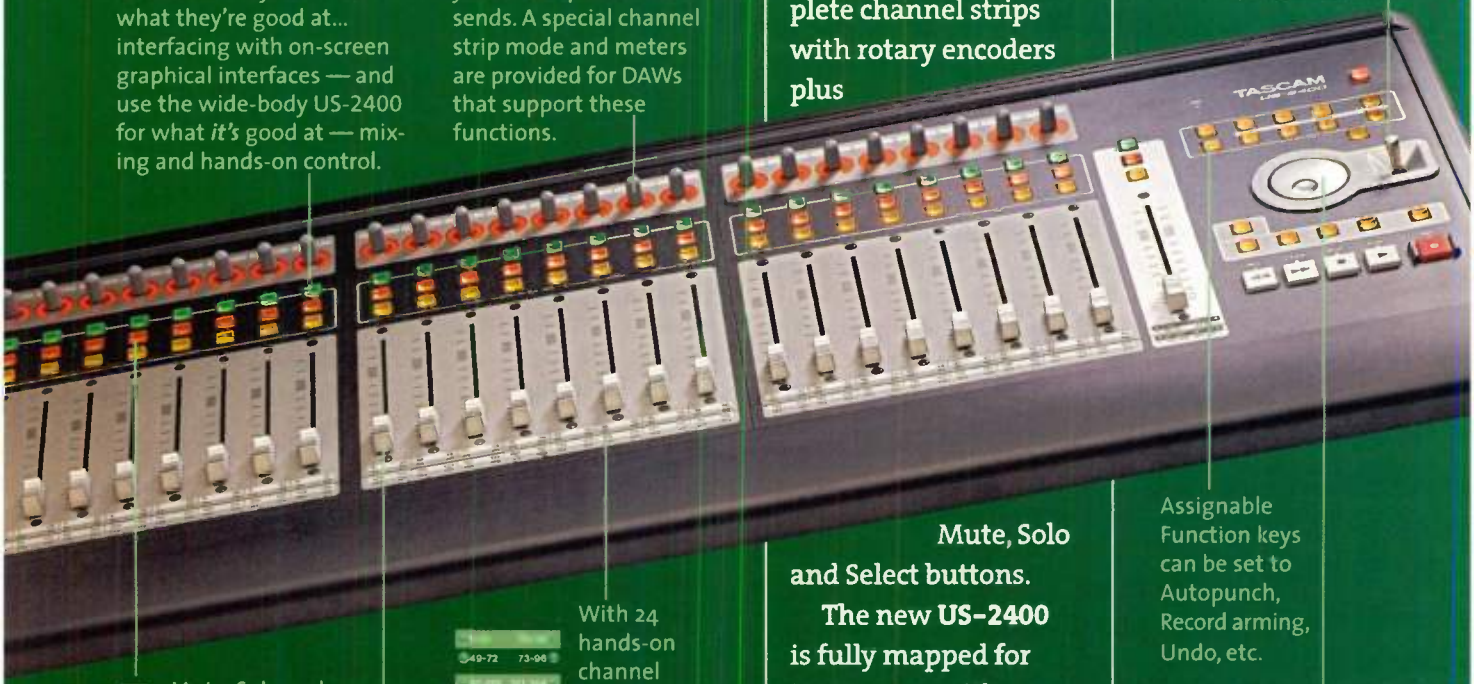

Mute, Solo and Select buttons on every channel beat the heck out of trying to mouse and click on-screen icons.

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Filter is available in VST and RTAS formats for Mac and Windows, in MAS format for the Mac, and in DirectX format for the PC. The plug-in supports both Mac OS 9 and OS X, and an Audio Units version is planned for future release. The package includes a CD containing all versions as well as a printed manual (which is also provided in PDF format). Filter uses challenge-and-response authorization, which can be carried out instantly online or within several days by e-mail or fax. Filter will run unauthorized for a ten-day grace period.

Although Filter will run on any system capable of running a compatible host program, four stages of filtering and delay can eat up a significant amount of processing power. To run more than one instance, you'll want a fast computer. When I ran four instances of the VST version in several hosts on a dual-processor Mac G5/2 GHz, the hosts' CPU meters registered about 35 percent.

FIG. 2: Filter's 12-row modulation matrix lets you route 39 modulation sources, including a variety of MIDI messages, to most of Filter's settings.

THE SIMPLE

Each filter can take on any of five shapes: lowpass, highpass, bandpass, notch (often called band reject), and flat. Flat simply lets you use the stage's delay line without filtering. Each filter's slope can be set to 12, 24, 36, or 48 dB-per-octave (also referred to as 2-, 4-, 6-, and 8-pole). With four multimode filters, a flexible signal path, and a little thought, you can create just about any filtering configuration you want. The thought involves analyzing how the filters will interact.

When you place two filters in series, only the frequencies that pass through both filters will be heard. When you place two filters in parallel, all the frequencies that pass through either filter will be heard. For example, if you place highpass and lowpass filters in series and set the cutoff frequency of the highpass filter below that of the lowpass filter, the frequencies between the two cutoffs will pass through. That gives you a bandpass filter, with the advantage that you can change the width of the band. If you place the same filters in parallel with the highpass cutoff above the lowpass cutoff, the frequencies between the cutoffs will be filtered out. That gives you a notch filter, and again you control the notch width. Place the parallel (notch) filters after the series (bandpass) filters, which is one of Filter's routing options, apply a little modulation, and you'll have a notch sweeping through an expanding and contracting band. **Web Clip 1** illustrates that effect.

Filter makes arranging the filters as easy as possible by providing an interactive display showing the active filters' response curves in bright, semitransparent colors. You can bring any filter to the top by clicking its corresponding button beside the display or by clicking any uncovered portion of its curve within the display. You can then change its frequency and resonance by dragging vertical and horizontal lines that appear in the display.

As I mentioned, by setting the filter types to Flat, you can use the delay lines by themselves. And just as filter routing affects the outcome, so does the routing of the delays. With the delays arranged in parallel, Filter becomes a standard four-tap feedback-delay line. If you arrange the delays in series and increase the feedback, Filter will produce cascading echoes as later delays in the series generate echoes of the echoes generated by earlier delays.

When the delay times are very short (below roughly 30 ms) and the feedback is set high, the delay lines act like resonators. In that configuration, the delay time controls the resonator pitch and the feedback controls the decay time. Resonators are especially effective with drums, turning ordinary drum sounds into tuned percussion.

Minimum System Requirements

Antares Filter 1.0

MAC: Mac OS 9 or OS X 10.2; compatible host

PC: Windows 98 or XP; compatible host

THE BIZARRE

Of course, multibank filters and multitap delays are not hard to come by. Along with Filter's modulation options, however, the interaction of the filters and delays provide the most unusual results. **Web Clip 2** is a percussion loop created from a single kick-drum hit processed using two Filter presets that combine delay and filtering with Filter's Function and Rhythm Generators.

Filter's four Function Generators each combine an LFO and an envelope generator. Although a Function Generator can display only one or the other at a time, they can be used simultaneously. The LFOs offer ten waveforms: sine, triangle, ramp up, ramp down, square, short pulse, long pulse, and three varieties of random—hold, slew, and ramp. Random ramp is an unusual and very useful construct that produces a ramp down from a different starting level on each cycle. It would be nice to have full control of pulse width, but the three choices provided are adequate for most tasks.

You can sync the LFOs to the host's tempo, in which case their rates are set in note values ranging from a 32nd note to 16 quarter notes. Triplets and dotted

notes are also provided. When an LFO is not synced to tempo, its rate can be set in either Hertz or beats per minute. When using bpm, the LFO cycle length is a quarter note at the chosen tempo.

Filter's envelope generators are quite flexible. They offer delay, attack, decay, sustain, and release stages and can be triggered by either of the two Rhythm Generators or by MIDI. When controlled by MIDI, the envelopes can be either triggered, with a fixed sustain time set by the Hold control, or gated. You can set all envelope time parameters either directly in milliseconds or as a percentage relative to the Rhythm Generator's step size.

The Rhythm Generators are 48-step gate sequencers. Both run on the same clock and must have the same step size. Separate clocks and step sizes would be better, but at least you can set the number of steps for each independently. Like the LFOs, the clock can be free running or synced to tempo.

You can use the Rhythm Generators either to trigger the envelopes or as modulation sources. In the latter case, any step that is turned on sends the maximum value and any that is turned off sends the minimum. The two-value limit is mitigated somewhat by the modulation amount controls, which themselves can be modulation targets. That capability allows you to create pseudo-step-sequences, in which the step values are set by, for example, an LFO controlling modulation amount.

Filter also has an Envelope Follower that generates a control value based on the level of the incoming audio. Its Attack and Decay controls allow you to set how rapidly it reacts to changes in level. The Envelope Follower has many uses, the most familiar of which is the auto-wah effect produced when it is used to modulate a bandpass filter's frequency.

Filter's modulation routing is controlled by a 12-row modulation matrix (see **Fig. 2**). Its 39 mod sources include the LFOs, envelopes, Rhythm Generators, and Envelope Follower, as well as MIDI Note, Velocity, Mod Wheel, Aftertouch, Volume, and Pan. Each MIDI source can be routed separately from four different MIDI channels. Destinations include all filter, delay, LFO, and envelope parameters as well as the modulation amount for the top four modulation routings.

FILTER SATISFIES

Filter ships with more than 100 factory presets divided into five categories roughly indicating their intended use: Beats, Effects, Instruments, Pad, and Spectral. You can get a good idea how Filter works just by exploring the presets. The manual is excellent and even the presets are fully documented.

Filter is an excellent piece of DSP software. Its filters are capable of everything from subtle coloring to modulated, resonant, slicing up of the frequency spectrum. You can use the delays as resonators, for reverblike echo effects, and as multitaps. The modulators are plentiful and offer some clever twists. The price might appear a bit high for a filter-and-delay plug-in, but it's justified by Filter's versatility and high sound quality.

EM WEB CLIPS

Len Sasso *is an associate editor of* **EM**. *He can be contacted through his Web site at www.swiftkick.com.*

PRODUCT SUMMARY

Antares
Filter 1.0 (Mac/Win)
filter plug-in
$199

FEATURES	4.5
EASE OF USE	4.0
DOCUMENTATION	5.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Sounds great. CPU efficient. Very flexible modulation matrix. Six signal-path configurations. Excellent documentation.

CONS: Crowded control panel is sometimes hard to read. Rhythm Generator functionality is limited.

Manufacturer
Antares Audio Technologies
tel. (888) 332-2636 or (831) 461-7800
e-mail info@antarestech.com
Web www.antarestech.com

RØDE NT2000

A modern mic with a vintage sound.

By Rob Shrock

I have had the opportunity review a number of Røde mics over the last few years, and I've been impressed enough to purchase several of them for use on my own recording projects. So when I read that Røde had spent $1 million to design the mic capsule for its new NT2000, my interest was piqued and I wanted to hear it for myself.

My chance came when **EM** asked me to review the NT2000. I received a pair of the mics and proceeded to test them under real-world conditions. But before going into details about how they performed, let's look at some of the NT2000's features.

THE LOOK

The NT2000 has a unique appearance, highlighted by silver control knobs on a black background on the front of the microphone's chassis (see **Fig. 1**). The three knobs control the polar pattern, pad, and highpass filter. Although similar controls have been featured on mic bodies before, I've never seen any as visible as those on the NT2000.

But the knob design of the NT2000 is not simply for aesthetics; these controls are continuously variable. The input can be trimmed anywhere from 0 to –10 dB, and the highpass filter can be set to engage anywhere between 20 Hz and 150 Hz. Best of all, the polar pattern can be varied continuously between omni, cardioid, and figure-8, allowing you to dial in, for instance, a setting somewhere in between pure omni and cardioid. I'll talk more about this powerful feature later. The knobs themselves have a notch cut out of them so that you can see a graded set of dots along the throw of the pot for easier replication of specific settings (see **Fig. 2**).

WORK IT

I first tried out the NT2000s while recording acoustic guitar for several artist demos I was planning to shop in Nashville. Record executives in Nashville know what a good acoustic guitar sounds like, so I knew that the tracks recorded with the NT2000 had to sound great or I wouldn't be able to use them.

I started by setting the mics up in a middle-side (M-S) configuration, which I like for stereo acoustic guitar. (For those unfamiliar with this technique, I'll explain briefly. M-S uses one mic in a cardioid pattern pointed directly at the sound source and a second mic in a figure-8 pattern with one of the null sides pointed at the source so the mic picks up the room. On playback, the figure-8 channel is duplicated to a third channel on which the phase is reversed; the two figure-8 channels are then panned left and right and mixed in behind the cardioid signal.) I also set up a ribbon microphone so that I would have another sonic choice while recording.

The sound of the two mic setups was quite different. Although the ribbon mic sounded good and quite detailed, it needed a lot more preamp gain and ultimately had to be moved closer to the guitar. In addition, I had to add a healthy amount of top end, something ribbon mics frequently require. The M-S setup with the NT2000s sounded much more "finished" right from the start. However, there was some obvious low-end rumble that needed to be filtered out, so I decided to try the rolloff on the NT2000.

That was when I noticed that it is impossible to know exactly where you're setting the NT2000's highpass filter, because only the extreme settings of 20 Hz and 150 Hz are labeled around the knob. There are 11 dots that help you find your way somewhat, but it's still a guess. I dialed the pot to the fourth dot from the 20 Hz side, hoping that it would be rolling off somewhere around 80 Hz. I can't be sure precisely how close I ended up getting, but the guitar sounded much better. I did the same for the second microphone and started recording tracks.

I recorded three songs in this configuration and was pleased with the results. I ended up using the NT2000 M-S configuration for the two up-tempo songs and for variety, I used the ribbon-mic track on the ballad. Not that the NT2000 tracks didn't sound great—they

FIG. 1: The large-diaphragm NT2000 condenser microphone incorporates Røde's newly designed HF-1 mic capsule.

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Jennifer & Scott Smith -- TAXI Members

For the longest time, my wife Jennifer and I were skeptical about TAXI. What songwriter, artist or band in their right mind would pay to have their material pitched?

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did. I simply wanted a different sound for the ballad, and the ribbon mic had a significantly different character that worked well for the slow song.

VOCALIZE IT

When it came time for cutting vocals, I set up the NT2000 alongside Røde's NT1-A, which I had previously used with great success when miking the female vocalist who was on this project. I also had one of Røde's K2 microphones on loan; the K2 uses the same HF-1 capsule as the NT2000, but in a tube-microphone design. To provide an additional point of comparison, I also set up a Neumann U 87.

The singer had become attached to the sound of the NT1-A, which has a nice lift in the 5 kHz range and a tame, neutral midrange that immediately sounds good in headphones. The NT2000's midrange is much more aggressive from about 400 Hz to 1 kHz; the U 87's midrange is similar. The NT2000 had more air in the 10 to 12 kHz range than the U 87, which on this particular singer sounded somewhat honky and not quite natural.

We quickly eliminated the U 87 and brought the Røde K2 into the session.

NT2000 Specifications

Capsule	1-inch gold-sputtered, custom design
Polar Patterns	omni, cardioid, figure-8; continuously variable
Frequency Response	20 Hz–20 kHz
Output Impedance	200Ω
Sensitivity	–36 dB, ±2 dB
Equivalent Noise	7 dBA
Maximum Output	+15 dBu (at 1% THD)
Dynamic Range	136 dB
Maximum SPL	147 dB (at 1% THD); 157 dB with pad at max.
Signal-to-Noise Ratio	84 dB
Power	48V phantom power
Pad	0 to –10 dB, continuously variable
Highpass Filter	20–150 Hz, continuously variable
Active Electronics	J-FET transformerless
Dimensions	9.15" (L) × 2.36" (diameter)
Weight	1.83 lb.

The K2 sounded quite similar to the NT2000, and it took a lot of listening back and forth between the two to discern the differences. The sonic signatures of each eventually emerged in the way that you would probably imagine: the K2 exhibited that slight sizzle on the top end and the larger-than-life character with which tube microphones often imbue a source. Granted, the differences were subtle, but they were noticeable nonetheless. Compared to the NT1-A, both the NT2000 and the K2 sounded like they would later need a slight boost at 5 kHz and a slight dip in the 600 to 700 Hz range—exactly what I usually do when working with vintage mics such as the Neumann U 67 or M 49 on vocals.

In fact, both the NT2000 and the K2 reminded me of mature, vintage microphones. The bottom end was much fuller on both the NT2000 and the K2 than the NT1-A, and overall, the sound was bigger and meatier. I do love the NT1-A; however, it is clear that the HF-1 capsule in the NT2000 and K2 sounds very much like an older, classic-mic capsule.

The NT2000 is also whisper quiet, with an extremely low self-noise rating of 7 dB. The only microphone I know that's any quieter is the NT1-A, which has a 5 dB self-noise spec.

TESTING ONE, TWO

Because we were only at the point of cutting rough vocal tracks while the artist got comfortable with the songs, I could conveniently try out the variable pattern of the NT2000. We tried a few passes at each of the three settings (cardioid, omni, and figure-8) and at different distances, with the singer "slating" the specific pattern and distance for each take.

With the NT2000 set to omni and the singer extremely close and singing through a pop filter, the sound was absolutely gorgeous. Omni mode doesn't

FIG. 2: Continuously variable controls for pad, pattern, and highpass filter give the NT2000 great versatility.

exhibit the proximity effect in the way that a cardioid pattern does, and the frequency response is typically flatter and more natural sounding. On playback through the big speakers, it sounded exactly like the singer was standing right there in the room. We both loved it.

We did get a few microphone overloads when she sang loudly while standing right up on the mic. Dialing the NT2000's pad to a –3 dB setting and boosting the preamp to make up the gain solved the problem, though.

PICK A PATTERN, ANY PATTERN

In the personal-studio environment, where the acoustics are often less than perfect, I've often wished for a polar pattern that gave me the purity of omni and the better isolation of cardioid. With its continuously variable pattern, the NT2000 makes such in-between settings as easy as turning a knob.

On the day we were recording test vocals, we found a sound we really liked by starting with the omni setting and slowly adjusting the pattern toward cardioid. We stopped when we'd removed just enough room sound to clean up the vocal sound while still retaining a lot of the naturalness of the omni setting. The setting we landed on in that particular environment was almost to the halfway point between the third and fourth dots from the omni side. The room sound was a little tamed down, and the singer was still able to get close up on the microphone without a ton of boomy proximity effect. Sweet.

Later, I tried some settings in between the cardioid and figure-8 with my own voice. These settings were a little less dramatic, but it was nice to add some sense of space to a primarily cardioid sound by introducing some of the microphone's backside. One problem with the knobs is that they're a little tight overall, and they aren't very easy to adjust when the NT2000 is seated in the shockmount, especially if you have big fingers.

During my rattle-the-car-keys test to check the high-frequency response, there was a noticeable drop in highs at the two side points of the NT2000 in omni mode. Although omni mode picks up sound in all directions, the sides don't sound exactly the same as the front and back of the mic (typical of all mics in omni mode). This could potentially present problems if you were, say, recording a group of background singers positioned around a single NT2000. But as long as you positioned the singers so that nobody was singing directly into the side of the mic, you'd be fine. And the off-axis rejection of the cardioid and figure-8 patterns is very good.

THE RØDE TAKEN

Apart from the two minor quibbles I mentioned previously—the shortage

PRODUCT SUMMARY

Røde
NT2000
condenser microphone
$899

AUDIO QUALITY	4.5
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Tone quality reminiscent of classic mics. Superquiet operation. Polar patterns continuously variable between omni, cardioid, and figure-8. Good off-axis rejection. Polar pattern, pad, and highpass-filter controls are on the mic chassis. Includes quality shockmount and carrying case. Affordable.

CONS: Knobs are a bit difficult to manipulate. Insufficient indicators for the high-pass filter control.

Manufacturer
Røde Microphones
tel. (310) 328-7456
Web www.rodemic.com

of indicators for the rolloff frequencies and the slight difficulty of adjusting the knobs when the microphone is in the shockmount—I have nothing but very positive things to say about the NT2000. I like the microphone so much that I intend to buy the two review models I was sent and continue to enjoy their vintage tones, flexibility, and quiet operation.

And here's the kicker: the NT2000 retails for $899, and it will probably have a street price that's significantly less. Considering that the NT2000 sounds like a composite of some of the best vintage microphones, only quieter and more flexible, it's quite a good value. (If you want the tube version, an extra $100 gets you the K2, but without a pad or highpass filter.) It's a wonderful world we're living in when great new microphones have become so affordable.

Producer-composer **Rob Shrock** *has worked with a host of world-class artists, including Burt Bacharach, Elvis Costello, LeAnn Rimes, Aretha Franklin, and Ronald Isley.*

EMAGIC

LOGIC PRO 6 (MAC)

More powerful, easier to use, and a megabundle.

By Brian Smithers

What happens when you take one of the most complex music-production applications to be found anywhere and add a bunch of new features? Would you believe it becomes easier to use? With the addition of new setup and project-management features and a couple of new tools, Emagic's Logic 6.4 manages to do exactly that. To top it off, Emagic has bundled TDM integration and *all* of its virtual instruments—which formerly sold separately for well over $1,500—for a measly $50 more than Logic Platinum 6.3 alone.

Gone are Logic's previous flavors of Platinum, Gold, and Audio or Silver, leaving only Logic Pro ($999) and Logic Express ($299). Logic Pro combines the application formerly known as Logic Platinum with ES1 and ES2 virtual analog synthesizers, EVP88 virtual electric piano, EVB3 virtual organ, EVD6 virtual clavinet, EXS24 mkII sampler, EVOC20 vocoder, Pro Tools|HD Extension, ESB TDM system bridge, and the new Space Designer convolution reverb. (You can read more about several of these plug-ins in the November 2000, May 2001, and June 2003 [see "Is It Real or Is It Emulated?"] issues of **EM**, online at www.emusician.com.) Because Logic's attributes have been covered in these pages before (most recently, version 5.5 in the March 2003 issue), I'll focus on what's new with version 6. Check out the sidebar "Space Designer" for details on that exciting new plug-in.

I tested Logic Platinum on a dual-processor Mac G5/2 GHz with 2 GB of RAM (see the sidebar "G5? Gee Whiz!"), which Apple graciously provided in order to demonstrate Logic's optimizations for the latest generation of Macs. The combination speaks well for host-based systems, to say the least.

ENVIRONMENTAL CHANGE

The first thing anyone learns about Logic is that its Environment is a powerful and daunting tool for configuring your studio. Long-time Logic users scoff at the idea that it's difficult to learn, but Logic newbies scoff at the scoffing. At best, it's still a big part of the program's overall learning curve.

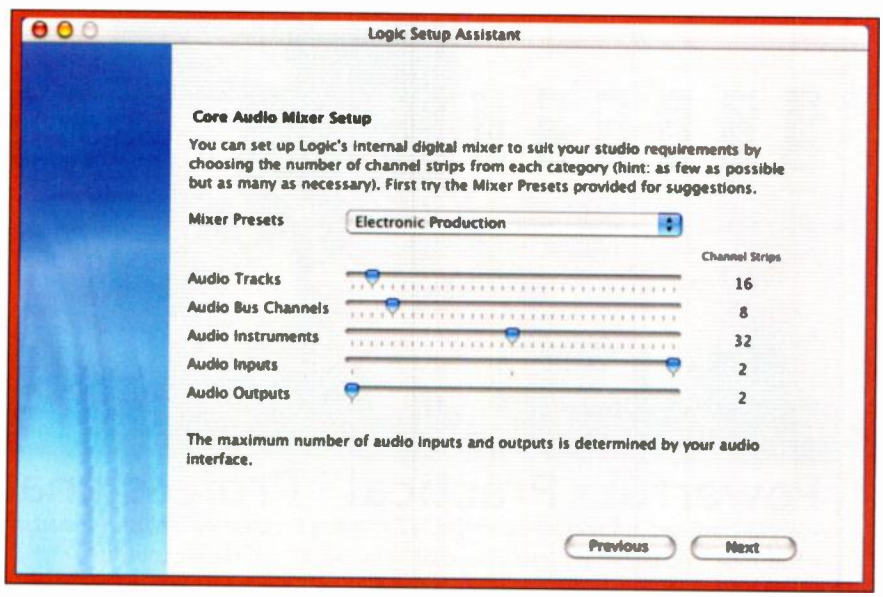

FIG. 1: The Logic Setup Assistant makes child's play of configuring a basic Environment. Its wizard-style interface quizzes you on the pertinent details of your setup and preferences.

Minimum System Requirements

Logic Pro 6.4

PPC/300 MHz; 256 MB RAM; 350 MB free disk space; Mac OS 9.1 or Mac OS X 10.2; free USB port for XSKey

Dumbing down the Environment would risk limiting its power, so the good folks at Emagic instead came up with the Logic Setup Assistant to lend a helping hand (see **Fig. 1**). The Setup Assistant is essentially a "wizard" that asks a series of questions to determine how to configure new sessions. When you launch the Assistant from the Logic Preferences menu, it promptly closes Logic, queries you about your needs, and then launches Logic again with a shiny new Environment customized according to your feedback. Unfortunately for some users, Setup Assistant doesn't work under Mac OS 9.

Having to launch Logic first seemed a bit cumbersome to me, so I decided to put the Assistant right on the Dock. Because it's a hidden file in a hidden folder, though, I had some trouble locating it; consequently, the easy solution was to fire it up from within Logic. While it was running, I Control-clicked on its Dock icon and chose Keep in Dock from the pop-up menu. I can now launch the Assistant directly from the Dock any time I want. Of course, most users will run the Assistant once or twice to set up an Environment and then leave it that way.

The Setup Assistant prompts you first to choose your audio hardware from CoreAudio's list, and then to choose how many audio tracks, buses, audio instruments, audio inputs, and audio outputs you'll need in your Mixer. Three presets are available as a starting point, including Electronic, Acoustic, and Standard configurations that differ primarily in the number of audio and audio-instrument tracks they provide. You can choose key-command sets, configure which displays you want to use in Screensets, and identify your MIDI devices.

Of course, such details are just scratching the surface of what's possible

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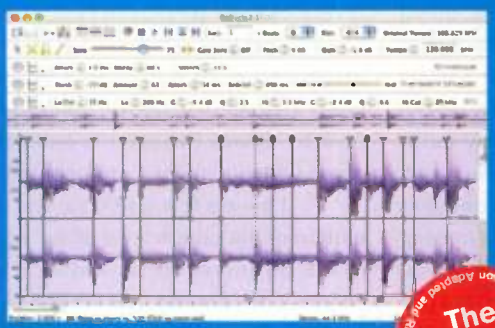

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propellerhead SOFTWARE

SPACE DESIGNER

Emagic's Space Designer is a real-time convolution reverb plug-in. It's one of the latest plug-ins that depends on a process that until recently was too complex to compute in real time. Convolution enables you to capture the reverberant characteristics of a particular space, such as a studio or concert hall, and apply them to an audio source. It accomplishes that by multiplying an impulse response (IR, the captured ambience) by the source audio, which requires some heavy number-crunching.

Because I tested Space Designer on a machine that's currently the fastest Macintosh with 2 GB of RAM, processor load was not an issue. A single instance barely moved the CPU meter, and four instances pushed the meter to about 20 percent of each processor. Using exceptionally long IRs or higher sampling rates will consume CPU cycles, however. Emagic recommends a G4 or G5 with 512 MB of RAM, but the more horsepower, the better.

Acting on impulse. Space Designer comes with over 1,000 IR files and many presets based on them, covering everything from stairwells to bathrooms to warehouses (see **Fig. A**). Because convolution can also model the behavior of effects processors (or any other audio device), many presets are devoted to emulating the sound of big-name reverbs; some of those, including some wonderful delays, are excellent.

The Outdoor category reproduces the ambience of forests and fields. When I applied those presets to classical concerts recorded in dry spaces, they sounded less claustrophobic. I was pleased that the outdoor sounds are useful, because the one glaring omission in the included IRs and presets is a complete absence of real concert halls.

You can record your own impulse responses and deconvolve them in Space Designer (under OS X 10.3). Because IRs can be AIFF, WAV, or SDII files, you can trade them with colleagues (even if they use other convolution reverb software). Space Designer also synthesizes IRs.

The plug-in includes a resonant filter that can operate in highpass, bandpass, or 6 or 12 dB lowpass modes. You can shape filter frequency with an envelope. The cutoff, which in the initial release had been expressed only as a percentage, is now properly expressed as a frequency.

FIG. A: Emagic's Space Designer reverb plug-in combines real-time convolution with filter and envelope controls. Its sound ranges from highly realistic to bizarre and unusual.

Most Logic plug-ins have lots of parameters you can automate, but Space Designer has only three: direct (dry) output, reverb (wet) output, and Stereo Crossfeed, which can narrow or even reverse the source's stereo image. The reason for so few automated parameters is that after each parameter change, Space Designer has to recalculate the convolution, making you wait a few seconds before you hear the effect of your tweak. Once in a while I even heard a pop or click at the end of the recalculation.

Keeping it real—or not. The sound of Space Designer at its best is quite good indeed. With the right IR, it creates lively, interesting delays; warm, lush, natural-sounding reverbs; and stimulating special effects. I imported an IR of an old church (downloaded from a competing plug-in's user site) and applied it to a recording that was too dry. With a quick adjustment to the wet/dry mix, I was able to place the performers in the hall they should have played in.

Space Designer's various controls, though much appreciated, rarely helped make a sound more natural. Rolling off some highs with the lowpass filter sometimes improved the sound, but everything else usually made the sound more artificial. That's not necessarily bad—for some sounds, shortening the envelope to gate the reverb or adding resonance to make a sound ring was very useful and appropriate. Using synthesized IRs generally yielded rich, dense, studio-style reverbs. Space Designer is equally at home with purists or experimentalists.

I have a few quibbles. The resolution of the reverb level slider works in 0.5 dB increments, which is too coarse. I got around that limitation by putting Space Designer on an Aux track, whose fader has slightly finer resolution; that is a more traditional use of reverb anyway. You can't change the delay time, because it's a function of the IR itself; that's not surprising, nor is it a design flaw, but it's a limitation unless you either create a set of IRs with different delay times or time-stretch an existing IR. I was also disappointed that Space Designer doesn't directly support surround sound.

All told, though, Space Designer is a welcome addition to Logic's effects palette. It's a different sort of reverb and requires some minor modifications to your expectations and approach, but it's worth the effort.

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FIG. 2: The new Save as Project option helps to ensure that all of your audio files, samples, and instruments are preserved as a group.

in the Environment, but that's the point. The Setup Assistant brings together the essential ingredients to get you started—no more, no less. To top it off, it saves your configuration as the Autoload template so you will start from the same point with every subsequent new session.

The new Project Manager is essentially a smart database applet that recognizes dependencies between song files, audio files, video files, samples, and more. It lets you see exactly what assets belong to a given Project across multiple volumes and even lets you add one or two comments per file.

Building on that idea, Logic now offers a simple means to collect and manage the files associated with a song. The Save as Project function assembles song files, audio files, samplers, and other related assets into a Project folder (see **Fig. 2**). It lets you specify whether files should be copied or moved to the Project folder or left in their original locations. There's even an option to include any songs that share files with the current song, letting you back up multiple versions of a song in one step.

MAKING ARRANGEMENTS

Various improvements have been made in the Arrange window that not only facilitate editing but also alleviate some of the window-hopping that was formerly necessary. The most obvious addition is the Arrange Channel Strip, a tiny slice of the Mixer window that now lives along the left edge of the Arrange window. It is literally a copy of the current track's channel strip that allows you to assign effects, change I/O, adjust pan and volume, and generally do anything you can do in the Mixer without having to toggle windows. In fact, you might find that the only time you'll need to open the Mixer is when you're actually mixing.

You can now zoom in to the sample level in the Arrange window, though sample-accurate editing still requires opening the Sample Editor. You can do time-stretching directly in the Arrange window, and Time Machine now has multiple algorithms optimized for different types of source material.

Smart Snap is a welcome enhancement to Logic's drag-and-drop behavior. The edit grid varies according to zoom level, with the assumption that the farther you're zoomed in, the smaller you want the grid. You can override the grid by pressing the Control key. With Control pressed, the grid is as fine as the current zoom will allow—essentially whatever time increment is represented by a single pixel.

Another new feature called Smart Loop Handling lets you split loops without messing up subsequent loops. For example, if you decide you want to snip a beat from the first iteration of a loop, Logic will first duplicate the looped clip and use that as the basis for the remainder of the loops.

The new Marquee Tool, represented by an unassuming little crosshair, allows you to select data in the Arrange window without regard for object boundaries. You simply lasso the data you want to copy, delete, mute, drag, or otherwise edit across multiple

G5? GEE WHIZ!

I always wince at Apple Computer's ads comparing the performance of its newest computers to that of PCs; they always sound shrill and almost desperate to me. So I approached the loan of a dual-processor Mac G5/2 GHz for this review with a tiny bit of skepticism. How could any machine live up to the hype that Apple has given the G5? After spending some time using one, though, all I can say is that there weren't enough hours in a day for me to drag this machine to its knees. No matter how many audio tracks, plug-ins, and virtual instruments I threw at it, the G5 hummed along as though it barely noticed me. I'm impressed—moreover, I'm more enthusiastic than ever about the future of host-based systems.

Apple briefed me on the technical details of the new machines. The short version is that between the new 64-bit architecture, the increased independence of the dual processors, and the increased memory and I/O bandwidth, there's a lot of oomph in that box. It will still take another OS upgrade to realize the full potential of the 64-bit processing, but significant gains are already in evidence.

The new enclosure is designed to be much quieter than recent G4s, but the review unit turned out to be the loudest computer in my studio. The main culprit was a pronounced whine that Apple suggested might be the fan on the upgraded video card it installed. It appears to be an isolated problem, though; the school where I teach has a classroom containing 25 Mac G5s, and it's pretty quiet in there.

More interesting is the inclusion of optical S/PDIF I/O on the back panel. It's capable of 24-bit audio at 44.1 or 48 kHz. If your software knows how to tell CoreAudio to sync to the incoming bitstream, it will lock up nicely for bit-accurate digital transfers. It also supports DTS and AC-3 streams.

"I Switched"

Who: "Prince Charles" Alexander

Occupation: Music producer, mixer, multi-instrumentalist; based at Quad Studios, New York City

Honors: RIAA multi-platinum award-winner; Governor, Grammy Committee Board of Governors

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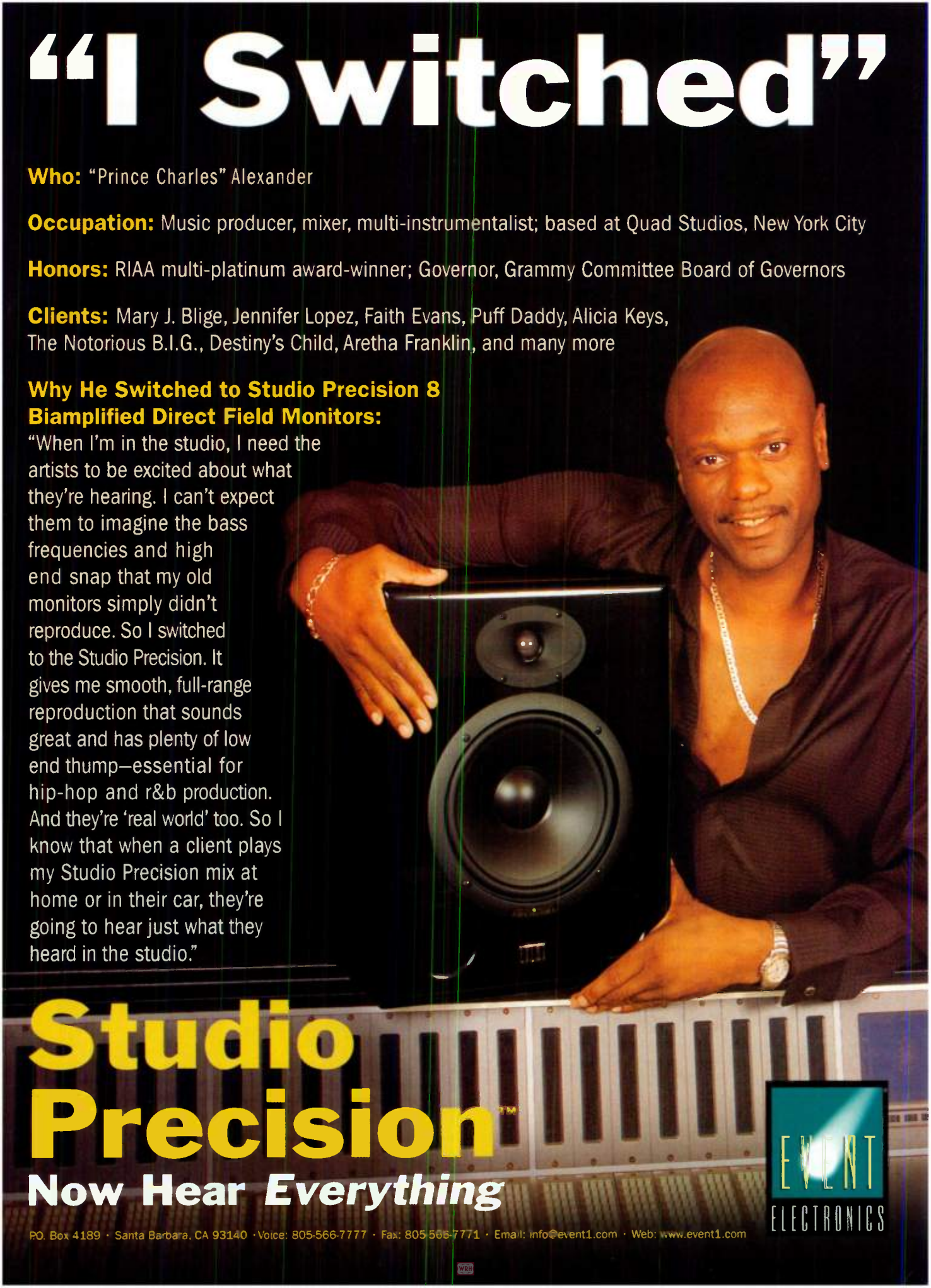

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objects and multiple tracks. By default, the Marquee Tool respects Smart Snap grid increments, but the Control key overrides that behavior, too. Shift-clicking lets you shorten or lengthen a selection, and Control-Shift-clicking allows you to shorten or lengthen with fine resolution.

Another handy new function is the ability to hide tracks in the Arrange window. If you click on the Hide button at the top of the window, each track sprouts a Hide button of its own. Click on a track's Hide button and then click on the main Hide button again, and the track will go away. Click on the main Hide button, and hidden tracks will reappear. Although this is a welcome added feature, I found the clickety-click implementation a bit unwieldy. A quick trip to the Key Commands editor to find the hot key for the track Hide button and to create a hot key for the main Hide button left me feeling much more efficient, and the phrase "There's nothing wrong with Logic that can't be fixed by what's right with Logic" started running through my mind. By the time I set up a couple of Screensets with various subsets of hidden tracks, I was feeling pretty darned good about life.

Any Logic user who works with video will be delighted by the Video Thumbnail Track, which provides a visual reference next to your audio and MIDI tracks. In Mac OS X, you can also play QuickTime movies in DV format through a FireWire device.

FIG. 3: Each audio, audio-instrument, and bus track now features an 8-band Channel EQ. Its intuitive interface and editable graphic display are well thought out and much appreciated.

MAXED-OUT MIX

The new Channel EQ provides eight bands of as much as 24 dB of boost or cut along with a cute overview of the curve displayed on the channel strip, even when the Editor window is closed (see **Fig. 3**). I rarely encounter something that really deserves to be called intuitive, but I found the Channel EQ to be exactly that. Its normal controls are laid out sensibly enough, but when

I clicked on the graphic display to see whether it was editable, something special happened: I discovered that it's not only editable, it's smooth, sensible, and simple to tweak. Just grab the curve and start moving it up, down, left, and right until you hear what you want to hear. The one thing that requires a fraction of a second's thought is the little circle that appears now and then depending on your mouse position. Grab the circle just once and you'll see that it adjusts the band's Q. Enabling the Channel EQ is as simple as double-clicking on the blank graphic display just above the Inserts. Channel EQ is also available as a regular Logic plug-in along with the usual EQs.

Controlling your mix is easier than ever with the host of control surfaces that Logic Pro supports. Everything from the CM Labs MotorMix to the Radikal SAC-2.2 can be put in charge. Others include the Yamaha 02R96 and Mackie Control, and you can mix and match control surfaces.

For those of us who don't have even one control surface, the new One Fader Automation lets you attach any MIDI controller to an automation parameter. When Automation Quick Access is enabled, any slider, wheel, or joystick on your MIDI keyboard is immediately in control of the active track automation parameter. Nothing could be simpler (or more appreciated).

To join tracks for editing or mixing purposes, 32 groups are available. The Group Settings dialog offers an impressive degree of customization for groups, right down to which of eight sends will function as a group. You can assign a track to more than one group by holding down the Shift key, and you can apply the most recent group assignment to a track by holding down the Option key. Click again while pressing the Option key and you will clear all group assignments on that track.

In addition to the newly bundled soft synths, three new virtual instruments are included with Logic: ES M (monophonic), ES P (polyphonic) and ES E (ensemble). All three are virtual analog synthesizers and can produce interesting and useful sounds. As for third-party plug-ins, the transition to Apple's Audio Units plug-in format means that under Mac OS X, Logic doesn't recognize VST plug-ins. Fortunately, FXpansion's VST to AudioUnits Adapter will make many of your favorite VSTs feel right at home in the brave new world of Audio Units.

UNDER THE HOOD

Logic 6.2 and later updates include a number of performance optimizations that enable the current version of the program to run more efficiently than prior versions. While some of the changes were aimed at Apple's new Mac G5 machines, some benefits were in store no matter which Mac you use. The code has been rewritten to take better advantage of dual processors on G5 and G4 machines. According to Emagic, having a second processor now comes closer to doubling the capabilities of having only one.

Of course, any processor eventually succumbs to a voracious appetite for processor-hungry plug-ins, so Logic lets you freeze tracks to reduce CPU load. Freezing entails bouncing a track with effects to disk, thereby trading CPU load for an increased disk-playback burden. Although some musicians have been doing that manually for years, Logic 6 was the first program to offer Freeze as a one-step function. What makes Freeze so cool is how simple it is: you click on the snowflake icon and hit Play, and Logic zips through the song, bouncing all newly frozen tracks. On the Mac G5, freezing multiple tracks is several times faster than recording in real time. You can specify that frozen tracks will be bounced to 32-bit floating-point format (so they will be sonically identical to their unfrozen state) or to 16- or 24-bit format (to keep down the strain on your hard drive). The track's appearance doesn't change, and you can unfreeze a track at any time.

MANUAL DEXTERITY

Logic Pro comes with no fewer than 13 separate manuals, including one for each bundled instrument and three for Logic itself. The Logic Pro Reference Manual is almost 800 pages long. To top it off, one of the three installation discs is devoted entirely to PDF files of those manuals in German, French, and English.

This impressive volume of documentation mitigates some of the quirks of the online Help. For example, when you search Logic Help, you also search Help for every other application that uses the Mac's Help Center. When I queried Help for information about MP3 support, I got much more information about Apple iTunes than about Logic.

LOGICAL CHOICE

Emagic provides frequent updates whenever Logic's feature set or stability can be enhanced. Logic 6.2 offered ReWire 2 support, MP3 import and export (in Mac OS X only), and the ability to bounce faster than real time. Version 6.3 included support for the Space Designer reverb plug-in, and 6.3.1 added compatibility with Digidesign's new HD Accel hardware. Logic 6.3.2 enhanced Panther compatibility, enabled recording on track numbers higher than 128, and made various other improvements. Logic 6.4, the most recent update as of this writing, includes several enhancements to Space Designer. Because this review has focused on the new features, I haven't even mentioned Logic Pro's excellent surround implementation, scoring capabilities, extensive set of included effects plug-ins, and other goodies.

I applaud Emagic for making Logic Pro more user-friendly than previous versions of Logic, but beginners will still find it daunting. I can't think of another digital audio sequencer with as many multilayered menus or one that requires the use of so many

PRODUCT SUMMARY

Emagic
Logic Pro 6 (Mac)
digital audio sequencer
$999

FEATURES	4.5
EASE OF USE	3.5
DOCUMENTATION	4.5
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Comprehensive MIDI sequencing and powerful digital audio production tools in an integrated environment. Extensive bundle of plug-ins, including several outstanding software synthesizers. New ease-of-use features. Track freezing and offline bounce. Extensive control-surface support. New editing tools and functions. Video Thumbnail Track and FireWire DV playback. Highly configurable and customizable. Integrated support for Pro Tools HD systems.

CONS: Many functions are still unnecessarily complex. Not all features are supported under OS 9. No native VST support in OS X.

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Apple Computer
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Web www.emagic.de

different windows to perform basic functions. For example, the Arrange window has 12 different tools—yikes!

The fact that Emagic has decided to essentially give away its entire line of soft synths with Logic makes Logic Pro 6 one of the best values anywhere. If you're a Logic veteran, the newest version provides plenty to keep you excited. If you're considering becoming a Logic user, it offers some good reasons to give it a try now. Either way, Emagic has once again brought a considerable amount of careful thought and intelligent design to the party, and that's good news.

Brian Smithers *is an Apple Pro-Certified Logic Trainer and a Digidesign-Certified Pro Tools Instructor. That's probably why his doctor says he's "certifiable."*

pedals, but not the sostenuto (middle) pedal. I needed a Pedal Sensor with a little more substance, though; my piano rests on a hardwood floor, and I had a little difficulty keeping the pedal sensor in place. I also needed a Pedal Sensor with a little more height. The device needs to be within an inch of the pedals, so I ended up sticking a book under it to position it correctly. The manual mentions an optional Pedal Height Adjustment Accessory that theoretically could have solved both my height and substance issues, but it wasn't yet available when I wrote the review.

You get one 15-foot and one 6-foot cable (both of which are essentially heavy-duty USB cables) for the connections to the Control Module. The long cable is intended for the Pedal Sensor, but I needed to put the Control Module on the right side of my piano, and the Sensor Bar's connector is on the left side. So I ignored the manual's warnings of data errors, and used the long cable for the Sensor bar. If that caused data errors, I never saw them, but I would definitely have preferred an additional connector on the other end of the sensor bar.

UNDER CONTROL

The PianoBar's Control Module has a black finish with a simulated ebony-wood-grain top (see **Fig. 2**). Notwithstanding the obligatory mass of cables that emanate from it, the Control Module complements the look of a nice piano reasonably well. The Control Module was a little too wide to sit on the shelf next to my grand piano's music stand, though, so it sat at an angle with one end on the piano's rim.

The Control Module has connections for the Scanner Bar, Pedal Sensor, and three expression pedals (which are not included). It also has two audio outputs on ¼-inch jacks, a ⅛-inch headphone output, and MIDI In and Out connectors. Except for the headphone jack, all connections are on the back of the Control Module.

The Control Module's front panel has a Volume knob, indicators for MIDI activity, and a 2-by-24-character LCD display with a bright blue backlight. In addition, it has four buttons to navigate through the menus, four buttons to quickly select MIDI channels 1 through 4 (for soloing or volume changes), and a rotary encoder knob to change values in the menus. Also on the front panel is a slot for inserting a Library Card (one blank card is included).

The Control Module's sound generator is 64-voice polyphonic and 16-part multitimbral, and it conforms to the General MIDI 2 specification. Onboard are 300 sounds that can be organized into 100 Setups. Each Setup configures the sound, settings, and effects for each of the 16 MIDI channels. The first 60 Setups are user-rewritable. You can also edit Setups 61 through 80, but those are the 20 available slots in the inserted Library card; if you want to copy a Setup to a card, simply copy it to an available slot between 61 and 80.

▼

The PianoBar passed my Velocity-sensitivity test with flying colors.

The remaining 20 Setups are factory presets and can't be rewritten. They contain duplicates of some of the more useful Setups in the first 60 User slots. The only way to preserve the User Setups to external media is to save them to memory cards, 20 at a time. The PianoBar offers neither SysEx-dump capabilities nor a factory-reset feature, so you'll need three memory cards if you want to back up the original PianoBar setups. Fortunately, Moog has priced PianoBar accessories quite reasonably; a pack of three Library Cards costs $34.95.

A photosensitive LED on the Scanner Bar allows you to manually change Setups. When you tap the LED with your fingertip, another LED lights up above the piano key that corresponds to the current Setup number. (If Setup 1 is active, for example, the LED above the lowest A would illuminate.) Press a different key to change to another Setup. You can't assign MIDI Program Changes to piano keys, and you can't select Setups 89 through 100 using this method—only 1 through 88.

Setup editing is comprehensive and reasonably intuitive. For each MIDI channel, you can specify parameters such as Velocity curves, keyboard zones, MIDI routing, and effects levels. Each Setup has two effects—reverb and chorus—but some reverb presets are actually delays and some chorus presets are flangers or delays.

FIG. 2: The Control Module, a GM2-compatible sound source, stores 100 multitimbral Setups in addition to converting sensor signals to MIDI data.

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EVENTIDE

CLOCKWORKS LEGACY BUNDLE 1.12 (TDM)

A little bundle of joy for fans of vintage effects.

By Nick Peck

In 1971, a small New York–based company called Eventide Clockworks designed and sold hardware phasers and compressors to the biggest rock groups and studios of the day. Flangers and Harmonizer-brand effects processors were soon to follow. More than 30 years later, that company, now called simply Eventide, has recreated its original groundbreaking processors in plug-in form. The Clockworks Legacy Bundle for Mac-based Pro Tools TDM systems brings a 1970s vibe to a desktop near you.

Each of the five plug-ins—Instant Flanger, Instant Phaser, H910, H949, and Omnipressor—has a smart graphical look and feel that is virtually identical to that of its hardware counterpart. The plug-ins are uniformly straightforward and easy to use. You can be on your feet quickly without reading the documentation, although the enclosed PDF manual is well written and helps clarify several details. Eventide also includes scans of the original hardware manuals.

Each plug-in comes with a collection of factory presets that make good starting points for experimentation. The programs are complete and bug-free—I didn't suffer a single crash or hang-up during my tests. All of the plug-in parameters can be automated, and they all feel fluid, particularly with a MIDI or Ethernet-based control surface. In fact, all of the plug-ins except Omnipressor have at least one parameter that responds to MIDI messages.

INSTANT FLANGER

The Instant Flanger plug-in (see **Fig. 1**) has a liquid, wobbly character that I haven't heard in other digital flangers. It imparts a distinctive sonic character reminiscent of '70s-style David Bowie and Genesis. This is not your typical "jet whoosh" type of flanger. Used sparingly, it could work well for vocal doubling. I liked it on acoustic guitar and sax, and it could also be a good choice for adding motion and vibrato to sounds that have fairly static pitches.

For modifying effects, Instant Flanger has Feedback, Bounce, and Depth knobs. Feedback mixes a portion of the output back into the input. Bounce simulates the effect of a tape motor changing speed. Depth adjusts the wet/dry mix. There are also four input controls: Oscillator Rate, Manual (for adjusting delay times), Remote Control (for accepting MIDI continuous-controller data), and Envelope Follower. The Envelope Follower has Threshold and Release knobs, letting you trigger flanging for loud passages while keeping quieter sounds unaffected. By simultaneously selecting multiple input controls, you can create complex, unexpected effects.

Minimum System Requirements

Clockworks Legacy Bundle 1.12

Pro Tools TDM for Macintosh, version 5.1 or higher (Mac OS 9) or 6.1 or higher (Mac OS X)

INSTANT PHASER

Instant Phaser's layout and rubbery, organic quality resemble Instant Flanger's. It's capable of extreme effects, but I preferred to use it sparingly, because overdoing it made instruments sound out of tune. I particularly liked using it to synthesize a stereo image from a mono source. The stereo image expanded and contracted in interesting ways with small adjustments of the settings.

Instant Phaser's controls are a subset of the controls on Instant Flanger. On the front panel are a Depth knob and Remote, Oscillator, Manual, and Envelope Follower input controls. Unlike Instant Flanger, however, Instant Phaser only lets you use a single input control at a time.

H910

A bow to Eventide's first Harmonizer effects processor, the delay-based H910 plug-in (see **Fig. 2**) is grungy and perfect for turning nice, mannered sounds ugly and strange. It would serve well for techno and industrial styles of music or for creating sound effects with a distinct '70s character.

The H910's controls are few and easy to use, allowing anyone to master this plug-in quickly. The knobs control input level, feedback level, manual pitch-shift amount (as much as one octave up or down), and antifeedback level. This last knob adds a small amount of changing

FIG. 1: The Instant Flanger (top) and Instant Phaser (bottom) plug-ins have a rubbery, organic quality that would sound right at home on a '70s David Bowie record.

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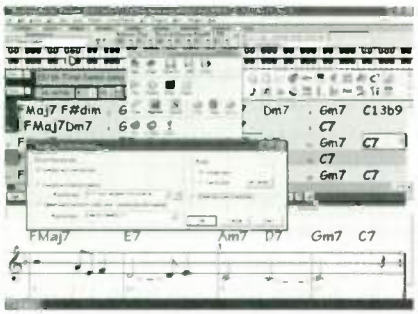

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Features listed are for Windows® version.
Macintosh® users: for a full list of features, please visit **www.pgmusic.com/band.htm**

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frequency shift to the output signal; it was originally designed to decrease the effect of room resonances in a live setting. Selection buttons along the bottom specify how the pitch shift is controlled: manually, with the anti-feedback setting, or using MIDI. Harmonizer H910 can also serve as a digital delay, allowing slapback echoes as long as 112 ms to be added to a signal.

This plug-in was not my favorite of the bunch, but I can see it being useful for creating specific strange pitch-based effects. For example, I like to crank up the feedback in tandem with downward pitch-shifting to create instant *Battlestar Galactica* laser-beam sound effects.

H949

H949 is a much improved version of the H910 Harmonizer effects processor. It includes many more controls and thus offers a wider range of sonic possibilities. Additional features include separate feedback levels for main and delayed signals, low- and high-shelving EQs, and a Repeat mode that freezes and recirculates the incoming audio at the time the button is pressed. Delay times can be a whopping 400 ms, and the pitch-shift range is three octaves (up one or down two).

In addition to standard pitch shifting, H949 has micropitch shift (for subtle doubling effects), standard delay, random delay, flanging, and reverse delay. Each of these quite distinct functions can be used to create a wide variety of interesting effects. H949 sounds better and offers far more control than H910, though H910's approachable simplicity does have a certain charm.

H949 is certainly not as clean or as convincing as modern pitch shifters like those in Eventide's current hardware line. But that's not its purpose. Like the rest of these plug-ins, H949 is meant to create a certain old-school vibe, and it does that well.

Unfortunately, that vibe comes at some cost. Both Harmonizer pitch shifters consume a lot of DSP power, and they're limited to a 96 kHz sampling rate. The rest of the plug-ins can work at sampling rates up to 192 kHz.

FIG. 2: Eventide's H910 (top) and H949 (bottom) plug-ins are good for creating wild vocal effects.

FIG. 3: The Omnipressor compressor's Function knob allows for a wide variety of compression types, including dynamics reversal.

OMNIPRESSOR

The crowning achievement of the Clockworks Bundle is the Omnipressor dynamics processor (see **Fig. 3**). This plug-in includes all the standard features of any compressor-limiter-expander-gate combo.

But what sets Omnipressor apart from the crowd is its compression-ratio Function knob. Turning this knob counterclockwise adjusts the ratio from 1:1 to 1:10, allowing the unit to function as an expander or gate. Turning the knob clockwise adjusts the ratio from 1:1 to ∞:1, causing it to function as a compressor or limiter. Pushing the knob past infinity (Can you do that? Where's Carl Sagan when you need him?) takes you into *negative* compression, where quiet sounds become loud and loud sounds become quiet. That can lead to all sorts of interesting sonic explorations. On a drum kit, for example, this setting creates a fabulous backward sucking cymbal effect—right in the psychedelic pocket.

Omnipressor sounds great with more pedestrian settings, too. It has an aggressive character perfect for rock 'n' roll or any recording that wants a bit of attitude. You have to watch your levels carefully, though. Omnipressor can easily add 30 dB of gain to a signal before you realize it. That can create some ferocious digital overs. You therefore must be mindful of your monitor settings, especially when switching factory presets.

Omnipressor handles input and output gain using a series of 10 dB calibration buttons. Various combinations let you lower your input level by 10, 20, or 30 dB, and to raise your output level by the same amount. While this system is true to the original, it's one case where I would have preferred form to follow function: input and output level knobs would offer a greater degree of control.

PRODUCT SUMMARY

Eventide
Clockworks Legacy Bundle 1.12 (TDM)
vintage effects plug-ins
$795

FEATURES	3.0
EASE OF USE	4.5
QUALITY OF SOUNDS	4.5
VALUE	3.5

RATING PRODUCTS FROM 1 TO 5

PROS: Terrific sound quality. Easy and fun to use. Authentic vintage vibe.

CONS: Available for Mac TDM systems only. Effects in H949 and H910 are DSP-intensive and limited to a 96 kHz sampling rate.

Manufacturer
Eventide, Inc.
tel. (201) 641-1200
e-mail audio@eventide.com
Web www.eventide.com

On the left side of the front panel are the standard Threshold, Attack, and Release knobs. A Bass Cut button is included along with meter-switching buttons that let you monitor the input, output, or gain reduction. The virtual meter's ballistics are animated realistically and look appropriately vintage. Because of Omnipressor's wide dynamic range and multipurpose operation, the default needle position is in the center, with a maximum throw of –30 dB to the left and +30 dB to the right. There are attenuation and gain-limiting knobs with accompanying LEDs; these two controls work with the Function knob to further restrict or enhance the amount of compression or expansion. Omnipressor also includes a sidechain input.

I can't praise this plug-in enough. It sounds unlike anything else I've heard, has loads of character, and can dramatically change the sound of an individual instrument or a full mix. If Eventide still made the hardware version, I'd buy two. As it stands, I'm satisfied with the plug-in.

LASTING LEGACY

In a market glutted with digital simulations of vintage instruments and processors, Eventide has done an outstanding job of creating something, well, "original." The Clockworks Legacy Bundle imparts an instantly recognizable '70s character that I haven't heard in any other plug-in, and these effects sound musical and interesting, allowing you to process sounds in unexpected ways.

There are a few drawbacks to the Legacy Bundle. I wasn't wild about the sound of the H910, especially considering that the H949 is included in the set. The $795 price tag is also a bit steep, although you do get five plug-ins for that price. However, the bundle's biggest limitation is its lack of compatibility with any system other than Mac-based Pro Tools TDM rigs. Having said that, if you own such a system, and you're involved in rock, techno, industrial, or any other effects-heavy production, Eventide's Clockworks Legacy Bundle offers lots of new old sounds that are worth a serious listen.

EDIROL

UA-1000 (WIN)

USB audio comes of age with this multichannel interface.

Brian Smithers

Before FireWire audio interfaces hit the scene, Edirol was among a small group of manufacturers offering USB audio interfaces. Defying the widespread yet erroneous view that USB 1.1 was unsuitable for quality audio I/O, Edirol helped break PCI-card dominance of the audio-interface market. When FireWire finally arrived, it capitalized on the one legitimate limitation of USB: low bandwidth.

Technology is a never-ending game of leapfrog, of course, and USB 2.0 broke the bandwidth barrier with a 20 percent bigger pipeline than FireWire (which has subsequently doubled its own bandwidth). Not surprisingly, Edirol is the first to take advantage of USB 2.0. Its UA-1000 offers ten simultaneous channels of 24-bit, 96 kHz I/O, along with such niceties as four microphone preamps, analog and digital connections, and hardware monitoring.

The UA-1000 is a one-rackspace device jam-packed with connections, controls, and LEDs on the front panel and still more connections on the back (see **Fig. 1**). It ships with ASIO, WDM, and MME drivers for Windows XP. Macintosh support is planned, and Edirol is currently working with Apple to develop a USB 2.0 audio standard for OS X. You can connect as many as four UA-1000s to your computer and use them simultaneously. For this review, I tested a single UA-1000 on an Athlon XP 2500+ with 512 MB of RAM.

Minimum System Requirements

UA-1000

Pentium 4/1.2 GHz; 128 MB RAM (256 MB recommended); Windows XP; USB 2.0 port

UP AND AWAY

Installing the UA-1000 requires the Windows XP Service Pack 1 as well as a software update, which is included on CD-ROM. An instruction sheet clearly explains the process of checking for the Service Pack and installing the update. Edirol explains that the update provides efficiencies in the USB stack that were developed by Microsoft after testing with the UA-1000. On my system, installation was a snap, and I experienced no instability during the review period.

The installation process places a UA-1000 icon in the Windows Control Panel. Clicking on it reveals an unassuming little applet (see **Fig. 2**) that offers a virtual patch bay for flexible I/O routing along with a mixer to facilitate such things as cue mixes during recording.

The control panel is divided into two patch bays and two mixer sections. The Wave In patch bay controls whether a stereo physical input, a virtual output, or the Monitor Output will be directed to your DAW's track inputs. (For example, you could choose inputs 1 and 2 in Sonar and actually be directing physical inputs 7 and 8 to that track.) This disconnect between the physical and virtual inputs can save a lot of time and trouble repatching gear or reassigning inputs. In addition, assigning a virtual output as a track input allows you to directly record the output of other applications such as the Windows Media Player. You can also record a real-time mix of all of the UA-1000's physical outputs by assigning Monitor Output as a track input. That lets you generate a reference mix as you're doing multitrack recording.

The Output patch bay gives you two sets of assignments, labeled "Direct Monitor: On" and "Direct Monitor: All Off." The first set is used when either the front panel Direct Monitor Soft Ctrl button is on or any of the Input mixer's channels are enabled. In all other cases, the second set is used. That allows you to listen to physical inputs on all channels when tracking, then to easily switch to monitoring virtual outputs when mixing.

As many as five configurations can be stored in nonvolatile memory. Settings for the Wave In patch bay, the Output patch bay, the Monitor, and the panel switches can be stored and recalled independently. That means that you can, for example, load the Wave In

FIG. 1: The Edirol UA-1000's front panel features four combination XLR/TRS inputs with preamps and phantom power. The back panel handles the rest of the connections, with analog audio, digital audio, word-clock, and MIDI I/O.

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patch-bay settings of Memory 4 without changing your current panel-switch settings.

Context menus are exceptionally well-implemented in the patch bay and mixer. I would expect them to reset a control to its default setting, but they do that and much more. Right-clicking on a fader allows you to set it or all faders to any multiple of 6 dB between –36 and +120 dB, including full attenuation. Right-clicking on a pan control lets you set it or all pan controls to hard left, hard right, center, or alternating left and right for stereo sources.

It's also nice that all patch-bay and mixer functions can be controlled from the computer keyboard, and about half of them have hotkeys. That's not just useful to mouse-phobic musicians like me; it also makes life easier for visually impaired musicians.

The UA-1000 documentation is exceptionally detailed. If you follow the instructions carefully, you probably won't need any tech support. My only real beef with the manual is its silly warnings to use only the power and USB cables provided. In fact, any good-quality cables will do, and the whole point of detachable USB and IEC-power cables is their universality.

GO WITH THE FLOW

The four combination jacks on the UA-1000's front panel accept balanced XLR, balanced TRS, or unbalanced ¼-inch TS plugs. Each of these jacks provides microphone preamplification. Phantom power (+48V, on the XLR connection only) and a 20 dB pad can be switched on and off for each pair. The front-panel channel 3 input can be switched to high-impedance operation for use with guitar or bass pickups.

The four front-panel inputs are duplicated on the back panel with unbalanced TRS jacks. Those can be switched in pairs to act as channel inserts, adding a good deal to the UA-1000's flexibility. However, the switches are on the back panel, and the front-panel inputs are functional only when the back-panel connections are set as channel inserts. That means you'll want to leave an open rackspace above the UA-1000 if you intend to change modes very often. The remaining four analog inputs (for channels 5 through 8) are balanced TRS jacks on the back panel. Sensitivity adjustments for each pair of analog inputs are made using four front-panel knobs. The front panel also has knobs for adjusting headphone and direct-monitor signal levels.

The UA-1000's eight analog outs are rear-panel balanced +4 dBu TRS jacks. You can use optical or coaxial S/PDIF to provide input for channels 9 and 10. The optical jacks also accept ADAT Optical; because ADAT Optical input renders the coaxial S/PDIF and the analog inputs and outputs inactive, the UA-1000 is limited to eight simultaneous channels of I/O when using the ADAT connections. One of the UA-1000's few real shortcomings is that it doesn't support Sample Multiplexing (SMUX). That means the maximum sampling rate for ADAT I/O is 48 kHz. Fortunately, the S/PDIF I/O can handle 96 kHz. BNC word-clock and MIDI I/O round out the UA-1000's rear-panel I/O.

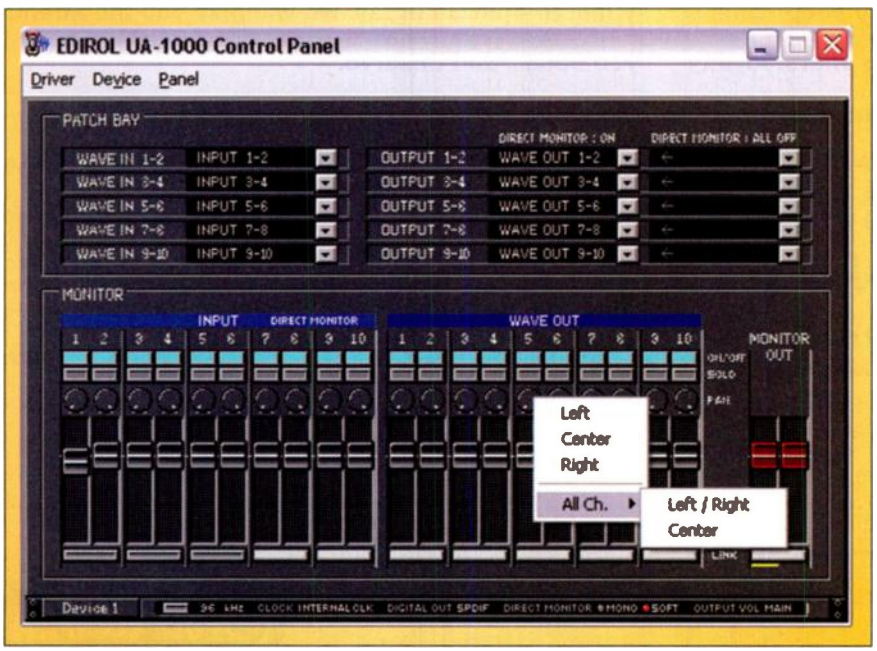


FIG. 2: The UA-1000 Control Panel offers a flexible virtual patch bay and mixer. Its context menu provides speedy access to common functions.

PRODUCT SUMMARY

Edirol
UA-1000
USB 2.0 audio interface
$945

FEATURES	3.5
EASE OF USE	3.5
AUDIO QUALITY	4.0
VALUE	3.0

RATING PRODUCTS FROM 1 TO 5

PROS: Low-latency ASIO 2.0 and WDM drivers. Four preamps. High-impedance input. Flexible software patch bay and mixer. Channel inserts (on channels 1 through 4). Multiple units can be used simultaneously.

CONS: Changing sampling rate requires power cycle. No 96 kHz ADAT Optical support. Can't use ADAT and analog I/O simultaneously.

Manufacturer
Edirol
tel. (800) 380-2580
e-mail sales@edirol.com
Web www.edirol.com

INTO ACTION

Using the UA-1000 is straightforward, as you'd expect, and I had no trouble getting it to do anything I wanted. The front-panel microphone preamps are very convenient for remote recording or for quickly setting up a mic or four. Mic cables hanging down the front of your rack may not be appealing in a

UA-1000 Specifications

Analog Inputs	(4) front-panel balanced XLR/TRS combo with phantom power and 20 dB pad (channel 3 switchable to high-impedance); (4) rear-panel unbalanced TRS (these duplicate front-panel ins); (4) rear-panel balanced TRS line
Analog Outputs	(8) balanced TRS; (1) ¼" headphone
Digital I/O	(2) optical S/PDIF (ch. 9–10) or ADAT Optical (ch. 1–8, 48 kHz max.); (2) S/PDIF coaxial (ch. 9–10); (2) BNC word-clock (in and out)
MIDI I/O	(1) In, (1) Out
Sampling Resolution	24 bits
Sampling Rates	44.1, 48, 88.2, 96 kHz
Drivers	ASIO 2.0, WDM, MME
Additional Features	software patch bay and mixer
Dimensions	17.0" (W) × 9.6" (H) × 1.7" (D)
Weight	6.2 lb.

permanent installation, but that's the trade-off for having a multipurpose unit.

The preamps, and indeed the whole analog front end, are the same as those found in Roland's VS-2480. Although I didn't compare them directly, I can say that the preamps are reasonably quiet and relatively transparent sounding, as I would expect in comparably priced interfaces and mixers.

My biggest issue with the UA-1000 is that to change the sampling rate you must power-cycle the unit. That's not as bad as having to open the box and move a dip switch, but if you change sampling rates frequently, you may find it annoying.

Everything I recorded and played back with the UA-1000 sounded quite good. At lower sampling rates the sound was clean and pleasing; at higher sampling rates there was a bit more detail and clarity. Sonically, I'd be happy to have the UA-1000 in my studio.

You can find similarly priced USB interfaces with more I/O, but probably at the cost of a couple of preamps, MIDI I/O, and channel inserts. If those features are important, the UA-1000 definitely deserves your consideration. If your PC has USB 2.0 but lacks FireWire, the UA-1000 is your ticket to high-quality audio. ❦

PRESONUS

EUREKA

A quality channel strip for the personal studio.

By Eli Crews

The Eureka is a formidable new channel-strip processor from PreSonus that features a mic preamp, a compressor, and a 3-band parametric equalizer. With a list price of $699, it falls in the low- to midprice range, an area in which PreSonus has previously succeeded with products such as the MP20 mic preamp and the BlueMax compressor.

PreSonus is also releasing the AD192, an optional 24-bit, 192 kHz digital-output card ($249) that goes with Eureka and will make the unit even friendlier for DAW applications. The card wasn't ready to ship at the time of this writing, but according to PreSonus, it should be available by the time you read this.

BOLD AND BLUE

The Eureka looks bold and elegant, with a brushed steel face and cobalt metal knobs (see **Fig. 1**). Its switches light up a nice, eerie blue when engaged, making it easy to see their respective states at a glance. Trying to see the knob settings is not as easy; they reflect so much light that it's difficult to see their indicator lines. (Ironically, it was only in low light, with the unit angled in a particular way, that I could easily see the settings. In a rack, with normal lighting, it was almost impossible.)

As you would expect, most of the inputs and outputs are on the back panel, but for convenience, there is a ¼-inch instrument input on the front. Above it are three LEDs (showing –20 dB, 0 dB, or Clip) that indicate signal strength at the instrument and mic inputs. The input section's three knobs control preamp gain, impedance, and saturation (which I'll discuss more later). Above these knobs are switches for activating the line input, the +48V phantom power, the –20 dB pad, the 80 Hz high-pass filter, and the polarity-reverse function.

The Compressor section offers a comprehensive set of controls: Thresh (threshold), Ratio, (makeup) Gain, Attack, Release, and Sidechain Hi Pass, a filter control for frequency-dependent compression applications such as de-essing. You can tailor the compression further with the Soft switch, which toggles between soft- and hard-knee compression curves. Finally, there's a Bypass switch for taking the compressor out of the circuit. The default, unlit state of the switch indicates that the circuit is active, which may be counterintuitive to some users.

In the middle of the Eureka's front panel is a brightly lit VU meter with the PreSonus squiggle logo printed on it. For engineers who are used to looking at analog tape machines, the VU meter is an especially nice feature, even if it adds slightly to the cost to the unit.

EQUAL TIME

To the right of the meter is a 3-band, fully parametric EQ. Each band has three knobs: Q (bandwidth), Gain, and Freq (center frequency). For each band, the range of the Q is from three octaves to two thirds of an octave, getting narrower as you turn the knob clockwise. The Gain knob allows up to 10 dB of boost or cut for each band.

The frequency range of the EQ is 20 Hz to 20 kHz, divided among the bands as follows: the Low band ranges from 20 Hz to 300 Hz, the Mid band from 200 Hz to 3 kHz, and the High band from 2 kHz to 20 kHz. The EQ circuit has its own bypass switch and a switch labeled EQ>Cmp that places it before the compressor in the audio signal path, which is a very handy feature.

To accommodate the many controls in the EQ and compressor sections, PreSonus staggered the knobs for each in a zigzag fashion. As a result, I found the knobs to be a little too close together to adjust comfortably. That said, the knobs do have a solid feel and nice turning action, and they are detented in steps, which is useful both for recalling settings and for making fine adjustments. Rounding out the front panel are a master volume knob, with a range of –80 dB to +10 dB, and a switch for assigning the VU meter to the task of showing the compressor circuit's gain reduction instead of output level.

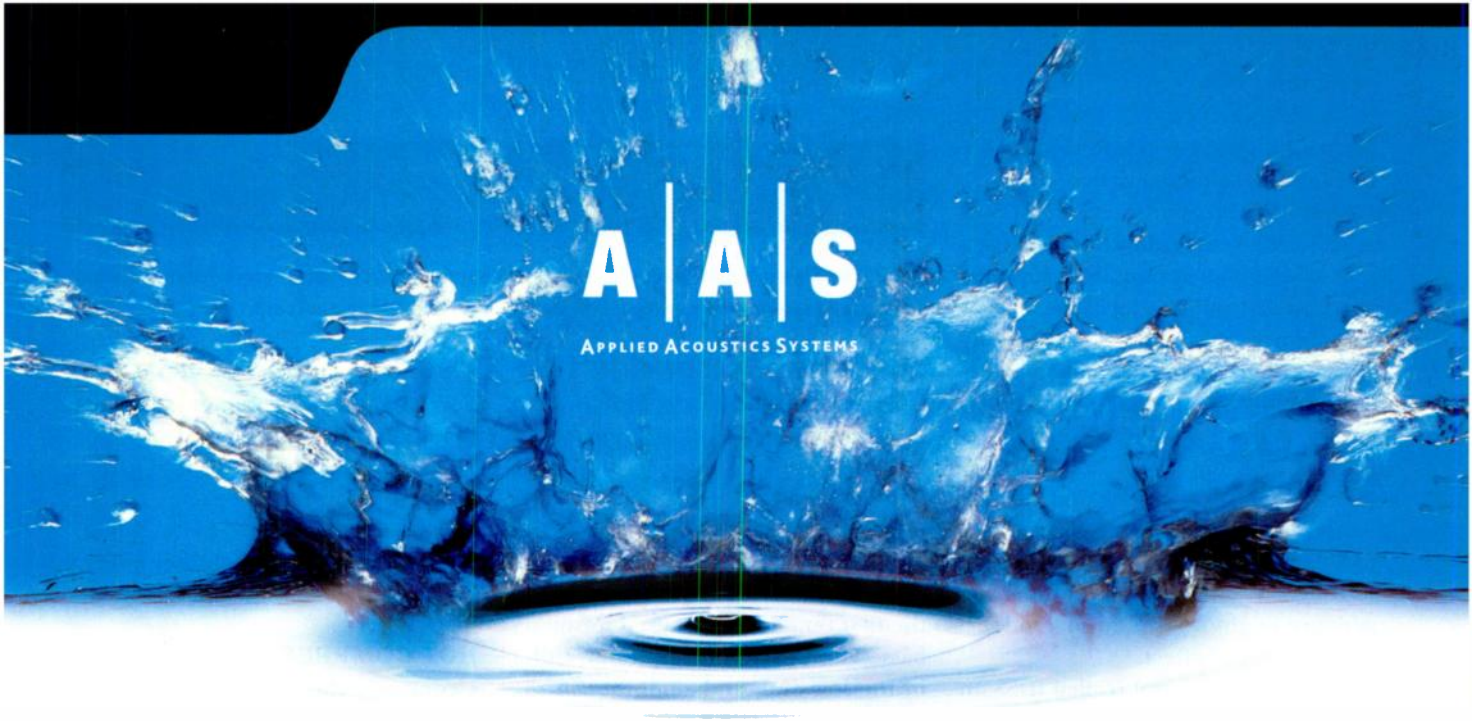
Eureka's back panel (see **Fig. 2**) features an XLR mic input, a ¼-inch TRS line input, separate balanced ¼-inch TRS send and return insert jacks, and both XLR and ¼-inch TRS jacks for the line-level output.

OHM ON THE RANGE

I tried the Eureka in a variety of tracking and overdubbing applications, and I compared it to other preamps in my rack, including the Focusrite ISA 428 PrePack (my current benchmark pre), the Focusrite Platinum VoiceMaster (a "channel-strip" box in the same price range as the Eureka), and PreSonus's own MP20 dual mic pre. I was surprised at how well the Eureka held its own.

The Eureka reportedly has a similar preamp design to the MP20, which I've used and loved for years: a discrete Class-A, transformer-coupled, dual-servo, twin-JFET preamplifier. The only difference is that the Eureka's input impedance is selectable, whereas the MP20's is fixed at 1.3 kΩ. This added feature lets you match (or mismatch) the output impedance of the mic you

FIG 1: The PreSonus Eureka is a single-channel mic preamp, compressor, and equalizer that offers quality sound and a high level of control.



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are using, giving you additional control for tonal characteristics.

I question PreSonus's choice of 50Ω as its lowest setting, however. That setting was completely unusable with every mic I tried, producing an extremely low and distorted output. The lowest setting on my other selectable-impedance preamp, the ISP 428, is 600Ω, which seems far more useful as a low extreme. Conventional wisdom says the input impedance should run about ten times a mic's output impedance. Most common mics have impedances between 50 and 300Ω, so it makes sense that a 50Ω input impedance might not sound so good, except as a special effect.

The Saturate control (called IDSS on the MP20) emulates the effect that tape saturation and tube warmth have on the even-order harmonics of a sound. I never touch this knob on my MP20; doing so tends to dull the unit's otherwise sparkly tone without any sonic benefit. But the Eureka's Saturate control is useful, smoothing the grit on electric bass, distorted guitar, and gruff male vocals.

PRODUCT SUMMARY

PreSonus
Eureka
channel strip
$699

FEATURES	4.0
EASE OF USE	3.5
AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Good sound quality. Separate, balanced send and return insert jacks. Selectable mic input impedance. VU meter that can show gain reduction. Natural-sounding compression. Good-sounding EQ offers three fully parametric bands.

CONS: Knobs are small and close together. Knobs' shiny surface makes it hard to see indicator. Lowest impedance setting too low for most uses.

Manufacturer
PreSonus Audio Electronics
tel. (800) 750-0323 or (225) 216-7887
e-mail info@presonus.com
Web www.presonus.com

FIG. 2: The Eureka's back panel features an integrated power supply; servo-balanced inputs, outputs, and inserts; and a space for the optional AD192 digital-output card.

COMPARE AND CONTRAST

I began my testing by comparing the Eureka against my other preamps for recording direct electric bass. The Eureka sounded better than both the MP20 and the Focusrite ISP 428 for this application. I was quite surprised, considering that the ISP 428 blew me away the first time I heard it used for direct bass. The Eureka was rounder, fuller in the low end, and smoother in the highs than the ISP 428. The MP20 sounded slightly flat in comparison to the Eureka. The Focusrite Platinum VoiceMaster doesn't have an instrument jack, so I didn't include it in the direct-bass comparison.

I also tried running the electric bass through a bass amp, miking the amp with an AKG C 414 EB, and patching the mic's output through the various preamps. In this test, the ISP 428 one-upped the Eureka. The Eureka sounded best at the 2.5 kΩ setting, but the ISP 428 on its High Impedance setting (a much higher 6.8 kΩ) was clearer and crisper. Both the MP20 and the VoiceMaster were a little dull in comparison, which makes sense given the higher input impedance of the Eureka. Adding a little of the Eureka's saturation effect, which helped to smooth out the fret noise, sounded good in this instance.

Next I ran clean electric guitar through an amp, miked it with a Shure SM57, and ran its output through each of the preamps. The differences between the preamps were so subtle that there was no way to pick a clear winner. Again, the Eureka sounded best when I used its 2.5 kΩ impedance setting. With clean guitar, I found that dialing in any of the saturation effect destroyed the sparkle in an unpleasant, muffling way.

On distorted guitar, the Saturate knob pleasantly smoothed out the grit. It might make the guitar seem a little dark in a mix, but on its own it rounded out the otherwise harsh sound. The ISP 428 also excelled in this test, but the Eureka sounded clearer and fuller than the MP20 and the VoiceMaster.

On male vocals, the ISP 428 had the greatest detail in both frequency and dynamics, but the Eureka (at 2.5 kΩ) and the MP20 were right behind, followed closely by the VoiceMaster.

HOT COMPRESS

I compaed the Eureka's compressor with a dbx 266A and various software-compression plug-ins for Digidesign Pro Tools and MOTU Digital Performer. The Eureka's line input makes it possible to use the compressor and EQ sections during mixdown. The line input and outputs are rated at 0 dBu = 0 dB, so theoretically a +4 dBu signal will be too hot coming in, and a –10 dBV signal will be a little too quiet. In practice a –10 signal drove the compressor and EQ perfectly well, but a +4 signal distorted the Eureka a little at the input stage. The unit's output was fine for both +4 and –10 inputs, but setting the input of my sound card to –10 resulted in a slightly better signal-to-noise ratio.

The Eureka's compressor was a bit more subtle than either the dbx or the plug-ins, even when pushed to its max. It seems better for gentle, musical compression than for extreme effect-type limiting. The controls are responsive yet gradual, and the included sidechain highpass filter is handy for de-essing or letting through low frequencies on drum overheads or bass tracks. The presets listed at the end of the manual offer good starting points for setting the compressor for various instruments.

As mentioned, you can monitor the gain reduction of the compressor on the VU meter, which is a capability usually found only on high-end units. Confusingly, however, the meter shows gain reduction even when the compressor is not engaged.

Also, when the compressor or EQ was

Eureka Specifications

Channels	1
Audio Inputs	(1) balanced XLR mic; (1) ¼" balanced TRS line; (1) ¼" TS inst.; (1) ¼" balanced/unbalanced TRS insert send
Audio Outputs	(1) XLR; (1) ¼" TRS; (1) ¼" balanced/unbalanced TRS insert return
Input Impedance	2.5 kΩ, 1.5 kΩ, 600Ω, 150Ω, or 50Ω (mic, selectable); 10 kΩ (line/insert return); 1 MΩ (inst.)
Output Impedance	51Ω
Dynamic Range	>115 dB
Headroom	+22 dBu
Frequency Response	10 Hz to 50 kHz
Gain	+12 dB to +52 dB
Total Harmonic Distortion	<0.005% (no saturation); <0.5% (full saturation)
Dimensions	19.00" (W) × 1.75" (H) × 7.00" (D)
Weight	8 lb.

switched in or out of the circuit, the unit emitted a loud pop. I realize that it's difficult to take components in and out of the signal path without some sort of click or pop, but it seemed louder on the Eureka than on other units I've worked with. (According to PreSonus, this was a problem found only on very early production units, and it has subsequently been remedied.)

That aside, the EQ circuit, like the compressor, was user-friendly and musical. It was slightly subtler and more transparent than the EQs in my DAWs and on my console, but that's generally a good thing. The inclusion of three bands of fully parametric EQ is quite remarkable for this price range, as is the smoothness of the high band.

EUREKA!

I was quite impressed with the Eureka. For tracking and mixing, it offers a winning combination of versatility, control, and excellent sound quality that's sure to appeal to seasoned engineers and less experienced recordists alike.

Eli Crews *is an Oakland-based engineer and musician. For information on Eli and his studio, go to www.newimprovedrecording.com.*

Quick Picks

SIBELIUS SOFTWARE

Sibelius 3 (Mac/Win)

By Jim Aikin

The Sibelius notation program has a strong reputation for combining an extensive feature set with an easy-to-use interface. My first experience with Sibelius was with version 3 ($599), and I'm happy to confirm that the program is as elegantly designed as it is powerful.

EM ★★★★★ hot pick

The manual is one of the best I've ever seen: it's well written, well organized, comprehensive, and thoroughly indexed. All of the basic tasks, from entering notes to repositioning symbols, are handled in a transparent fashion. The commands you'll need most often—flipping a stem up or down or changing an accidental to its enharmonic equivalent—use the fewest keystrokes, and in most cases that means one keystroke.

Version 3 is primarily a maintenance update writ large; it's not a floor-to-ceiling overhaul (none was needed), and it's not packed with innovations. However, the list of new features and fixes in the *Upgrading to Sibelius 3* booklet fills 11 pages. Rather than wade through every single detail, I'll just hit a few high spots.

Duly Noted

Sibelius's Arrange feature provides a way to intelligently copy music from one set of staves to another. That lets you do things like expand a piano score into an arrangement for string orchestra, or collapse a string orchestra arrangement into a piano score. In version 3, there are 18 new jazz Arrange styles based on the orchestration practices of well-known jazz composers and arrangers.

Beaming has been improved. It's now possible to beam over rests, and beam grouping and subgrouping is more flexible. Guitar-chord diagrams can now be transposed, and you can create your own libraries of favorite diagrams.

Font changes can now be applied to all objects of a given type in the score. When the guitar player in the band I work with complained that the chord symbols in a lead sheet were too small, I was able to increase their point size globally with one command.

When editing a score that includes multiple instruments, you can hide and show various staves independently, making it easier to work on one area (such as the woodwind section) at a time. In addition, the ability to import files in MakeMusic Finale format has been improved. Speaking of which, Sibelius Software offers a competitive cross-grade price of $199 for users of Finale, Passport Designs Encore, and MOTU Mosaic.

Sibelius 3 includes 30 new plug-ins, covering a wide range of chores such as Add Accidentals to All Notes, Add Schenkerian Scale Degrees, Realize Figured Bass, and Convert Folder of Scores to Web Pages. In addition, symbols can now be attached to the system rather than to an individual staff. This is useful for symbols that should be shown in all parts after part extraction. The ability to export TIFF graphic files, with resolutions up to 1,200 dpi, has also been added.

Using the Kontakt Silver sample-playback synth in Sibelius 3 gives you a better idea of how your scores will sound in the real world.

Making Kontakt

For some musicians, the big news with Sibelius 3 may be the inclusion of a sample-playback synth called Kontakt Silver. Although it is far less powerful than Native Instruments Kontakt, its parent program, Kontakt Silver is a big improvement sonically over the software synth that lives in your computer's operating system. Consequently, you can get a reasonable idea of what your score is going to sound like before you present it to the ensemble that will play it.

You can capture as many as eight tracks of audio playback from Kontakt Silver to your hard drive. Although the sound set is limited (20 sounds in all, with very few sound-editing features) and playback is rather wooden (compared to a dedicated sequencer), the feature is there if you need it.

Final Bar

At $599, Sibelius will appeal primarily to those who are serious about creating scores (although it is priced in line with its biggest competitor). In a few tweaky areas, Sibelius might not be as flexible as Finale, but in my experience it's much easier to use.

Make no mistake: Sibelius's feature set is fully professional. The program supports cross-staff beaming, compound time signatures, floating cue staves, Web publishing, the ability to import graphics, and more. The specialized needs of teachers and songwriters are well taken care of. After years of burying my head in the piano-roll display of a sequencer, I'm actually starting to look for excuses to do *more* printed scores—Sibelius is that good.

Overall EM Rating (1 through 5): 5

Sibelius Software, Ltd.; tel. (925) 280-0600; e-mail infousa@sibelius.com; Web www.sibelius.com

TRILLIUM LANE LABS

TL EveryPhase 1.1 (Mac/Win)

By Nick Peck

Trillium Lane Labs TL EveryPhase 1.1 ($249) is a cross-platform phaser plug-in

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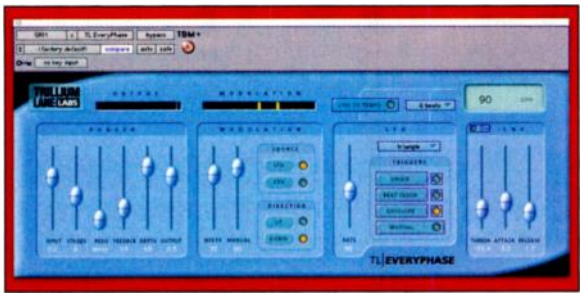

Trillium Lane Labs' EveryPhase is a cross-platform phaser plug-in for Pro Tools that offers a few unusual twists.

for Pro Tools. On Pro Tools|HD Accel systems, it offers a full 5.1-channel mode, creating phasing effects that swirl all around the listener. The installation disc provides RTAS and TDM versions for Mac OS 9, Mac OS X, and Windows. TL EveryPhase uses iLok copy protection and thus requires a USB dongle.

Phasers on Stun

TL EveryPhase is designed cleanly, with a window that displays every parameter at all times. The graphics are somewhat spartan, emphasizing substance over style. Rather than knobs, most of EveryPhase's controls are faders, which work better in a computer environment. The parameters are laid out in four panes: Phaser, Modulation, LFO, and Envelope. The plug-in also provides a tempo indicator and Output and Modulation meters. I controlled EveryPhase with my Digidesign Control|24, which made tweaking the parameters a snap. (I also used a dual-processor Mac G4/867 MHz, Pro Tools TDM 6.2.2 software, and Pro Tools|HD 3 Accel hardware.)

The Phaser section consists of Input and Output level controls and four faders that control the effect: Stages, Resonance, Feedback, and Depth. The Stages fader sets the number of phase taps, from 2 to 18. Feedback sends a portion of the phased output back to the input, changing the sound's timbre; adding too much feedback creates a rather cold-sounding whine. Resonance alters the phaser stage from which the feedback is taken. Depth is the most interesting timbral parameter: positive depth values increase the amount of phase relative to the dry signal, as you'd expect, whereas negative values create a markedly different, less conventional type of phased tone.

In the Modulation section, you choose whether it's the LFO or the envelope follower that controls the phase change, and by how much. The Manual fader lets you determine a starting offset to the modulation sweep, and Up and Down buttons let you determine the phase modulation's direction. The Modulation meter is particularly novel and useful: in a single bar graph, it displays the modulation width, offset, and amount.

The LFO section features an LFO Rate control and waveshape selections—triangle, ramp, sine, half-sine, square, and a series of stepped waveshapes. Rather than gliding smoothly, the stepped waveshapes jump from one value to the next, which can be particularly effective when they're synced to the session's beat clock. The LFO can operate in free-run or single-shot mode, but it can also be triggered to run from the beat clock, from an envelope generator, or manually. The trigger sources can run separately or in tandem, offering a good deal of creative flexibility.

The Envelope section triggers phasing with the amplitude of the incoming audio signal. The Threshold, Attack, and Release settings behave just as they would on a compressor, but instead of altering the dynamic range, they control the amount of phase shift. Of particular interest is Sidechain mode, which lets you specify a different audio track as the envelope source. The Envelope section is great for altering the timbre of percussive sounds.

The Very Next Phase

On first listen, I was not bowled over by EveryPhase's sound. It seemed a bit thin, and many of the factory patches had a distinctively metallic quality. You can't always judge a plug-in by its default presets, however. Once I began to twiddle the faders, I coaxed EveryPhase into producing more and more sounds that I really liked. The Envelope function worked well to trigger short phasey bursts on snare drums, creating a swooping pitch bend during the snare's posttransient ring-out. Acoustic guitar took on a shimmery, rubbery quality when modulated continuously with a slow LFO. I also found EveryPhase useful for sound design, sweeping the tail of a rifle shot to create an element for a sci-fi type of weapon.

It's a Contender

TL Labs has thrown its proverbial hat into the phaser ring with EveryPhase. Its interface is uncluttered, and it offers some novel twists on well-trod ground. The Stages parameter, the Modulation meter, and the stepped LFO waveshapes are all interesting and useful. Cross-platform compatibility and support for surround sound are also quite welcome.

Though EveryPhase isn't the only phaser I would ever use, I did find situations in which I really liked it, particularly when used subtly. Once I really delved into the parameters, the sound began to grow on me. If you're in the market for a Pro Tools–based phaser plug-in, download the demo version of EveryPhase and try it yourself.

Overall EM Rating (1 through 5): 3

Trillium Lane Labs LLC; e-mail info@tllabs.com; www.tllabs.com.

I3 SOFTWARE

DSP-Quattro 1.5 (Mac)

By Len Sasso

DSP-Quattro ($129) reaches beyond the boundaries of traditional audio-editing software. In addition to the typical sample-editing tools, it features multitrack recording, simultaneous playback of multiple audio files, real-time hosting of virtual instruments, and real-time effects processing of audio input. Its fully configurable signal path allows you to mix any combination of audio sources, apply separate insert effects to each source, apply master effects to the mixed output, and record the results to a new audio file. Along with all this flexibility, you get a full-featured sample editor.

EM ★★★★★ hot pick

DSP-Quattro can edit files in AIFF, WAV, and Sound Designer II formats at any sampling and bit rate. It can also import any file for which there is a QuickTime translator, which includes MP3 and AAC

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Produced by veteran sound designer Jennifer Hruska. Recorded by Emmy Award winning engineer Antonio Oliart and RIAA award winning engineer John Bono. Processed and programmed with the utmost in playability, this collection is a joy to create with.

Articulations:

Legato
Marcato Legato
Melodic Legato
Flutter Tones
Half Step Trills
Whole Step Trills
Staccato
Double Tongue "ta"
Double Tongue "ka"
Muted Legato
Muted Staccato
Muted Flutter Tones
Rips
Sforzando
Mute Sforzando
Stopped Horns
Horns Bells Up
Trombone Slides
Trombone Pedal Tones
Effects

Instruments:

French Horns
Solo Melodic French Horn
1st & 2nd chair French Horn Section
1st through 4th chair French Horn Section

Trumpets
Solo Melodic Trumpet
1st chair Trumpet
2nd & 3rd chair Trumpet Section
1st through 3rd chair Trumpet Section

Trombones
1st chair Bass Trombone
1st & 2nd chair Tenor Trombones
1st through 3rd chair Trombone Section

Tuba
C Tuba
Eb Tuba

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DSP-Quattro offers a full-featured sample editor (top), convenient marker, region, and loop management (bottom right); support for Playlists and CD burning (bottom left); and real-time hosting of virtual instruments (bottom-left insert).

compressed files as well as audio-CD tracks. For audio files that for some reason don't load or import using those alternatives, there is a Raw Data import option that will try to translate any file into audio based on parameters (bit size and order, sampling rate, and so on) that you choose. You can simultaneously open multiple audio files for editing, and DSP-Quattro will maintain a separate, complete undo/redo history for each one, in effect making all editing nondestructive.

As with most sample editors, you can set regions, markers, and loops; you can copy and paste between files or within the same file; and you can apply standard DSP processes like gain change, reversing, trimming, removing DC offset, fading, pitch or time shifting, and so on, to any portion of the audio file. Furthermore, you can instantly apply any of DSP-Quattro's plug-in effects as well as any VST or AU plug-in effect that you have installed. Built-in plug-ins include a chorus, delays, reverbs, and various filters.

DSP-Quattro has the full spate of looping features. You can set loop points manually or by designating a tempo, length (for example, four bars at 120 bpm), and starting position. A sample-accurate loop editor allows you to match loop start and end points precisely, and a handy X-Fade Loop feature will place a copy of the looped region after the original and create a crossfade between them. By copying a loop into a new audio file and repeating the X-Fade Loop feature you can create one long audio file with as many perfectly crossfaded copies of the loop as desired.

Plug In and Play Along

In addition to instant offline processing of a single DSP effect, you can insert plug-in effects after each audio generator (audio file, audio input, software instrument, Playlist track, and so on). There are four insert slots per audio generator and four more insert slots for the Master output. In case the audio generator is an audio file, the insert effects can be printed to the same or a new file, with or without the Master effects. For other audio generators, you can record the result of effects processing by using the MasterOut recorder.

Playlists are fully implemented in DSP-Quattro, including the ability to assign different plug-in effects to each Playlist track. You can add tracks to the Playlist either from audio files on your hard drive or from audio files already open for editing. In the latter case, the track can be any region, loop, or selection within the audio file. You can set nondestructive gain change, pause time, and crossfades between Playlist tracks. Once set up, your Playlist can be exported as an audio file, exported as an audio-CD image in Roxio Toast format, or burned directly to an audio CD in Redbook format (OS X only). You can also simply use the Playlist as an audio-file sequencer, with which to play along.

DSP-Quattro can interact with hardware or software samplers in three ways: by exporting and importing audio files, by using the MIDI Sample Dump Standard (SDS), and by using the SCSI protocol. A variety of sampler-specific formats for popular models from Roland, Akai, E-mu, Kurzweil, and Yamaha are supported.

DSP-Quattro is an absolutely outstanding piece of software and an incredible value. It offers features and performance on par with or exceeding sample-editing software costing four or five times as much. The documentation is complete and accurate. The user interface is well thought out, and key commands are provided for the most often used operations. Any one of its five basic functions—sample editor, soft-synth rack, real-time effects box, mastering tool, or audio-CD burner—is easily worth the $129 price tag.

Overall EM Rating (1 through 5): 5

i3 Software; e-mail support@dsp-quattro.com; Web www.dsp-quattro.com

SONY PICTURES DIGITAL

Rhythm and Twang (Acidized WAV)

By Marty Cutler

Country music offers nearly as many stylistic variations as jazz or pop, having taken on facets of both. If you're looking for country tracks infused with a shot of rock, Sony Pictures Digital's *Rhythm and Twang* ($59.95) might be just the ticket. The single CD-ROM of 16-bit, 44.1 kHz monophonic samples is devoted to stringed instruments: guitar, bass, fiddle, and pedal steel. There are no drum grooves in this collection, although a companion CD of drums would not be amiss; good country-style drumming is deceptively simple and often difficult to program convincingly.

The CD comes with a number of additional files, including a copy of Sony Acid Xpress and a demo song. A Loops folder holds subfolders for each instrument. Every loop carries the tempo and original key signature in its file name—a good thing, because no other documentation is included. Files of the same key and tempo can often hold different feels, so a 125 bpm shuffle can be one file away from a 4/4 rocker in the same tempo. A little more information would be useful to sort out the loops.

All of the tracks confine themselves to three basic keys: C, E, and G. Those are reasonable choices, since they allow the guitar and fiddle to take advantage of open and closed position riffs in typically native keys, although tracks in D and A would be welcome for the same reason. You get four starting tempos: 65, 95, 125, and 155 bpm, and the Acid formatting can stretch or compress the files for alternate tempos. I

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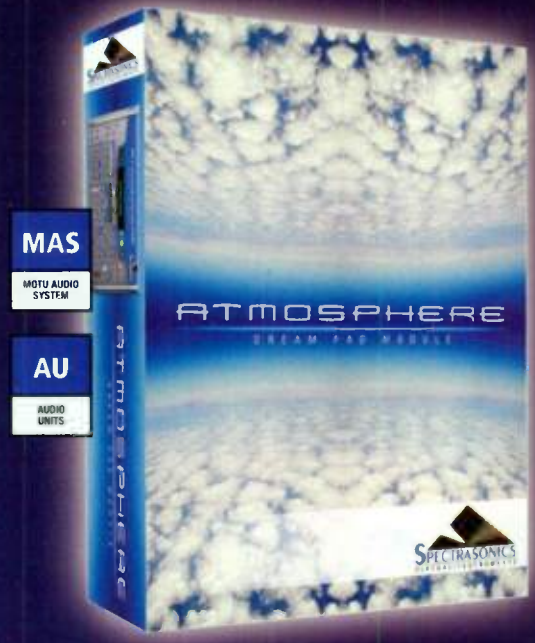

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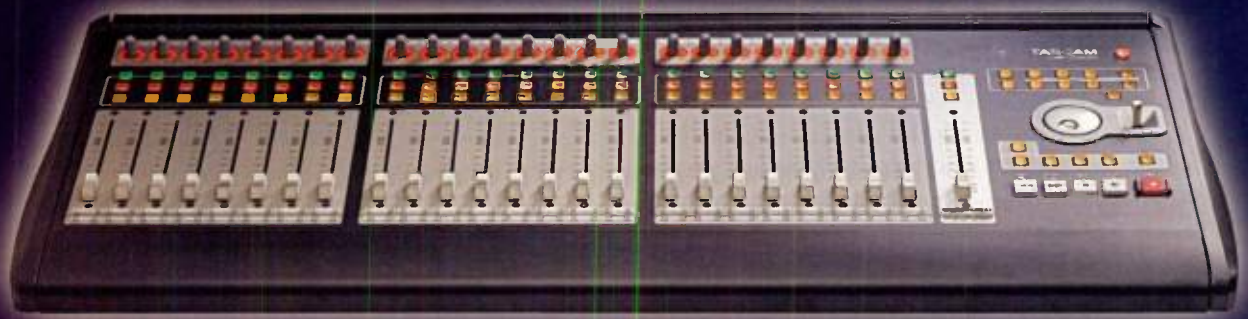

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By Larry the O

In Defense of the Lone Arranger

The late Jerry Garcia, while recuperating in the hospital from a nasty car wreck that killed one of his friends, debated the merits of working alone artistically, as compared with working in a group. His conclusion was that working in a group is the way to go, and that when people interact with each other, things often happen that surpass what an individual can do alone. Garcia's career makes a strong case for his argument.

In large part, I agree with his assessment. But I differ with him in that I don't think working in a group goes beyond where an individual can go; it simply goes different places. I feel that working alone as an artist has been unfairly criticized by many people as self-indulgent and lacking the balance in perspective that comes from having outside input. Indeed, much of the world's greatest art has been produced by one person working alone. Working alone allows for deep, considered exploration of an interior world and maintains a purity of vision.

When I work alone, especially in the studio, it is often a meditative process that feels like sculpting, as I shape one detail at a time, working toward a big picture that only I can envision. The pace can be extremely slow, and that's generally fine with me. Working alone, I have the freedom to wander, unrestricted, in any direction. I can pursue a specific idea, meander all about the musical countryside, mix the two approaches, and even try both at once—if I can figure out how to do it.

Some artists, like Frank Zappa, choose to record alone because they don't want to compromise, accede to anyone else's wishes, or deal with other people's personalities. As misanthropic as that might sound, it is completely understandable to anyone who has endured the expected, yet remarkably stupid, band squabbles.

Lest you think that all I want to do is sit alone in a windowless room with a bunch of computers, please understand that I love collaboration and playing in bands, and I think that all of the criticisms about an artist working alone are valid; it is very easy to putter around and not complete anything when working alone, and it is equally easy to overedit until all the life is drained from a piece. You might accept something that isn't all that good because you want to think that it's good. I agree that all those dangers are present.

I don't agree, however, with the assertions that one can always go to a better place in collaboration, and that working solo rarely produces excellent, fully realized, and balanced work. Some creative pursuits, such as portrait painting and sound design, don't lend themselves as readily to collaboration as music does. There are always those artists that find ways to work with others on such things, but in many ways, those types of works often come out better when created by one person.

There comes a time when I want to take what I've created out of my little world and see how people respond to it. If I'm creating single-handedly, though, I rarely share anything that is unfinished. While I am creating it, it is only mine. When it is complete, it becomes the world's.

Does that seem self-indulgent? Perhaps, but so is a hot fudge sundae. I work by myself on a much more regular basis than I eat hot fudge sundaes, yet I gain no more weight than when working in collaboration. On the other hand, eating hot fudge sundaes has never resulted in my tearing out my hair.

The point here is that determining when a work is complete or, more practically, when it has reached an acceptable degree of completion, is not subject to pressure from others or influenced by their agendas. Sometimes the input of others can bring useful alternative viewpoints; other times it can result in decisions that may be regretted in the long run.

I am not here to bury the idea of working by oneself, but to praise it. I don't, however, praise it over and above collaboration. Perhaps I'll write in favor of collaboration in another column.

Larry the O *is accustomed to spending hours alone in windowless rooms, during which time he fondles a small gold ring and mutters "My Precious!" a lot.*

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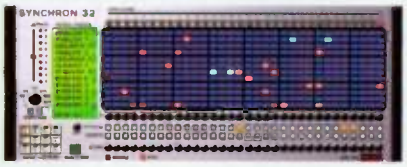

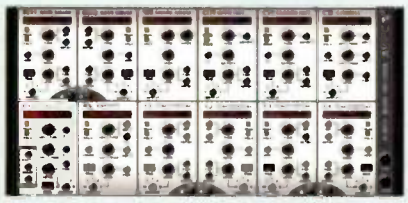

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