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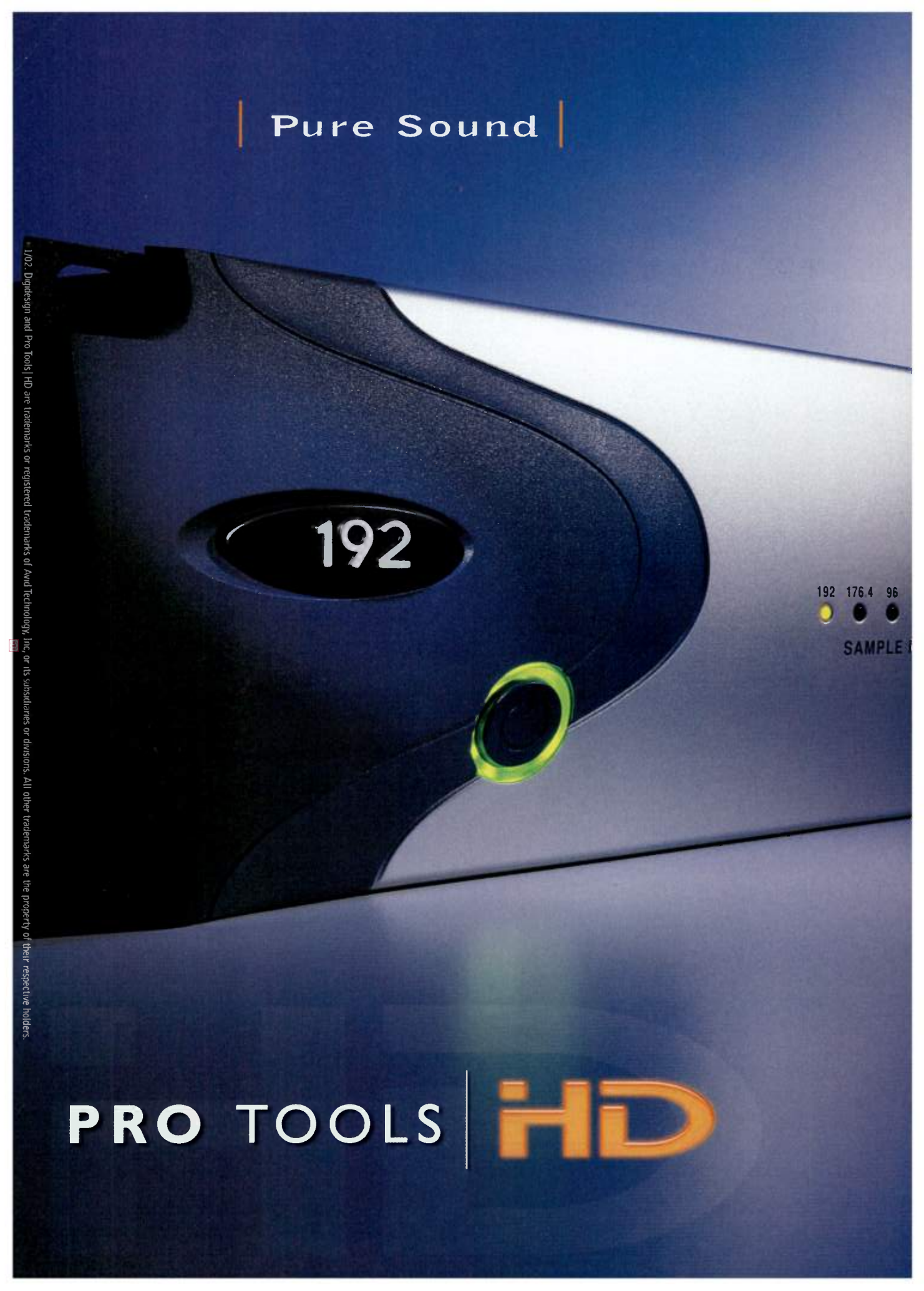

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DEPARTMENTS

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COLUMNS

REVIEWS

Contemplating a Rat's Rear


ANTHONY PIDGEON

When I was an undergraduate student, one of my professors had a saying that has helped me again and again over the years: Periodically, you have to step back from whatever you are doing and ask yourself, "Who gives a rat's ass?"

Pithy, yes, but words to live by. If no one except the creator cares about a project, there's little point in pursuing it for public release. Of course, one can still create art for art's sake or just for fun; I'm talking about projects meant for public consumption.

That applies to any sort of project, product, or service. For instance, I've directed **EM**'s editors to apply the "rat's ass" test to our articles: each story should tell you, directly or indirectly, why you might care about a particular product, technology, or applications tip.

We can apply the relevancy test to new technology, as well. The answers aren't always clear, but the question is always worth asking. For instance, in this issue's "Tech Page" column (see p. 30), Scott Wilkinson calls to our attention Calimetrics' new MultiLevel (ML) technology (commercialized by TDK), which potentially will triple the capacity of CD-Rs. Unfortunately, countless existing CD drives won't be able to play ML discs, though some CD-R/RW burners apparently could be upgraded to handle the new format. But DVD-R already offers better compatibility and larger capacity than a triple-capacity CD-R, so who gives a rodent's rump about ML CD-R? On the other hand, Calimetrics also intends to apply its technology to double the capacity of DVD-R, a format for which the user base is still small, so the compatibility issue is less troublesome. I find that remarkably rat-rear relevant.

Let's briefly apply the relevancy test to 192 kHz recording. Obviously, many great recordings have been made at 44.1 or 48 kHz. Let's even assume, for the moment, that going to 96 kHz is a wise decision (an assumption that is open to question). Maybe some people really need to go to 192, but I suspect the vast majority cannot tell the difference at that point and simply figure that more is better.

Maybe it isn't better right now, though. Sampling at higher rates proportionately increases the amount of data. The resulting need for more storage is just the tip of the rat's tail; processing and busing this large amount of data can put a serious hurt on your computer, even with a supplemental DSP card. Some products simply fail under those circumstances. So you have to decide whether going to 192 will make a practical difference in your work. If not, then we're just having a war of specifications, and we ought not give a rodent's rear about that.

We should ask the same question about 32-bit recording, about DVD-A, about surround mixes for music-only projects, and so on. But most of all, we should reflect on the relevance of our own projects because we, too, have product for sale.

Most often the answer will be that the project is relevant for some folks but not for others. Sometimes nobody gives a rodent's rump except those who market the product or service in question. Clearly, we will be successful only if our customers care about our work. So whenever you are considering a proposed project or purchase, take time to contemplate the timeless question of the rat's rear.

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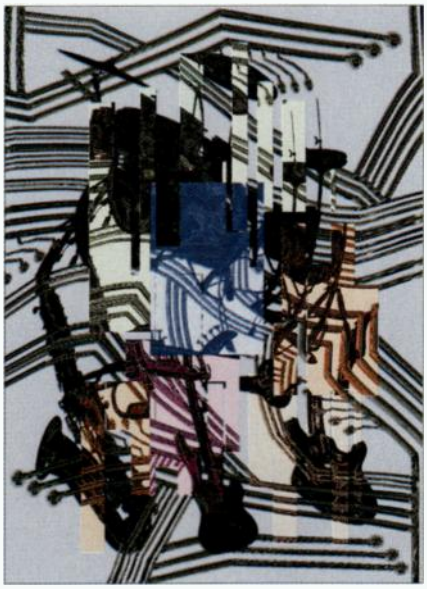

MAE LAROBIS

AN EARFUL ON EMBEDDING

I read David Battino's article ("More Than Meets the Ear," March 2002) and did not find the one piece of information I thought would have been the most obvious—how do I embed simple titles, tracks, and copyright information so that they appear in a computer media player such as Windows Media Player and CD Player? Did I miss something?

Randy Tobin
via e-mail

Author David Battino replies: Randy—Embedding CD-Text is simple, but because the information is written directly to the subcode area of the CD, both your CD-burning program and your CD-R drive must support CD-Text creation. On Windows, Roxio's Easy CD Creator and Ahead's Nero can do the trick; on the Mac, you can use Emagic's WaveBurner or Ahead's NeroMax. Note that Roxio Toast and Jam, the leading Mac burners, don't yet support CD-Text, which is why the process may not have seemed obvious. With the other programs, you simply select a menu item or click on a checkbox and enter the information you want to display. For links to those programs (and many more enhanced-CD resources), visit www.emusician.com.

Unfortunately, few CD players support CD-Text; in fact, the majority of software players actually grab the information from CDDB.com, not the disc itself. Uploading information about your own projects to CDDB is possible but needlessly clogs the database if you'll be distributing only a few hundred copies.

AND THE WINNERS ARE . . .

I have been subscribing to **EM** since its beginning. Most of it is good most of the time. In real life, everything cannot be a "10" all of the time, but you sure had a pair of winners in the March 2002 issue. Gary S. Hall's article "Burning Ambitions" explains everything you should know about CDs and CD burning. Hall is totally brilliant. I read all the replication publications in addition to other related publications, and it has taken me years to gather the knowledge that Hall condenses into one, albeit long, article. Anyone who runs a recording studio should have that article tattooed on his or her arm. It is truly fine writing and presentation.

The other article, "Desktop Musician: Windows Washing" by David Roach, was of the same quality. I've been in the business a long time and have gathered much of that same data, but I have never seen it all in one concise article. That's quite an achievement for **EM**.

I often pass my magazines on to my friends. Can I obtain those articles electronically or point people to the articles on your Web site without having to resort to photocopying? These articles should be saved on the computer hard drive of every electronic-music geek on the planet.

Clark Ferguson
via e-mail

*Clark—**EM** makes the most recent two years' worth of articles available online in the back-issue section of www.emusician.com. At this time, we can provide only the text, but we are considering ways to include the graphics.—Steve O*

LOST IN CYBERSPACE

I just finished reading the March 2002 issue and wanted you to know how valuable it is. I tried to provide my feedback through the magazine's Web site, but it appears to be in a state of disarray. I received script errors on the first page and found many links that seem to send me off into space.

However, the March issue's focus on CD construction ("Burning Ambitions") was well represented and appropriate to **EM**'s audience of technolust gearheads. Bottom line: Wow. Great issue, but the Web site needs work.

J. Perry Lamb
via e-mail

I'M WITH THE BAND

I enjoyed reading Marty Cutler's February 2002 article "Virtual Bandmates." I have been using Soundtrek's Jammer Professional and PG Music's Band-in-a-Box programs on my PC with a Yamaha CLP 560 MIDI keyboard since 1998. Band-in-a-Box has one of the easiest user interface windows. In the article, however, Cutler states that Band-in-a-Box has a significantly more complex interface than other programs.

My first reaction was astonishment. When I saw the Band-in-a-Box interface window in Fig. 2, I saw a window that I had never seen before. I wouldn't have recognized it except for the program logo at the top.

I agree that the interface window shown in the article is a complex layout of buttons, icons, and menus, but it doesn't look anything like the Band-in-a-Box versions that I have been using. I

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use version 11—what version did the author use?

Gerald Losser
Gloucester, Virginia

Gerald—The screen shot you refer to shows Band-in-a-Box version 8 for the Mac, which is the most recent version for that platform. However, I spent a good deal of time with version 10 for Windows. (Version 11 wasn't announced until the article was being proofed.)

I never stated that the user interface is difficult. I did say that it was more complex and that the array of buttons, menus, and icons were confounding at first glance. Part of that problem is redundancies between menu commands and onscreen buttons that take up screen real estate. I agree that the program is a breeze to use; that said, the user interface found in the most recent Windows upgrade doesn't appear to be any less complex than the versions I reviewed. —Marty Cutler

THE MEEK SHALL INHERIT

Barry Cleveland's article about Joe Meek ("Production Values: Meek First," February 2002) was one of the most enlightening articles I've read in a music magazine in a long time. It was great to learn about Meek's story—both the technical and personal info—and to discover that he was behind two songs from my musical childhood that left distinct impressions on me.

"Telstar" was a song I always remembered hearing but never knew anything about—not even its title. "Have I the Right" was another one of those songs I recall jumping right out of the radio, stark and clear.

I was amazed to find out that Joe Meek had a hand in this and was also so far ahead of his time as an innovator of music and technology. Why I haven't heard more about him, other than the Joemeek line of products, baffles me. Thanks for a great article. It's definitely a keeper.

Rob Chanter
via e-mail

Rob—You got that right! Until recently, Joe Meek was sadly underappreciated on this side of the Atlantic. In fact, the first time I heard his name, I thought the person was referring to Joe Meek, the famous mountain man who trapped furs with Kit Carson and Jim Bridger in the 1830s. I admit it was an ignorant mistake, but at least both Joe Meeks were extremely talented mavericks!—Steve O

HOLD ON A MINUTE!

In the "Chain, Chain, Chain" article in the February 2002 issue, the information regarding the typical configuration of TRS signal flow is incorrect. The majority of mixers and other pro-audio equipment typically send signals through the tip and receive signals through the ring of the patch cable.

Eric Hold
via e-mail

Eric—There is no standard or "typical" way to wire mixer inserts. Soundcraft, for instance, makes some consoles with TRS inserts wired tip = send and some that are wired ring = send.—Steve O

BLOW THAT HARP, SON

Thanks for finally having an article about recording harmonica ("Recording Musician: Taming of the Shrill," January 2002). I am a harmonica and keyboard player, and it was great to see the harmonica given the same treatment enjoyed by other instruments in your fine series. It's also great to know that author Brian Knave is a harmonica player.

Erik Eisen
via e-mail

WE MEAN IT, MAN

I have been a subscriber since October 1987. The magazine is really top-notch. Although at times it seems that much does not apply to me, I constantly find that it really does. I refresh my memory by rereading articles long after they appear. For example, the stuff about compression, sound cards, EQ, reverb—it is all so relevant. Unlike many other magazines, **EM** has never fallen into the trap of publishing senseless articles. Whereas other publications recycle boring and fictitious debates that carry on for months, **EM** takes a professional, sincere approach.

Cooke Harvey
via e-mail

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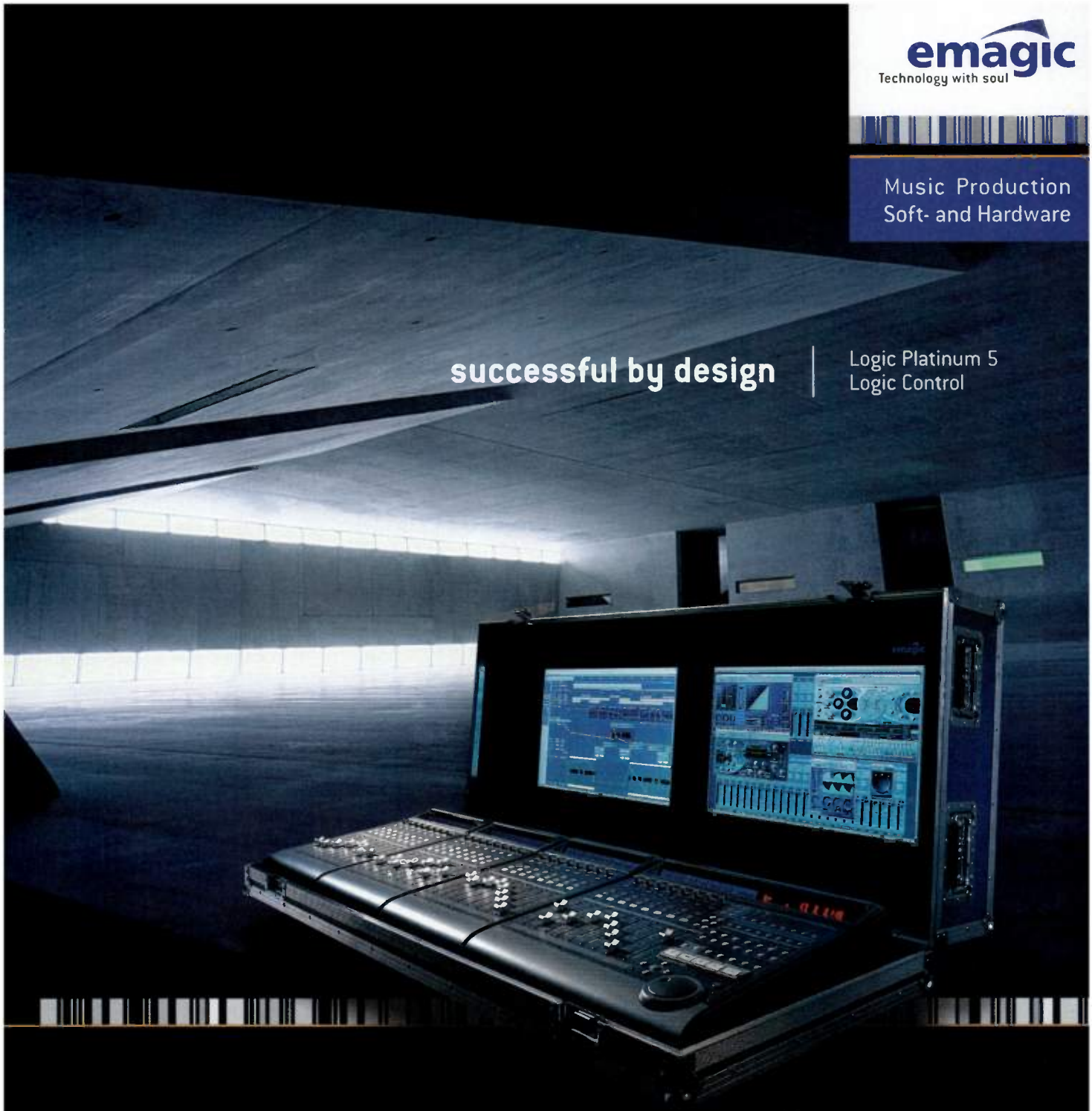

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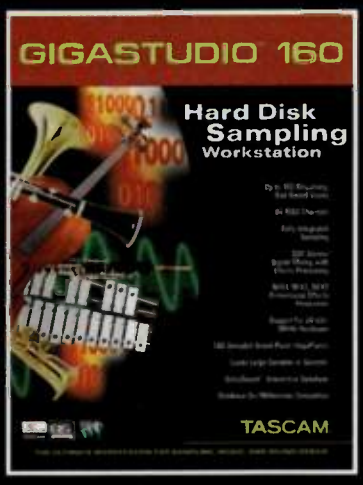
Technology with soul



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GIGASTUDIO 96

GIGASTUDIO 32

Four Reasons Why Fastest and

As the first software solution to use streaming technology for playing huge samples off your PC's hard drive, **TASCAM GigaStudio™** quickly became the world's leading sampler for professional composers and musicians. But today, there are a few more choices in samplers that stream audio. Why is Giga still considered the best sampling tool for power users? Here are four reasons to keep in mind:

1. Kernel-Level Functionality. Giga offers something that no plug-in sampler can provide: it operates at the kernel level of your PC, beneath the point where Windows® functions. This patented technology allows GigaStudio to offer the very best latency specs, even with huge amounts of polyphony.

2. Sample Buffering. No hard drive on the planet is fast enough to stream samples with the blazing speed needed for real-time performance. Giga pre-loads a tiny portion of your samples into RAM to accomplish this task. Plug-in samplers have to load a much larger section of the sample into RAM, which can greatly affect the number of instruments that can be used in one session.

3. Standalone Application. GigaStudio can be used on a PC by itself as a standalone software application, or can be used on the same machine as your PC-based sequencer. Plug-in samplers need to be accessed within a DAW application, which can severely limit their professional performance capabilities.

4. The Best Libraries = The Best Sound. Put all the technology aside for a moment, and all that really matters is the sound. Giga libraries were specifically designed for the Giga platform. They take advantage of Giga's Dimensions functions like key switching, crossfading, velocity control and more. Plus, their incredible detail and sonic quality is optimized only in GigaStudio...with nothing lost in the translation.

So if you're into the very best that sampling technology has to offer, get into Giga. Visit your TASCAM dealer or check it out online at www.tascam.com, because when it comes to sampling, bigger and faster is always better. Period.

Giga Sound Library Spotlight

TASCAM/Christian Lane Ultimate Marimba & Vibes

NEW! With four sets of mallets to choose from, this 2-disc library features a great-sounding Marimba One 5-octave instrument and a Musser set of vibes. Multiple velocities, rolls, muted strikes and lots of extras truly make this the ultimate!

TASCAM/Jim Corrigan Nashville High-Strung Guitars

One of the coolest, most playable acoustic guitar collections for Giga. Recorded with incredible quality, this totally authentic collection of up and down strums and dynamically playable single strings for solo parts represents the sound of Nashville at its finest.

TASCAM/Christian Lane Ultimate Timpani

NEW! A 6 CD library built from a set of Hinger Touch Tone timpani with calf heads. Five mallets, eight velocity levels, individual right and left hand samples and incredible detail give this library a 5/5 star rating in Electronic Musician magazine!

TASCAM/Peter Ewers Symphonic Organ

The entire, historic, grand La Madeleine, Paris cathedral and the Cavaillé-Coll organ for GigaStudio! For the first time ever in any sampled pipe organ, the original cathedral ambiance is included via release triggered samples.

TASCAM/Conexant GM150/GM500 General MIDI Kits

You've never heard General MIDI like this! Two different collections (150MB and 500MB) of multi-megabyte instruments, including a complete set of acoustic instruments and synthesizer textures.

TASCAM/Gary Garritan GigaHarp

The sound of angels! Every string of a Salvi Pedal Harp sampled in stereo representing seven pedals, four attacks/velocities per string, two harmonics per string, glissando, hand-dampening and muffling.

GigaStudio is the Biggest, Best Sampler...Period.

There are hundreds of sample libraries that have been developed specifically to take advantage of Giga's streaming technology. Here's a small selection of the best.

TASCAM/Scarbee J-Fingered Bass

NEW! 1046 samples are dedicated to each of the 3 pick-up settings, providing a total of 3138 samples (1.15 GB)! The musicality of this handmade Celinder J Update 4 is expressed in every hammer-on, pull-off, grace-note, staccato-release and slide. Amazing!

Bigga Giggas/ Bigga Orchestral Brass

4 CD set that includes 341MB of tuba, 441MB of trombone, 302MB of french horn and 348MB of trumpet. All samples were recorded with multiple velocities at 24-bit/96kHz resolution.

Q Up Arts/David Torn Pandora's Tool Box

70 tracks of highly unusual sounds, divided into textural, riffic and rhythmic loops. Great for adding edge and unique personality to music production, sound for picture and multimedia projects.

TASCAM/Larry Seyer Acoustic Bass

Over 500 MB in size, every note of every string sampled in stereo at 4 velocities with no loops. Features finger-damped staccato release resonance samples that will play on the note-up (release) and body resonance volume control, fast and slow up/down slides, riffs, special effects, and more.

Dan Dean/Solo Brass

A 10 CD set of incredible solo brass instruments, with up to 8 levels of dynamics per note. Includes varieties of trumpet, trombone, french horn, tuba, cimbasso, euphonium and more.

Sonic Implants/ Hammond B3 Collection

A gritty, powerful B3 in all its glory! 45 B3 sounds are included, all run through Leslies and played by one of the world's best organists. Includes complete drawbar settings.

TASCAM/Matt Ragan Maximum Strength Steel String Acoustic Guitar

The beautiful, clear tone of a massively multi-sampled Martin 000-16. More than 1,200 discreet samples are dedicated to the instrument, providing more than a gigabyte of incredible realism with hammer-ons, pull-offs, palm mutes, release-damps and more.

Garritan/Orchestral Strings

The biggest (and possibly best) Giga library available today... a 16 CD-ROM set of orchestral strings recorded at Lincoln Center, using the world's most renowned instruments performed by world-class virtuosos.

VR Sound/Hip Hop

Over 1000 kicks, snares, hats, 3D live loops, kits, funky wah guitar, big distorted bass and more. Make instant phatt grooves...just add Giga!

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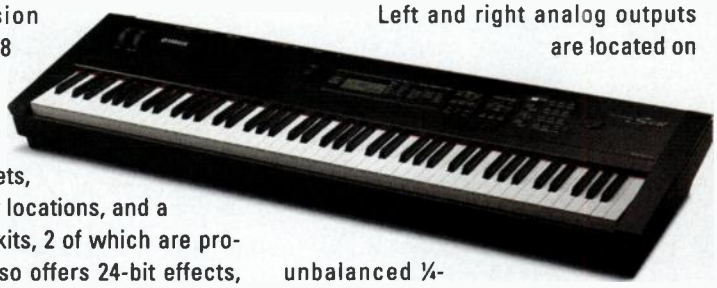
WHAT'S NEW

By Marty Cutler

YAMAHA S08

The S08 ($1,495) from Yamaha has a weighted, hammer-action keyboard and offers 64 notes of polyphony. The instrument holds 16 MB of sample ROM, which yields 25.2 MB when converted to 16-bit linear format. (Some of the samples use a proprietary data-compression scheme.) The S08 is 16-part multitimbral, and it offers 493 XG sounds, 128 presets, 128 user memory locations, and a total of 31 drum kits, 2 of which are programmable. It also offers 24-bit effects, including 17 reverb types, 17 chorus types, and 49 additional effects such as delay and distortion.



The S08 can load and play back type 0 Standard MIDI Files. A slot for SmartMedia cards lets you store as much as 128 MB of voice data or sequences. Yamaha bundles its SQ1 sequencing software and S08 Voice Editor software for editing and storing patches, along with Card Filer (Mac/Win) for porting data from your computer to the SmartMedia card.

A rear-panel USB port allows the S08 to send and retrieve MIDI data. MIDI In, Out, and Thru ports are included, as well. Left and right analog outputs are located on unbalanced ¼-inch jacks, and there are also ¼-inch TRS jacks so that you can hook up a footswitch and an expression pedal. Yamaha Corp. of America; tel. (714) 522-9011; e-mail info@yamaha.com; Web www.yamaha.com or www.yamahasynth.com.

STEINBERG WARP VST

Warp VST (Win; $299) is Steinberg's entry into the arena of guitar-amp-modeling plug-ins. The software offers three classic amp models and three virtual speaker cabinets. Latency is negligible when the plug-in is used with ASIO.

The plug-in delivers simulations of the Roland Jazz Chorus, the '60s Plexi Top, and rectified amps. Speaker cabinet simulation gives you an open-back combo amp with a 12-inch speaker; a British-style cabinet with four 12-inch '80s-era speakers; and a model of the Greenback cabinet with four 12-inch speakers. You can mix and match amps with cabinets.

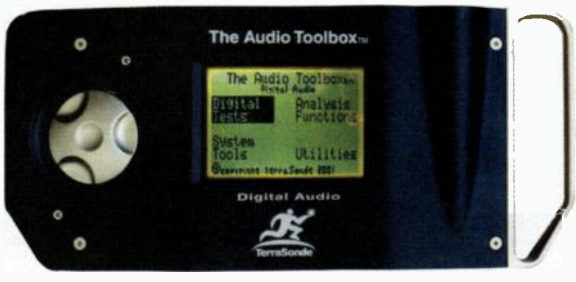
If you want to use Warp VST, you'll need a Pentium II/500 MHz computer with 256 MB of RAM, a VST 2.0 host program, and, of course, a low-impedance input device, such as an electric guitar. Steinberg North America; tel. (818) 678-5100; e-mail info@steinberg.net; Web www.us.steinberg.net or www.cubase.net.

TERRASONDE DIGITAL AUDIO TOOLBOX

The Digital Audio Toolbox ($2,499) from Terrasonde assembles a host of troubleshooting utilities for digital-audio systems in a handheld device. It offers dual inputs and supports 24-bit, 96 kHz audio.

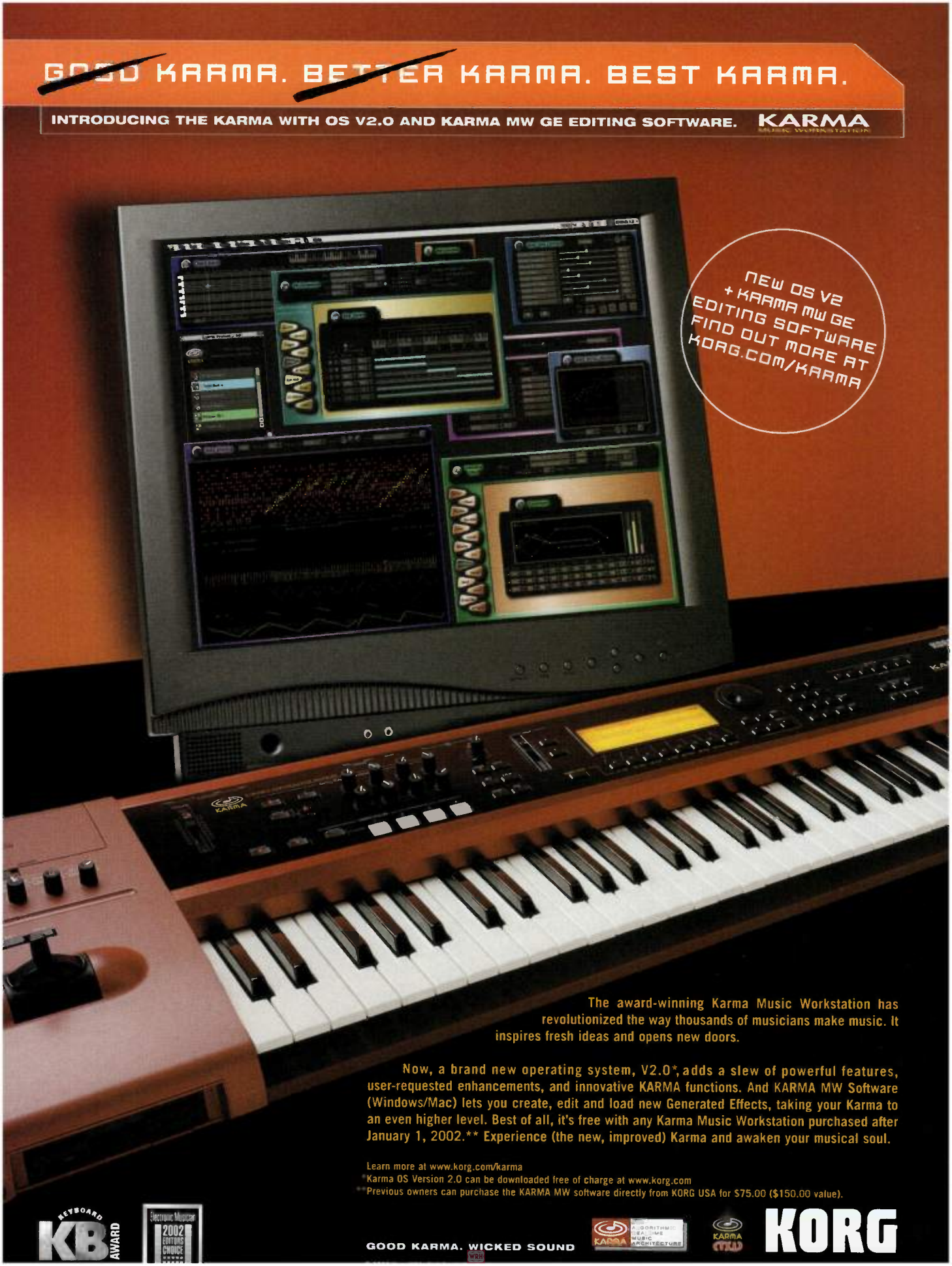
The Digital Audio Toolbox has four sets of digital-audio I/O connectors: AES/EBU, S/PDIF (optical and coaxial), and ADAT. Word-clock I/O and a serial port are also provided for sending data to computers. There are balanced ¼-inch and unbalanced RCA mono analog outputs for monitoring the digital input signal and test tone output. You can also monitor through the ¼-inch headphone jack or the built-in speaker.

More than 25 test, analysis, and utility functions are available in the Digital Audio Toolbox. The test functions include Transparency, which can detect modifications to the data stream; Lock, for testing phase lock between signals; a Jitter Meter that measures any jitter caused by the digital-audio interface; and Cable Quality, which checks for jitter induced by a faulty cable.



The Bitstream Analyzer displays bit depth and header information, and the Digital Watchdog detects errors. Tools include a sampling-rate converter and a Digital Generator, which provides a variety of digital-audio test signals. You also get a tool for measuring digital signal levels in a variety of formats.

So that you can monitor and generate machine-control signals, the Digital Audio Toolbox includes a Sony 9-pin jack. The unit also comes with a rechargeable battery system, serial computer interface cables, and a foam-lined hardshell case. TerraSonde; tel. (888) 433-2821 or (303) 545-5848; Web www.terrasonde.com.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SOUND ADVICE ▲▲▲▲

KID NEPRO

Kid Nepro offers two collections of programs and combinations for the Korg Karma workstation. Although the programs and combinations originated as Triton patches, sounds have been revamped to make use of the KARMA technology's Generated Effects and other real-time controls.

Soundtrack/Film Mix ($35) offers dramatic pads and effects that are geared toward film and TV scoring and feature complex rhythmic patterns and slowly evolving timbres. Classic Synths ($35) re-creates vintage analog synthesizers. The first 21 programs are monophonic bass and lead patches modeled after the Minimoog, the Sequential Circuits Prophet-5, and the Oberheim OB-8.

More than 100 programs consist of synth brass, pads, strings, hits, more bass and lead sounds, and grooves created by Generated Effects. Many grooves are patterned after the styles of Emerson, Lake, and Palmer; Kraftwerk; and other '60s and '70s progressive bands. Kid Nepro; tel. (246) 420-4504; e-mail info@kidnepro.com; Web www.kidnepro.com.

▲ BARDSTOWN AUDIO

Samples of the 9½-foot Bösendorfer Model 290 Imperial Grand piano in GigaStudio and Halion formats are offered on *Bösendorfer* ($199). Kip McGinnis sampled the instrument in a concert hall to give it a natural ambience.

The sound set provides samples with pedals up in four velocities, an equal number with pedals down, and additional release samples. You get about 1.5 GB of samples. According to the company, *Bösendorfer* will soon be available in EXS24 format as well. Bardstown Audio; tel. (502) 349-1589 or (800) 814-0820; e-mail info@bardstownaudio.com; Web www.bardstownaudio.com.

▲ ILIO

John Lehmkuhl (aka Skippy) produced and played on his new collection of drum loops, *Skippy's Noizbox* (Akai, E-mu, Giga, Roland, $199; CD-Audio, $99). The grooves offer a more aggressive sound than his previous effort, *Skippy's Big Bad Beats;* processed sounds are more prevalent on this collection, and the grooves are tailored toward European dance-music styles.

More than half of the collection exceeds the tempo of 120 bpm, but if you need to slow things down a bit, the Groove Control feature can change the tempo without changing pitch. You can also change the mix and layer or subtract groove elements in real time.

In addition to full mixes of the loops, the collection includes sample groove-construction kits. If you want, you can substitute sounds or change the pitch of any instrument.

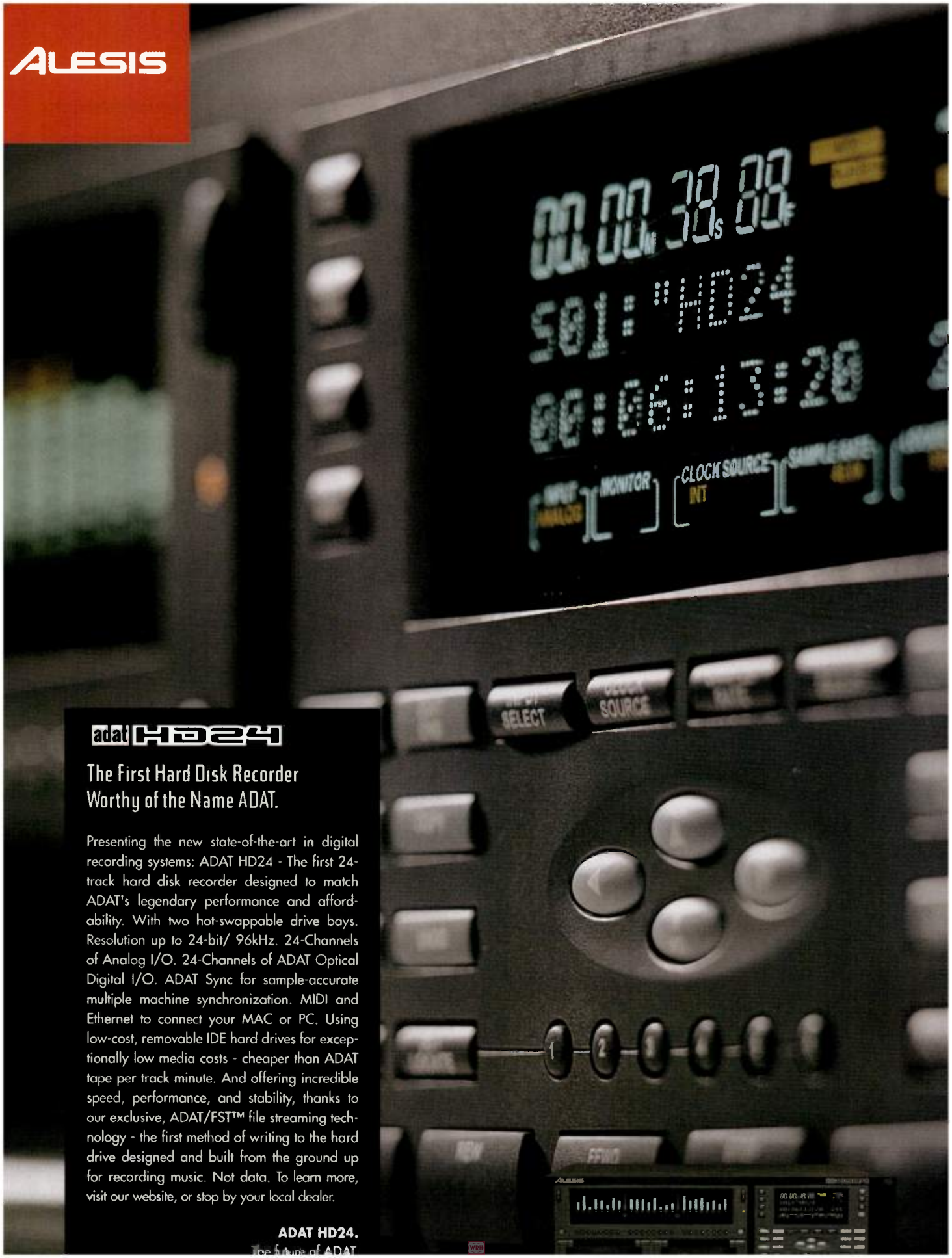
Aretha Franklin, Otis Redding, Steve Winwood, and Peter Gabriel have all called upon the talents of Wayne Jackson and Andrew Love, collectively known

as the Memphis Horns. Ilio offers *Memphis Horns* (CD-ROM for Akai, Giga, E-mu, Roland, $299; 2 CD-Audio discs, $149), a collection of sampled phrases from the duo. Phrases are played on trumpet and trombone as well as baritone and tenor saxes.

The collection includes hundreds of phrases in multiple keys and tempos. Phrases are offered in complete form and are broken up into shorter pieces. Phrase tempos range from 70 to 140 bpm, and each phrase is sampled in five harmonically related keys. Multisampled unison lines; major, minor, and major seventh chords; and characteristic brass effects—such as flutters, falls, tone clusters, and swells—are also included. Ilio Entertainments; tel. (800) 747-4546 or (818) 707-7222; e-mail ilioinfo@ilio.com; Web www.ilio.com.

PRO-REC

More than 300 Akai S1000 format instruments are offered on the *Synth-Sonic 2* ($149) CD-ROM. You get 500 MB of Korg Triton samples taken from Pro-Rec's original program libraries. The sounds are compatible with Kurzweil K2000, BitHeadz Unity, Steinberg Halion, and other samplers. *Synth-Sonic 2* includes pads and lead sounds, 42 synths, 75 basses, 26 keyboard instruments, 33 individual drum sounds, and 6 drum kits. Pro-Rec, Inc.; tel. (805) 705-3217; e-mail sales@pro-rec.com; Web www.pro-rec.com.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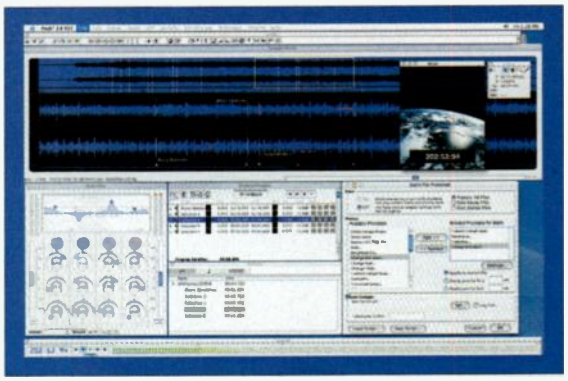
BIAS PEAK 3



Peak 3 (Mac; $499) is a fully native Mac OS X version of BIAS's digital-audio editor. It supports carbonized VST plug-ins and takes advantage of other Mac OS X features, including the Aqua user interface, protected memory, and CoreAudio support for multiclient hardware. However, Peak 3 will also run under operating systems as early as Mac OS 8.5.

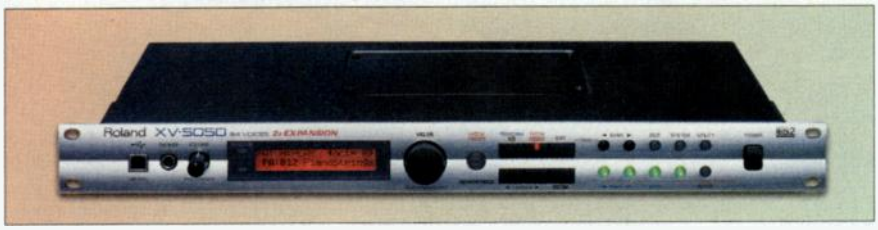
BIAS's Vbox plug-in-processing matrix has been integrated into Peak 3, allowing you to create complex multi-effects patches using VST plug-ins. Vbox is also integrated into the playlist, so you can control routing, level, mute, bypass, solo and effects presets for each playlist event. Other features include built-in high-quality MP3 encoding and dithering algorithms. User interface enhancements comprise a cursor-palette tempo calculator and an expandable Contents window.

Peak 3 offers unlimited Undo and Redo functions with a graphic edit history and automated Red Book CD burning. The package comes with more than 25 VST plug-ins, including BIAS's new Freq 4-band EQ. (Owners of Peak 3 can upgrade to the SuperFreq 10-band EQ plug-in for $49.)

Peak 3 requires a PPC G3/200 MHz Macintosh with 64 MB of RAM. With operating systems earlier than version X, users will need Apple Sound Manager 3.3 or later. BIAS (Berkeley Integrated Audio Software); tel. (800) 775-BIAS or (707) 782-1866; e-mail sales@bias-inc.com; Web www.bias-inc.com

ROLAND XV-5050

The XV-5050 synth ($995) is a 1U module from Roland that features the same synthesis engine as Roland's XV-5080. The XV-5050 gives you 64-note polyphony, combines four stereo oscillators for each program, and can accommodate two SRX-series expansion boards for additional waveforms and programs. The module also offers 11 filter types and two LFOs per patch.



The XV-5050 includes all of the XV-5080 presets, adds a batch from the Fantom workstation, and also offers a host of new presets. The module has five independent effects processors, including 24-bit reverb, chorus, delay, and guitar-amp modeling. You can also assign 3-band EQ to each output.

The front panel's USB port offers direct MIDI communication with computers and allows the module to serve as a 16-channel MIDI interface. Roland also bundles a CD-ROM (Mac/Win) with USB drivers.

On the rear panel, you get MIDI In, Out, and Thru ports. Analog outputs are a pair of unbalanced ¼-inch jacks for individual outputs and a second pair for a stereo mix. The synth features coaxial and optical S/PDIF digital output jacks. Roland Corp. U.S.; tel. (323) 890-3700; Web www.rolandus.com.

ENSONIQ HALO

Thirty-two MB of sample ROM and real-time controls lead the feature set for Ensoniq's 61-key Halo ($1,295). The 64-note polyphonic instrument includes three slots that can hold a maximum of 128 MB using E-mu Proteus ROM cards. The synth delivers its 16-bit, 44.1 kHz sound set through a built-in 24-bit dual-stereo effects processor and 24-bit D/A converters.

Halo holds 640 factory and 512 user presets along with 64 user-programmable multitimbral slots. A single patch can consist of four oscillators. Sound-shaping capabilities include more than 50 different digital filter types and more than 100 synthesis parameters. The dual-effects processor offers a host of parameters that sync to MIDI Clock.

Four real-time control knobs let you control synthesis parameters, and the top panel of the keyboard has an array of 16 buttons that can trigger MIDI sequences or any of the unit's programmable arpeggiators. According to the company, the instrument's MIDI response time is several times faster than that of the Proteus 2000.

Four unbalanced ¼-inch analog outputs are divided into two stereo pairs. You get a ¼-inch TRS stereo analog effects return. Quarter-inch TRS jacks accommodate a footswitch and an expression pedal, and MIDI In, Out, and Thru ports are provided. E-mu/Ensoniq; tel. (831) 438-1921; e-mail info@emu.com; Web www.emu.com.

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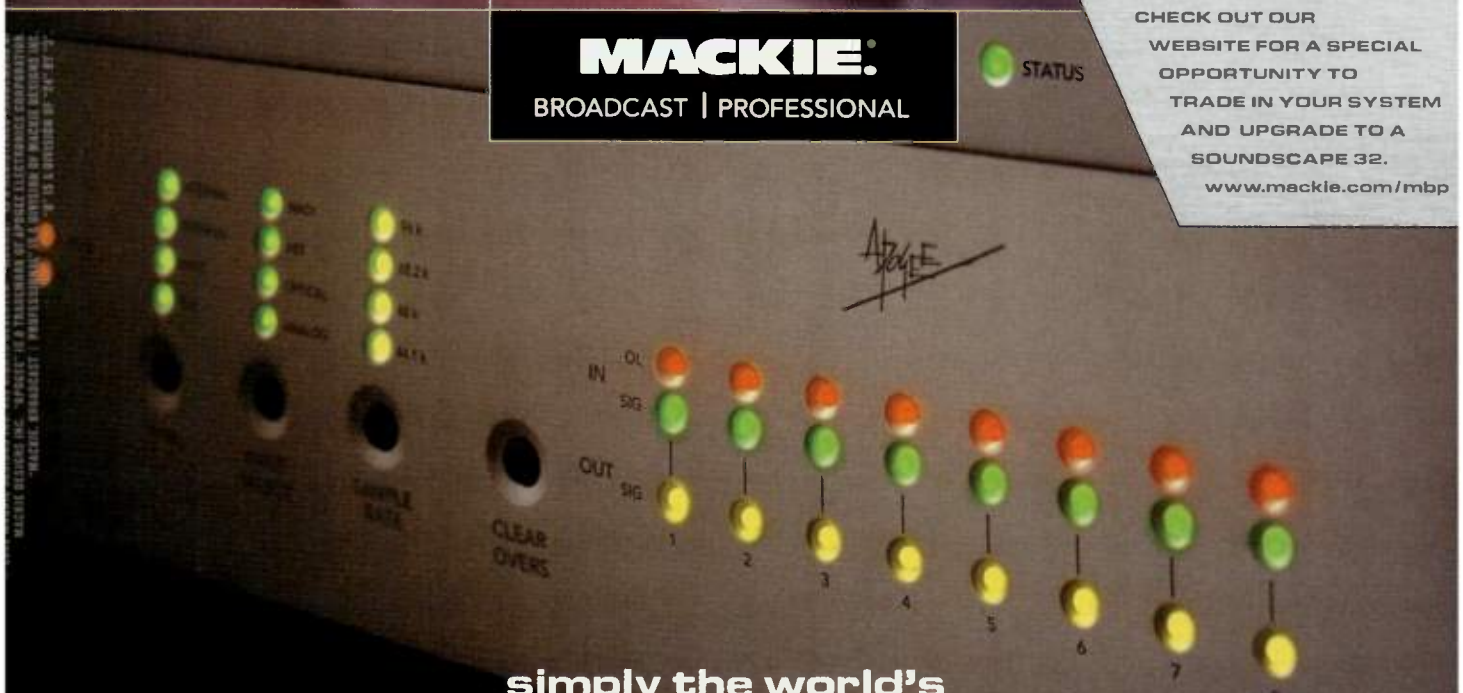

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BITHEADZ UNITY SESSION

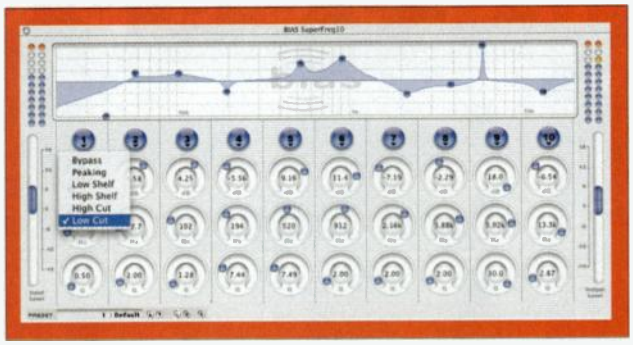
BitHeadz Unity Session (Mac; $649) has a new plug-in architecture that lets different synthesis and sampling types interact under one hood. Unity Session includes the AS-1 and the DS-1 software and adds physically modeled clarinet and flute, as well as hammered- and bowed-string, models. Oscillators from one synthesis type can interact with other oscillators in real time. For example, you can use a DS-1 sampled oscillator as a carrier or modulator for an oscillator with a Retro AS-1 waveform.

MIDI and audio effects processors in Unity Session appear as plug-ins. Audio effects processors include standards such as reverb, chorus, delay, and flange; an assortment of dynamics-processing tools; Degrade for bit reduction; and Pitch Bend.

The DS-1 can play samples loaded into RAM while simultaneously streaming larger samples from the computer's hard disk, and support for GigaSampler files has been added. The program can automatically map Akai S1000, DLS, SoundFont 2.0, and SampleCell II Instruments. The package ships with several gigabytes of samples and programs to get you started. Unity Session now provides an enhanced keyword-search function; you can quickly locate and load samples stored on networks and on removable media. Unity Session can automatically detect MOTU Audio System, VST, Real Time AudioSuite, or DirectConnect.

Unity Session requires at least 128 MB of RAM and a PPC G3/333 MHz computer. The program is optimized to run on Macintosh OS X, but you can use versions as old as OS 8.6. BitHeadz; tel. (401) 886-7045; e-mail info@bitheadz.com; Web www.bitheadz.com.

BIAS SUPERFREQ



SuperFreq ($79; upgrade from Freq, $49) is a VST plug-in for Macintosh. The software provides ten bands of parametric EQ and is driven by OS X Carbon events to work within any Carbonized VST host application. SuperFreq can also work under Mac OS versions 8.5 through 9.2.*x*, but the interface will have a brushed-metal look rather than the aqua look of the OS X version.

SuperFreq offers –18 to +18 dB gain values and a bandwidth Q that's adjustable from 0.1 to 30. You can sweep frequencies from 20 Hz to 20 kHz and bypass individual bands. On-screen up and down arrows allow you to shuttle through your library of presets. Clipping is indicated by two rows of virtual LEDs.

The filters include high and low shelving, peak, notch, high cut, and low cut. You can also launch SuperFreq as a 4-, 6-, or 8-band parametric EQ. BIAS (Berkeley Integrated Audio Software); tel. (800) 775-BIAS or (707) 782-1866; e-mail sales@bias-inc.com; Web www.bias-inc.com.

PRESONUS DIGIMAX LT

The PreSonus DigiMax LT ($999.95) is an 8-channel mic preamp featuring dual-servo preamps with 48V phantom power and a –20 dB pad. The unit offers ADAT Lightpipe digital output and handles sampling rates of 32, 44.1, or 48 kHz.

The front panel provides controls for gain, pad, and phantom power, and LEDs that indicate the presence of signal. The DigiMax LT sports Neutrik combo connectors for each channel, and you can select line or microphone level input.

The 1U device also includes ¼-inch TRS inserts for each channel and a pair of BNC connectors for word-clock In and Out. The DigiMax LT is housed in a steel chassis and has a built-in power supply. PreSonus Audio Electronic; tel. (800) 750-0323 or (225) 216-7887; e-mail presonus@presonus.com; Web www.presonus.com.

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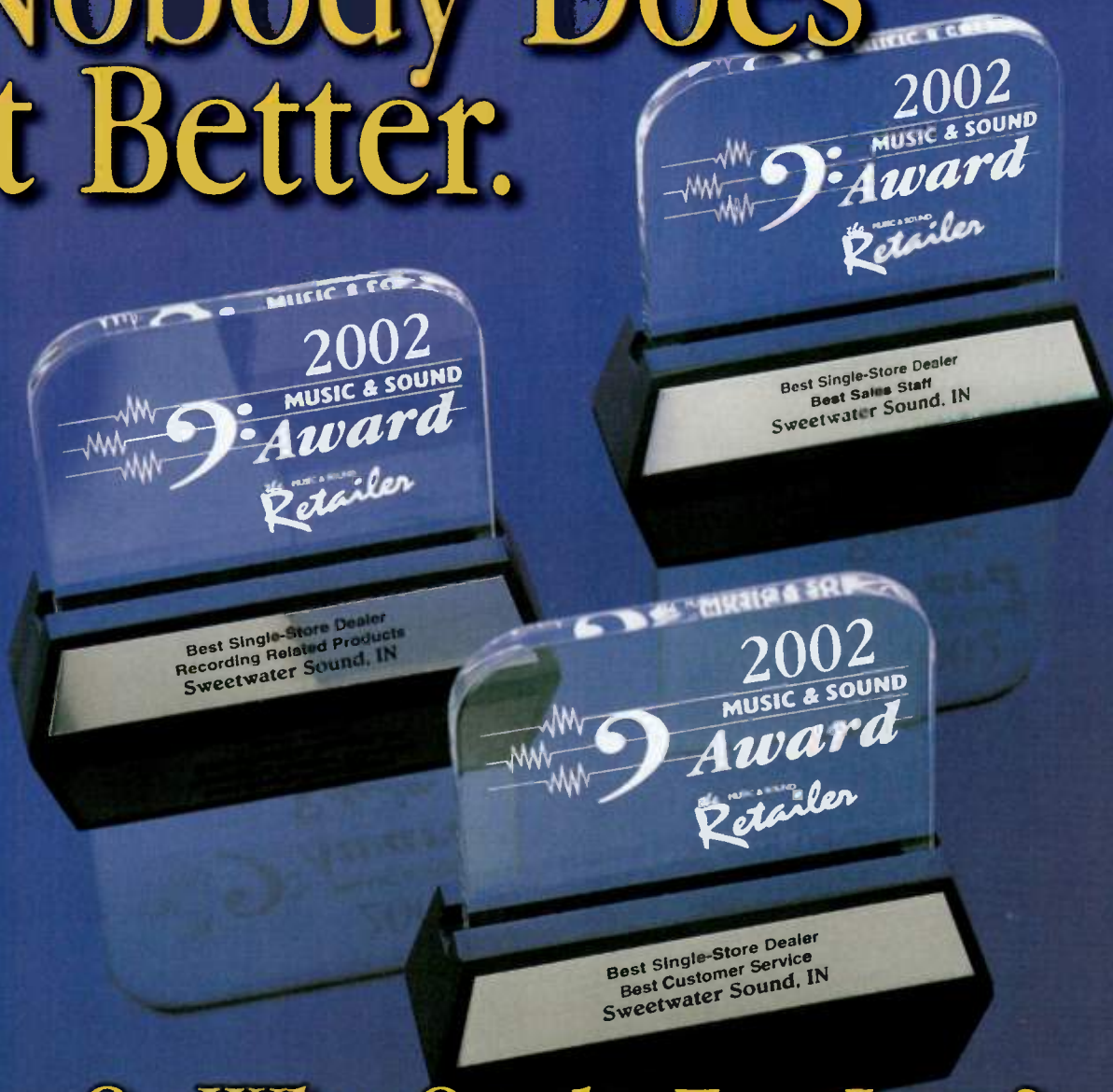

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KEY CHANGES

Midiman/M-Audio has announced that it will distribute a new line of Groove Tubes microphones. Along with several new models, the company will reissue the Groove Tubes MD1. . . . **DUY** has ported its TDM plug-ins to **Digidesign**'s new Pro Tools HD system. The plug-ins will now offer sampling rates as high as 192 kHz. . . . **Ableton** announced a free update of LIVE for Mac OS X. . . . **TC Group** has launched TC-Now .com, an online shop offering direct sale of hardware as well as a brand-new service that lets you rent algorithms on a per-project basis. . . . **Mackie** has announced a strategic partnership with **Universal Audio.** Mackie will be distributing Universal Audio's products, including the UAD-1 Powered Plug-Ins system; the UAD-1 card now supports Windows 2000 and XP. . . . **Emagic** has announced the availability of Windows XP–compatible drivers for its AMT8, Unitor 8 MkII, and MT4 Plus MIDI interfaces, and the EMI 2|6 multichannel audio interface. Download the drivers at www.emagic.de/german/support/download/toolswin.html. . . . **Access Music GmbH** has announced the Virus Indigo TDM, a software synthesizer that offers up to 160 voices and a 96 kHz sampling rate when used with Pro Tools HD systems. . . . The **ATI Group** will handle North American distribution of U.K.-based **Audient PLC**'s products, including mixing consoles, surround-sound processors, and graphic EQs. . . . **ArtistPro Publishing** will launch a new line of Pro Audio Press instructional titles on DVD for the professional and home recording musician. . . . **Propellerhead Software** is releasing Mac OS X versions of its entire product line.

APPLIED ACOUSTICS SYSTEMS LOUNGE LIZARD

Lounge Lizard (Mac/Win; $199) from Applied Acoustics Systems uses proprietary physical models to replicate the sounds of classic electric pianos. The manufacturers claim that the software replicates all of the mechanisms in the real instruments, including hammers, pickups, tone bars, and tines.

Lounge Lizard comes with a variety of presets for vintage electric pianos; in addition you can create sounds outside the realm of vintage instruments. Classic presets emulate several varieties of Rhodes, Wurlitzer, and RMI electric pianos. All controls respond to MIDI Control Change and other messages in real time. Onscreen knobs let you adjust parameters for mallet, fork, and pickup.

Built-in effects processors include wah-wah, a six-stage phaser, tremolo, and delay. You also get knobs for main volume as well as bass and treble adjustments. The software supports adjustable bit depths and sampling rates.

Lounge Lizard can be used as a standalone synthesizer, but it also supports DXi, VST, DirectConnect, MOTU Audio System (MAS), ASIO, EASI, DirectX, and WDM. Macintosh users will need a PPC G3/300 MHz computer with at least 32 MB of RAM. For Windows, you'll need a Pentium III/450 MHz processor with Windows 98SE, ME, 2000, or XP, and at least 32 MB of RAM. Applied Acoustics Systems; tel. (888) 441-8277 or (514) 871-4963; e-mail info@applied-acoustics.com; Web www.applied-acoustics.com.

ROLAND MMP-2

Roland's MMP-2 ($695) microphone preamp offers two channels of built-in microphone modeling; 4-band EQ with nine types of filters; modeled tube compression; an enhancer; a de-esser; and digital and analog I/O. A/D/A conversion is 24-bit with selectable sampling rates of 44.1, 48, 88.2, and 96 kHz.

The MMP-2's mic modeling feature is optimized for input from the AKG C 3000 B condenser mic, Roland's DR-20 dynamic mic, and four additional microphones. Mic models include a dynamic type suited for vocals, a large dynamic, and large and small diaphragm condensers.

The MMP-2 ships with a plug-in that models nine types of vintage mic preamps. The preamp uses Neutrik combo jacks for mic inputs, provides 48V Phantom power, and has –20 dB pads for each input. The rear panel sports a pair of XLR analog outputs. You get an AES/EBU and a coaxial S/PDIF output jack as well as S/PDIF In, so you can process digital audio through the MMP-2. A rear-panel switch lets you change the output level from –16 to + 4 dBu.

The USB port allows you to edit the MMP-2's EQ and compressor settings. Roland bundles editing software for Macintosh and Windows and promises that future updates, including new effects plug-ins, will be available for download. Roland Corp. U.S.; tel. (323) 890-3700; Web www.rolandus.com.

Looking For Some Serious Action?

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The S08 gives you professional quality sound and features at an affordable price, just $1199*. It has a ton of features you'd only expect to find on synthesizers costing twice as much. For starters, it's equipped with an 88-note weighted action keyboard that offers an incredibly realistic piano touch. In fact, it's the same Balanced Hammer action as our top-of-the-line MOTIF 8 synthesizer. What's more, many of its 749 on-board voices are based on our award-winning S80 synth. We've even added a complete XG and General MIDI Level 2 voice set to complement the built-in, direct from SmartMedia™, SMF playback sequencer.

If you're looking for the feel of a real piano and the sonic flexibility of a cutting edge synth, check out the S08 at a Yamaha dealer near you.

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TECH PAGE

By Scott Wilkinson

MultiLevel Marketing

Tripling a CD-R/RW's capacity is easier than you think.

When I attend the Consumer Electronics Show (CES) in Las Vegas each January, I don't expect to find new technology that is applicable to music production. This year, however, I was surprised to discover something that could become ubiquitous in standalone and computer-based digital audio workstations. TDK (www.tdk.com) was demonstrating a new recording technology that promises to triple the capacity of CD-R/RW discs—that's right, a standard-size CD-R or CD-RW that can hold as much as 2 GB.

This technology, called MultiLevel (ML), was developed by Calimetrics (www.calimetrics.com), a small Silicon Valley firm that has entered into a marketing partnership with TDK. The beauty of ML is its simplicity; a conventional CD-R/RW drive can be upgraded to ML capability with the addition of a single controller chip and some firmware modifications. The drive's optics and mechanics remain unaltered. ML does require special blank discs, which TDK calls ML-R and ML-RW. These discs will be available for about $2 and $3, respectively.

Instead of burning pits of different lengths into the disc's recording layer, ML defines "data cells" that are a uniform length of 600 nanometers (nm). By comparison, conventional CDs burn pits that range in size from 833 to 3,000 nm. The write laser burns a nonreflective circular spot into each cell, and the diameter of the spot determines the amount of light reflected from the cell as the data is read. The current implementation can accommodate eight spot sizes that result in eight levels of reflectivity. Thus, each data cell represents three bits ($2^3 = 8$) instead of one as it would in a conventional CD, tripling the capacity of the disc (see Fig. 1).

ML offers other advantages over conventional CD-R/RW. For example, the new format writes data sequentially rather than interleaving it, which avoids any buffer-underrun problems. That requires a different sort of error-correction code (ECC) to recover lost data during playback, and ML uses a Reed-Solomon product code that is similar to the ECC used on DVDs. In addition, ML does not require finalization; the table of contents (TOC) for each session is written separately, and ML players read those TOCs to find the data on the disc.

ML-R and ML-RW blanks use organic dye and phase-change alloy recording layers, respectively, much like CD-Rs and CD-RWs. However, the new materials are slightly different from their conventional counterparts because ML marks are smaller than the marks on a CD, so the recording layer must be "tuned" for that size range. Still, blanks can be manufactured with existing facilities, greatly reducing production costs.

In addition to standard-size ML blanks that measure 12 cm in diameter, TDK is making two smaller sizes: 8 cm, which holds 650 MB of data, and 6 cm, which holds 200 MB. At CES the company also announced a computer-based internal ML drive that can write data at more than 5 MB per second, which will fill a 2 GB disc in less than six minutes. The drive can also read and write normal CD-R/RW discs, but ML discs require an ML player.

Calimetrics is working to apply ML technology to recordable DVD; recent experiments increased the data density by a factor of 2.1 using 12 reflectivity levels. (DVDs record data and correct errors more efficiently than CDs, so the relationship between data density and number of levels is not equivalent.) The result is a capacity of 10 GB using current technology, and that could easily increase to 30 GB or more using blue lasers. In any event, ML is an exciting new technology that could have a major impact on optical media storage. ❁

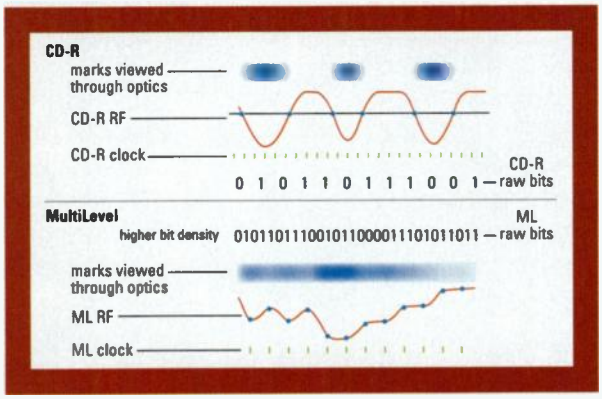

FIG. 1: ML burns spots of different diameters into each data cell rather than pits of distinct lengths, which yields eight levels of reflectivity, effectively tripling the storage capacity of a CD-R/RW.

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WEB PAGE

By Gino Robair

WEB SITE OF THE MONTH

From Demo to Deal is the first in a series of music-education programs by **InsideSessions** (www.insidesessions.com). The ten lessons, referred to as sessions, are in the form of interviews with industry professionals, including well-known individuals such as Sting, Elton John, and George Martin, and other highly influential industry types, such as Glen Ballard, Bob Clearmountain, Jimmy Iovine, and Tommy Mottola.

The program is designed to let students work at their own pace. When a person has finished the course, he or she will receive a Certificate of Completion. In addition, the A&R department of Universal Music Group (the parent company of InsideSessions) will review the student's demos. If the student opts for MusicSessions Plus package ($99.90), he or she will also receive written feedback from a Universal Music Group A&R exec.

The ten sessions are How to Get In, Writing the Songs, Making the Demo, How to Get a Label Deal, Making the Right Deal, Building a Team, Inside a Record Company, Inside the Recording Studio, Getting Your Music Out There, and The Future. Each session is divided into related subsections, and you can easily move between them using on-screen buttons.

Subsections in the Writing the Songs session, for example, include Inspiration (featuring Elton John and Enrique Iglesias) and Heavy on the Hooks (with Shaggy, Barry Gibb, and Nelly). Under Making the Demo is the topic How Raw Can It Be, featuring Sheryl Crow and Tony Brown. Even if you don't like the music or style of the artists being interviewed (the list of interviewees is extensive, covering every popular style), what they have to say is universal when it comes to succeeding in today's music industry.

From Demo to Deal requires that you use the CD-ROM in conjunction with the Internet, and a high-speed connection is recommended (though dial-up users can view the lessons from the disc). I viewed the lessons using a Mac G4 and a Power Macintosh 8600/300 MHz with a T1 connection, and surprisingly, the viewing experience from the Web was much better than directly from the disc in both cases.

The minimum requirements for using From Demo to Deal are Windows 98/ME/2000 and Microsoft Internet Explorer 4.0 on the PC, and Mac OS 9.0 and Netscape Communicator 4.5 on the Mac. Windows Media Player 7 and Macromedia Flash Player 5 are also required on both platforms, and all of the required applications are available on the CD-ROM.

DOTDOTDOT.COM

Whether you are looking for a musician to tour or an engineer for an upcoming studio session, check out **iMusicWorks** (http://imusicworks.com). The service is free, and it gives you a safe and secure place to find audio professionals for your project or to offer your services to others ....

Theremin enthusiasts have a number of quality resources on the Internet. Start out with a visit to **ThereminWorld** (www.thereminworld.com), where you can learn more about the instrument and network with other like-minded theremin fans. The site's Photo Gallery and RCA Victor Theremin Registry are personal favorites. From ThereminWorld, you can move quickly to other theremin-related sites using the Theremin Ring buttons at the bottom of the home page. Highlights of the Ring are **Theremin.info** (http://theremin.info), which includes scanned manuscripts of Percy Grainger's rare score for theremin, *Free Music No. 1;* and **Art's Theremin Page** (http://home.att.net/~theremin1), which offers articles about

building a theremin, schematics, and related technical advice. No theremin tour would be complete without a visit to the home page of the instrument's reigning virtuoso, **Lydia Kavina** (http://postman.ru/~tvox). And all levels of theremin player are welcome to subscribe to **Levnet** (www.korrnet.org/mailman/listinfo/levnet), the online discussion group that is named after the instrument's inventor, Lev Termen (aka Leon Theremin).

DOWNLOAD OF THE MONTH

If you are looking for an introduction to granular synthesis, you could hardly pick a better place to start than Marcel Wierckx's Real-Time Granular Synthesizer (RTGS) 2.1 (Mac; $45; www.wxs.nl/~menti049/software.html). RTGS puts the basic granular techniques at your fingertips.

RTGS was created in Cycling '74 MAX/MSP (which is not required in order to run RTGS) and comes with very clear HTML documentation and balloon-style onscreen help. The demo version is fully functional except for audio recording, which is limited to one minute.

Granular synthesis involves breaking a sound file or live-audio stream into small fragments called *grains,* and then sequencing the grains. RTGS gives you independent control of the pitch of the individual grains, the size of the grains, and the time between grains. If the sound file is buffered in RAM, you can also control the speed and direction of the sound-file scan that is used to create the grains. RTGS will buffer sound files as long as eight seconds (by simply loading the first eight seconds of any AIFF or SDII file). It also works with live audio, which can be streamed from disk or input through your sound card.

RTGS's Main control panel is where you choose the source for granulation and set the basic granulation parameters. All the parameters except grain density (for example, the time between grains) offer real-time randomization. In a nice twist, transposition (for example, grain pitch) can be microtonal or quantized to semitones and restricted to a user-defined scale. When buffered input is used, you can automate the Scrub control at speeds ranging from half speed to ten times. That allows you to create interesting time-stretching and time-compression effects without changing the pitch of the individual grains.

One of RTGS's more unusual features is its binaural processing of RAM-buffered sound files. With RAM-buffered files, you always start with two mono sound files. RTGS alternates grains from the two files but applies the same parameters to each. (You can work with stereo by first splitting the stereo files.) When the two files are unrelated, the results are unusual, intermingled effects. The MP3 example Worry illustrates several granulations of a pair of spoken-word files. RTGS's minimum requirements are a Macintosh G3/233 MHz, 64 MB of RAM, OS 8.1, and, if you plan to use MIDI controllers, Open Music System (OMS) 2.3.8.

—Len Sasso

WEB APP

The eXtensible Music Format (XMF) is a new standard designed to facilitate the delivery of a wide range of audio-file types to various software applications and handheld devices. Published by the MIDI Manufacturer's Association (MMA; www.midi.org/xmf/xmf_faq.htm), the XMF specification outlines a

highly flexible file format that Internet developers and cell-phone manufacturers can use to combine MIDI, WAV, DLS, and MP3 data into one tidy package. The standard is open and free, and it can be incorporated into operating systems and multimedia products without license or patent fees.

An XMF file is made up of two basic parts: a hierarchical container, or *metafile*, and the audio content. Actually, it's very much like the files-and-folders system that you find on your Mac or PC. The container is defined as a series of nodes; each node can contain a specific file (such as MIDI data) or another node (such as a set of DLS instruments used by the MIDI file). A node can even contain a URL link to a file on a Web server, allowing an XMF file to play audio data that continually changes, such as a weather report.

Metadata that contains information about the audio file's content is also stored. This can assume the form of a standard Resource ID (MIDI, DLS, and so on), copyright notices, or customized data such as composer's notes or musical-style indicators. XMF files are geared toward low-bandwidth applications; therefore, the specification also includes optional compression and encryption algorithms for fast, secure downloads.

Beatnik (www.beatnik.com) has already launched a new version of its Audio Engine, which supports XMF. The mobileBAE, as it is called, is optimized for delivery of multimedia messages and ringtones to third generation (3G) cell-phone technologies. It uses a Type 1 XMF content format that combines standard MIDI files (SMF) with DLS instruments and is the functional equivalent of an open-standard, nonproprietary version of a Rich Music Format (RMF) file.

But XMF files are highly expandable and could also function as, say, a header file containing all the sounds effects and digital music files for a game level. At the 2001 Fatman's Bar-B-Q (a yearly interactive audio conference held in Texas), Chris Grigg, chair of the MMA's XMF Working Group, dreamt up an overview of what an interactive audio development environment might look like, using the XMF spec as a starting point (www.projectbarbq.com/bbq01/bbq01r5.htm). However, though the MMA has published example code for parsing XMF data, there are no publicly available tools for actually creating XMF files. That may change if the standard is adopted by a burgeoning mobile-Internet industry.

—Peter Drescher

BAND ON THE WEB

Few musicians go to Mark Trayle's level of musical and technological extremes (http://shoko.calarts.edu/~met). Whether it involves installing a tiny video camera inside a flute or placing a photoresistor in a tin can, Trayle is not content with simply putting musical notes on paper: he insists on designing his pieces from the ground up.

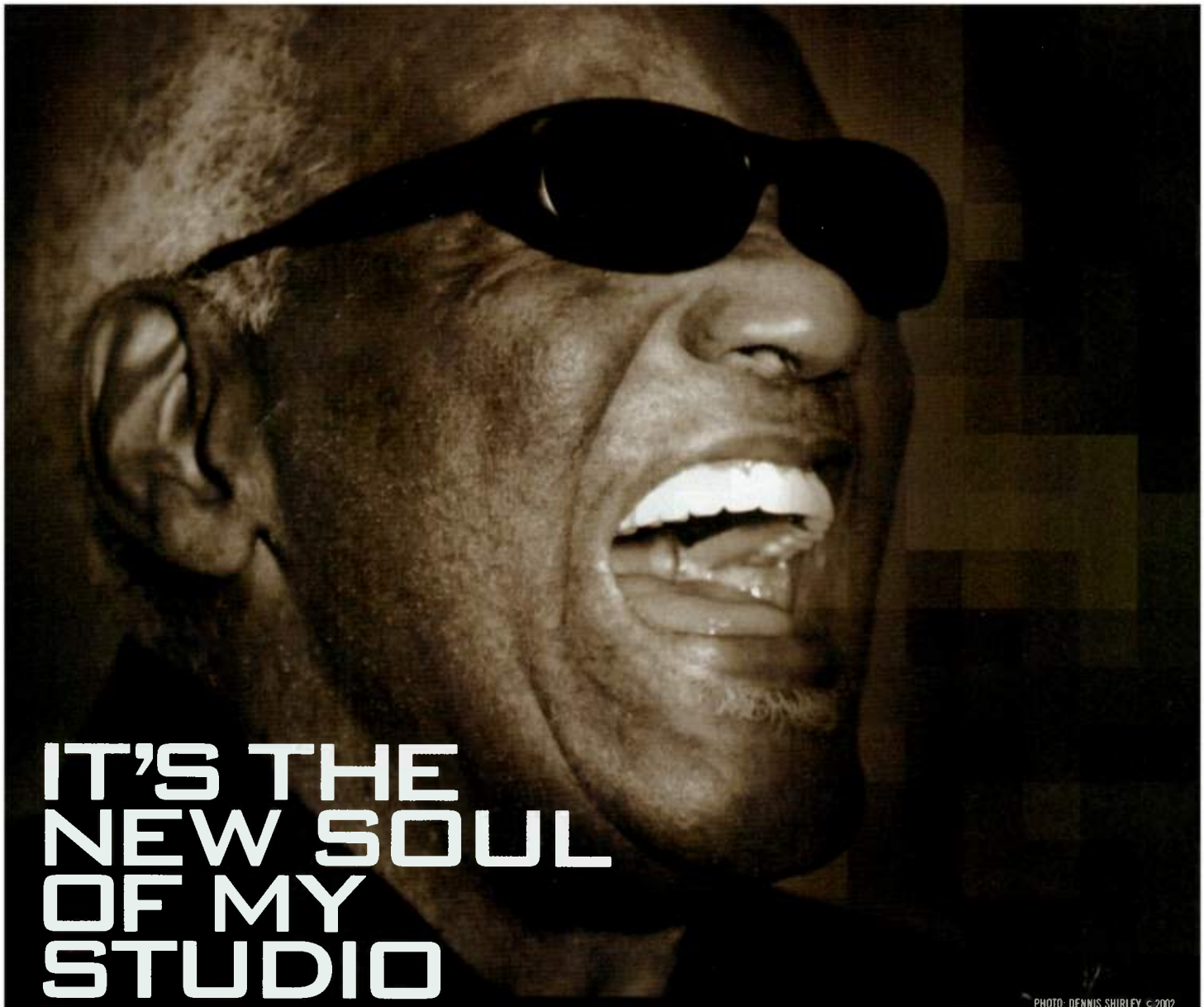
For example, on his most recent release, *RPM::MHZ* (Artifact Recordings, 2001; www.artifact.com), Trayle built a credit-card reader, for the piece "capital magnetic," that plays microcompositions based on the information culled from the magnetic stripe on a performer's credit card.

"Each time a card is swiped," Trayle says, "the contents of its magnetic stripe are captured and parsed to form the melodic, rhythmic, and timbral motifs of a short musical composition. Using genetic algorithms, compositions compete in a simulated music marketplace. Some become dominant, others less 'popular.' Some combine to form new 'styles' or 'genres' that in turn influence the more popular ones, and so on."

For the piece "Arcana 33 1/3" on *RPM::MHZ*, Trayle transformed an old gramophone turntable into an interface for a computer and used a tin can, with built-in photoresistor inside, as a MIDI controller. The final work on the disc, "Ciprocal," is a bubbling, rhythmic work created as Trayle navigates through a highly complex "web" of digital circuits of his own creation.

Trayle is also an agile improviser. On his recent release with multi-instrumentalist Vinny Golia, *Music for Electronics & Woodwinds* (Nine Winds, 2001; http://members.aol.com/ninewinds), Trayle used the software synthesizer SuperCollider (www.audiosynth.com) running on a Mac Powerbook to both manipulate Golia's sounds (particularly on "lazy Third Eye," "Cheapman and the sweater he cherished," and "Behind the Fifty-five dollar face") and generate new material. The result is a remarkably fresh electroacoustic interaction.

We welcome your feedback. E-mail us at emeditorial@primediabusiness.com.



IT'S THE NEW SOUL OF MY STUDIO

PHOTO: DENNIS SHIRLEY ©2002

> RAY CHARLES

When recording legend Ray Charles and Grammy-nominated engineer Terry Howard (Barbara Streisand, Stevie Wonder, Fleetwood Mac) decided to upgrade Ray's recording studio, they chose SONAR as their central production technology. Ray and Terry could have had any software they wanted. And they both agreed that SONAR could do it all.

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Peering Into Cyberspace

Peer-to-peer networks can help your music reach new listeners.

By Todd Souvignier

Peer-to-peer file sharing has become a controversial subject. Napster and its successors have fostered a new type of digital community, which is built around the sharing and swapping of digital assets. The very existence of this frontier zone—much of it beyond the easy reach of legislation, litigation, or digital-rights management—has sparked an ongoing battle over the control of digital media. ● It is just plain wrong to steal somebody else's possessions. It is every bit as wrong to redistribute copies of people's work—such as music, video, or software—without their consent. **EM** serves content creators and copyright owners; we always support our readers, our advertisers, and the companies we cover as they attempt to protect (and profit from) their hard-earned intellectual property. ● At the same time, it is easy to see substantial legitimate and beneficial uses for peer-to-peer file sharing. Some software programmers write freeware or shareware programs that are meant to be given away and spread around. Likewise, many musicians actively choose to share their music, posting free MP3 files on Web sites to promote their acts and garner new fans. High-traffic file-sharing networks have created a big opportunity for creators to get their work out into the world.

Illustration by Dmitry Panich
Images of astronauts courtesy of NASA

Peering Into Cyberspace

This article is about maximizing that opportunity. I'll discuss general ways of increasing upload traffic (such as the number of files you are distributing) that are applicable to any peer-to-peer network and then look at some specific techniques for working with the most heavily traveled systems. But first, why would you want to share files?

TAKING IT TO THE STREETS

Taking your music to where the people are is a timeless strategy. Performers have traditionally set up in the town square or other public spaces and performed to the passing crowds. These days you see buskers in the subway or at the local bookstore.

The largest Internet music crowds are in the file-sharing space, looking for songs. Most are just music fans and computer enthusiasts who live off the land, so to speak. But although many people who swap files just want a free ride, there are still opportunities for copyright owners to make legitimate use of these public file-sharing venues and to enforce their copyrights.

It boils down to this: unknown artists can easily and routinely distribute as many as 100 songs per day through file-sharing networks. Ask yourself: are you getting those kinds of hits on your band's Web site or MP3.com page? If you handle it right, this exposure can pay off directly in additional Web-site traffic, press reviews, gigs, and CD sales.

BEING A GOOD HOST

Each computer in a peer-to-peer network can be a client and a server. Users designate one or more shared folders, which contain files other users can access. Peer-to-peer software handles the file-serving tasks and provides a search-engine function that lets users find and download files from other users' computers.

There are some simple things you can do when running a server to increase network exposure and get more traffic.

Have a fast, current computer. Macintosh users should be on a G3 or G4; Windows users want a Pentium III or 4 (or comparable) processor. RAM and hard disks are cheap, so bulk up! Relics belong in museums; you need a real computer.

Get broadband. Transferring a typical three-minute MP3 file can take as long as half an hour over a 56 kbps modem. With a cable modem or DSL, it's just a couple of minutes. Nobody wants to download from servers that are on dial-up connections; people gravitate toward hosts that are on cable, DSL, or T1 lines.

Leave the computer on. Stay logged in. People won't find your files if the computer is turned off, if you have disconnected from the Internet, or if a file-sharing program is not running.

Maintain a reliable, stable system. If the computer crashes, nobody can get at your files. Perform maintenance and file-repair routines on a regular basis, remove unneeded programs, and make sure you're properly backed up so you can reinstall the operating system and get a fresh start when things get shaky.

One network at a time. You'll probably just crash if you try to run more than one peer-to-peer file-sharing program at any given time. Rotate between several popular networks for optimum reach.

THE NAME GAME

In a simple peer-to-peer search engine, the user sends a query by typing a word or phrase into the search tool and the results, called *query hits*, are returned to the user. This list of results shows all of the available files on the network that include the word or phrase in the file name.

Song files typically have names that contain the name of the artist, song title, and album title; for example:

artist name_album title_song title.mp3

If acts are not well known, chances are slim that someone will search for them by "artist name." If they don't already have a hit record, it is unlikely

DATA ABOUT METADATA

Compressed audio files can include *metadata*. Metadata means "data about data" and refers to optional text fields included in a file header. These text fields can contain the song title, album title, artist name, copyright information, detailed track descriptions, and even URLs.

The metadata in MP3 files is called *ID3 Tags*. There are two types of ID3 Tag: ID3v1 and ID3v2 (see **Fig. A**). ID3v2 allows far more description, including composer information and URL. Most MP3 encoders let you populate these fields with custom information of your choosing. It's somewhat time-consuming—you need to tag your files by hand—but fortunately, the data is retained when the file is copied.

The more advanced peer-to-peer systems search through metadata as well as file names. Diligently filling out tags is as important as naming your files correctly.

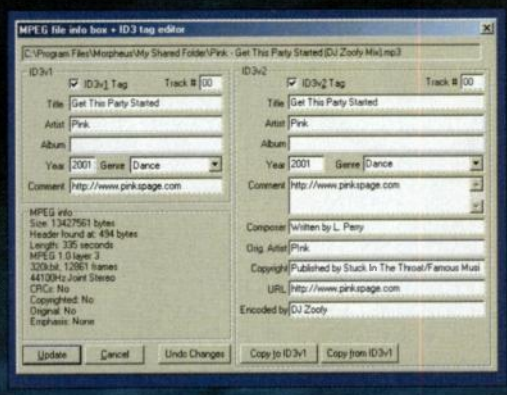

FIG. A: This figure, from Winamp's Tag editing dialog, shows ID3v1 on the left and ID3v2 on the right. Note that the ID3v2 tag offers additional documentation.

TWO HEADS ARE BETTER THAN ONE!

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that anyone is looking for their song or album title. So where does that leave you?

People who have done Web design are probably familiar with *keywords*, which are words or phrases that are included in a page for the purpose of getting more search-engine hits. I'll apply the keyword idea to song files, for the purpose of hitting more queries.

EVERYBODY IS A STAR

If you, any member of your group, or even your sideman have played with well-known acts, you can harness that star power. Make a copy of your audio file and rename it something like:

band name_song title_featuring famous player_former group name.mp3

That's a long name for a sound file, but Windows will allow it. Mac users must resort to placing long descriptive titles in ID3v2 tags or creating folders with those names, which is something Windows users should also do. (See the sidebar "Using Folders.")

Along the same lines, maybe you or your group is produced, engineered, or otherwise assisted by someone with worthwhile credits. If so, work that connection! Make another copy of your song, and give it a name such as:

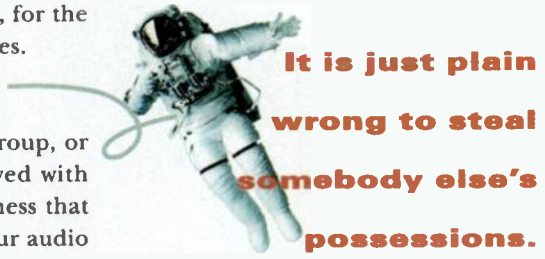
band name_song title_recorded by_also recorded group one_also recorded group two.mp3

Now your song files will appear in searches for the artists, groups, or individuals that you are associated with. Many variations on this affiliate-naming scheme can be worked, such as "on tour with . . . ," " protégé of . . . ," and "former roadies for . . ."

To take things a step further, try *affinity* naming. You can have file names that include your city, state, or region. You can also include musical genre or style in the file name. You may even make mention of unrelated but possibly similar-sounding acts. For example:

if you like the beach boys, jefferson airplane, mothers of invention, captain beefheart, or wild man fischer_download band name_song title.mp3

You've spammed into the query space of prominent, albeit unrelated, artists. Hopefully, your act sounds something like the bands you're piggybacking.



Another approach is to create file names that make use of common search terms; for example:

download band name_song title_new_anime_windows xp_simpsons.mp3

I've seen a lot of content online with this type of spam-packed name. Check out www.gnutellameter.com for a current list of the most popular query terms on the Gnutella network. When labeling files, label the Desktop folders and metadata Tags (see the sidebar "Data About Metadata.")

PEER SOFT TWEAKS

Running a reliable server and thoroughly naming and tagging your shared files will increase your uploads into any peer network. What's more, each file-sharing program has settings you can fine-tune to hot-rod its performance.

FastTrack. Created by a team of Europeans, the FastTrack network is better known by the names of its client applications: Morpheus, Kazaa, and Grokster. Boasting an installed base of more than 60 million copies (according to figures at CNET's www.download.com), FastTrack is the most popular file-sharing network, eclipsing Napster's one-time reported peak of 50 million units. (As with all "installed-units" figures, the actual number of users is more like half or a quarter of the number reported, as most users wind up installing the program multiple times.)

Morpheus, Kazaa, and Grokster are virtually identical programs, and all participate on the same network. Choose any one of the three and stick with it; you don't need to install or run multiple FastTrack programs on the same machine. I'll just refer to Morpheus in this article for the sake of convenience, but the following information applies equally to Kazaa and Grokster.

FastTrack uses the concept of *SuperNodes*, a way of indexing the network in a distributed fashion. The FastTrack software looks for users with the best setups and appoints those machines to create lists of content found on computers nearby. Users can opt in or out of becoming SuperNodes.

For optimum hosting, select Options in Morpheus's Tools menu. In the Options dialog, click on the Traffic tab. The Bandwidth settings are there—set

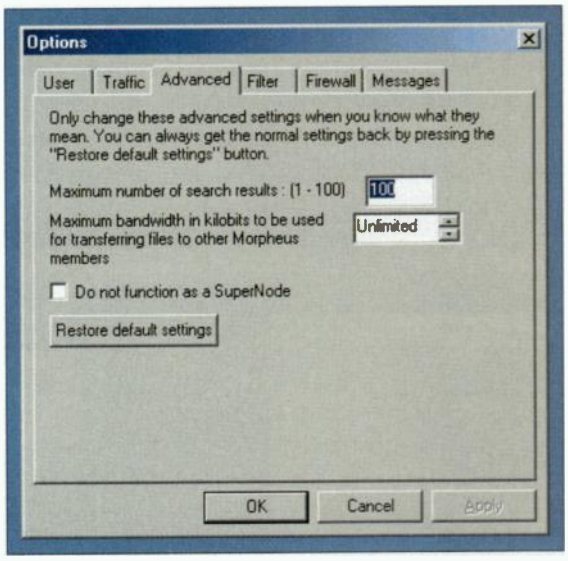


FIG. 1: Use the Advanced Options tab in Morpheus to allocate upload bandwidth.

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1 x WORD CLOCK I/O (BNC)
1 x ANALOG 24/96 LINE/HEADPHONE OUTPUT
1 x MIDI I/O, 16 CHANNELS HIGH-SPEED MIDI

ALL PORTS SIMULTANEOUSLY AVAILABLE:
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the Maximum Number of Simultaneous Uploads to Unlimited.

At the bottom of the Traffic tab is a checkbox called Disable Sharing of Files with Other Morpheus Members; make sure it is not selected because you do intend to share files.

Next, under the Options dialog's Advanced tab (see **Fig. 1**), set the Maximum Bandwidth to Unlimited; that keeps the upload throttle wide open when transferring files to other Morpheus members.

Finally, under the Tools menu, select Find Media to Share. That opens the File Import dialog. Click on the Folder List tab and navigate through your file structure to select the music folders you intend to share.

Audiogalaxy. Created by a group in Texas, Audiogalaxy is arguably the No. 2–ranked file-sharing network, with 28 million copies installed. Audiogalaxy is a music service—you don't find software or videos here.

A small client application (known as a *file agent*), called the Audiogalaxy Satellite, is downloaded and run on members' computers. A somewhat cumbersome Web interface (viewed through Netscape Navigator or Microsoft Internet Explorer) provides search and community features, such as member bios, message boards, and member playlists. For a peer-to-peer network, Audiogalaxy is fairly centralized. Like Napster, there is a central server that indexes users' collections as well as the audio files at Audiogalaxy's own Web site.

Audiogalaxy Satellite must be running for file sharing to occur from your computer. To specify which folders are to be shared, select Edit Shares from the Audiogalaxy Satellite File menu and then add the desired folders.

Next, go into the Options menu and select Bandwidth Throttle. Move the slider bar all the way to the right to select More Bandwidth (But Faster). Note that downloading from Audiogalaxy is limited to one file at a time, unless you have agreed to share 25 or more files.

Audiogalaxy has a hosting service that provides space on its own server. It allows musicians to continuously share their authorized files, even when their individual computers are offline. At one time, there was a Web-based interface, and musicians could easily upload songs to the server, quickly establishing an official presence on Audiogalaxy.

Unfortunately, the upload mechanism has been eliminated; a company representative says it was too much work and expense to maintain and suggests that musicians simply host from their own computers. However, bands can still apply for Audiogalaxy hosting in a retro fashion: send a CD to an address in Texas and then wait some number of weeks or months for the music to be approved, digitized, and uploaded—not exactly a step in the right direction.

Gnutella. First devised by rogue elements within America Online (AOL), Gnutella was promptly reverse engineered and improved upon by an array of participants. Although the original Gnutella program is no longer in general circulation, its descendants use the network protocol that continues to bear its name. The most popular Gnutella network clients are BearShare and LimeWire, with 10 million and 7 million installed units, respectively.

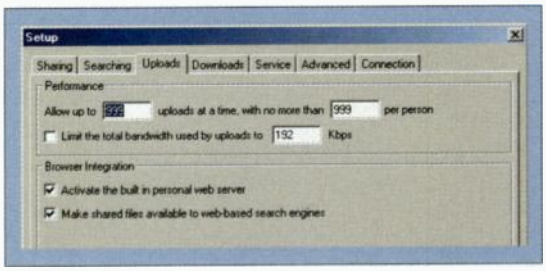

FIG. 2: In BearShare's Uploads Setup tab, be sure to deselect Limit the Total Bandwidth.

The Gnutella network has no central server, and it establishes a distributed, branching structure. Any single computer is visible to the five or ten computers to which it is directly connected, the next five or ten computers each of those machines are connected to, and so on, up to a limited number of *hops* (a hop is a direct host-to-host connection).

Although a quarter-million computers may be running Gnutella-compatible applications at any time, each machine can reach only roughly 2,000 to 10,000 other computers at a given time.

The Gnutella protocol has a feature called Time to Live (TTL) that determines the number of hops each query (or query hit) can propagate along the network. If queries had unlimited TTL, they would recirculate endlessly, and the network could quickly gridlock. The TTL settings directly affect the length of your computer's reach within the network. Increasing TTL lets more computers find your content and allows you to search further downstream.

BearShare offers a great deal of customization, and although there are many controls, they're a little spread out and there is no documentation for some of those controls are not documented. BearShare spokespeople

USING FOLDERS

Some file-sharing applications look at the names of shared folders, as well as the files contained within, and will return query hit results that include the folder names. If your shared folders are called Music or My Shared Files, they aren't helping you market and distribute your work. Certain peer-to-peer networks, such as Audiogalaxy, expect subfolders to be labeled with artist or band names. Greg Bildson, CTO and COO at LimeWire, suggests creating a folder called NewArtist in your shared directory. He says, "This will allow general searches for new-artist material."

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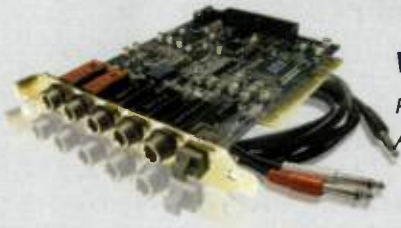

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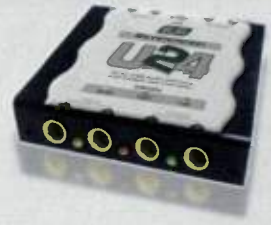

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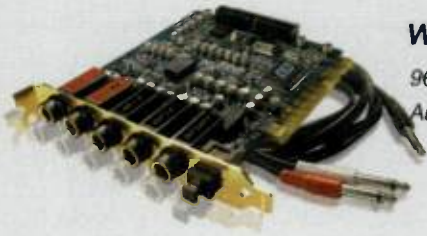

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approved the following recommendations, yet they repeatedly refused to explain their TTL implementation:

Open the BearShare Setup dialog and select the Sharing tab. Deselect and disable the parameter called Limit the Number of Results to ___ Results per Search. Next, for Ignore Searches with Less than ___ Characters, set the value to 1.

Click on the Uploads tab (see **Fig. 2**) and find Allow up to ___ Uploads at a Time, with No More than ___ per Person. Input the maximum value 999 in both fields. Then, deselect (uncheck) Limit the Total Bandwidth Used by Uploads and select (check) Make Shared Files Available to Web-Based Search Engines. Still in the Setup dialog, go to the Service tab and check on Always Accept Incoming Service Connections.

In the Advanced tab set the Message TTL, Agent TTL, and Max Hops to ten. FreePeers, the maker of BearShare, recommends not altering the TTL settings, but I say go for it.

Under the Connection tab, select the appropriate connection speed from the Network Interface section. Now you can close the Setup dialog and go to the Hosts panel. In Hosts, make sure that Automatic is checked on and select appropriate Minimum/Maximum Hosts values. Modem users should try 5/5; DSL users can crank it up to 25/25 before running into bandwidth issues. Values as high as 999 are allowed, but even settings well under 100 can leave you in an unstable condition, possibly resulting in the crashing or freezing of your computer. Finally, go to the Uploads panel and make sure that Share Files is checked on.

LimeWire is noteworthy for its short "warm-up" period and Mac OS version. A company official says Gnutella is working on a distributed local-indexing feature, similar to FastTrack's SuperNodes, that would be the first of its kind in the Gnutella network. The feature is called Ultrapeers and should be online by the time you read this.

LimeWire hides its TTL settings but has other useful controls and should be configured thusly: in the Monitor panel, check on Incoming Searches to enable people to find your content. In the Connections panel, input the highest allowable value in the Keep Approximately ___ Connections Up data-entry field.

Then, under the Tools menu, select the Options Dialog. Under the Uploads section (see **Fig. 3**), enter the highest allowable value, 999999, for Max Upload Slots. Place the Upload Bandwidth at 100 percent.

Finally, under the Options Dialog's Advanced section (in the Preferencing area), find Files You Must Share to Not Be a Freeloader. Set that to zero, because you want to let everyone get your files, including freeloaders. Under the Allow Freeloaders option, move the slider to Always.

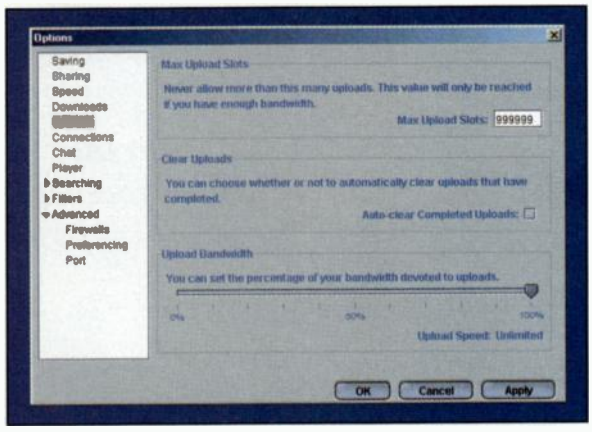

FIG. 3: Upload Slots and Bandwidth should be set to the maximum in LimeWire's Uploads Options.

IMesh. Out of Israel comes iMesh, which claims it has 19 million installed units and 7 million "registered" users. IMesh is the least stable of the prominent file-sharing systems. Moreover, version 2.20 could not be uninstalled from the computers that I tested it with. I alerted iMesh to the problems, but the company did not respond.

IMesh has a simplified set of controls. From the Preferences menu, open the Options dialog and select the Upload panel. Set Maximum concurrent uploads to 99, the highest allowable value. Next, in the Share section, make sure Allow Other Users to View All My Shared Files is checked on. After that, go to the Connection display to select your modem type.

DON'T FORGET THE URL

Here are two final suggestions applicable to any file-sharing network as well as conventional Web-site downloading or streaming:

The Windows Media Encoder and the WinRip Studio MP3 encoder allow you to add URL scripts to audio files. URL scripts cause Web pages to open automatically whenever a file is played (in a supported player). The use of URL scripts can directly convert shared files into additional Web traffic each time that the file is played. (Windows Media Encoder URL scripts are covered in detail in "Desktop Musician: Power Windows," in the February 2002 issue.)

Also, always make sure to include your Web site URL somewhere in the metadata when you're tagging songs: ID3v2 includes a dedicated URL field; in ID3v1 you can use the Comment field. It won't launch a pop-up, but it could help somebody find you. After all, helping people find you and your music is what it's all about.

Todd Souvignier *is a cofounder of Exploit Systems, Inc. (www.exploitsystems.com) and wrote the second edition of* Musician's Guide to the Internet, *forthcoming from Hal Leonard Corp.*

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Sequencing with Style

Compared with live recordings, MIDI sequences often lack warmth and expressiveness. That is in part because many electronic musicians regard MIDI sequencers merely as recording devices with a few tools for correcting notes, timing, and basic dynamics. I can't count the number of times I've seen MIDI musicians play the notes, tighten up the parts, and simply move on to the next track.

In fact, MIDI recording offers a deep synergy between the sequencer and the inner workings of the synthesizer's sound-shaping capabilities. The ability to change virtually any aspect of a performance at any phase of the creative process is an immensely powerful creative tool.

Expressive sequencing is achieved using three main elements: the synth architecture, the sequencer, and the controller. All too often, articles about MIDI sequencing focus on just one of those aspects. For a truly animated musical performance, it's vital to consider all three components as a whole.

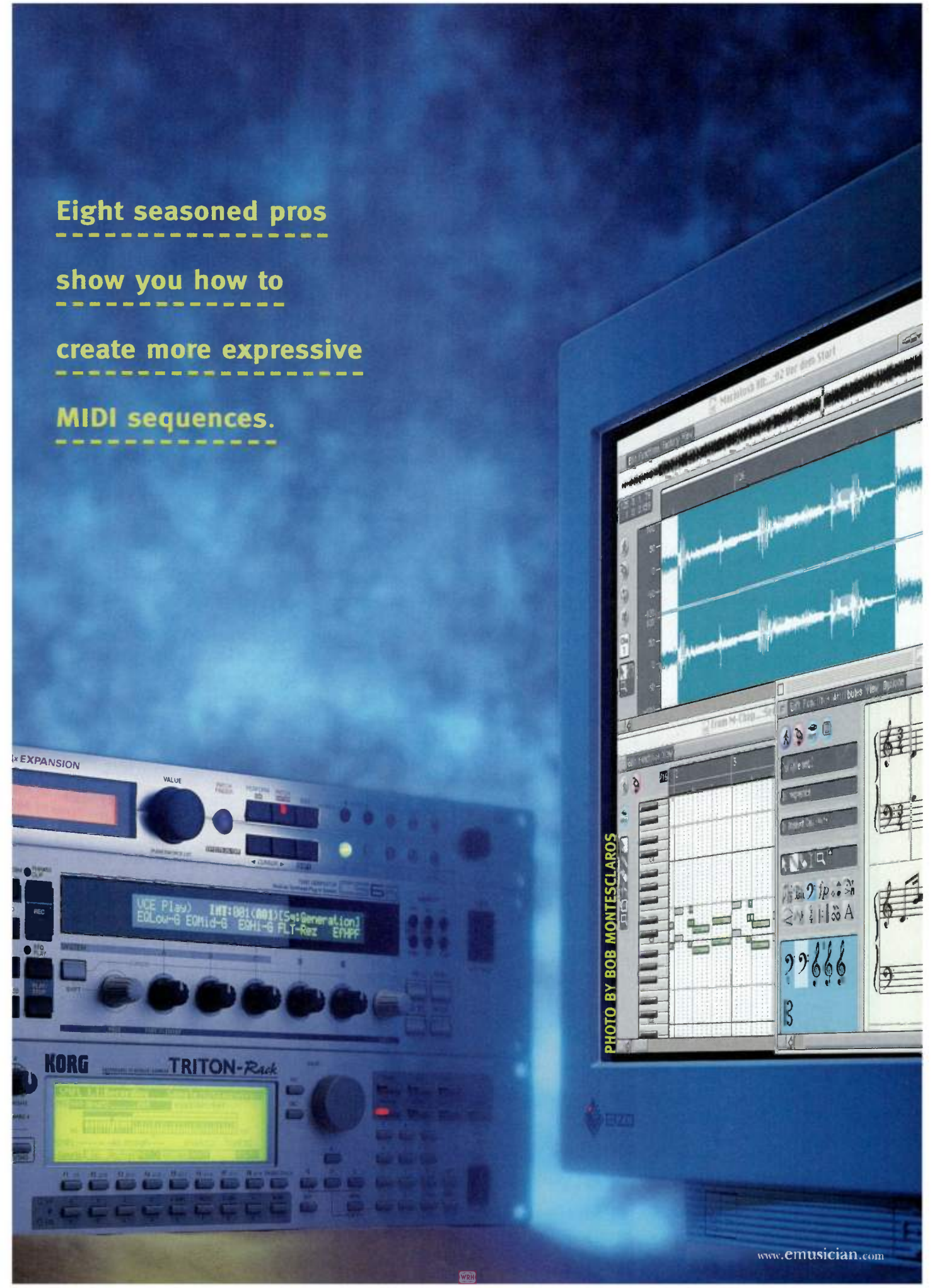
By Marty Cutler

To that end, I enlisted the help of artists whose work reveals a deep understanding of MIDI and sound design coupled with stylistic know-how (see the sidebar "Getting to Know You" for a bio of each contributor). The result is a wide-ranging pool of ideas from the standpoints of synthesizer programming, sequencing, and control options.

INSIDE YOUR SYNTH

Whether the synthesizer is a sample-playback unit, a physical-modeling synth, or something else, it has many features that are common to all synths, including envelopes, low-frequency oscillators (LFOs), and other modulation capabilities. Those features primarily control timbre, loudness, and pitch. Modulation features such as LFOs and envelope generators (EGs) can run free, but your best option for lively, nonrepetitive sequencing is to bring those capabilities under real-time control.

Eight seasoned pros show you how to create more expressive MIDI sequences.



Sequencing with Style

For example, LFOs are great candidates for modulation with Aftertouch. You can supplant the periodic effect of LFOs with a more humanized effect by controlling their depth or speed in real time. Many late-model synths offer knobs, sliders, and other controls that govern a variety of modulation features. Those controls often transmit Control Change (CC) messages instead of less efficient, bandwidth-consuming System Exclusive (SysEx) messages. If your synth offers such controls, you can capture and manipulate them in your sequencer.

ACOUSTIC BASIS

Once you've grasped the capabilities of your synth's sound-shaping tools, what do you want to do with them? Whether his synth sounds are emulative or not, Rob Mounsey looks for elements that evoke acoustic instruments. "I always try to make sounds that suggest that they could be some sort of real instrument that you haven't run into," he says. "I try to create the illusion that you've found an unusual instrument that people haven't heard yet—one that could actually happen in an acoustic space with acoustic materials. The way to get there is to carefully analyze acoustic instruments that you like to hear."

Lyle Mays also finds inspiration in the behavior of acoustic instruments. One of his signature sounds is a swooping, ocarina-like synth patch. Mays explains the acoustic orientation of that sound: "It reflects the way pitch responds when a string is plucked; the harder you pluck it, the more out of tune it is at first before it settles. The other acoustic principle is the way ensembles, especially young children, start things out of tune and then gradually end up more in tune. I was thinking specifically of a grade-school choir of ocarinas, and the pitch attacks are just all over the place. The kids are listening, so they eventually get closer in tune with each other.

"That's an oversimplified version of what I'm talking about. It's much subtler in the synth sound, but one of the oscillators does start sharp and then comes down in pitch, and the other one hits the pitch. There's pitch information on every attack." Routing Velocity to control oscillator pitch adds a bit more acoustic behavior in that acoustic instruments, particularly plucked strings, stretch and go further out of tune the harder they are hit.

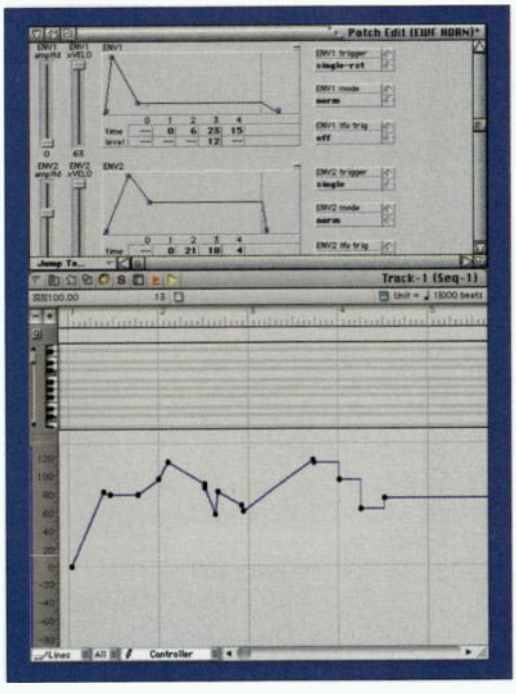
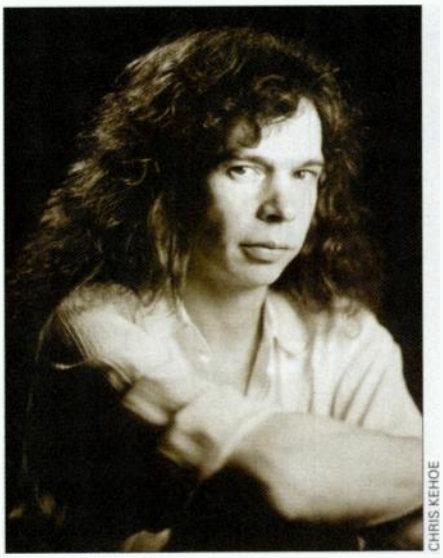


FIG. 1: If you contrast a typical synthesizer envelope generator with a hand-drawn envelope in your sequencer, you'll see that your sequencer offers much more subtlety and detail.

QUESTION REALITY

Sampled instruments often provide a superficial realism, but sustained listening can be boring. The static nature of samples often works against a natural feel and sound, but using sampled instruments doesn't have to be a sonic dead end. If you understand your synth's architecture reasonably well, you can find ways to imbue samples with new life and realism.


CHRIS KEHOE

Lyle Mays says he believes that it's important for synthesizers to behave like acoustic instruments. A grade-school ocarina choir inspired one of his best-known patches.

Sometimes all that's missing are the imperfections that occur naturally in acoustic sounds. George Whitty enhances the realism of his sounds by using waveforms from unrelated instruments. "I used to create my Hammond sounds by putting the [frequency modulation (FM)] part of a Yamaha SY99 through a SansAmp to dirty it up, but that messed around with the bottom end too much," he says. "The most suitable thing to create Leslie grit is a high-passed alto-saxophone wave. The gritty grunge of a real Hammond through a Leslie cabinet creates an aggregate effect that's not just a bunch of sine waves added up, but a kind of dirty, tubey thing. In trying to simulate that dirt, the high end of the saxophone samples works great; I filter out most everything below. I can make a sampled string section play more expressively by assigning a bit of bandpassed distorted guitar to the expression pedal to add some bite as things get more intense."

OUT OF RANGE

Occasionally, the right sound exists in the outer regions of a wholly unrelated

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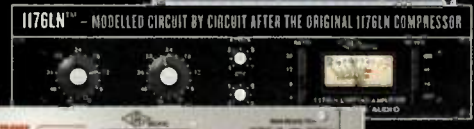

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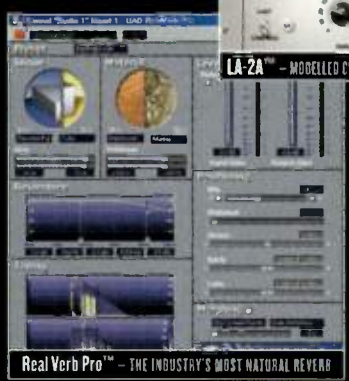

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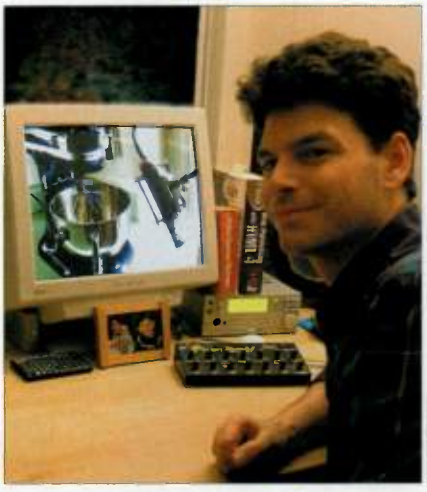
instrument. Jimi Tunnell carefully tests his sounds outside as well as within the usual playing range suggested by a patch. He finds that the categories suggested by preset titles can often lead you to overlook material that's viable for completely different applications.

"Don't look at the name of the sound," Tunnell says. "Just because a patch is named 'Flaming Gibbons' doesn't mean its only possible use is to imply monkeys on fire. Forget the names and listen first to the general shape and timbre of the sound."

I have a background in bluegrass and country music, and I've often sought the perfect pedal-steel-guitar sound.



OTTO TEIMER

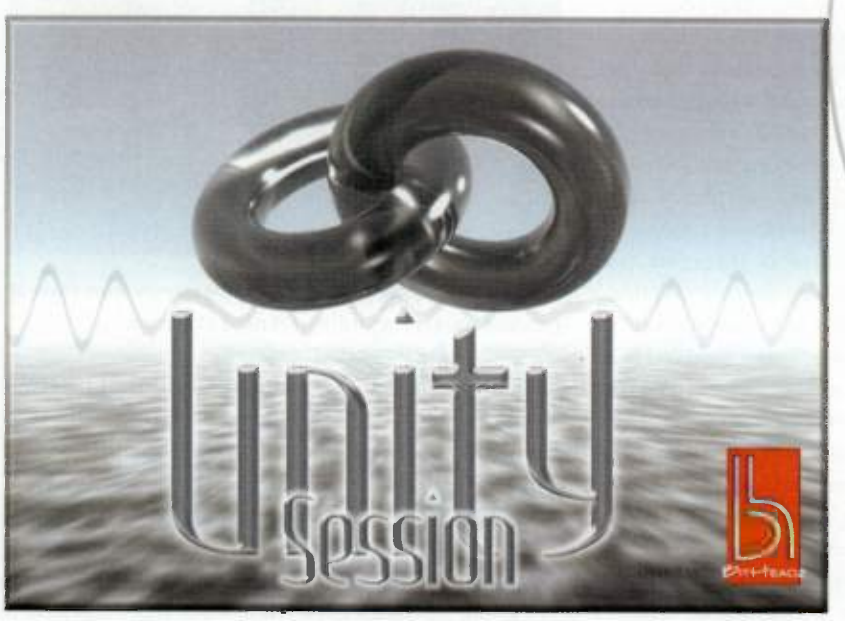
David Battino provides realism to MIDI tracks by adding acoustic-instrument artifacts such as fret squeaks and scrapes.

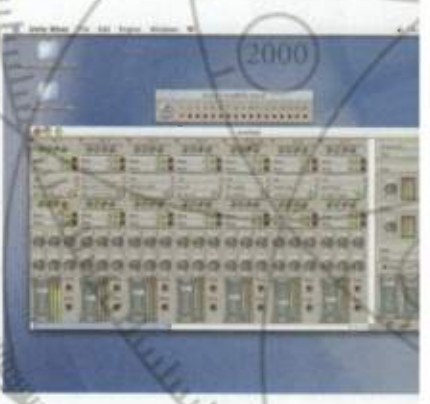
I've heard patches that approximate the instrument's slow, weepy characteristics, but I've rarely heard a patch that captures its higher registers or one that conveys the fast staccato soloing techniques I've heard from some steel players. However, when I accidentally sent the wrong Program Change message to my Roland Sound Canvas, I heard just the right sound from its fretless-bass patch. To help complete the country tune, I found an effective Telecaster-like sound in the General MIDI (GM) Clavinet patch; it was perfectly nasal, though a tad synthetic sounding. With a bit of adjustment to the filter's cutoff frequency, I found just what I needed.

David Battino takes his cue from movie sound design. "Often, technically accurate samples sound wimpy and unrealistic in context, so you need to exaggerate them, subtly adding timbres the mind expects to hear," he says. "For a movie soundtrack, I had to create an electric-bass solo for an actor to match during filming. I set up a layer in a Korg T3 to trigger a fret-squeak sound in a very limited Velocity range—something like 55 to 64 out of 127 possible values. That meant the squeaks appeared almost randomly.

"When I saw the final cut of the movie months later, I initially thought they'd replaced my performance with a real player. I doubt a real bass would

Obsession.





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have produced those squeaks, but they lent a certain organic realism to the performance. The Roland SC-8850 Sound Canvas and the Yamaha Motif, among other synths, include numerous performance artifacts such as scrapes and breath noises that you can use to desterilize a track."

Mounsey likes to beef up sampled sounds with analog synthesizer waveforms. "People have been layering samples with analog stuff for a while; that's an old trick. You can hide deficiencies in the sample that way and make it more even or full. I like to take a sampled sound and mix it in with something different that's filling in certain holes, maybe rounding out frequency ranges that I miss or coloring the sound a little differently."

One reason that sampled single-instrument sounds usually fall short is that they don't evince the complex timbral changes that acoustic instruments go through. Simply layering another waveform with the original isn't going to do the job; you need to continuously vary the balance between one layer and the other. More importantly, you need to do it in a way that the sequencer can capture.

Stock fretless-bass samples sound a bit too muddy and static for my taste, for example. Instead of relying on those samples, I use a dual-oscillator patch with a sampled, fingered electric bass on one oscillator and a tuba sample on the other. (Other sampled brass instruments such as French horn also work.) I control the second oscillator's amplitude (and to a lesser extent, its filter frequency) with Aftertouch. Bearing down on the keys brings up the tuba waveform, producing that horn-like Jaco Pastorius tone. You can also use Aftertouch or Modulation to bring in a light, slow LFO to get that characteristic slow, wide vibrato, but be careful not to overdo it.

Even if your goal is a replica of an acoustic instrument, don't forget to listen carefully to unabashedly synthetic waveforms; you never know when a little fine-tuning with filters or envelopes will yield the basis for a perfect instrumental sound. For example, to imitate the nasal qualities of a fingered electric bass, I've had great success using pulse waves at roughly 25 percent pulse width. By subtly modulating pulse width, you can vary the virtual picking hand's distance from the bridge; as pulse width approaches 50 percent, you can simulate the rounder, more hollow tone achieved by playing a bass closer to the neck.

COURTESY ROB MOUNSEY

Rob Mounsey builds ensemble arrangements from several instruments, each with different rates of vibrato.

It's a good idea to become acquainted with your synth's raw, unprocessed waveforms. Familiarity with your palette of waveforms can suggest new sounds or offer alternatives to old favorites (see the sidebar "The Naked Synth").

COOL WIND

Some wind instruments are among the most problematic instruments to bring to life. Listen to any decent saxophone player, and you'll realize that the number of timbral changes that occur in a short time is just impossible to capture with any sampler, much less a sample-playback synth with a limited ROM sound set.

Fortunately, you don't have to resign yourself to static saxophone snapshots. Frequency modulation is a potent technique for animating sampled wind instruments. You don't need a DX7 or the like to use FM; many synthesizers provide LFOs that creep up into the audio-frequency range, which should be enough for this trick. Take a boring, static sax sample and route Aftertouch to control LFO level. Set LFO speed to maximum. When you press down on the keys, the sample vibrates rapidly enough to produce sidebands that should effectively simulate an overblown effect. Adjust the LFO speed to taste and experiment with different LFO waveforms for different sidebands.

COURTESY GEORGE WHITTY

George Whitty adds bite and animation to sampled string sections by fading in distorted guitar samples processed through highpass filters.

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Sequencing with Style

An old ploy for imparting realism to sampled saxophones, flutes, and other wind instruments is to record them with breath noise in the attack. The problem is that even legendary sax player Ben Webster (noted for his breathy sound) took a break from that technique now and then.

Jack Hotop explains how to conquer the sampled breathy-saxophone sound: "I've used [Korg] Triton and [Korg] Karma highpass filters on bottles and pan flutes and then added them to saxes and other woodwind sounds to create a breathier quality. I often will keep them at a low level initially so that they can be mixed in using Velocity, the ribbon, or the *y*-axis of the joystick. That provides you with more control over breathiness. Besides, constant blowing can make you pass out after a while."

It's difficult for a sampled instrument to duplicate the attack transients of the original. When you play a sample above or below its original pitch, you transpose the transient's pitch and envelope. In addition, the transient spectrum needs to modify in response to variations in the attack's intensity. Again, a little creative frequency modulation goes a long way.

From programming my Casio CZ-1000, I learned that you can use pitch envelopes to provide artificial attack transients. Program an envelope generator so that the oscillator quickly rises above normal pitch during the attack and immediately falls to normal pitch during the decay. Experiment with the pitch envelope's attack level to tune the fake transient's frequency. To keep the transient's pitch consistent regardless of which note you play, don't assign note number or key position to modulate the pitch envelope's rate or level. On the other hand, modulating the pitch envelope's depth with Velocity can add a stronger snap when you dig in.

BUG OR FEATURE?

Surprisingly, a synthesizer's bugs or quirks can provide realistic artifacts. If you've ever programmed a Korg M1, for example, you might know that certain samples overload the instrument's outputs when you play them in a raw, unfiltered state. With the amplitude and filter wide open, a few samples actually produce an aliasing, fizzling sound and quickly shut off the outputs.

By reducing the oscillator level and filter settings, I discovered a way to creatively use that idiosyncrasy. If you route Aftertouch to control oscillator level, you can selectively add distortion and aliasing to the M1's static sax sample. When you press down on the keys, the otherwise unpleasant artifacts provide a fine simulation of an overblown saxophone's squealing harmonics.

A synthesizer's quirks can also add a unique, less realistic touch. "While reviewing a cheap General MIDI keyboard, I became curious [about whether] it would respond to external MIDI Control Changes," says Battino. "So I hooked up my trusty Keyfax PhatBoy and spun the knobs while the $300 synth played one of its demo songs. Apparently, the manufacturer had skimped on the microprocessor, because the additional data just bamboozled the keyboard, causing it to spew horror-movie sounds. I've noted similar effects when overtaxing synthesizers, but this one was notable because I only had

GETTING TO KNOW YOU

Since attending Oberlin College, **David Battino** has worked at Village Recorder in Los Angeles, spent a year in Tokyo as a Henry Luce Scholar, and worked with Roger Linn on the MPC 3000. Battino is the founding editor of *Music and Computers* and **EM**'s *Desktop Music Production Guide*. He currently runs Batmosphere, a music and media-production service.

Jack Hotop studied at the Berklee School of Music and the Boston School of Electronic Music. He has performed and recorded with Todd Rundgren, the Drifters, Gloria Gaynor, James Brown, Richie Sambora, John Entwhistle, and many others. As senior voicing manager, Hotop has provided sound design for Korg since 1983.

Lyle Mays has been part of the Pat Metheny Group since its inception in 1977, cowriting much of its music. In his youth, he studied with Rich Matteson and Marian McPartland, then studied composition and arrangement at North Texas State University, and eventually toured with Woody Herman's Thundering Herd. Mays has received four Grammy nominations for his solo work.

Rob Mounsey attended the Berklee School of Music and went on to work with Steely Dan, Paul Simon, Lyle Lovett, Milton Nascimento, Eric Clapton, Chaka Khan, and many others. He was recently nominated for a Grammy Award for his arrangement of Puccini's "Nessun Dorma," performed by Aretha Franklin.

As Burt Bacharach's arranger and music director, **Rob Shrock** appeared on the live album *Burt Bacharach—One Amazing Night*. He recently toured with Bacharach and Elvis Costello in support of their collaboration, *Painted from Memory*. Shrock toured with Dionne Warwick for many years and has performed with Sheryl Crow, Elton John, Gladys Knight, Frank Sinatra, and Stevie Wonder.

Guitarist **Jimi Tunnell** studied at North Texas State University. He has toured with Yukihiro Takahashi and was a member of Steps Ahead. Tunnell has also performed and recorded with Laurie Anderson, Carly Simon, Malcolm McLaren, Adam Holzman, Omar Hakim, Tom Coster, and the Bob Belden Ensemble.

George Whitty spent several years as a member of the Brecker Brothers band. He recently began pre-production on a Michael Brecker CD and produced and played all keyboards for Randy Brecker's current release. Whitty is featured on Santana's *Supernatural* and has toured with David Sanborn and Peter Erskine. He also writes and produces music for television.

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to twist a single knob to bring it to its knees. The sounds were so tortured."

Whitty was irritated when his Yamaha EX5 wouldn't play back a Mono-mode sample perfectly legato. "I'd intended to sample all my favorite 12-oscillator Oberheim stuff into the Yamaha and try to take the Oberheim off the road," he says. "When you play legato, there's a really obvious dip in level between the notes. I spent quite a while futzing with the EX5, trying to fix it, until I figured out that it's really kind of great the way it is. I can entertain myself for quite a while just holding one note and tweaking it by playing other notes briefly. That resets the held note to different Velocities, each of which is preceded by a little volume dip and pitch swoop. The result is a sort of tweezed ethnicity."

KATHY SCHMIDT

Master synthesizer programmer Jack Hotop layers stock GM sound-effects patches with other sounds to add more punch. Here Hotop gets an assist from his friend Nureyev.

ALL TOGETHER NOW

You might be tempted to use sampled or synthesized ensemble patches because that's easier than building ensemble performances instrument by instrument. Unfortunately, though, that might make your music less expressive.

Why are ensemble sounds appealing? Mounsey says that "our standard paradigm for a big, warm orchestral sound is a string section playing chords. One thing that we love about a string section is that there are a lot of individuals playing with different vibratos—not just different depths, but different speeds. They're coming from different places in the stereo field. So one magical way to make a lot of beautiful space is to have very subtle, multiple vibratos going on. With sustained orchestral-instrument sounds, what we should be using is continuous controllers."

To simulate an ensemble, Mounsey relies on multiple synthesizers and string

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sounds, each with a different modulation rate and depth. "Normally, I make very careful edits on volume curves and modulation to create vibratos," he says. "I do a live pass and then edit it."

Polyphonic Aftertouch is also useful; each note can have its own vibrato, and you can vary it continuously. It's also helpful to route Polyphonic Aftertouch to filter-cutoff frequency and maybe add a touch of control over resonance; after all, in a real ensemble, no two players have exactly the same tone.

"If you're using samples of string ensembles, try layering a little bit of a solo-instrument sample on top of them; that's a pretty standard trick," says Mounsey. "If your samples are legato without much attack, add a tiny bit of a staccato sample; just dial it in slightly to give a little definition to the sound."

SHROCK THE BAND

Rob Shrock uses samples of smaller ensembles to compose orchestral parts. "My basic orchestral template consists of over 60 MIDI tracks, which I adjust as needed for each piece," he says. "The string section takes up the largest number of tracks. I split the strings into sections much like a real orchestra: first violin, second violin, viola, cello, and bass. The device that adds the most impact to the overall sound of simulating a string section is adding several solo instruments to the ensemble sounds.

"For instance, when sequencing a large section of violins playing a melody, I will typically sequence a large ensemble sound (12 or more players) followed by a small ensemble sound (4 to 8 players). I will then add three or four individual solo-violin tracks playing the same part. Each part is played as a separate pass—the variations in performance impart density and interest to the musical line. If there are second violins, I will repeat the process again for those parts, usually with duplicates of the same sounds, but always on different MIDI channels. The same process applies to the cellos. Although I use a wide variety of samples depending on the specific articulations I'm going for, this technique works well even without a massive sample library.

"It is critical to manipulate the dynamics of the lines as you are sequencing. Learning to manipulate a volume pedal is probably the single most important factor in creating good orchestral MIDI parts. I highly recommend using expression [CC 11] for manipulating your volume pedal, slider, or wind controller. That frees up MIDI Volume for overall balancing of the parts.

"As is common, a lot of my sounds are tweaked so that faster Velocities create shorter attack times. However, I also use a few other techniques to help provide variations to articulations. I often layer three or four different sounds that are Velocity switched, which provides immediate access to several articulations instantaneously. Because a lot of the actual dynamics of the performance are coming from the manipulation of Expression and Volume data rather than Velocity, I can use Velocity to switch articulations.

"I tend to group articulations into basic categories—for instance, melodic, marcatos, pizzicatos, and sordinos for strings. Each category gives me several selections based on how hard I play. When applying this technique, I usually keep it simple, dividing Velocity into soft, medium, and hard ranges that are easy to play on the fly. If I accidentally play a note out of the intended range and trigger an unintended sound, it is a simple matter of editing that Velocity in the sequencer. That technique speeds up the sequencing process.

"With woodwinds and brass, I tend to stay away from ensemble sounds, opting to build up sections by sequencing each instrument individually using solo sounds. For big, thick brass and French horn sections, I will occasionally layer ensemble sounds underneath for added density and power."

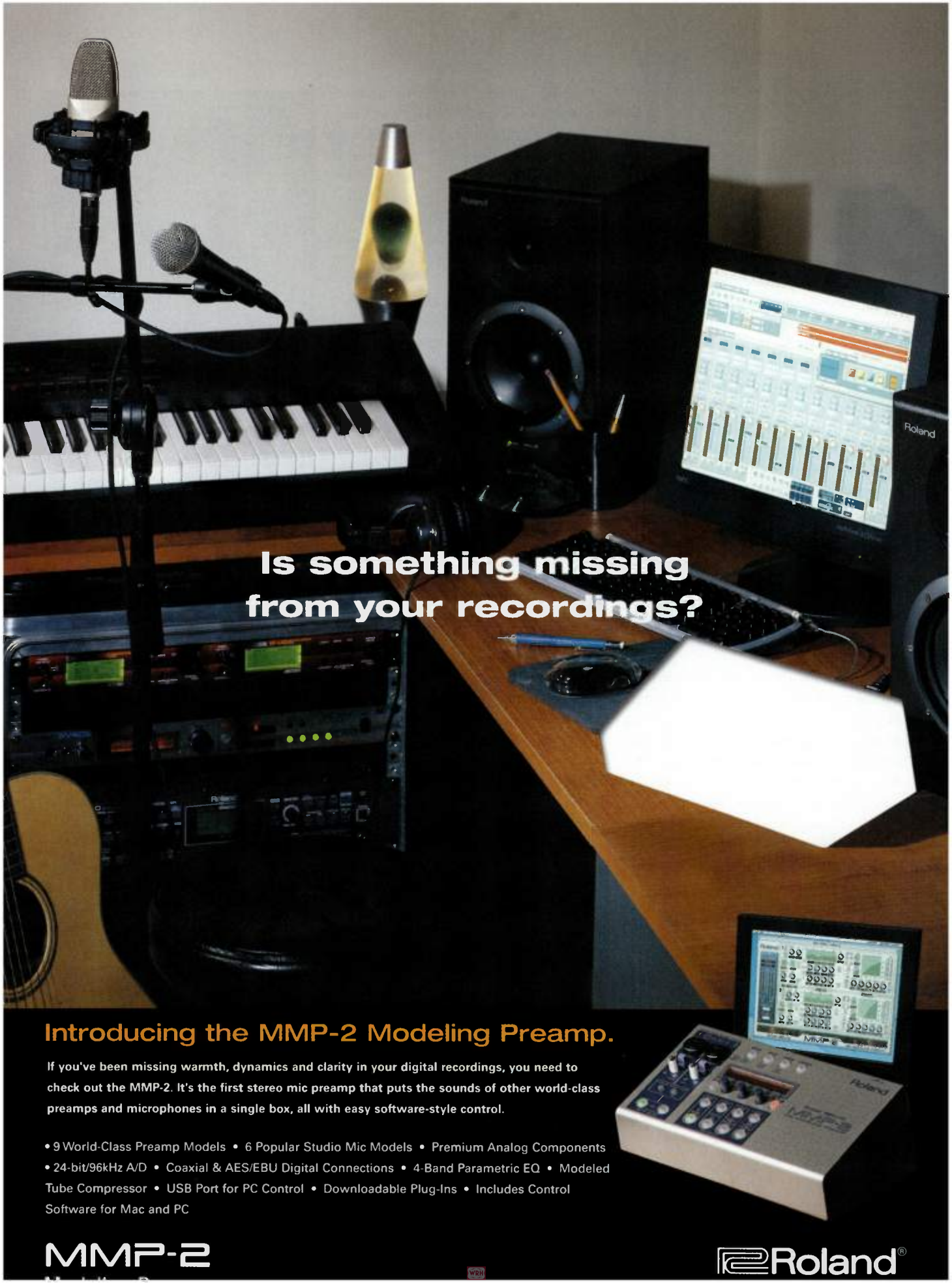
BILL DOUTHART

Although he is proficient with keyboards, Jimi Tunnell finds that MIDI guitar controllers offer extra dimensions of expression, especially for sequencing bass tracks.

MAYS'S WAYS

Mays takes advantage of the multitude of tracks offered by sequencers. "This may be obvious," he says, "but I use a ton of tracks. They're free. If I have to do some kind of brass function, ideally, I would write out the section as I would score it for a brass section and perform the individual parts. I use trumpet, second trumpet, and so on and do different performances of each. I also like to use a slightly different sound for each section, with its own volume rides. You avoid that pianistic 'chord, chord, chord' sound, where you have a whole bunch of notes at once, played on one patch. I'm very skeptical about doing keyboard-style parts for brass instruments."

Preprogrammed envelopes often pose particular problems. Mays cites the example of sforzando-brass patches with preprogrammed envelopes. "There's just a uniformity to them that bugs me," he says. "An envelope may only fit a certain section in one piece; it's just not a practical use of one's time." Instead, he prefers to program envelopes inside the sequencer, drawing changes in amplitude and timbre by hand. "I think that's a superior way to go about it," says Mays. "Basically, synth sounds are not complex, and they don't change over time unless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you program them to. Acoustic instruments, on the other hand, just naturally change in time. Even if you try to bow a string exactly the same way twice, it's going to come out differently."

In fact, Mays prefers to send control information from his sequencer rather than preprogram synth patches. He feels that preprogrammed sounds don't work as well in the studio when they're played in conjunction with acoustic instruments.

There are several ways to control synthesizer sounds with your sequencer. For example, you can program your synth to get louder and open up a filter with Aftertouch, Expression, Modulation, or any of a slew of registered and nonregistered parameter numbers. Editing MIDI messages within your sequencer can afford more control than many synths can offer.

Compared with the EGs built into a typical synth, hand-drawing envelopes in your sequencer can provide a great deal more power, continuity, and detail (see **Fig. 1**). If that's too labor intensive, there's a middle ground: use an expression pedal to record the changes and then fine-tune the performance by hand in the sequencer. Most current synths offer a wealth of knobs and sliders that can achieve the same ends.

"When you're sequencing, never copy and paste," Mays says. "The more detailed the work you do, the more detailed the final results will be. More is always better in this department."

SURREALISTIC GIZMO

One of the most significant aspects of synthesis is that you can create sounds that have no precise counterparts in the acoustic world; just the same, the sounds usually duplicate functions of their acoustic relatives. Synth pads often duplicate the work of string or piano parts, and you can substitute completely artificial synth leads for blazing guitar solos.

Even special effects or bursts of noise can supplant or reinforce some acoustic element of a sequence. Hotop explains: "Occasionally, I've used the Triton's GM sound-effects program Car Stop for a rap-style scratching effect, and I've layered Explosion with Orchestra Hits to add a little extra bang. I have even used the Heart Beat for a low, muted bass drum."

On his most recent album, *Solo: Improvisations for Expanded Piano,* Mays augmented the sound of his MIDIfied grand piano with samples; he left no part of his piano untapped (or unscraped). "Most of the prominent sound effects on my *Solo* record started with acoustic piano," he says. "We spent a day crawling around the piano, recording various hits and scrapes—a lot of them with the sustain pedal held down to take advantage of the natural resonance of the chamber. And then those sounds were massaged beyond recognition. One of the reasons they sound so rich is that the actual samples are high-quality recordings of the full envelope. They take up a lot of space; they're not faded out."

Sampled electric and acoustic guitars and pianos are prime source materials for interesting, slowly evolving pads. The more continuous animation you can bring to the sound, the more intriguing the timbres will be. Offering his programming expertise, Hotop says, "The Karma and the Triton PCM [pulse-code modulation] ROMs have layered piano samples, which are called EP Pad 1, 2, and 3. These samples use long loops. It's easy to slow down the envelope attacks, add sustain, and then apply a tiny bit of filter modulation using the filter LFOs at slow rates. Add insert and master effects to further animate the pad if necessary."

CONTROLLING SHARE

MIDI is often regarded as a keyboard-oriented technology. Nonetheless, **EM** readers are surely aware of the wealth of MIDI controllers with strings, pads, mouthpieces, and other nonkeyboard appendages. Those instruments offer capabilities that are not readily available from keyboards. Consider adding nonkeyboard controllers to your MIDI studio.

COURTESY ROB SHROCK

Rob Shrock builds large ensemble arrangements from individual-instrument and smaller-ensemble samples.

Tunnell often switches between keyboards and MIDI guitar controllers. "Although I put in lots of data from the keyboard, I have found that triggering certain sounds from a guitar synth can turn a horribly stilted and unusable patch into something much more organic," he says. "Bass sounds, in particular, are drastically improved that way. Some of the glisses, grace notes, and the like are just impossible to emulate from a keyboard. Also, I occasionally like to layer bass sounds with different Velocity curves, which gives the effect of randomizing the attacks a bit. A pan flute or recorder sound that sounds incredibly stock really comes alive when triggered that way.

"I have to stress, however, that you really do need to study the phrasing of these instruments to pull it off. Don't play your Jimmy Page licks with this stuff. That being said, the inherently fluid articulation associated with the guitar seems to serve these sounds really well."

Sequencers allow musicians to clean up their playing, but take care not to oversanitize a performance. MIDI guitars can often glitch or play notes that were caused by accidentally brushing an adjacent string; sometimes those imperfections can provide a little extra realism and funk to the track.

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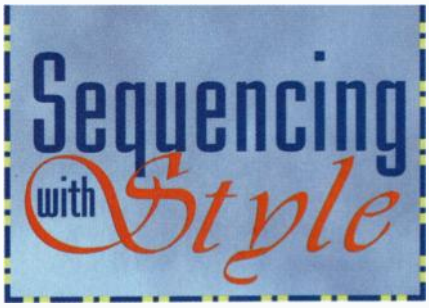

A MIDI guitar with Pitch Bend enabled can be invaluable for sequencing realistic string or brass ensembles, but as stressed earlier, avoid the temptation to use generic ensemble patches. Instead, with Pitch Bend enabled, record each instrument in an orchestra one at a time. Even if you think you've hit the note dead-on, your controller will send subtle amounts of Pitch Bend to each instrument. The process might seem a bit laborious, but when you've sequenced the entire ensemble, the performance will be more realistic and animated.

It's easy to overlook some of the gestural possibilities that a synth's real-time controllers offer. For example, I've always regarded ribbon controllers as an excellent way to create different types of smooth and continuous modulation. Battino and **EM** associate editor Gino Robair point out that you can tap the ribbon at divergent locations to create discrete, noncontiguous data. You can achieve a wonderful effect by tapping rhythmic patterns at different points on a ribbon controller that's controlling filter frequency; your synth will respond with drastic changes in timbre, all in time with your gestures.

SEQUENCER AS SYNTHESIZER

Many people have wished for synthesizers with unique features; they're certain that one extra feature will be just the ticket for breathing fire into their sequences. You might not realize that many untapped capabilities lie under your sequencer's hood and that the sequencer you already use may offer more flexibility than the latest gizmo will allow.

For instance, many musicians have lusted after synths that can morph from one timbre to another; among the synths with that capability are the Sequential Circuits Prophet-VS and the Korg Wavestation. Both employ *vector synthesis,* which is nothing more than a bit of creative volume-crossfading and panning. You can easily achieve vector-type effects in your sequencer by performing volume crossfades between distinct synths and sounds. With your sequencer, you can go well beyond the limit of any single device's complement of features to create long, animated timbral changes.

Some synthesizers have the ability to scan a list of waveforms, often resulting in a rhythmic pattern of evolving timbres, percussive grooves, or both. You can simulate that effect using the arpeggiator that's built in to many sequencers.

For example, one neat feature I discovered in Digital Performer's arpeggiator is a small checkbox that reads, "Cycle through device group assignments." A Device in Digital Performer is simply a group of instruments that are assigned to a track; any data sent to or from that track plays those instruments. With the box checked, the arpeggiator sends successive notes down the list of instruments, creating burbling, rhythmic grooves or wavetable-scanning-type effects, in the tradition of the venerated PPG synthesizers and Korg Wavestations. Best of all, playback is automatically synchronized to your sequence's tempo.

If your sequencer doesn't offer a similar feature, try this trick from Battino: "Select every nth note of a track and drag [or cut and paste] the selection to another track. Repeat this several times and then assign a different patch to each track. The effect is reminiscent of the old Yamaha TX802's Note Rotate feature."

IT'S ALIVE

Given the wealth of ideas offered here, you no longer have any excuses for creating dull, lifeless sequences. Your sequencer is far more than a mere word processor for MIDI data. If you understand and take advantage of the close relationship between your sequencing software and your synthesizers, your sequenced music will take on a life of its own.

Assistant editor **Marty Cutler***'s second computer for sequencing was a Commodore C-64; he shorted out his first one as he was connecting the video cables. Thanks to Kate Andrews of Shorefire Media and Gilles Amaral at the Ted Kurland Agency for their assistance.*

We welcome your feedback. E-mail us at emeditorial@primediabusiness.com.

THE NAKED SYNTH

One task that's fundamental to assembling your palette of synthesizer colors is auditioning and evaluating the entire range of your synth's waveforms. Playing any synth patch by patch can give you hints about the unit's abilities, but presets are often just the tip of the iceberg.

The object is to train your ear by listening to the raw sound of the waveforms. You can audition raw waveforms by initializing a patch. Most editor-librarian programs provide the ability to do that, and some synths offer a patch-initialization button. If you do not have an editor-librarian or if your synth doesn't provide patch-initializing amenities, you'll just have to dig in to your synthesizer's programming menu.

Remove all effects and examine the sound with a simple envelope: instantaneous attack, no decay, full sustain, and instantaneous release. The filter and amplitude envelopes should resemble a square wave. One handy shortcut is to start with a generic organ patch, because organs are usually programmed with simple on/off envelopes and no Velocity sensitivity. Conversely, you can set all envelope depths to zero, making sure to open up filter cutoff frequency all the way and eliminate all modulation control, including Velocity, Aftertouch, and LFO modulation.

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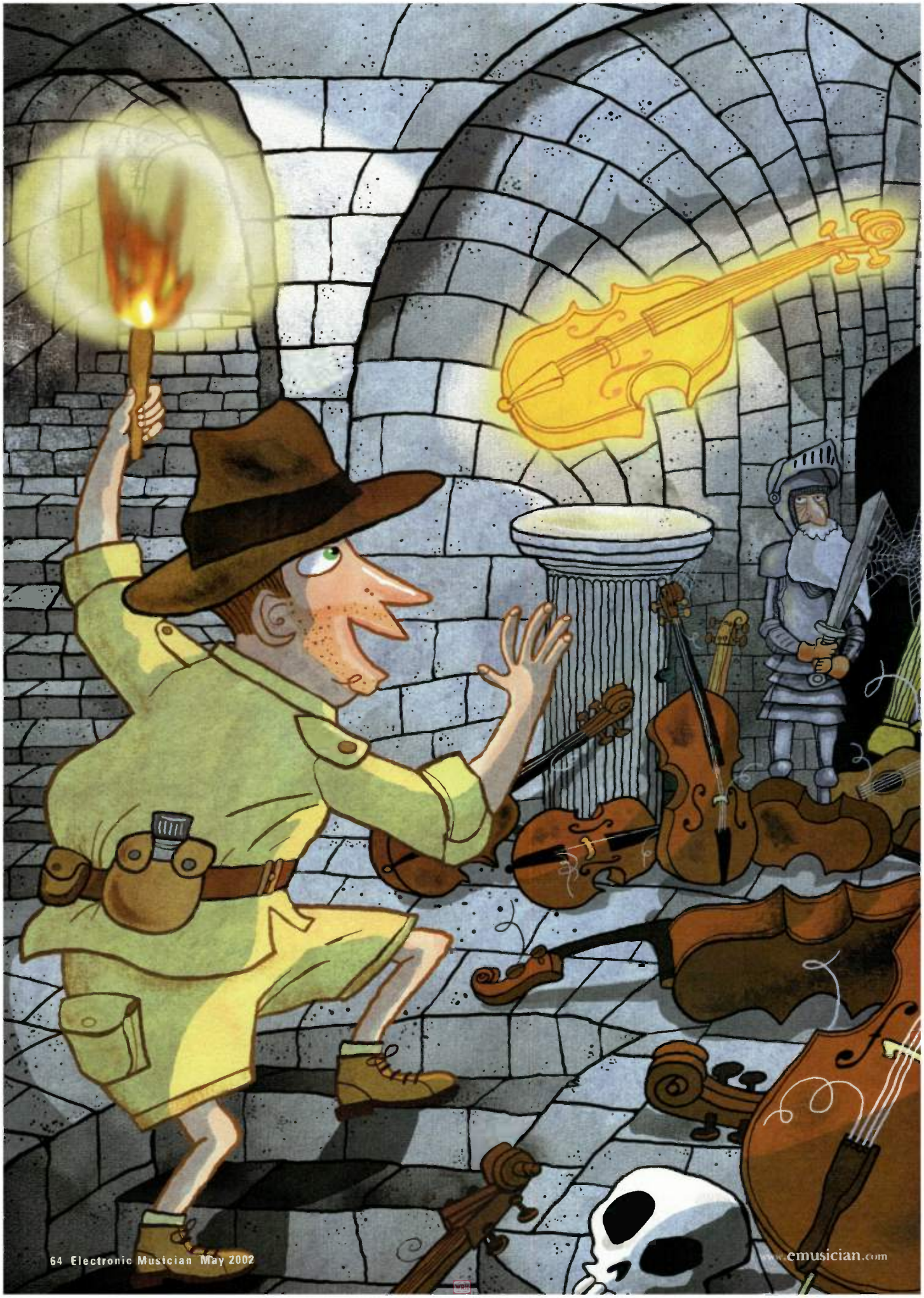

String Quest

Join the adventure: The search for the ultimate orchestral string library.

By Gary Garritan

As a composer and a recording artist, my favorite part of the arranging process has always involved the string section. Strings are the backbone of a symphony orchestra; they have a wider dynamic range, are more versatile, and possess greater expressive power than any other section of the orchestra. Unfortunately, recording with a full complement of strings is prohibitively expensive and highly problematic on a number of levels.

During the past two decades, many composers and arrangers turned to sample libraries to realize their musical goals, and a realistic and protean string library soon became the elusive Holy Grail of the sampling world. I, too, dreamed of having the perfect string-sample library at my fingertips, and before long I had embarked on a quest to develop such a library.

The endeavor to produce the ultimate string library was indeed like the pursuit of the Holy Grail. The adventure was beset with trials, tribulations, and occasional disappointments. I planned the journey, made the commitment, set out on the mission, encountered unexpected detours, and met many interesting and amazing people along the way.

ILLUSTRATION BY JEFF SHELLY

String Quest

THE JOURNEY BEGINS

Garritan Orchestral Strings (GOS) is the product of years of creative work. Planning for the library began in 1998 when I engaged the Moravian Philharmonic Orchestra in the Czech Republic to record some of my music. (See "The Real Thing" in the October 2000 issue of **EM** for more on my experiences working with Eastern European orchestras.) Earlier that year, I had purchased several of the leading string libraries, but none of them met all of my criteria: impeccable sound quality, the finest instruments available, chromatic sampling, wide-ranging multiple dynamics, numerous articulations, and greatly expressive capabilities. Although I had resigned myself to hiring a real orchestra for the project at hand, the idea sprang to mind that someday I should develop my own string library based on my criteria.

In scouring various newsgroups and forums, I learned that many other musicians were also dissatisfied with their string libraries and wanted much more. So I began to imagine what the ideal sample library would encompass. Recent advances in high-resolution audio technology and the increased memory capacity in PCs were making it possible to overcome the limitations of existing libraries. With the advent of Tascam's GigaSampler and GigaStudio (which was originally developed by Nemesys), I could sample the string sections chromatically at multiple Velocities and I could sample every articulation and playing technique imaginable. I was beginning to see the light; the odyssey had begun.

GARY GARRITAN

FIG. 2: The Italian-made Gagliano, Guarnerius, and Stradivarius violins shown here (left to right) are among the rare and valuable instruments from the 18th century that were included in the GOS recording sessions.

MAPPING THE TERRITORY

To guide me through my quest, I needed a reliable map. Obtaining the help of others who knew the orchestral and sampling terrains would prove invaluable as I charted the journey. I spoke with many composers, arrangers, and players, and each of them told me what he or she wanted most in a string library. String players were consulted to determine which articulations to record and to explore the capabilities of their instruments.

I analyzed existing libraries to ascertain what was already available and what was lacking. I consulted many other specialists, including orchestrators, string experts, conductors, sample developers, audio engineers, university professors, and sound designers to glean as much information as possible before the recording sessions. I even returned to the method books that I used as a young violin student and read various textbooks on orchestration. Another part of the research involved attending concerts and listening to great orchestral string recordings. I also sat in on orchestral recording sessions to gather ideas about string-recording techniques. The more I sought the advice of the experts, the more prepared I felt.

STEPHANIE BERGER

FIG. 1: Lincoln Center for the Performing Arts is a large complex of theaters and auditoriums well known for hosting world-class artists.

THE CASTLE

The most important attribute of any sample library is the quality of its samples. In my pursuit of the best string samples, I knew that I would have to search for the finest stringed instruments. My initial plan was to record an orchestra in Eastern Europe. I investigated several orchestras in various countries and inquired about their instruments. Unfortunately, the countries were still struggling after the Cold War, and the players didn't have the high-quality instruments that I needed for my library. In addition, European orchestras frequently tune to a different standard (A 443) than American orchestras (which typically tune to A 440). It was clear that I needed an alternate plan.

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Frank Spitznagel, the musical director for the project and a veteran artist at New York's Lincoln Center for the Performing Arts, arranged for a rare opportunity to record some of the world's finest stringed instruments in a large rehearsal hall at the Lincoln Center complex. I was thrilled. As a young boy, I was deeply moved by the sound of a Stradivarius violin played by Itzhak Perlman at Lincoln Center; little did I know how far that inspiration would carry me.

Lincoln Center is the largest and most respected performing-arts complex in the world (see **Fig. 1**). It is the Camelot of culture. To record there meant my quest was off to a great start. As good fortune would have it, several other factors guided me along the way, including the necessary contacts, the latest recording technology, and many good and generous people who were willing to assist me.

THE COURT

As with the knights of the Round Table, each string player for the GOS library was handpicked. Finding top-notch performers with fine instruments was a challenge. Dave Eggar, the conductor, and Pauline Kim, the concertmaster, worked diligently to select and book world-class string players with incredible instruments.

In time the instrument roster grew to an impressive size: two Stradivarii, a Guarneri, three Gaglianos (see **Fig. 2**), two Testores, a pair of Montagnanas (see **Fig. 3**), a Vuillaume, a Klotz, a Calcanius, and a Betts, among many others. These famous instrument makers are revered for the quality of their workmanship and represent the pinnacle of instrument design. Collectively, the stringed instruments in the GOS library are worth millions of dollars. Despite being hundreds of years old, the instruments possess a powerful and beautiful sound. I'm still amazed that we were able to assemble such fine players with their exquisite instruments in the same room at the same time. Providence was indeed smiling down on this quest.

THE WEAPONS

Assembling the appropriate tools for the recording sessions and attending to the technical aspects of those sessions were vitally important tasks. Choosing the right recording equipment, microphones, recording format, sampling rate, and resolution demanded considerable planning. When you're sampling exceptionally fine instruments, you must capture each instrument's full spectrum and tonal characteristics or you'll seriously compromise the samples. That meant using superior equipment and paying careful attention to recording technique. The subtle details, overtones, and resonances would all be lost if we opted for anything less.

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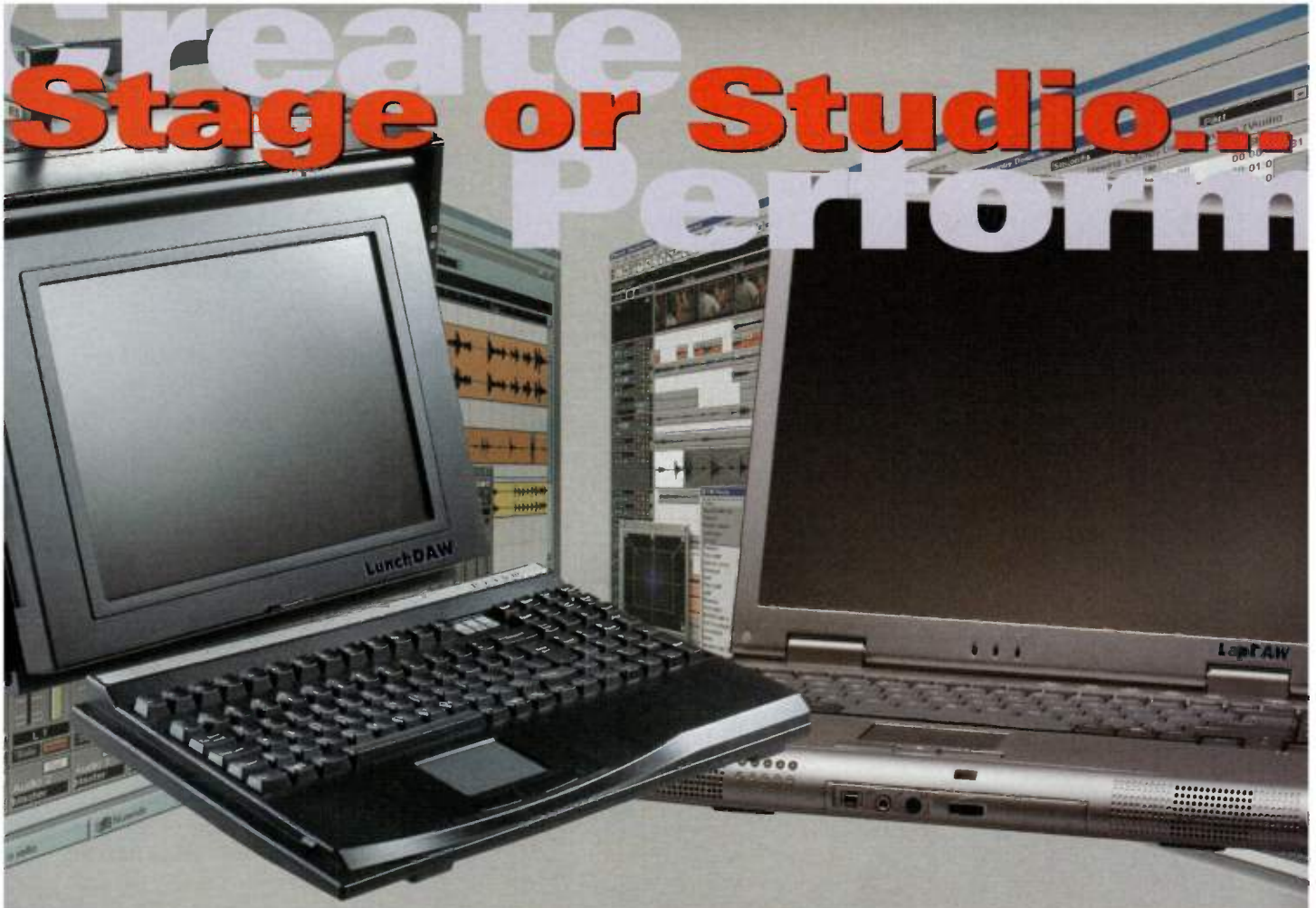

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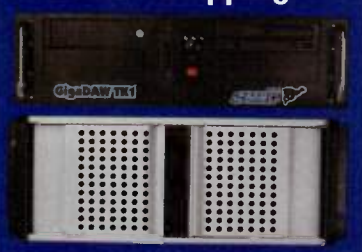

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Finding the ideal placement for the mics—the "sweet spot"—required experimentation and careful listening. Locating the microphones too close to the players would cause one or two instruments to dominate the ensemble, destroying the ambience of the sound. Too far away would result in excessive room noise, muddiness, or loss of detail. It would also make it harder for the sample user to control the sense of space. We therefore settled on a more moderate mic placement to optimize the audio quality and character as well as flexibility.

GARY GARRITAN

FIG. 3: This is one of two valuable Montagnana cellos from the 18th century that added to the sound of the cello section.

In order to avoid phasing and to achieve a proper stereo spread, we placed a Jecklin Disk between the two microphones. In addition, we put two Neumann KM 84 cardioid microphones farther back in the hall. (The four-microphone configuration allowed for possible 4-channel or surround-sound releases in the future.) We also used a Crown SASS-P stereo microphone to capture room ambience and impulse responses.

FIG. 4: The Northern Sounds GigaSampler Users Forum maintains an active and informative Web page devoted to the GOS library.

Once the microphones were in place, we routed the signals to an Apogee PSX-100 analog-to-digital converter and recorded directly to hard drive with 24-bit resolution and a sampling rate of 88.2 kHz. Stringed instruments have a rich harmonic content that 16-bit, 44.1 kHz equipment cannot adequately record. A higher sampling rate captures the full range of the music more accurately with less chance of distortion (or aliasing) and fewer phase problems.

Moreover, a higher sampling rate better preserves the transient responses such as the attacks. (The high-resolution recording also allows the string library to stay current when 24-bit sampling becomes the industry standard.) We opted to use the 88.2 kHz sampling rate instead of the 96 kHz rate to avoid errors in downsampling. We felt that a sample conversion based on a 2:1 ratio instead of a 2.177:1 ratio would be more accurate (less prone to error).

I was concerned about what to do if anything went wrong, particularly in the event of a computer failure. We therefore devised a backup plan, adding a separate external archival drive and a Tascam 24-bit DAT recorder to the rig—just in case.

SLAYING THE DRAGON

The thousands of samples in the GOS library were obtained after many days of intense and demanding recording sessions at Lincoln Center. We arrived early to set up the equipment and troubleshoot any problems that might arise. The conductor, the musical director, and the concertmaster attended every recording session. Each day we informed the players of what to expect. In preparation for the sessions, the musicians played various classical selections and some of my original music while the recording staff performed the requisite sound checks. Once everyone felt comfortable, was warmed up and in tune, and had a good "group" frame of mind, the sampling began.

Dave Eggar, a Harvard- and Juilliard-trained musician, was the conductor in charge. Although I had prepared extensively, he had his own expert opinions on common usage and on the articulations and ranges that we should sample. He led the string sections as though he were conducting an actual performance. He interpreted and approached the project as a major musical work, tying the various bits and pieces of the project into a cohesive whole. Eggar had a good rapport with the players, which brought out the best in everyone and made for a better performance. He was brilliant, and my recording sessions were clearly in good hands.

The musicians played through many scales using different articulations. We recorded myriad long bowings and

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short bowings as well as additional techniques and effects. At times the players started at the lowest note of their instruments and worked their way up the scale chromatically. We recorded as many as four dynamics levels for almost all articulations, bowings, and playing techniques. We sampled alternative bowings and various finger positions. Occasionally, the musicians played through the circle of fifths to break up the monotony. They also played intonation notes followed by the desired target note. That technique not only allowed for proper tuning but also provided a sense of connectedness from note to note. We recorded second and third takes to obtain alternative sets. Throughout the process, we paid constant and close attention to monitoring in order to obtain the best recording levels and signals.

At the time of recording, we had to decide how best to use the limited and expensive time available. Of course, we had business matters to take care of, such as signing releases and payments, and we kept a detailed logbook of the recordings. Periodically, we took breaks, which helped to avoid fatigue and provided a wonderful opportunity to talk with the players. Their patience and stamina were amazing; imagine being a virtuoso player and having to play scales for most of the day. Overall, the recording sessions were a success. After investing so much effort and money, the major challenge was over, but a great deal more work lay ahead.

THE NEXT BATTLE

Producing a playable sample library from a series of source recordings proved to be a daunting challenge involving a mountain of editing and programming work. It was a laborious and tedious task, requiring a great deal of care and unfaltering attention to detail. Each of the thousands of individual notes had to be sliced up, trimmed, and named—one at a time. Because multiple takes were recorded, the editing team had to choose the best takes

THE PROGRAMMER'S TALE

When developer Gary Garritan first described to me the extent of the original recording sessions, I could see that this project presented an unusual opportunity to develop the kind of string library that I as a composer had always wanted. The challenge was to take a massive number of raw samples and organize them into a usable and expressive collection of specialized Giga-format Instruments.

From a technical standpoint, construction of the library's Instruments took place at four levels. At the sample level, programming included things such as tuning, editing, and looping. The note-region level involved mapping the samples across the keyboard, making note-to-note volume adjustments, creating the amplitude envelopes, and so forth. At the Instrument level, I worked with layer crossfades, response curves, filters, key switches, and controller assignments, among other things. Finally, there was the overall organization of the Instruments into groups that were intuitive to use and that were practical for CD replication.

With its array of programming aids, the GigaStudio Instrument Editor made the job easier (see **Fig. A**). To assist in the programming process and to ensure consistency, I devised a number of MIDI files to test the evenness of attacks, note-to-note levels, and Velocity-split transitions. To keep Garritan abreast of my progress, I improvised short music files that demonstrated each new feature and sent them in as MP3 files. Eventually, these short "progress" files were posted as demos on the Garritan Web site. (A link to these and other demos is available at the **EM** Web site.)

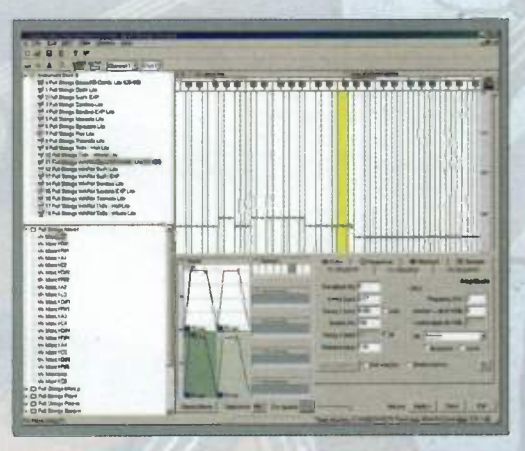

FIG. A: In the GigaStudio Instrument Editor, Tom Hopkins spent a great deal of time editing and mapping out the samples to create often complex Instruments.

The beta-testing process led to the further refinement of the Instruments and the addition of new ones; it turned the evolution of the library into a real team effort. Perhaps the most interesting phase of the project involved Jeff Hurchalla's programming of the MaestroTools utility, which allowed us to extend the capabilities of GigaStudio.

From the start, Garritan and I were of one mind about trying to retain as many human qualities as possible in the final product. Because the library was so large (with numerous articulation and layer choices), we felt we had the luxury of allowing many small realism-enhancing idiosyncrasies to remain in the library. The beta testers helped us walk the line between human variability and distracting flaws.

—Tom Hopkins

for each note. We made decisions and compromises on a note-by-note basis. We had to deal with false starts, bow scrapes, coughs, moving chairs, level inconsistencies, mistakes, tuning problems, and more.

For editing, we used several advanced programs, each capable of 24-bit, 32-bit, and 64-bit processing. When necessary, we applied noise reduction using the HDA AudioCube to process the samples with 64-bit floating-point precision, thereby preserving the sonic fidelity. Sonic Foundry Sound Forge 5 was our main editing program, though it was in beta at the time; needless to say, this library put the program through a rigorous test. Other editing programs included Steinberg WaveLab, Syntrillium Cool Edit Pro, and Sek'd Sequoia.

Neither equalization nor compression was applied to the sound files, but we did use some normalization to provide smoother transitions in GigaStudio. As the final step, we used Apogee's UV-22 process to dither the samples from 24-bit to 16-bit, preserving as much detail as possible. During the editing phase, I was fortunate to have access to state-of-the-art equipment and to have several good people by my side. It took more than eight months just to edit the samples—a Herculean feat!

We now had meticulously edited samples of the finest stringed instruments and the most comprehensive selection of articulations and techniques available. Still, the journey was far from over. We knew we had to make the samples musically playable and expressive. Mere notes cannot convey the act and art of musical performance. When emulating the sound of a real string section, the more control options you have, the more successful you will be at creating a realistic performance. In the past, sampled instruments were considered the antithesis of expressive performance. The time had come to change that perception.

Real string performers play their instruments in a variety of ways. For example, they bow and pluck the strings, they change articulations and bow strokes at a moment's notice, they alter dynamics instantly or gradually over time, they slide to a note or across many notes, and they impart a lush, expressive vibrato to a tone. What we needed was a way to allow a keyboard player to emulate those and other performance actions with relative ease. That would require an innovative solution. I therefore thought it best to assemble a roundtable of experts to help me succeed in my quest.

KNIGHTS OF THE ROUNDTABLE

As I continued my journey to produce the ultimate string library, I enlisted the help of a creative beta team. In the pursuit of musical excellence, as in other areas, collaboration often beats working in isolation. Fortunately, the modern age permits us to link electronically to a great many people. So I extended an invitation to members of the Northern Sounds GigaSampler Users Forum (www.northernsounds.com) to join a beta-test project. Previously, I had asked the forum members what they wanted in a sampled-string library, and many shared their wishes and provided valuable input (see **Fig. 4**).

Seventy-five forum members applied for the test project. From the initial group, I carefully chose beta testers based on their knowledge and experience. The resulting beta team consisted of 20 professionals from six countries. Their backgrounds were diverse—the group included Hollywood composers, educators, computer programmers, string players, professional musicians, engineers, scientists, and even one person working on the Human Genome Project. They were a gifted and special team and maintained a good group dynamic throughout the beta-test period.

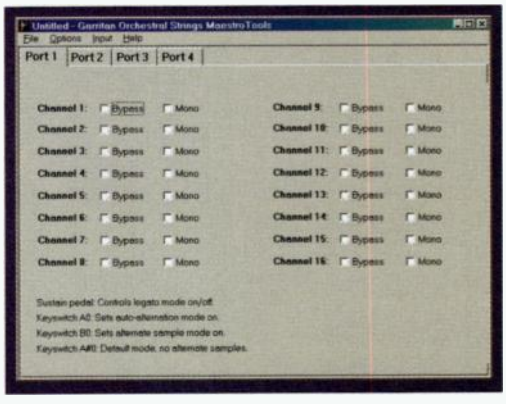

FIG. 5: Jeff Hurchalla's MaestroTools presents a single window for assigning its activity to different MIDI channels. It then appears as an option within GigaStudio.

THE MAGICIAN

With all of the creative input from the beta testers, I needed some exceptional programming and perhaps a few miracles. Enter Tom Hopkins, a fine musician, composer, and programmer extraordinaire. Hopkins was originally contracted to do what we thought would be several weeks of programming. As new possibilities and new horizons began to emerge, however, it soon became evident that this project was not what we had thought it would be.

Hopkins developed a workable version of the string library, which we sent to the beta testers. They experimented, played, and composed with the library, submitting comments and suggestions almost immediately. We soon established a unique private forum on the Northern Sounds site, where all of the beta testers exchanged ideas and interacted on a daily basis. Hopkins's musicianship and programming magically transformed the library into a coherent and musical whole. (For more on the programming, see the sidebar, "The Programmer's Tale.")

To my delight, the beta testers were amazed by the library's comprehensiveness and the quality of its sounds. The energy they devoted to their appointed tasks was astounding and unexpected. They put the library through its paces, scrutinized every note, and even used it in their films, television shows, and compositions. They were brutally honest in telling us what they liked and what they didn't like.

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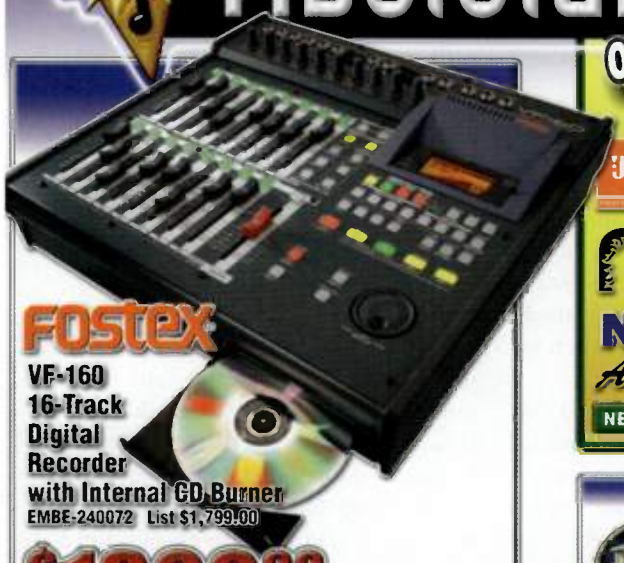

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ideas, as well as criticisms, all of which we wanted so that we could further improve the library. Bug fixes, new instruments, additional experiments, and more suggestions soon followed. We tried to examine all of the ideas the team members had to offer, adopting the best ones while continuing to implement new ideas of our own. Some excellent ideas were simply not practical (at least not initially), but most could be incorporated into the programming and were therefore included.

A period of many weeks ensued that could best be described as intense brainstorming or "idea jousting." Three weeks became five months. With the help of the beta team, we began to think about sample libraries in new ways. As we delved into the library, we systematically considered what makes strings sound the way they do. A beta member who uses the GOS library in a television series mentioned that the strings, being as detailed as they are, tended to stand out a bit during underscoring. We responded by implementing Warmth Control to let the user adjust the warmth of the strings as needed. Some beta members wanted tighter pizzicatos, whereas others wanted the looser, more natural-sounding pizzicatos as recorded in Lincoln Center. Undaunted, Hopkins programmed both options, allowing users to select from between the two.

The beta team pointed out that one telltale sign that a performance is sampled (or synthesized) is that each note sounds exactly the same regardless of its context. They stressed that variation is essential to any musical performance. In response to that idea, we developed up-bow and down-bow Instruments. At first the presets were separated on the keyboard by three octaves to permit manual alternation of bow strokes. However, that was before the development of a standalone MIDI program that was called MaestroTools.

Jeff Hurchalla, who was one of the beta testers, brilliantly programmed MaestroTools with an Auto-Alternator feature to supplement GigaStudio's built-in capabilities (see **Fig. 5**). The Auto-Alternator feature lets you alternate up-bow and down-bow samples automatically and effortlessly, like real string players do. It also enables you to switch between tight and loose attacks for certain instruments or to switch between other types of samples.

Some beta testers also observed that an important problem with samples was the difficulty of playing legato. One beta tester posted, "If we can get [GOS] to yield an easily playable and realistic legato phrase, we will have achieved what no one has done before." Legato is a natural technique for string players, but it's difficult to emulate with a keyboard because of the lack of a naturally smooth connection between successive tones. Playing string samples on a keyboard usually creates an awkward, forced effect, especially if each sampled tone has a conspicuous swell.

The beta-team members put their heads together over the legato problem for weeks. What we needed was a method whereby only the first note of a legato phrase would have a full, natural attack, allowing the subsequent notes of the phrase to sound more closely connected. We explored and tested several options until we came up with an effective solution: the MaestroTools Legato Mode feature, which overlays "masking-attack samples" and smooths out the transitions between connected notes.

With over 888 postings spanning 160 threads, we continued to collaborate, strategize, and discuss and to follow interesting tangents. There was a constant flow of ideas. We witnessed outcomes previously thought unattainable and began to think that anything might be possible. It almost became a game for the beta testers to try to stump Hopkins and me to see what we couldn't deliver. We were continually asking, "What can we do to make this even better?" The beta team became a driving force behind some of GOS's best features, and we often relied upon the expertise of individual beta testers. It's hard to imagine how we could have pulled this off without the beta team. I thoroughly enjoyed the process.

THE HOLY GRAIL

As a result of my odyssey into the world of sample-library development, a unique and special orchestral sample collection has emerged (see **Fig. 6**). While on the journey, I learned what it takes to make a fine string library, though the newly formed friendships were perhaps the best consequence of all. The quest, however, won't end here, because I have chosen to make this an evolving library rather than a final one. In my journey, I came to realize that the Holy Grail was not simply an end result. The journey itself, the process, was the true Holy Grail.


GARY GARRITAN

FIG. 6: The GOS library was originally released on 16 CDs. As it continues to grow, however, new update discs are added to the collection.

Gary Garritan *is the creator of the* GigaHarp *sample library and inventor of the MIDI Harp. You can reach him at gary@harps.com. For more information on the Garritan Orchestral Strings library, visit www.garritan.com.*

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Illustration by David Ball

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By Daniel Fisher

six-string synthesis

The marriage of guitar to synth can be either rocky or blissful.

MIDI guitar might conjure up the image of a guitarist wailing away as a synthesizer obediently translates every nuance of picking, scraping, and bending into astounding synth leads or profoundly realistic orchestral masterpieces. Unfortunately, dreams quickly turn to disappointment if you don't understand how guitars, MIDI converters, and keyboards interact. That understanding is crucial to experiencing the full potential of MIDI guitar systems.

It's a popular myth that poor tracking and data-conversion delays are the primary reasons guitarists can't attain MIDI nirvana. Although MIDI guitar systems are much better than they used to be, the guitar-to-MIDI connection still has technical limitations. Even if MIDI guitars could instantaneously track every gesture, you'd still have to deal with the way devices transmit and respond to MIDI.

Many guitarists have exaggerated expectations of what MIDI guitar converters can do; at the same time, they underestimate the tremendous power and flexibility of synthesizers and samplers. In this article, I'll examine the relationship between MIDI guitar and synths and samplers (see the sidebar "Guitar à la Mode" for information about the modes that determine how synths respond to MIDI messages).

TAKE NOTES

Many performance techniques used by guitarists are lost in the translation to MIDI. When you play a keyboard, it initially conveys pitch and whatever results from the keystroke's Velocity—typically, volume and brightness.

On a MIDI keyboard, press a key very fast and then very slow, and notice the difference in the sound. What you hear is the only difference you can make when you play a note on a MIDI guitar. You can't palm-mute a keyboard or pick it at an angle as you can a guitar, and you certainly won't hear any effect if you strike the keys with your nails or finger pads.

Real-time MIDI performance is analogous to telling a story by writing it instead of using your voice. You cannot employ your favorite vocal tricks such as inflection, whispers, or shouts to get your point across; you need to focus on tools such as word choice and punctuation. To extend the analogy, the exact meaning of the story is also subject to interpretation on the receiving end. Depending on a receiving device's MIDI implementation, it can reinterpret, misinterpret, or ignore certain parts of the message.

AROUND THE BEND

As one of the most important control parameters, Pitch Bend can enhance or ruin your MIDI guitar experience. Pitch Bend commands tell a synth or sampler whether to bend pitch sharp or flat, but not always how far to bend it; the precise interval depends on the receiving instrument's range settings. Most of the time, you should set your MIDI sound source's Pitch Bend depth to the greatest interval you normally bend your strings and then add one more half step.

To avoid disaster, make sure your guitar-to-MIDI converter's Pitch Bend range matches your synth or sampler's Pitch Bend range. Most synth programs default to a range of two half steps, so you'll probably need to change them. Some synths let you change the range globally so that with one parameter tweak, you can match the maximum Pitch Bend range to that of your MIDI guitar converter. Some MIDI guitar converters are able to transmit Pitch Bend Change messages (CC 6, CC 38) that automatically set up the receiving synth with a matching Pitch Bend range. With other synths, you should resign yourself to setting each patch's Pitch Bend range individually.

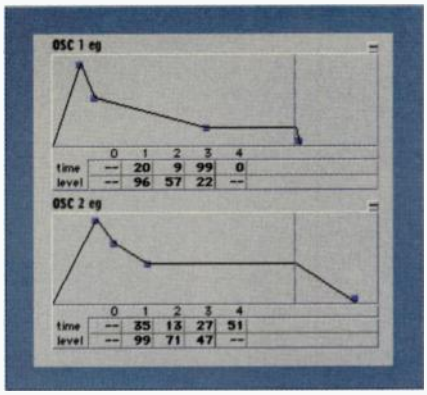

FIG. 2: Sounds with long releases offer new sonic territory for guitarists. This two-oscillator patch has been programmed with envelopes that quickly fade out a full-sounding sawtooth wave, leaving a brighter waveform to linger.

Picked or plucked guitar strings start slightly sharp and then settle into a stable pitch over time, which can adversely affect your sound source's pitch by creating Pitch Bend data. Keep in mind that some keyboard instruments, including pianos and organs, cannot bend notes, and pitch bending can sound unnatural. Although you may hate to give up one of your guitar's natural emotive tools, consider eliminating Pitch Bend information altogether for those instruments. Still, if you feel particularly adventurous, you may want to bend notes on pianos and organs anyway.

You can filter out Pitch Bend messages at several locations in the MIDI stream. Because MIDI guitars can generate a constant stream of Pitch Bend data, disabling Pitch Bend in your guitar-to-MIDI converter delivers the best results. If your MIDI converter doesn't allow you to store a Pitch Bend range for each program, you can program your synthesizer patches to ignore Pitch Bend messages by setting their ranges to zero.

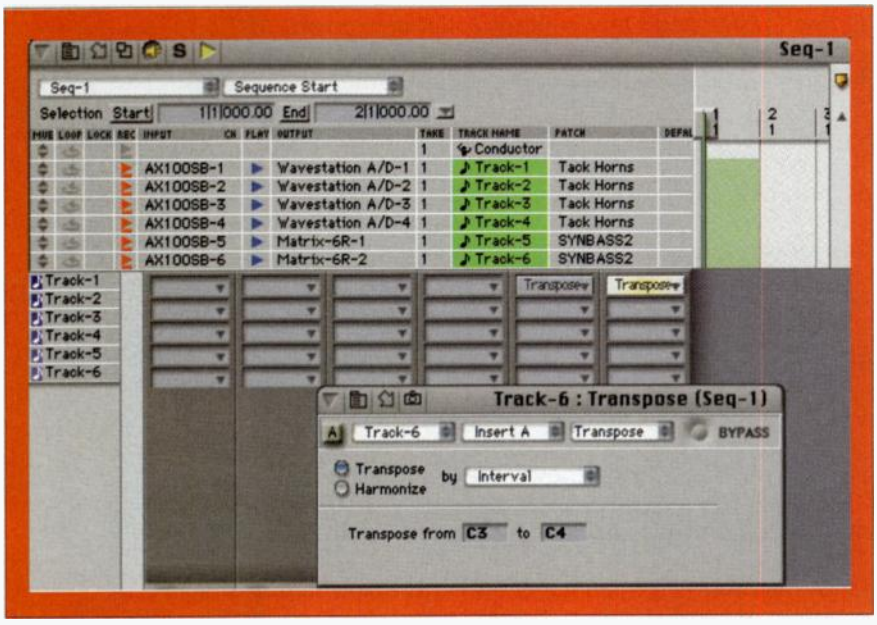

FIG. 1: Many software sequencers offer real-time processing features that can extend your MIDI guitar's capabilities. Here the MIDI guitar's strings are routed to different multitimbral synths in Digital Performer. Signals from the fourth and fifth strings are transposed an octave higher.

DIGITAL DEXTERITY

MIDI guitar controllers let you create vibrato manually; when you wiggle a string, the controller transmits Pitch

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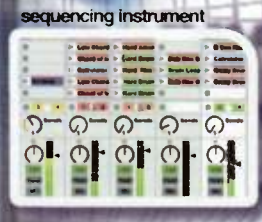

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Bend messages that translate to vibrato. Keyboard players rarely produce vibrato with their pitch-bend wheels, though; it's just too hard to do gracefully. Instead, they use modulation wheels to create low-frequency-oscillator (LFO)-induced vibrato. Sending Modulation messages to impart vibrato might seem like a weak substitute at first. However, you can smoothly change vibrato depth and speed, and a synthesizer's vibrato goes above and below the original pitch as it does with most instruments.

If you're trying to sound like a keyboard player, vibrato is usually produced with Modulation messages, but they're not the only way to add vibrato. You can control an LFO with an expression pedal or a breath controller. You can even map Pitch Bend messages to change vibrato depth so that as you bend a string farther, the vibrato gets deeper and faster.

Given that your hands are busy most of the time, any modulation effects that you can automate are a godsend. The easiest type of automation is programming your synth patch to create vibrato with a free-running LFO that fades in after you've held the note for a certain length of time.

SWEET RELEASE

Most MIDI guitar systems can easily take advantage of other MIDI Control Change (CC) messages. One immediately useful device is the sustain pedal. If your MIDI converter has a footswitch input, you can connect a sustain pedal or another momentary footswitch. Most MIDI guitar systems offer built-in pedals or input jacks for pedals that can perform sustain, hold, or other functions.

Synthesizers usually respond automatically to Sustain Pedal messages (CC 64). Such messages are great for adding a natural release to acoustic- and electric-piano sounds, for example. For indefinitely sustaining sounds, a Sustain Pedal message can hold any notes that are playing until you release the pedal. If you're careful not to overuse it, a sustain pedal can give your synth a life apart from your guitar's sound.

Even if your guitar lacks a footswitch input—as do the Parker MidiFly, Godin guitars, and Brian Moore guitars—you can still incorporate footswitches (see the sidebar "Instrument-Specific Tricks"). Most MIDI keyboards offer footswitch and expression pedal jacks, and you can automatically merge their inputs with your guitar's MIDI data. MIDI-mapping devices can turn a footswitch into the MIDI message of your choosing.

Another useful controller message is Sostenuto (CC 66). If you step on a pedal that's sending Sostenuto, the synth will continue to hold the notes that you're playing, but unlike Sustain, any notes played afterward will not be held. That means, for example, that you can hold a pad and then solo over it; you can even play a different sound.

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Sostenuto is available with several variations on many MIDI guitar systems, including Blue Chip Axon and Roland units. For instance, the Roland GR-33 provides looped drum and percussion grooves; hitting the pedal lets you sustain the loop while you solo with a different patch.

Expression pedals and breath controllers are some of the best tools for taking advantage of a synthesizer's expressive capabilities. Analog synthesizers are famous for huge, beautiful filter sweeps. Use CC messages or delayed envelopes to sweep resonant synthesizer filters. Assign Pitch Bend to change your synth filter's cutoff frequency. Bending notes and simultaneously sweeping the filter is a potent and expressive trick, and adding a delay effect with regeneration will enhance the sweeps.

Filter sweeping isn't the only trick for creating animated timbres; try mapping Pitch Bend to control pulse-width modulation, for example. Most contemporary synthesizers support CC messages in ways that can drastically alter your synthesizer's sound. Consult the MIDI-implementation chart in your synth manual to discover a wealth of possibilities.

SENSITIVITY SESSION

Adjusting your MIDI guitar controller's sensitivity is crucial to good communication between your guitar and your synth. Ideally, each string should register a range of picking dynamics from your softest strokes to your hardest strums with a continuously increasing value.

To that end, most MIDI guitar controllers offer some form of level meter to provide a visual aid when you adjust string sensitivity. Attempt to set your MIDI converter's sensitivity so that a consistent level of soft plucking triggers each string, and then ensure that the peaks match each other when you play harder so that they just barely hit the highest Velocity. That should result in a uniform Velocity response from each string.

Even with the sensitivity calibrated, it is possible for MIDI Velocities to jump unexpectedly to a high or low value. If that occurs, adjust your synth in order to reduce its amplitude and filter responses to Velocity. That will give your synth performance a more cohesive feel, but be careful not to completely squash the dynamic range.

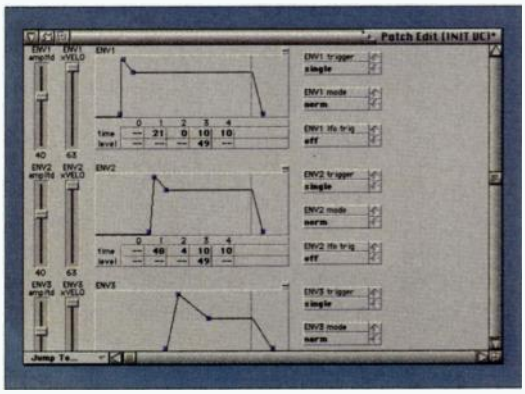

FIG. 3: You can delay your synth's response to a picked guitar string by programming amplitude envelope generators with a delayed attack. For a rhythmic effect, try creating a sound in which several oscillators have increasing delays.

REACH OUT AND TOUCH

As a guitarist, you are intimately familiar with the feel of the strings as you play them and the close relationship between your touch and the amplified guitar sound. Unless your amp volume is very high, you can also hear the vibration of the strings. Your brain uses that sensory information even before the sound travels from the speaker to your ear.

With a MIDI guitar, however, that tactile feedback is minimal or even nonexistent at first. To compensate, wear headphones or place the monitor speaker as close to your ears as possible to make it easier to ignore the physical guitar sounds. On a patch-by-patch basis, you will eventually develop a more intimate sense of touch with your MIDI guitar, and the experience will feel more tactile.

Emulating the performance styles of other instruments in terms of their note ranges and typical phrasing is an important aspect of playing synthesizers and samplers. Those elements are often more important than the degree to which the timbre accurately matches the real thing, and common guitar techniques can often run counter to an emulative performance.

For example, when reproducing piano, organ, or ensemble performances, strumming is rarely a desired effect. You should become proficient at plucking all of the strings simultaneously. Although MIDI allows you to go beyond the capabilities of any instrument's acoustic counterpart, let your ears and sense of taste be your guide.

THE OUTSIDE PITCH

By nature, most pitch-to-MIDI converters create a lag in the output of MIDI data generated from lower-pitched strings. Almost any synthesizer will transpose pitch an octave or two, however; you can usually transpose your synth's pitch globally or change oscillator frequencies at the patch level. Transposing lets you use your MIDI guitar's higher strings to play lower pitches on your synth—the higher the guitar note, the faster the MIDI converter can recognize the pitch. Playing an acoustic-bass patch from a different position on the fingerboard might take some getting used to, but tracking accuracy will definitely improve. If your MIDI converter can transpose notes, you can try this technique without having to do any synth programming.

In earlier times, some guitar controllers offered the ability to tune the output of individual strings to any pitch, allowing alternate tunings such as DADGAD or CGCGCE. Lamentably, because that feature costs more to implement and complicates the user interface, manufacturers have stopped providing it; few, if any, current MIDI guitar systems permit you to transpose individual strings by semitones. Fortunately, you can find a number of work-arounds for that shortcoming.

Any decently implemented multitimbral synth will transpose individual patches in semitones as well as allow each patch a distinct MIDI channel. After you have matched the sound source's MIDI channels to your guitar

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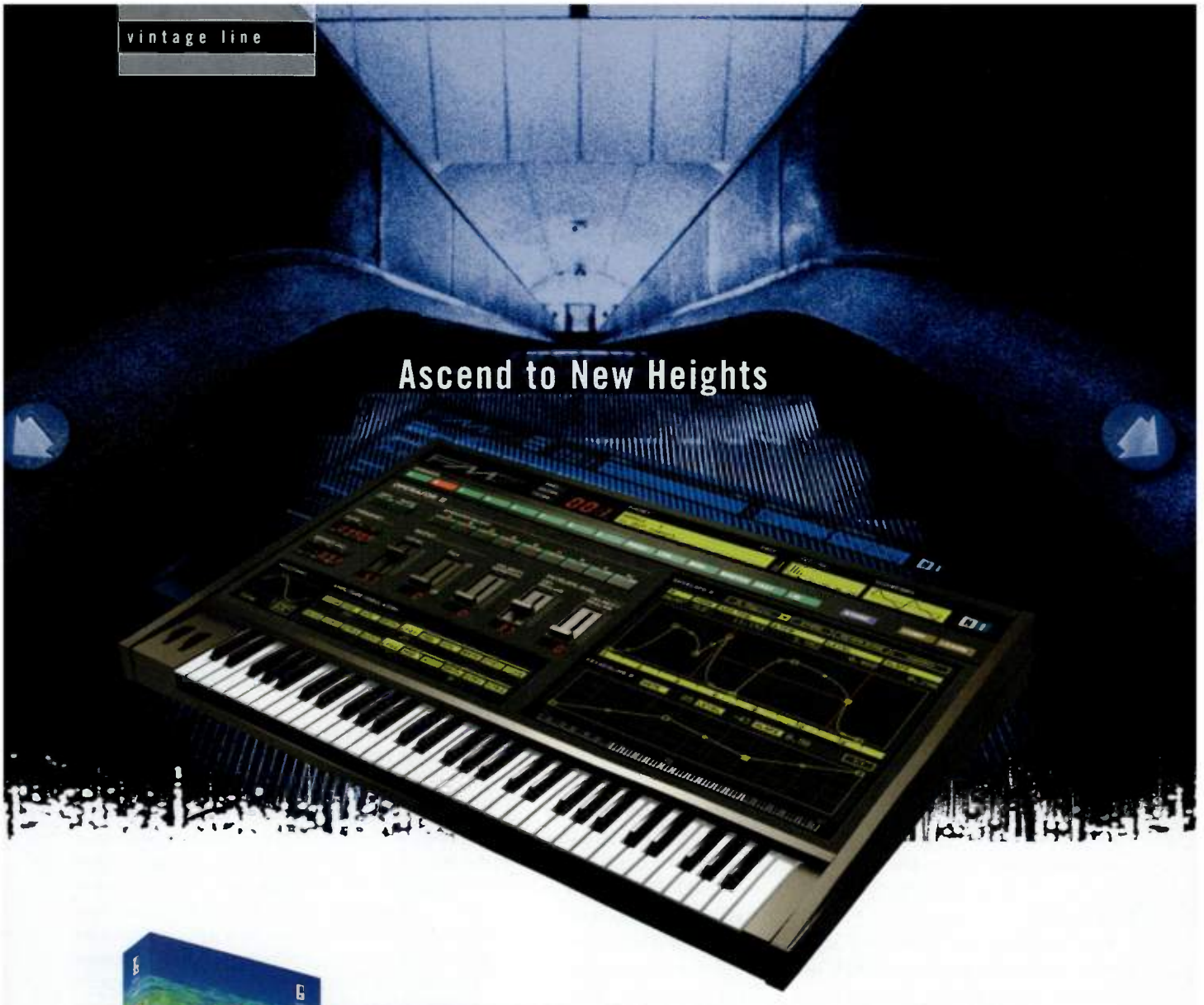

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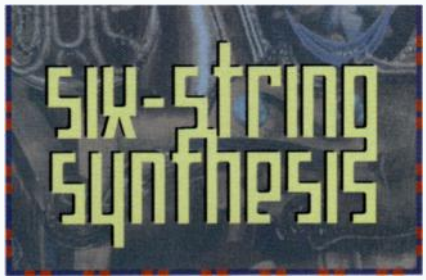

controller's string output, simply transpose each patch to taste. By programming and storing a different tuning in each multitimbral location, Program Change messages allow you to change your MIDI guitar's tuning without touching a single tuner.

If you transpose each string's MIDI output by octaves, you can create incredible chord voicings. Your guitar's six notes can be widely scattered over several octaves or bunched together into the same octave, allowing you to play cluster chords that were once the exclusive domain of keyboard players.

GET OUT THE MAP

Because samplers can remap notes, they can transpose every note from your controller to play any note you choose. That means that you can find comfortable fretting and picking patterns and then choose the pitches or sounds that result from each fret your fingers play.

For example, suppose that strumming the six open strings of your MIDI guitar sends the following notes: low E (MIDI Note Number 40), A (Note 45), D (Note 50), G (Note 55), B (Note 59), and high E (Note 64). You could assign samples in your sampler's keymap so that strumming, brushing, or picking across the strings produces a melody or riff. Although remapping notes is time-consuming, you can devise amazing parts that you control in real time.

Furthermore, most MIDI sequencing software allows you to transpose incoming MIDI data in real time, either by MIDI channel or by track. In Mark of the Unicorn Digital Performer, for example, you can specify each track's transposition value. You have to set up an individual input, track, and MIDI-channel output for each string. Be certain to save your setup as a template; once you've done that, you can quickly create a variety of alternate tunings at any time.

It's also simple to assign different sounds or different synths to any string (see **Fig. 1**). Even if you don't perform with sequenced tracks, a sequencer's potent real-time MIDI-processing capabilities make a strong case for taking your computer to a gig.

Whatever method you use to transpose your MIDI guitar's output, you'll probably want to minimize the acoustic and amplified output of the strings. Alternatively, you can create interesting harmonization effects by blending the sound of your guitar with the synth transposed to a unique interval.

ENVELOPING CONCERNS

Longer attack and release times can create musically useful atmospheric effects. The majority of synthesizer players are comfortable with the fact that some patches develop slowly when you press down the keys. Other patches start more quickly but continue to sound long after you release the keys. As a MIDI guitarist, you should learn to continue playing even if you don't hear your synthesizer the moment you pick a string.

With a little programming, you can make slow envelopes work for you in dramatic ways. Try creating a two-layered synthesizer sound with a bright, sparkling sound on one layer and a fuller sound on the other. Program a longer release time for the bright layer and a shorter release for the fuller layer. The fuller sound will fade away and leave the bright, sparkling sound to linger, thus avoiding sonic mud (see **Fig. 2**).

Some synthesizers have envelope generators that tack on a delay before the attack stage, delaying the time between a MIDI Note On and the sound's onset. By using subtle amounts of delay time (from 20 to 80 ms, perhaps), you can achieve a doubling effect between your real guitar sound and the synth. If you create multiple layers, each with a longer initial delay time (50, 100, 150, and 200 ms, for example), you'll get a very musical echo effect. Changing the waveform for each delayed layer will result in rhythmic echoes with a different timbre for each layer (see **Fig. 3**).

GUITAR À LA MODE

Perhaps the most distinguishing feature of MIDI guitar controllers is their ability to send a different stream of MIDI data from each string. Understanding that capability is crucial to a healthy relationship between your guitar and your synthesizers. The MIDI protocol conveniently offers four modes that determine how synthesizers respond to incoming MIDI messages. A MIDI guitar can transmit over a single MIDI channel or six channels at a time; synthesizers respond in one of the four modes.

MIDI Mode 1. In Omni On, Poly, your synth responds to MIDI data without regard to the originating MIDI channel. MIDI Mode 1 is a remnant of the early days of MIDI, when the major concern was simply controlling one synthesizer with another. Few modern MIDI devices implement Mode 1. As it pertains to MIDI guitar, the mode will work as long as your controller is set to address a single synthesizer with Pitch Bend disabled. If you are integrating your MIDI guitar and synthesizer into a sequencing system, avoid this mode.

MIDI Mode 2. In Omni On, Mono, your synthesizer can play just one note at a time, and it responds to incoming data without regard to MIDI channel. Again, this mode is best avoided in a sequencing system or when you have multiple MIDI devices.

MIDI Mode 3. Omni Off, Poly is the most commonly used of the four MIDI modes. Your synthesizer responds only to MIDI data on selected channels, and it can play more than one note at a time. Mode 3 is great for using your MIDI guitar with a sequencer because you can play chords.

MIDI Mode 4. Omni Off, Mono is the performing MIDI guitarist's mode of choice. Your synth responds selectively to individual MIDI channels and plays one note at a time from each channel, and your guitar can play one note per string. Monophonic performance minimizes the audible effects of glitching and prevents notes with long decays from interfering with subsequent notes played on the same string and MIDI channel. Mode 4 also works well for sequencing guitarlike performances with separate Pitch Bend from each string.

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POWER GLIDE

Portamento imparts a wonderful gliding effect to your synthesizer sounds. It's even more dramatic if you combine a synthesizer sound that has portamento with the output of your guitar; the guitar will hit a note dead-on while your synthesizer swoops into the pitch. Another benefit of portamento is that it can help smooth over possible glitches. Experiment with various types and amounts of portamento in your synth sounds.

It might take some getting used to, but playing a monophonic synthesizer patch that uses portamento is an enjoyable way to learn new expressive techniques. Brushing your pick across all six strings from low to high or vice versa, for example, causes the synthesizer to create wide pitch sweeps. You can also create an effect similar to Eddie Van Halen's powerful combination of hammer-ons and fret tapping by picking a string and then hammering on the same string or another one. Include a subtle amount of portamento to make the note transitions as smooth as possible. Because you can tap any string at any fret, the technique takes on added dimension and range. In order to get the full effect of a monophonic synthesizer, assign your controller to a single MIDI channel.

INSTRUMENT-SPECIFIC TRICKS

The Parker MidiFly and Brian Moore MIDI guitars can switch between three modes: Single-MIDI-Channel mode without Pitch Bend, Single-MIDI-Channel mode with Pitch Bend, and Multichannel-MIDI-mode with Pitch Bend.

The first mode lets you disable Pitch Bend at the source with the flip of a switch, which is handy for emulating pianos or organs. The second mode allows Pitch Bend as long as you're playing a monophonic line. If you play a second note, Pitch Bend is disabled; that's good, because sending Pitch Bend from more than one string over a single MIDI channel causes ugly, out-of-tune confusion from your synthesizer. Multichannel mode assigns a separate MIDI channel to each string.

The MidiFly provides a MIDI In jack that can merge MIDI signals from another controller with those generated by your guitar. That lets you add a MIDI foot controller, breath controller, or MIDI Program Change device.

Axon Tricks

One of the most fascinating features of the Blue Chip Axon MIDI converters is their ability to sense a pick's position between the bridge and the neck and send that data as a parameter value. The Axon manual makes several suggestions about where you might assign the data—including the mod wheel, filter cutoff, or pan—but you could use it in many ways. For example, you could send a Control Change (CC) message that changes a rack-mounted signal processor's effects depth. Try anything and everything you can imagine, and you might discover some worthwhile results.

Yamaha G50

The Yamaha G50 has several features in common with Axon MIDI guitar converters. One unique trick is its Touch Control feature, which has the ability to send any CC message you choose, based on the force of your picking. Try assigning Touch Control to the filter's cutoff frequency and crank up the resonance a bit to give your synth an envelope-follower effect.

Although the Axon and Yamaha units send CC messages by pick position, they send a discrete value for each note you play. Some CC messages are meant to be continuous (Modulation Wheel, for example) and do not fare well as discrete messages. However, both controllers let you set upper and lower modulation values, enabling a more subtle modulation effect. Better yet, the Axon provides two programmable expression pedal jacks that you can use for continuous forms of modulation. For the Yamaha G50, factor in an extra $350 for the optional MFC10 foot controller.

CONTROL FREAK

Why not let your MIDI guitar control other devices besides your synths? Program Change can radically alter your MIDI-instrument setup with the push of a button, but a Program Change sent to your synth won't necessarily select a musically meaningful patch on your effects processor.

Fortunately, most MIDI effects processors offer patch-mapping features that convert one Program Change into another. You can remap patches so that if you send Program Change No. 32 from your MIDI guitar to call up an eerie atmosphere on your synth, for example, your effects processor could call up Program Change No. 73, a complementary echo effect. With just one Program Change, your synths, guitar rig, and vocal processors can all jump to a new, custom-tailored sonic scene.

Some synthesizers can process external audio signals. Try routing vocals to your synthesizer and then modulate the filter cutoff with your MIDI guitar. Some synths can even modulate an incoming analog signal's pitch so that your voice will follow notes you play on your MIDI guitar.

Many synthesizers have built-in sequencers, pattern players, or arpeggiators that start when you play a specific note. If you set the trigger note above the normal range of the guitar, you can send a Start message by playing a harmonic. That lets you start your sequence or arpeggiator at will and still play the full range of your guitar.

A BEAUTIFUL RELATIONSHIP

I hope that this article has provided you with an improved understanding of the relationship between your MIDI guitar and your synthesizers. Once the kinks are worked out, an entirely new sonic universe is yours to explore—and it's one that can't be visited by someone who plays only guitars or keyboards.

Daniel Fisher *is an associate professor of music synthesis at Berklee College. He plays keyboards and backup guitar and sings with the tribute band Pink Voyd (www.pinkvoyd.com).*

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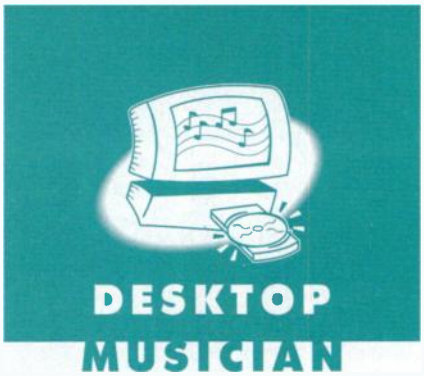

Music in a Flash

Produce bandwidth-friendly Flash soundtracks using loop mixes.

By Hayden Porter

Because of the ubiquitous presence of Macromedia Flash on the Web, many desktop musicians are interested in composing soundtracks for Flash movies. However, composers often become frustrated with the limited nature of short repetitive loops and the poor quality of Flash's streaming audio, which breaks up because of network congestion. Fortunately, there is a lesser-known type of backing track called a *loop mix*. Flash loop mixes are bandwidth-friendly (40 to 100 KB downloads) long-form compositions (typically 30 seconds to 2 minutes in length) that don't suffer from streaming dropouts.

BRUCE MCPHERSON

Flash loop mixes take advantage of the software's multitrack mixing capabilities to present an evolving soundtrack. Rather than compose a standard linear composition, you create several loop tracks and simulate a sequence by muting, unmuting, and crossfading the loops over time. The combined volume changes for each track determine the "linear" nature of the composition. Listen to audio examples 1 and 2 (drumjam.swf and 70slounge.swf, respectively) on the **EM** Web site 

to hear two styles of Flash loop-mix compositions.

In this column, you will learn how to convert a MIDI loop mix into a Flash-based loop mix. I will assume that you have the Flash authoring software (see **Fig. 1**) and are familiar with how to use the Sound panel and the Edit Envelope and Sound Properties dialog boxes. If you're new to Flash, you can download a 30-day trial version from Macromedia (www.macromedia.com).

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LOOPING LIMITS

There are two Flash authoring limitations to consider while preparing your composition. The first is that Flash Player can play a maximum of eight sounds simultaneously. That means you can have as many as eight tracks in your loop mix. Flash movies, however, often have other sounds in addition to the soundtrack. Therefore it is advisable to limit your composition to three to five tracks, allowing for playback of other audio resources, such as streaming voice-over narration and interactive button sounds. The second limitation is that the Flash authoring software supports only eight changes in volume per sound, which means that you can mute and unmute a track just twice using square-wave-shaped volume envelopes.

To maintain a reasonable file size, each looping track should be short in duration. The music should rely on the mix to reuse material and create musical variety from a small number of looping ideas. As long as you work within the necessary constraints, you can create compelling, bandwidth-friendly soundtracks for any Flash movie.

SEQUENCING SECRETS

To create a loop-mix composition, use a MIDI sequencer to compose multiple looping parts and record volume changes in separate but corresponding volume tracks. Loop tracks are commonly a few beats or measures long and are not long enough to hold all of the volume-change information. Thus, your MIDI file must have two tracks for each musical part: one to contain the loop and a second to hold the corresponding Volume controller data (Control Change 7). Download audio example 3 (70slounge.mid) from the **EM** Web site to see and hear this concept in a MIDI file.

Layering musical parts one at a time is the most natural process for composing loop mixes. It's an easy way to create variety and growth while working within the given limitations. Polymetric loops are another way to create rhythmic variety with minimal file size, because the combined pattern is longer than any one of its parts. I used that technique in Audio Example 1.

You can use other techniques, as well. In Audio Example 2, I created the intro by offsetting the bass loop by four beats in my MIDI sequence. The eight-bar bass loop emphasizes a B♭ tonality in the first four bars and an A♭ tonality in the last four bars. I wanted to solo the organ accompaniment for four bars and then introduce the bass on the B♭ tonality. To solve that problem, I offset the B♭ bars to the end of the bass line, so that when I unmuted the bass part four bars into the song, the loop was in the B♭ tonality. I also offset the melody so that its phrases would coincide with the altered bass part. That is just another example of how to compose your loop mix to maximize musical effect.

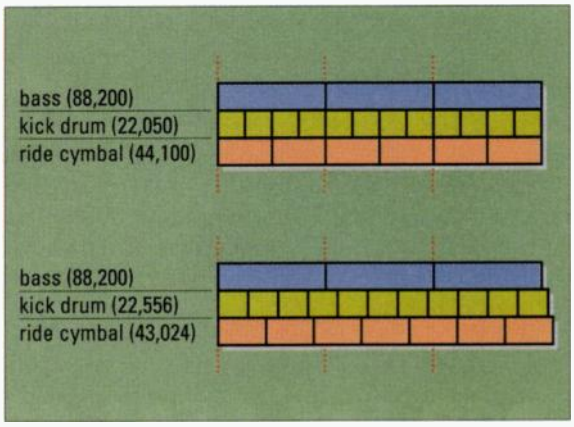

FIG. 2: The upper diagram shows how the beats line up in the different instrument tracks when the loops are properly edited. In the bottom diagram, the beats are out of alignment because the loops were not carefully trimmed.

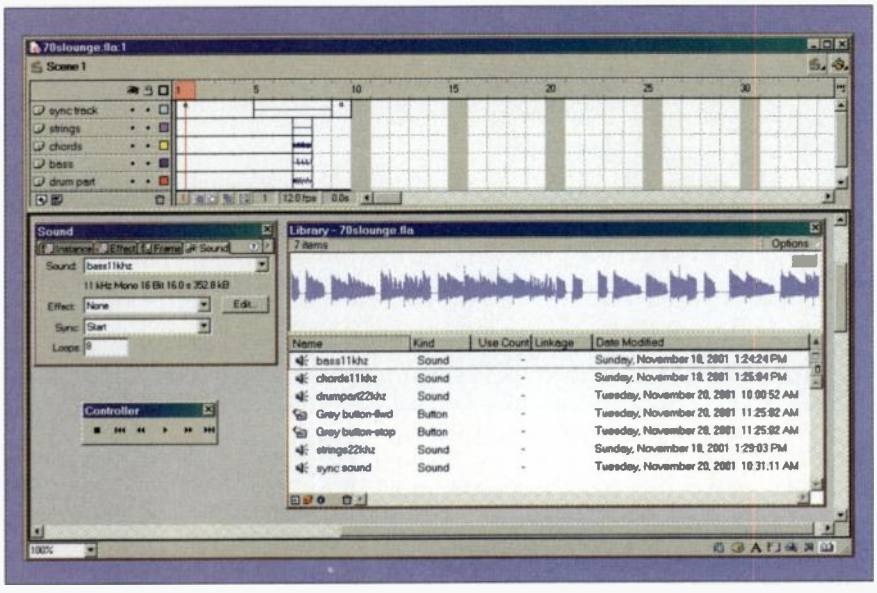

FIG. 1: In Macromedia's Flash authoring program, sounds and other elements are assigned to layers along a frame-by-frame timeline. In this example, the Sound panel, Library panel, and transport controls appear below the Timeline display.

CONVERTING WITH CAUTION

Once you finish the composition, you can begin the process of converting it into a Flash loop mix. First, record each loop track from your MIDI loop-mix composition into an audio-editing program. Be sure to disable looping for each track and mute the corresponding volume tracks before you record.

Next, edit your loop tracks for importing into Flash. The most important audio-editing issue is precisely measuring and setting the lengths of each loop. Accurate loop lengths determine whether the Flash player can sync your loop mix.

Flash uses "trigger" sync to synchronize multiple tracks; the program starts multiple sounds simultaneously and lets them play without subsequent resynchronization. If you don't accurately edit each track length, eventually your tracks

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will drift out of sync, because the downbeats won't properly coincide. Therefore, you must be precise in how you edit each loop length (see **Fig. 2**).

Use the following formula to determine the length of a loop in samples, the most accurate unit of measurement for digital sound:

(sampling rate × 60) ÷ tempo = number of samples per beat

Take the real-world example of a four-beat bass loop recorded at 120 bpm that uses a 44.1 kHz sampling rate. First, calculate the number of samples in one beat:

(44,100 × 60) ÷ 120 = 22,050 samples per beat

Next, multiply the samples per beat by the number of beats in the loop to determine the total number of samples in the loop:

4 × 22,050 = 88,200 samples in the bassline loop

In your audio editor, delete any extra samples at the beginning of the loop and set the selection range to start at 0 and end at 88,200. When you crop the selection, your loop will be the exact length necessary to remain in sync with your other loops as long as you follow the same process with the other loops.

Mix each loop to mono and downsample the loops to 22 kHz or 11 kHz before importing them into Flash. Save each edited loop as an uncompressed WAV (Windows) or AIFF (Mac) file.

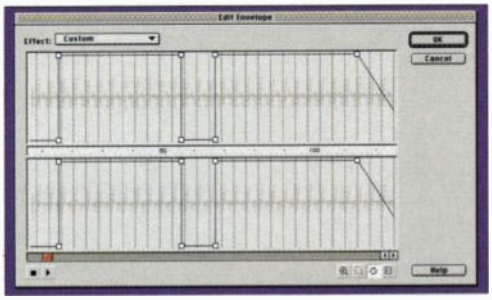

FIG. 3: The Edit Envelope dialog box lets you adjust an audio file's playback volume with as many as eight grab handles in each channel.

Your last audio-editing task is to create a short, silent 11 kHz sync sound to aid in the synchronization of the loop tracks. By setting this short track to Sync:Stream in the Flash Sound panel and having it play before and during the start of the loop tracks (overlapping the other tracks by at least a couple of frames), you can force Flash Player to improve its timing consistency. This work-around engages the player in Stream mode before the loops start to play so that all the tracks are triggered simultaneously instead of with a ragged start. Save the sync sound with your loop tracks.

MOVIEMAKING

Now you are ready to start creating your Flash movie. Start up the Flash authoring software and import each loop and the sync sound as tracks in a new Flash movie. The audio files should appear in your movie's Library window.

The following steps explain how to set up your movie's timeline. See **Fig. 1** for an example of how your timeline

Flash Bit-Rate Guide

Track Sampling Rate (kHz)	Channels	Flash MP3 Bit Rate (kbps)
44	mono and stereo	160 to 80
44/22	mono/stereo	64
22	mono and stereo	56 to 48
22	mono	32
22/11	mono/stereo	24
11	mono and stereo	20
11	mono	16 to 8

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But that's just half the story. Keyboardist arianna Hetrick also controls r synths, sound modules, and tual instruments via MIDI using the Zbus's rather deep (not that we want to ag) software control surface functions. e's able to adjust virtually any parameter any instrument (hardware or software) real-time from the EZbus's front panel and she can do it all while simultaneously erating the group's recording software th the EZbus's transport controls. And all the ile the drummer is triggering samples via the Zbus's second MIDI input. (The samples, of urse, play back through the EZbus over USB.)

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should look. Download Audio Example 4 (70slounge.fla) to see how I set up the sync sound and volume envelopes for each track.

1. Import each track and the sync sound into Flash.
2. Create and name a layer for each loop and the sync track.
3. Add a key frame to the sync-track layer and insert four more frames for a total of five.
4. Attach the sync sound in frame 1 of the sync-track layer.
5. Choose a number of loops in the Sound panel that will extend the duration of the sync sound across all five frames of this layer.
6. Choose MP3, 8 kbps for stream-sound compression in the Flash tab of the Sound Properties dialog box.
7. Return to the main window, add a sixth key frame to the sync-track layer, and attach a Stop action to prevent the entire movie from looping and prematurely replaying your mix.
8. Add a key frame to frame 3 of the remaining layers.
9. Attach each loop track to frame 3 of their respective layers.

MIXING AND MUTING

The final part of the conversion process is re-creating the volume mix for each track in the Edit Envelope dialog box (see **Fig. 3**). At this point, you should have your MIDI sequence and Flash movie open at the same time. Set your sequencer to display time positions in SMPTE time code.

At maximum zoom, the Edit Envelope display shows time values in hundredths of a second. (Click on the Watch icon to show elapsed seconds.) SMPTE time code shows durations in a similar timing system based on hours, minutes, seconds, and frames. One frame at SMPTE 30 or 29.97 fps is approximately 0.03 seconds. You can determine the corresponding hundredth of a second position in the Edit Envelope display by multiplying the number of frames in a given SMPTE position by 0.03. Follow these steps to complete the conversion process:

1. Set each track to Sync:Start in the Sound panel.
2. Set each track's loop value in the Sound panel to the same number of loops as the original MIDI track.
3. Add handles to the sound envelope in the Edit Envelope dialog box to re-create each track's Volume controller data. Make sure that the envelopes of both channels are identical.
4. Choose an MP3 bit rate in the Sound Properties dialog box based on the sampling rate for each track (see the table, "Flash Bit-Rate Guide").
5. Listen to an uncompressed version of your loop mix by clicking on the Play button in the Flash transport. Select Test Movie in the Control menu to hear the compressed version. You can stop sounds in both modes by selecting Mute Sounds in the Control menu. (For more tips on optimizing your loop mixes, see the sidebar, "Flashy Tricks.")

Deliver your finished Flash loop mix to Flash developers in SWF format or provide the original FLA file so that they can remix your music in a way that fits the needs of their presentation. Flash developers can load your sound into their existing presentation without alteration only when it's in SWF format.

The techniques that I've described are compatible with versions 3.0 and greater of Flash Player; the current version is 5.0 (as of this writing, version 6.0 is in beta). Flash Player 4.0 (and greater) supports MP3 compression; Flash Player 3.0 supports ADPCM compression only. However, the majority of Web users have at least Flash 4.0 capability, so most of the time, you can use MP3 compression without concern.

Because of the widespread nature of Flash Player, a potentially large audience exists for your Flash loop mixes, and there is also commercial potential for selling music libraries online. In addition, you may want to submit your loop mixes or Web site to Flash sound-collection sites such as Flash Kit (www.flashkit.com/loops) or Were Here (www.were-here.com).

Finally, feel free to ask further questions about this topic in the Flash Audio forums that I moderate at Sonify.org (www.sonify.org/cgi-bin/ultimatebb.cgi).

FLASHY TRICKS

To get the best results from your Flash soundtracks, follow these tips.

Audio-Quality Tips

Compression noises are often covered up by other parts in the mix.
Downsample before importing tracks into Flash to avoid its poor-quality algorithm.
Reduce the volume of a track in the Edit Envelope dialog box to minimize compression noise.
Test your mix on typical computer multimedia speakers.

Compression Tips

Bass tracks can be longer and can be compressed more than melody and accompaniment tracks.
Low-frequency timbres compress more than high-frequency timbres.
MP3 compresses stereo and mono sounds to approximately the same file size and with similar quality.
Use 3-bit ADPCM 22 kHz compression to improve the quality of cymbal sounds.

File-Size Tips

Faster tempos can shorten audio track lengths and reduce file size.
Having fewer tracks decreases the file size.
Keep the track lengths short.
Merge MIDI tracks with the same length and volume mix into a single WAV or AIFF file.
Rely on the mix to create musical variety.
Create more variety with one track by crossfading stereo channels in the Edit Envelope dialog box.

Hayden Porter *is a Web developer and musician who specializes in sound for the Web. He also writes "Remixology," a biweekly column published at Sonify.org.*

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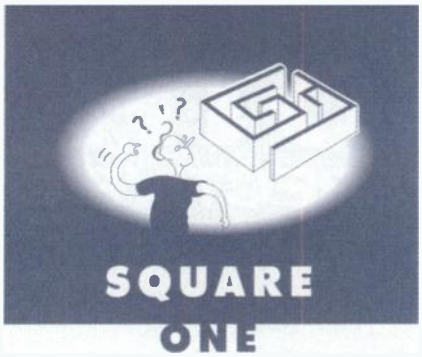

Let's Get Physical

The ultimate marriage of left brain and right brain.

By Brian Smithers

Analog synthesis can make interesting sounds, and sample-based synthesis has its charms, but here's another approach: What if you were to analyze the acoustic behavior of a few of your favorite instruments and then express all of that information as a complex set of physics equations? You'd have to consider the interaction of bow and string, reed and wood, and mouthpiece and brass. You'd also have to account for every conceivable performance technique that your favorite virtuosos have at their disposal. Then, you would solve those equations in real time in response to input from your favorite MIDI controller and sit back and enjoy the beautiful sounds that result.

Insane? Not really. That is the aim of physical-modeling (PM) synthesis, and though it is a bit of a miracle, it's actually not a new concept. The first commercially available PM synthesizer was Yamaha's VL1, which was introduced in 1993, and PM's roots date back more than a century.

PM synthesis exists at the nexus of several fields of science. It involves converting into physics equations the acoustician's observations on what makes instruments tick. Mathematicians come up with ingenious ways to solve those complex equations, and computer engineers develop the hardware and software to crunch all of those numbers quickly enough to respond immediately to a performer's input. Speaking of input, PM's multiple-variable nature lends itself to the intricacies of wind controllers and other imaginative MIDI devices, bringing one more field of endeavor to the table.

In this column, I'll take a simplified look at the fundamental logic of PM synthesis and explain why it's such a

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powerful and flexible method. I will spare you the gory details of equations and such, so the math-shy among you can relax.

GETTING EXCITED

To construct a mathematical model of an instrument's behavior, conceptualize what occurs when the instrument is played. For example, when a violin is bowed, the bow is dragged across a string, pulling it away from its resting position until the string's tension overcomes the bow's friction. At that point, the string slips back toward its resting position until the bow's friction overcomes the string's tension and starts pulling it again. This pull-and-slip cycle causes the string to vibrate, inducing vibration in the instrument's body. That vibration is the source of the violin's sound.

In PM terms, the bow scraping across the string is an *exciter*, and the violin's body is called a *resonator* (see **Fig. 1**). The exciter-resonator interaction is the core ingredient in PM, and it's easy to see how the idea applies to other instruments. A drum resonates in response to the excitation provided by a stick or a mallet striking the head, and a guitar resonates when a string is excited by a plectrum (a scientist would lose all credibility by calling it a one-syllable name such as *pick*).

Any fiddle player knows there's more to a good violin sound than a bow and a string, and your physical model continues with a look at the rest of the instrument. When the string vibrates against the bridge, it passes some of its energy on to the bridge and then to the violin body. The bridge's density causes it to vibrate less readily than the string—it has a greater *impedance*—so some vibration is reflected back to the string instead of being passed on. For the same reason, only part of the vibration is transmitted from bridge to body, affecting the way the vibration propagates.

Last, but certainly not least, the body of the violin has its own acoustic properties, and instead of merely amplifying the string's sound, the body also acts as a *filter*, shaping the tone of the string. The body's shape and size—along with the position, size, and geometry of the f holes—determine which frequencies are enhanced or attenuated, thereby determining the character of the filtering effect.

A SAXY MODEL

Now I'll take a closer look at how the four components—exciter, resonator, impedance, and filter—can be applied to model a saxophone. The excitation is initiated by the player blowing into the mouthpiece. That rush of air creates an area of lower pressure along the top surface of the reed. The low pressure pulls the reed toward the tip of the mouthpiece, restricting the airflow into the mouthpiece and allowing the pressure difference between the top and bottom surfaces of the reed to equalize. The tension in the reed from being bent is then able to pull the reed back toward its original position, where

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its tension is relieved and the airflow is great enough to start the process all over again. This tug-of-war is the source of the reed's vibration.

Therefore, the physical model of a saxophone starts by borrowing equations from aerodynamics to evaluate the force bending the reed toward the tip of the mouthpiece. Next, turn to structural engineering to quantify how strongly the reed resists being bent. Now calculate the amount of air that passes through the mouthpiece with each oscillation of the reed, and you can evaluate the intensity of the excitation.

Consider the number of variables in the reed model even before it interacts with the resonator. As the player blows harder, the reed deflects farther before it snaps back and lets more air pass through the mouthpiece, resulting in a more intense excitation. Sax players know that reeds of different stiffnesses respond differently to a given air pressure. (Saxophonists also realize that if you start trying to model what makes a reed "good" or "bad," everyone will go mad.)

Clearly, the complexity of any musical instrument defies comprehensive description. It's true that the more thoroughly you can model an instrument, the more realistic your synthesized version will be, but some degree of compromise is necessary. You can settle for a less thorough model, or you can find shortcuts in the calculations. Of course, limiting what behaviors you model can affect a virtual instrument's sound and playability.

FEEDBACK FROM A SAX?

The reed's vibration excites the resonator, the column of air contained within the body of the saxophone. As the player closes and opens different combinations of keys, the saxophone effectively changes length. The longer the column of air, the lower the fundamental frequency at which it resonates.

Think back to the exciter for a moment. If the reed's vibration excites the column of air to resonate but the resonant frequency of the column changes with its length, shouldn't the vibration frequency of the reed change correspondingly? After all, it doesn't make sense for the exciter to vibrate at one frequency and the resonator to vibrate at an unrelated frequency, does it?

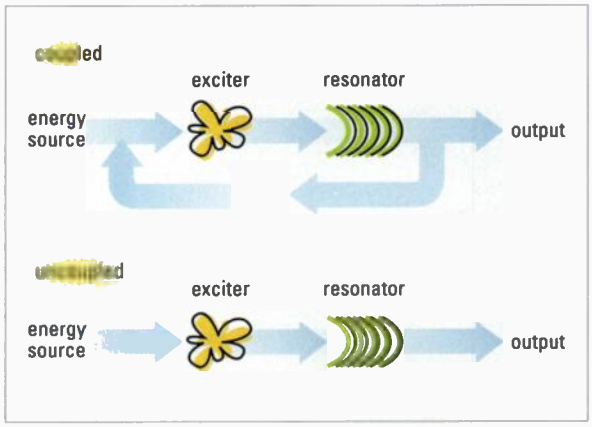

FIG. 2: Physical models demonstrate either coupled or decoupled exciter-resonator pairs. A coupled exciter-resonator pair exhibits a feedback relationship, in which the resonator influences the behavior of the exciter. A decoupled pair follows a feedforward structure, with no such effect on the exciter.

In fact, the vibration of the air column is the primary factor in determining the vibratory frequency of the reed. This is an example of a *feedback* relationship, inasmuch as the resonator's response to excitation manages to influence the behavior of the exciter (see **Fig. 2**). This type of *coupled* exciter-resonator pair is typical of wind instruments and bowed strings, in which an excitation is continuous and the resonance is periodic.

In contrast, if you strike a drumhead with a stick, the head doesn't feed any information back to the stick beyond bouncing it away. That is called a *feedforward* relationship or a *decoupled* exciter-resonator pair. Similarly, if you pluck a guitar string, the string doesn't have any opportunity to influence the behavior of the pick.

An additional aspect of the exciter-resonator relationship is that resonators are generally *linear* in their behavior, whereas exciters are generally *nonlinear*. If you apply more force to the resonating air column inside the sax, it responds with proportionately more output, or volume. A nonlinear exciter such as the saxophone reed, however, reaches a point at which additional air pressure causes it to close off completely, muting the sound, or to vibrate in an unpredictable way, causing a squeak.

RESISTANCE IS FUTILE

Just as the intersection of a violin's bridge and strings presents an impedance to the exciter's vibrations, so do changes in the diameter of a saxophone's body (or any wind instrument). However, a saxophone's conical shape results in an essentially infinite series of minute impedance changes rather than the single large impedance presented by a violin bridge (or a sudden, abrupt change in a wind instrument's diameter). The tonal effect of those tiny continuous changes in impedance is greater resonant support for even and odd harmonics. By contrast, the clarinet is essentially cylindrical with a closed mouthpiece and open bell, and its tonal spectrum consists primarily of odd harmonics.

In addition, the bends in the neck and bell of a saxophone also affect its tone. Calculating the impedance of a

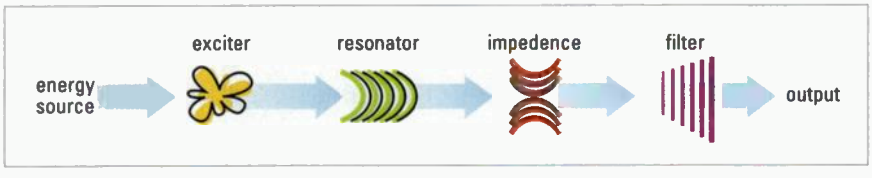

FIG. 1: This shows the four basic components of a physical model. The exciter initiates a vibration in the resonator. As the vibration passes through the instrument, it may encounter various impedances at points of transition. The shape and material of the instrument may also filter the vibration, dampening certain modes of vibration.

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saxophone, therefore, requires computing a continuously changing set of boundaries as well as the effects of several curves in the tube.

Once you've created the impedance equations, you can easily build an alto sax that's straightened out like a soprano sax or even build a straight bass sax if you wish, just by subtracting curves from the model. You don't even have to worry about the physical constraints that would make such an instrument unbelievably awkward to play. Among the many attractions of PM is its ability to create those sorts of what-if variations on instruments (see **Fig. 3**).

Any wind musician knows that the body of an instrument comes alive with the sound of the notes it plays. Whether it's made of wood, brass, or silver, it absorbs some of the energy of the vibrating column of air, transmitting part of that energy to the surrounding air and absorbing some with its own inertia. That filtering effect in the saxophone model is computed by analyzing the energy transfer that dampens the vibration of the air and contributes to the overall timbre identified as a saxophone. At the same time, you must calculate the filtering effects of the sound radiation pattern of the instrument's bell and the rest of the body.

CHOPS TO SPARE

If all of this were only a way to create convincing synthesized versions of acoustic instruments, that might be enough. After all, people build elaborate multisampled instruments that Velocity-switch from a hard to a soft attack. A PM instrument's equations allow its attack to change continuously according to Velocity, a feat that would require 128 Velocity layers in a sampler. Similarly, a PM synth doesn't merely pitch-shift a sample to sound different pitches—it recalculates the exact length of string or tube required to produce that pitch, resulting in a consistent timbre and realistic transitions from note to note.

As a saxophonist, I vary the lip pressure on the reed to bend the pitch and create vibrato. A model of that behavior has only to factor that change in reed position into the pressure/tension equation to create a realistic response to the pressure of my lip on a wind-controller mouthpiece. No more cheesy low-frequency oscillator vibratos!

FIG. 3: Applied Acoustics' Tassman is a modular virtual synthesizer built on physical-modeling principles. The Player view (top) and Builder view each show the primary building blocks of a PM synth. Impedance effects are built into the plate and controlled by the Decay knob. Each module features modulation inputs, and various other parameters can be controlled by MIDI.

If a violinist bows nearer the bridge *(sul ponticello)* or over the fingerboard *(sur la touche)*, the tone changes considerably. If your model properly represents the bow and string relationship, that performance practice can be reproduced properly, whereas a sampler would require two entirely new patches to be created. Similarly, the effect of a trumpet mute can be modeled by modifying impedance and filtering equations instead of starting a whole new program from scratch. That flexibility permits PM to reproduce the wide range of performance techniques employed by acoustic musicians in an easily controlled way.

A physical model can, by its very design, reproduce some of the most human effects of acoustic instruments, from the breathiness that saxophones get at low volumes to the squeaks they produce when overblown. The number of sample layers required to achieve that range of expression is beyond the capabilities of even the best modern samplers.

SONIC HYBRIDS

Some of the most interesting aspects of PM, however, arise when you reach beyond the re-creation of existing instruments and apply a bit of imagination. Have you ever wondered what it would sound like to bow a bassoon? Once you've reduced exciter and resonator to their component equations, who says you can't cross-pollinate them?

Why not build a trombone whose size decreases as its pitch rises? By the time you get into the treble clef, you're playing something more like a trumpet. Speaking of trumpets, the instrument used by Wynton Marsalis is designed to be much more rigid than other trumpets, greatly reducing the energy radiated from anywhere other than the bell for a more intense sound. Why not test that theory by designing a PM trumpet made entirely of granite?

By the time you get around to designing imaginary exciters and resonators, you'll have a palette of amazing complexity and potential. However, that complexity turns out to be PM's biggest limitation. The "classical" approach I've described is elegant in its conception, but it requires solving an enormous set of interrelated equations to create each sound; in fact, it's often called the *brute force* method. Even with the latest processors, you can't quite pull off PM in that manner in real time.

As a result, several simplified approximations have been developed. The most common approach is called *waveguide synthesis*, which is employed in currently available PM synthesizers. Another approach is called *modal synthesis*, which divides complex physical and tonal structures into smaller, more manageable substructures. In any event, PM synthesis opens a whole new world of sonic generation and manipulation for electronic musicians to explore.

Brian Smithers *is associate course director of MIDI at Full Sail Real World Education in Winter Park, Florida. You can reach him through his Web site at http://members.aol.com/notebooks1.*

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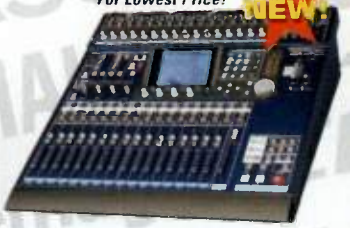

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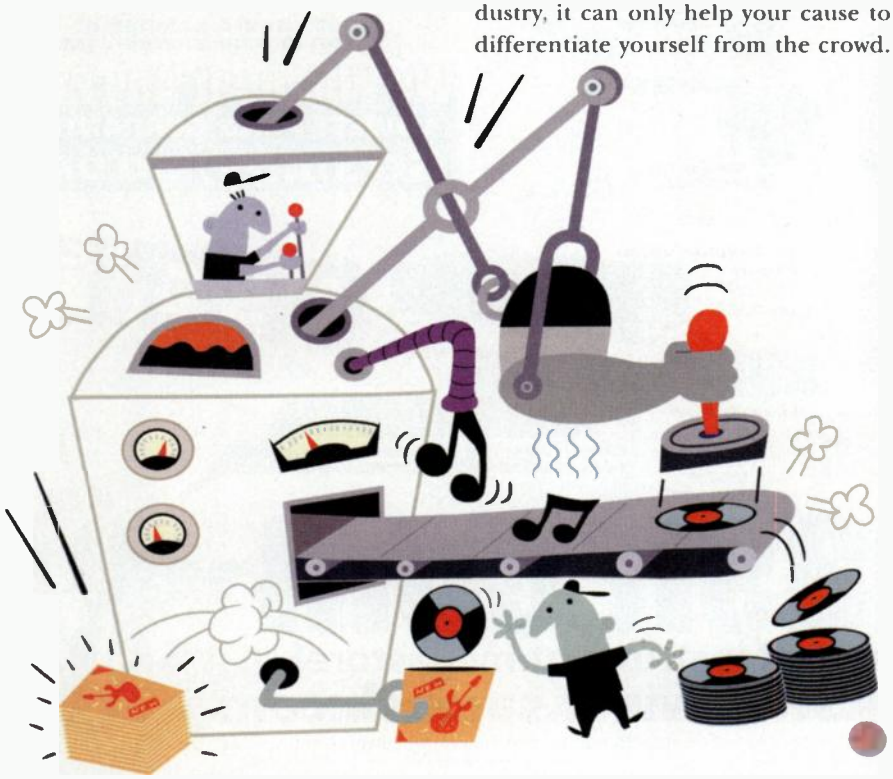
Pressing Matters

Add a new dimension to your career by pressing a 12-inch.

By Markkus Rovito

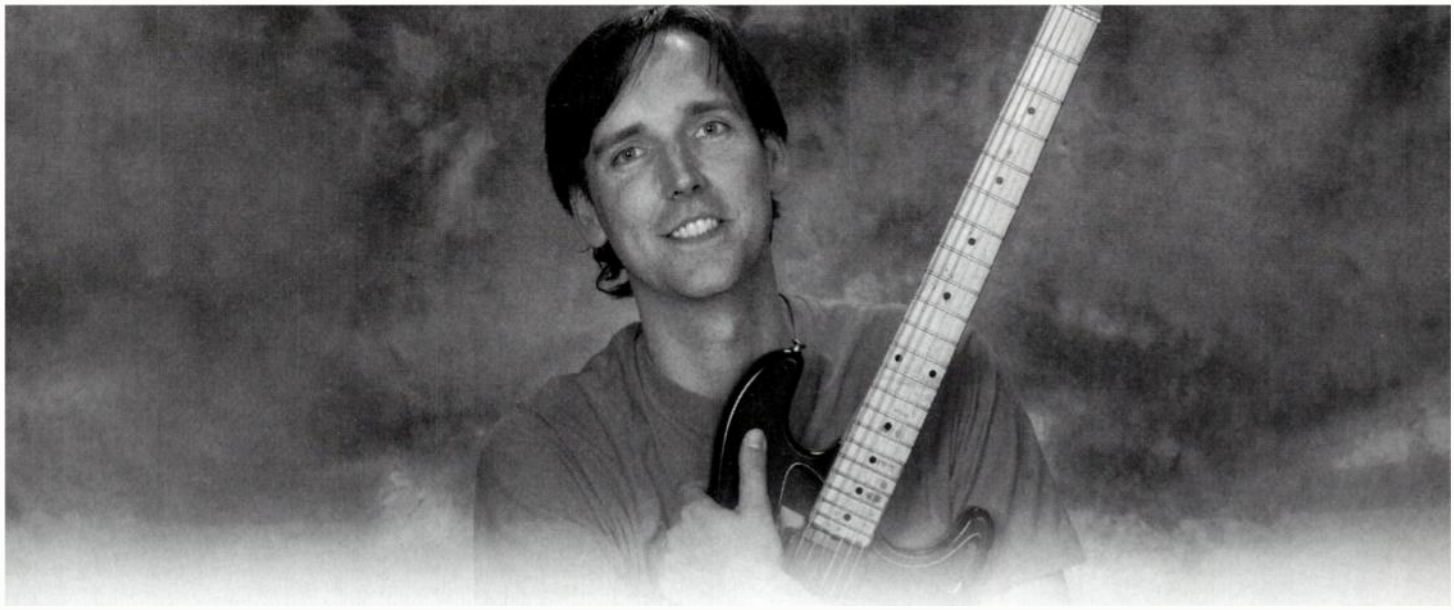
The days of the demo tape are finished. CDs have taken over as the preferred demo- and DIY-album format, and producing them is easier and cheaper than ever before. Almost every musician has a CD of his or her music—and that's the problem. When you reach the stage of promoting your music to the public and the industry, it can only help your cause to differentiate yourself from the crowd. Pressing some of your music to a vinyl record is one way to do that.



JACK DESROCHER

These days burning a CD is so simple that it no longer automatically indicates that an artist has a serious commitment to the music. Pressing a 12-inch record, on the other hand, shows at the very least a financial sacrifice and suggests that time was taken mastering the music to a high standard. If you've been paying attention to record labels lately, you know that most of them are no longer interested in cuts that are "demo-quality"; labels want demos to be completed or nearly completed.

Pressing a vinyl record should not be thought of merely as a ploy for attention. There is a serious market for vinyl. Turntables have definitely reentered the mainstream consciousness as a result of the explosion of the DJ culture, and vinyl never went out of style with the punk and indie underground. For aspiring hip-hop or dance-music artists, making a 12-inch record is an almost inevitable rite of passage into a more established place in the business. Sure, you could wait hopefully for the labels to bite, but for the price of a new synthesizer or multitrack digital recorder, you could press 500 to 1,000 12-inch singles suitable for sending to DJs, radio stations, and record labels. Your marketing



"Four Major Labels Came to See Me Because I Joined TAXI"

Lizard McGee -- TAXI Member

Most musicians never get a chance to meet an A&R person in the flesh. I had A&R guys from Columbia, Dreamworks, Maverick and Hollywood all come to see my band, Earwig, play live.

I spent the next day hanging out with one of them at his house. I played more songs, and we talked one-on-one for hours.

All this happened as a direct result of becoming a member of TAXI.

Ironically, I almost didn't join. Like so many other people, I didn't know a lot about TAXI, and I wondered if it was really legitimate. It just sounded too good to be true.

But I spoke with a few friends who were already members, and they explained how TAXI worked. It made sense.

I began to think about not only getting my music to record labels and publishers, but also pitching my songs to TV shows and movies to make some extra money with my music.

So, I joined, and it's already paying off big-time. Earwig is building a huge buzz because of all the contacts we've made through TAXI.

We haven't signed a deal yet, but we've definitely penetrated the so-called "inner circle" of the music industry. And that's exactly where you need to be to get yourself signed.

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Just ask for their free information kit, and get yourself signed up in a hurry.

I did, and my only regret is that I didn't do it sooner. TAXI has turned out to be the best investment I've ever made in myself.

strategy is an entirely different subject, but here's a look at some of the logistics of pressing a record.

MASTERING FOR GROOVES

Record pressing involves two mastering processes: the mastering of the source material to DAT, CD-R, or another format that a pressing plant will accept; and the transferring of the source material to a lacquer and then to a metal stamper that presses the vinyl. You need only concern yourself with the first part. Some pressing plants double as mastering houses and will create a master for you from your mixes. That will add considerable cost to the job, but certain companies will include mastering as part of an entire record-pressing package.

If you are mastering your own source material, here are a few suggestions that can help you prepare a digital source such as DAT and CD-R for pressing to an analog record. (For detailed information about mastering, see "Mastering on a Budget" in the October 1998 issue of **EM** and "Mastering Continuity" in the October 1999 issue.)

One thing to keep in mind is that vinyl records reproduce the frequency spectrum differently from the way that CDs do. As you would suspect, the system you use to play back the record (including the turntable and the speakers) plays a critical role in determining what you hear. The better the system, the more of the frequency range you can recover from the record.

Although some people will argue that the theoretical frequency range of a record surpasses that of the compact disc, it's difficult to get mastering engineers to agree on the exact limits of the range. Engineers concur, however, that the practical upper limits of a vinyl record are in the range of 16 to 18 kHz for albums destined for audiophile-quality systems and 8 to 16 kHz for the average reproduction system. The upper limit depends on the physical position of the music on the record itself as well.

Distortion will result if the reproduction system cannot adequately handle the frequency range of the record that is being played. Make sure to keep this in mind before boosting anything in the 8 to 16 kHz range as you mix. You should also consider using a de-esser on high-frequency sources such as vocals and cymbals.

On the flip side, a well-mastered record can reproduce lower frequencies better than a CD. For this reason, bass-heavy music is particularly well suited to vinyl—that's why DJs often prefer it for thumping club music. Problems can occur, however, if musicians, wanting to hear the bass the way they like it, overcompensate in the bass frequencies when they mix to CD-R. When such mixes are pressed to vinyl, the bass information can end up so loud that the stylus skips during playback.

In addition, higher bass levels require more physical space on the record, so for bass-heavy music, you need to limit the amount of music on each side of the record. Typically, the length of each side of a 12-inch record should be 10 minutes (optimum) to 12 minutes (maximum) at 45 rpm, and 16 minutes (optimum) to 24 minutes (maximum) at 33⅓ rpm. For dance and hip-hop with heavy bass, however, I recommend that you keep it to 6 to 8 minutes (optimum) or 9 minutes (maximum) at 45 rpm, and 8 to 10 minutes (optimum) or 12 minutes (maximum) at 33⅓ rpm. Longer times may necessitate a drop in levels.

GINO ROBAIR

This engineer's cutting lathe at Fantasy Records in Berkeley, California, is used to master a lacquer from the artist's source.

CHILL ON THE INSIDE

Another element that plays a part in the sound quality of the pressing is where a track sits physically on a record. A groove at the outer edge of the record can have a circumference as great as 36 inches, whereas the circumference of grooves toward the inside of the record can fall to fewer than 15 inches. These distances are covered in the 1.8 seconds it takes for one turntable revolution at 33⅓ rpm. As the grooves of a record approach the center, the same amount of information must be squeezed into a smaller space. As a result, music at the beginning of a record side sounds more pristine, and the odds that you'll get treble distortion increase the closer the grooves are to the record's center.

Dance-music 12-inches often have the banging club mix on the first side, with a more mellow, chill-out version on the second. You should follow suit on your own record by putting the louder, busier, more bass-heavy music at the beginning of a side and quieter music at the end of the same side.

Although you should always check with the record plant about how to prepare your master source for pressing, a few details are customary. If you want any changes made to the levels or dynamics of the source, be specific about what those changes are and type out the details on a sheet of paper. Also include a track sheet with the names, track numbers, start IDs, and lengths for each song, as well as the total side time, which should include the pauses between songs. The pause time between songs on the record will be whatever it was on the master.

MARK SMITH

Colored vinyl helps records stand apart from the rest. The additional cost for pressing a colored vinyl record can run from $0.25 to $0.50 per record.

CUTTING COSTS

Depending on the record-pressing service you use and the number of options you choose, the final pressing cost should fall between $750 and $1,300 for 500 12-inch 130-gram (130g) records, and between $1,100 and $2,000 for 1,000 records. Generally, these prices would include the master lacquer, metal processing, record pressing,

test pressings, labels, shrink-wrap, jackets, and possibly paper sleeves. The cost per record is much lower with greater quantities because the most expensive parts of the process, mastering and plating, are fixed costs that occur regardless of the number of records pressed. Ordering another pressing of the record is usually much cheaper, sometimes by as much as 50 percent. You could also spend a bit less by cutting down on the number of extra options.

EXPRESS YOURSELF

Using colored vinyl, including swirled or marbled vinyl, adds a distinctive look to your record, and colored vinyl is also known to sell better than black vinyl. Colored vinyl comes at a cost as well. Plan to spend $0.25 to $0.50 per record extra for colored vinyl. White, patterned, and custom colors are the most expensive. Depending on the company, there may also be a setup fee for ordering colored vinyl.

The aforementioned pressing prices are for the standard vinyl weight of 130g. Stepping up to a heavier-grade vinyl results in all-around improved sound. With 180g vinyl, you can count on a better stereo image, boomier bass, sharper treble, and less noise and distortion. Be prepared to pay dearly, however. Again, costs vary, but if the cost per 130g 12-inch is $0.78 to $0.89, 180g vinyl costs around $1.50 to $2.10 per record while fixed costs remain equal. Slightly higher grade 140g vinyl is another option, with prices sometimes only a few cents per record higher than the 130g price. Whereas 180g records are considered audiophile quality, some outlets do make 220g records available.

If sound quality is a top priority, make sure you order records made from 100 percent virgin vinyl. Some pressing plants will use recycled vinyl unless the customer requests otherwise. Virgin vinyl is more consistent and offers better audio performance. Just be aware that ordering virgin vinyl will put your cost in the upper tier of the aforementioned price ranges. But as with everything else in life, you get what you pay for.

Some record-pressing companies guarantee that you'll get virgin vinyl or they offer audiophile packages with virgin vinyl and, usually, a heavier-weight pressing. Other companies may offer only one set of prices, so check with them to find out whether they use 100 percent virgin vinyl.

Remaining considerations include labels and dust jackets. Most package deals include a white or black dust jacket with holes in the center and white labels with black print. If you have the money to go all out, you can order full four-color artwork for the jackets and labels. These prices are often quoted by the pressing plant for each customer.

VINYL THOUGHTS

Even with the improvement of CD recording and playback technology, vinyl records still offer advantages to DJs, favorable sound for bass-heavy music, and an intangible cool factor in a number of music scenes. Because they sell well at gigs and suggest a pro attitude to industry types, vinyl stands to be a major part of any artist's self-promotion. Although its mainstream days are long gone, the 12-inch record's cult status with DJs worldwide ensures that it will be a viable option for years to come.

KEEPERS OF THE LATHE

Here is a selected list of record-pressing services.

Acme Vinyl Corp. (Ontario, Canada)
tel. (905) 470-2937; Web www.acmevinyl.com

Disc Makers
tel. (800) 468-9353; Web www.discmakers.com

Disco Press (Naples, Italy)
tel. 39-81-839-2522 ; e-mail info@disco-press.net; Web www.disco-press.net

Europadisk
tel. (800) 455-8555 or (718) 407-7300; Web www.europadisk.com

Independent Pressing Company (London)
tel. 44-208-762-9988; e-mail info@independentpressing.com;
Web www.independentpressing.com

MaxDisk
tel. (866) 629-3475; Web www.maxdisk.com

Musicol Recording
tel. (800) 240-5963; e-mail info@musicolrecording.com; Web www.musicolrecording.com

Rainbo Records
tel. (310) 829-3476; e-mail info@rainborecords.com; Web www.rainborecords.com

Recordpressing.com
tel. (415) 701-9436 ; e-mail help@recordpressing.com; Web www.recordpressing.com

Record Technology
tel. (805) 484-2747; Web www.recordtech.com

Techlyn Records Pressing LLC
tel. (718) 369-7606 ; e-mail techlyn@techlyn.com; Web www.techlyn.com

Trutone, Inc.
tel. (201) 489-9180; e-mail sales@trutone.com; Web www.trutone.com

United Record Pressing
tel. (615) 259-9396; e-mail united@urpressing.com; Web www.urpressing.com

World Media Group
tel. (800) 400-4964 or (317) 549-8484; Web www.worldmediagroup.com

Markkus Rovito *(mrovito@earthlink.net) is a senior editor for* E-Gear *magazine and a bedroom studio musician.*

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REVIEWS

YAMAHA

9000 PRO

Is auto-accompaniment finally ready for the big time?

By Julian Colbeck

Yamaha's 9000 Pro keyboard workstation is a grown-up arranger-style keyboard. At its heart, it has auto-accompaniment patterns, but with the added sophistication of sampling, superb effects processing, a plug-in-board option, vocal harmonization, computer-based editing, and a multitrack sequencer with which you can create your own backings and styles.

The 9000 Pro is based on the PSR9000, a home keyboard that has a built-in speaker system. Even without built-in speakers, the 9000 Pro is still very much an auto-accompaniment instrument—a breed that EM has seldom mentioned in the past.

Auto-accompaniment instruments deal in Styles—complete musical backings (bass, drums, chords, and obbligato parts) whose voicings you can manipulate in real time by playing different notes and chords on the keyboard. Onboard the 9000 Pro are 125 Styles, from rock shuffles to an entire category called Ballroom.

Germany is the place where Styles rule supreme; whole aisles of the Frankfurt Musikmesse are taken up by companies that produce nothing but third-party Style content for instruments from Yamaha, Roland, Technics, and others. Accordingly, among some reasonably hip and "American-sounding" beats and grooves is an awful lot of oompah-ish material, the

FIG. 1: Yamaha has years of experience making the PSR series of auto-accompaniment keyboards. The 9000 Pro is an offshoot of that product line and has a feature set designed to appeal to professional musicians.

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appeal of which beyond the *bierkellers* of the Black Forest is questionable. The 9000 Pro is tailor-made for the German market.

As a concept, Styles are definitely useful to their proponents, ranging from the pianist at a Holiday Inn to church organists to budding songwriters in Nashville. Honestly, most keyboardists and composers could use a little help from time to time.

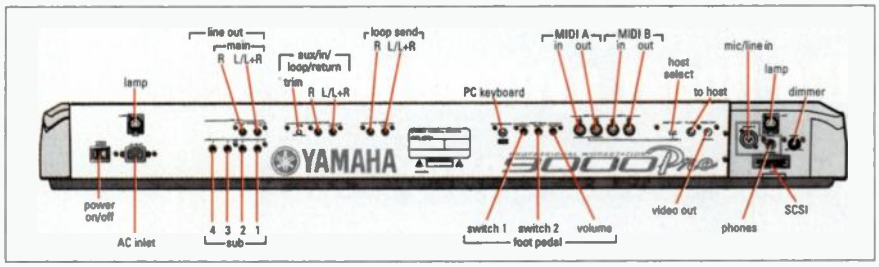

FIG. 2: The 9000 Pro's crowded rear panel has jacks for analog audio I/O, two gooseneck lamps, a PC keyboard and video monitor, footswitches, a volume pedal, SCSI, MIDI, and a direct connection to a computer.

ON THE OUTSIDE

Silver is the "in" color scheme, and the 9000 Pro sports a two-tone silver livery breaking up a dense front panel (see **Fig. 1**). Weighing 45 pounds, the instrument is surely one of the biggest in Yamaha's Portable Keyboard division, and it feels like it's going to last. Panel hardware is excellent, featuring large squishy buttons, almost all with LED indicators. The display is huge and attractively blue, though it doesn't respond to touch; it just looks like a touch screen. You navigate using the data wheel, increment/decrement buttons, and buttons adjacent to the display.

I had no problem maneuvering around the panel. My only niggle is the location of the ¼-inch stereo headphone jack, which is tucked (and I do mean tucked) in back beneath the socket for one of two optional gooseneck lamps. I actually had to consult the manual to find the headphone jack, which is a bit silly.

The 9000 Pro's rear panel offers a wealth of I/O possibilities, including two unbalanced ¼-inch main outputs, four unbalanced ¼-inch individual outputs, and a pair of ¼-inch aux sends and a pair of aux returns with a single trim knob to adjust their sensitivity, but no digital audio I/O (see **Fig. 2**). Inputs for a PC keyboard, two footswitches, and a volume pedal are also provided. In addition to two pairs of MIDI ports, a To Host port with a selector switch lets you connect the 9000 Pro directly to a Mac or PC. To the right of those are a video output, a combo XLR mic and ¼-inch line input for sampling, the headphone output, and a SCSI port. XLR jacks for the aforementioned gooseneck lamps are located at the extreme right and left sides, along with a Brightness knob on the right.

The specifications in the owner's manual claim that the 76-note keyboard has weighted keys, but it's not weighted in any sense that I could recognize. Still, it feels terrific—fast enough for all organ, synth, and orchestral tones, yet with enough feel for reasonably serious piano playing. (Keyboard feel is an intensely personal issue that's heavily influenced by the quality and Velocity response of the sounds.)

The 9000 Pro comes with a music stand that clips into two holes on the front panel. Sure, it looks a little homey (as in domestic), but it's handy, and if you don't want it, you don't have to attach it.

PIANOS AND POLYPHONY

The 9000 Pro's main piano sound is nothing less than a joy to play. Yamaha has made huge strides in piano sounds during the past two or three years. In the 1980s, Technics ruled that particular roost, but Yamaha is hitting a purple patch with the S80, Motif, and 9000 Pro (though they don't all use the same samples). I simply can't imagine why anyone would need to spend thousands on a set of piano samples these days.

Pianos are an important consideration, so an examination of all of the options is in order. In addition to the stellar Live! Grand (Yamaha's exclamation point, indicating a sound that uses stereo samples), the 9000 Pro features a marginally brighter and lighter Grand Piano and a much brighter Brite Piano. The single- and double-octave harpsichords (GrandHarpsi) are so good, they're almost edible. Whereas most synth harpsichords are lame and lifeless, these ring; they have depth and presence. All but the pickiest of purists are going to have great fun with those two selections.

Along with the obligatory honky-tonk piano comes the highly authentic Rock Piano (a Yamaha CP70), a MIDI Grand (a piano with added sparkle), a two-octave piano, a four-octave piano, and a CP80. (The CP80 is a little fuller than the Rock Piano, but the difference isn't earth-shattering.)

The Electric-Piano category boasts 18 more sounds, ranging from lovingly recreated Rhodes and Wurlitzer patches to some DX clones and a wonderful Clavinet. Considering the high quality of the harpsichords and Clavinet, it's a shame there aren't a few more. Fortunately, you can create your own, if for no other reason than to add Aftertouch-controlled pitch bend, which is an effect that a real Clavinet does automatically and that the 9000 Pro allows you to program.

Polyphony maxes out at a very acceptable 126 notes. The number of notes you can play simultaneously is reduced when you use auto-accompaniment, of course.

OTHER VOICES

The 9000 Pro's 48 MB of sound data span 13 main banks and a complete XG bank (Yamaha's turbocharged General MIDI [GM] sound set). Aside from the XG bank, which encompasses everything, the main categories are clearly slanted toward the auto-accompaniment player, and toward the European one at that. Hence, you'll find 27 entries in the Organ bank, 11 in Accordion, 62 in Guitar and Bass, 19 in Trumpet, 52 in Choir and Pad, and 33 in Synthesizer.

The scope may be focused, but the

Mixing on normal monitors is like trying to enjoy a gourmet meal with a head cold.

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overall quality is quite astounding. I was especially impressed by the stunning realism of the Sweet Flute, which has an overblown high harmonic when you add Velocity (the vibrato, even if it is gorgeous, is unfortunately in the sample). Other standouts include the Live! Nylon (guitar), with its mellifluous basic tone and Velocity-dependent harmonic, as well as the brilliant Aloha Gtr (Hawaiian) and sturdy Folk Gtr, which is crisp and without a whisper of tinniness in the higher registers.

An interesting subcategory of sounds called Organ Flutes kicks in a modeling-based sound engine whose features can be applied to one of the instrument's multitimbral Parts. Organ Flutes parameters are separate from normal Voice-editing parameters and are specific to organs: organ type, rotary speed, vibrato, harmonic volume and envelope, and so on. The regular organ presets are not bad, but the Organ Flutes take both realism and control to a whole new level. It's a shame that there was no room to add some physical drawbar controllers.

The 9000 Pro offers a healthy 24 drum kits, including three Live! kits from the PSR9000: Cuban, Pop Latin, and Brush. A few years ago, I thought Yamaha drums were a little weak compared with drums from Korg and Roland, but not any more. Thanks to meticulous Velocity mapping, generous sample times, and stereo recording, many of the 9000 Pro's drum kits are as good as anything you could put

9000 Pro Specifications

Sound Engine	AWM2 (sample playback); physical modeling
Keyboard	76-key; transmits Velocity, Channel Pressure
Polyphony	126 notes + polyphony of any plug-in boards
Multitimbral Parts	28 + parts from any plug-in boards
Voice Memory	ROM: 342 preset + 480 XG + 26 drum kits + 10 Organ Flutes; RAM: 32 user + 10 Organ Flutes
User Setup Locations	512
Waveform ROM	48 MB, Linear Predictive Coding (LPC) compressed
Sample RAM	1 MB standard; 65 MB max.
Sampling Rate	44.1 kHz; 16-bit mono
Sample Import Formats	AIFF; WAV
Effects	(6) global effects (54 types); (2) insert effects (164 types); (1) user-configurable (global/insert) effect; (1) global 5-band EQ
Sequencer	(16) tracks; (38,000) notes
Real-Time Controllers	(1) pitch-bend wheel; (1) modulation wheel
Audio Outputs	(2) unbalanced ¼" TS main; (4) unbalanced ¼" TS sub; (2) unbalanced ¼" TS aux send; (1) ¼" stereo headphone
Audio Inputs	(1) combo connector: XLR mic; unbalanced ¼" TS line; (2) unbalanced ¼" TS aux return
MIDI Ports	(2) In, (2) Out
Additional Ports	(1) 50-pin SCSI; (1) RS-232; (1) Mini DIN ASCII keyboard; (1) RCA NTSC/PAL composite video out; (2) XLR gooseneck-lamp sockets (with dimmer knob); (2) footswitch; (1) volume pedal
Expansion Board Slots	(2) Modular Synthesis Plug-In System (PLG series)
Removable Storage	(1) 3.5" HD/DD floppy drive
Optional Storage	(1) internal 8 GB hard drive
Display	240 × 320-pixel backlit LCD
Dimensions	49.9" (L) × 5.5" (H) × 16.1" (D)
Weight	45.2 lb.

together from a top-quality sample CD, but without the effort and expense.

SOUND CONTROL

Yamaha isn't marketing the 9000 Pro as a professional synthesizer, but the instrument offers several pro-level features. In addition to being able to quickly edit sounds internally, you can do more precise editing using a software editor supplied on CD-ROM.

Elements, which are effectively complete little synthesizer modules, are at the heart of Voices. Each Voice uses as many as eight Elements. For each Element, you can select from a pool of waveforms (including user samples) and tonally adjust them with two separate filters and three multistage envelope generators (EGs) that control filter cutoff, pitch, and amplitude. The four filter types are lowpass, highpass, bandpass, and band-reject. The solitary LFO offers only sawtooth and triangle waveshapes, which was a bummer when I needed a square wave to replicate a trill for the 9000 Pro's excellent mandolin Voice.

Although the instrument offers two types of Voice editing—Easy Editing (in which you dial straight into filter cutoff and resonance, abbreviated EG controls, and vibrato) and Full Editing—I doubt that 99 percent of 9000 Pro users will go anywhere near the Edit button. If Yamaha ever produces an intelligible step-by-step guide to Voice editing, users might dare attempt to edit Voices. Without that, there's not a chance in the world.

VOICES WITHIN VOICES

The 9000 Pro's vocal processor does a good job of emulating different styles, sexes, ages, and numbers of harmonizing singers. It can also add vibrato and usually keep you on key, because the Vocal Harmony feature is linked to your left-hand chord voicing.

I could probably write an entire review of the 9000 Pro's vocal-processing features, as the number of permutations borders on ridiculous. Briefly put, you plug in a microphone (there is mic/line level input selection and input volume control) and hit the Vocal Harmony button, and a whole world of spooky control and vocal accompaniment opens up to you.

The parameter list offers some delightful names and features, including gems such as Lower Gender Depth, Lead Pitch Correction, and Auto Upper Gender Threshold. You can set the male/female harmony type and where harmonies occur, as well as their volume, panning, tuning, vibrato, and delay factor. There's also a 3-band EQ, a noise gate, and a compressor for Vocal Harmony use.

Does the vocal processing sound natural? I'm sure I'd get better results with careful parameter selection and manipulation, but basically, it sounds great. Hey, now I can sing!

STYLIN'

Why it is that the concept of auto-accompaniment is terminally unhip, whereas the notion of retriggering backings that other people have produced (such as sampled loops) has become totally cool, is one of life's little imponderables. Auto-accompaniment began life in fox-trot and beguine territory, of course, but Styles have progressed tremendously since then, largely thanks to Korg's i-Series of instruments.

The 9000 Pro provides 125 preset Styles and, once you've loaded them, 87 more that you can access from the instrument's flash ROM. Some of the presets are way cool, no question. I grooved away for minutes on W.EndShuffle, a silky, atmospheric backing highly reminiscent of the Pet Shop Boys' "West End Girls."

I've noticed the arranger-keyboard fraternity expects Style providers to trawl popular songs for inspiration. That fixation on existing song material is annoying and one of the reasons why such keyboards have a bad rap among pro musicians. If arranger-type instruments provided feels and grooves that were fresh and free, their appeal would be broadened. Of course, my opinion is one of personal taste; if you play dinner sets in a hotel lounge, you'll probably be extremely happy with what the 9000 Pro has to offer.

Each Style comes in four main groove permutations, with three intro options, three endings, and four short fills or breaks, with each item slightly more complex than the previous one (Intro 3 is far more developed than Intro 1, for example). You can choose to use only the beat from each Style and play everything else yourself, or you can push the blue Auto Accompaniment button to bring in bass and chordal backings as well. What you play with your left hand determines what auto-accompaniment parts are played.

Accessing, playing, and adapting Styles is sophisticated on the 9000 Pro. In addition to the preset and flash Style locations, you can load new styles from floppy disk into flash ROM (despite its name, you can overwrite its contents).

Style writing is incredibly difficult; I know because I've done it. The difficulty usually stems from the instrument maker's algorithmic requirements and restrictions, as well as from the artistic slant that the programmer is required to provide or to avoid.

You can write your own Styles on the 9000 Pro using either Easy Edit, which lets you tweak a Style that's pretty close to what you want, or Full Edit, in which you dive into the world of CM7 scale tones for recording your bass and phrase tracks. Although it sounds strange when you're recording to base a groove on only the notes C, D, E, A, and B, it will allow the finished Style's fingerings and voicings to play properly. As I said, style writing is not for the fainthearted.

SEQUENCING AND SAMPLING

If arranger keyboards do nothing else for your creativity, they'll start, if not finish, a thousand and one new songs. Then you can record them into your favorite sequencing software and get down to some serious sequencing. But for those times when you're, say, stuck in a hotel room, the 9000 Pro does offer its own 16-track sequencer.

The sequencer is geared toward recording music that utilizes the auto-accompaniment features, but you can edit events to an acceptable degree—not just notes and durations but also things like chords and part levels. You can even record in step time if you have the leisure.

You can sample your own Voice material into the 9000 Pro with relative ease, and you can import WAV or AIFF data. Once you've recorded a sample, you can normalize, loop, retune, or even resample it. Sampling is 16-bit mono at 44.1 kHz, and you can upgrade the standard 1 MB of internal RAM to a maximum 65 MB using two 32 MB SIMMs.

No one would purchase the 9000 Pro for its sampling capability, but at least it offers more than just import and playback facilities, and sampling certainly opens the door to new textures and types of music. To that end, a SCSI port is provided for connecting an external storage device such as a CD-ROM, Zip, or Jaz drive.

WORLD VIEW

A glance at the price tag should tell you that there's a lot going on in the 9000 Pro. Yamaha has wrapped up a vast number of features and facilities in a capable package. Yamaha's experience in the field is unparalleled, and in terms of providing new material, its support is unending.

Yamaha is hoping that the omission of built-in speakers and the addition of some cool new sounds will help the 9000 Pro appeal to professional musicians ranging from working performers to budding film composers. It will do so to an extent, but I'm not entirely sure the specter of cheesiness has been eradicated for the average American pro player. If you've long admired auto-accompaniment instruments but never quite had the guts to admit it, however, now is a great time to jump in.

In addition to writing articles and books about music technology, **Julian Colbeck** *runs Keyfax Software, the company behind the TwiddlyBits MIDI sample libraries and PhatBoy MIDI performance controller.*

PRODUCT SUMMARY

Yamaha
9000 Pro
keyboard workstation
$3,995

FEATURES	4.5
EASE OF USE	4.0
QUALITY OF SOUNDS	4.5
VALUE	3.0

RATING PRODUCTS FROM 1 TO 5

PROS: Everything under one roof. Excellent screen. Beautifully built. A great blend of preset and customizable Voices and Styles.

CONS: Looks and behaves like a home keyboard. Style and Voice data are Eurocentric.

Manufacturer
Yamaha Corp. of America
tel. (714) 522-9011
e-mail info@yamaha.com
Web www.yamaha.com

I'M BIASED

David Torn aka splattercell
Guitarist/Texturalist/Producer

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Soundtrack work includes: traffic • a knight's tale • three kings • heist • the velvet goldmine • simone

Photographed by Karjean Ng at the studio of film composer Carter Burwell, New York City

David Torn is a genuine musician's musician — one of the most respected of our time.

Whether working with David Bowie on his latest album, or creating trademark textural soundscapes for *Traffic* and other blockbuster movies, or crafting a new splattercell CD, David's aesthetic for raw sonic exploration goes far beyond a conventional approach to music, let alone guitar. And to help him make his discoveries, David turns to BIAS software.

As he puts it, "I'm not much interested in what's been done before, especially when it comes to my own work. I need to keep uncovering new ground — and I love how BIAS products help me do that so intuitively, with critical speed & stability. Like my guitar, they feel like they were built just for me, letting me create a vocabulary for the language of my music."

It only makes sense that BIAS software is an integral part of David's creative process. After all, we share a common focus: the intersection of technology and art, where creativity flows on a path of least resistance. And it's from this place we create tools to help *you* define your own unique vocabulary.

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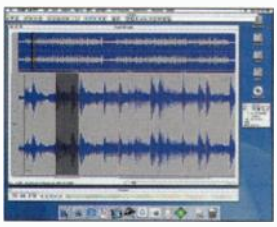

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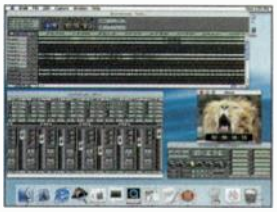

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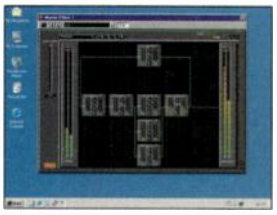

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SONIC FOUNDRY

ACID PRO 3.0 (WIN)

It's often imitated but never duplicated.

By Zack Price

When Acid was introduced, it was hailed as a breakthrough product, and rightly so. The program simplified the process of creating music using sample loops by enabling audio files with contrasting tempos and pitches to play together in time and in key. Gone was the tedious pre-production process of recalculating loop lengths and pitch changes that was necessary before assembling loops into a song. Acid calculated those changes on the fly and in real time.

Since then other companies have introduced programs that let you create music by sequencing sample loops with different pitches and tempos. Some of them are quite good, and some are merely adequate, but none does exactly what the latest version of Acid does.

ACID REFLEX

Acid allows you to arrange audio loops and hits through a simple drag-and-drop interface (see the November 1998 issue for a review of Acid version 1.0). It doesn't matter what the sample's original tempo or pitch is; Acid automatically stretches and transposes it to fit the current project's tempo and key. Like the original version's, Acid Pro 3.0's interface is divided into three main sections: the Track List, the Track View, and the multifunction Accessory panel (see **Fig. 1**).

When you first open the program, the Track List and Track View areas are empty, and the Tempo and Tuning parameters are set to default values of 120 bpm and the key of A, respectively. The lower part of the screen displays the Explorer, where you view and preview the various files that can be used in Acid. In addition to 16- and 24-bit WAV and AIFF files at sampling rates as high as 192 kHz, Acid Pro 3.0 now supports other audio file formats, such as MP3, Ogg Vorbis, and Windows Media (export only). Just click on any file, and Acid will preview it at the tempo and key specified in the bpm and Tuning settings.

Acid Pro 3.0 has independent transport controls in the Explorer area that let you control preview playback independent of the main transport. You can also route preview material to a separate audio bus if you have a multichannel audio card (see **Fig. 2**). That allows you to control the preview volume independent of the main mix, which can be an ear saver if you happen to select a particularly loud loop to preview.

Once you pick a loop or other audio file, drag it to the Track List or Track View area to create a new track. (Each file is assigned a separate track.) Tracks contain Events that indicate how and when the assigned file should be played. Acid typically loads short audio files into RAM, which improves performance in playback. (Users can set a duration threshold above which files are streamed from disk.) A numeric indicator in the lower-right corner of the screen indicates how much memory is used for samples compared to overall system memory. When a file is longer than the user-defined limit, the Beatmapper wizard goes to work. Beatmapper lets you add tempo information to a file, making stretching the file easier.

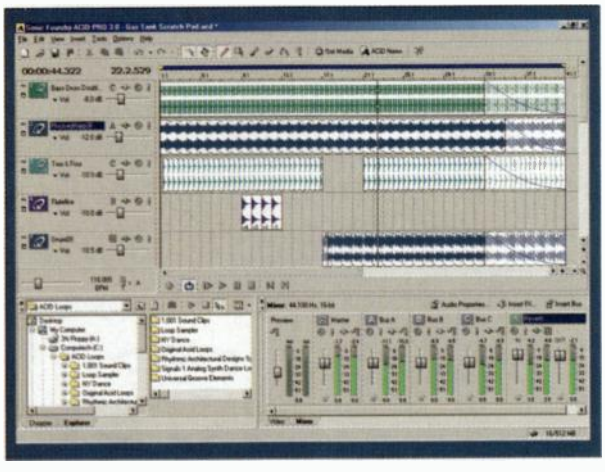

FIG. 1: Acid Pro 3.0's interface is divided into the Track List and Track View (the upper portion of the screen). The multifunction Accessory panel on the lower part of the screen displays the file Explorer, Chopper tool, audio bus assignments, and video display.

Minimum System Requirements

Acid Pro

Pentium II/300 (Pentium II/400 for video scoring); 64 MB RAM; Windows 98SE/ME/2000/XP; DirectX 8.0; 16-bit sound card

Once you've created a new track, you tell Acid where in time you want the file to play. Using the Draw tool, left-click and drag the mouse to insert Events into a track. To edit an Event, simply grab its end point and drag it to the right to expand or left to contract it. You can also click on an Event to highlight it and then move it to a different start point in the timeline. However, you can be more selective about what to play in a track, as you'll see shortly.

Although Acid works with sample data, you can improve performance if you first Acidize the file. That can be done directly in Acid, Sound Forge 5, or Sound Forge XP 5.0, which comes in the bundle. That process adds an extra chunk of data to the file's header that contains settings specific to Acid, including the type of sample, its number of beats, its root note for transposing, and its stretching properties. You can also specify whether the file should be looped, played as a one-shot, or beatmapped and whether it should include beat info and a base note for transposition, assuming you want the file to be transposed. Drum loops, for example, don't usually contain a base note, because you don't want the pitch to change whenever the tempo changes.

ON THE BEAT

The first time you import any given large audio file into Acid, the Beatmapper wizard starts if the program does not detect any tempo information in the file.

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Run the wizard, and Beatmapper synchronizes the file's tempo to the tempo of the current project. If you select No, the file maintains its original length regardless of your project's tempo.

Beatmapper works by identifying the downbeat of the first full measure and placing a marker at that point (any pickup beats before the first full measure are ignored). Then, it estimates the length of the measure and highlights a region in the file's waveform display to indicate where the first measure is. You can move the marker for the first downbeat to the correct position if it's incorrect or adjust the size of the region if needed.

The final step is to ensure that each measure in the file has been marked off properly by the Beatmapper. As far as I'm concerned, that's a useless exercise, because most long audio files rarely play back perfectly in time from beginning to end—the measure markers are going to be off after a while. However, as long as the downbeat for the first measure is in the right place and you have the true length of the first measure, the Beatmapper keeps the remainder of the track in tempo pretty well.

When Beatmapper finishes, you can change the project tempo to match that of the beat-mapped track. You can also preserve the pitch of the beat-mapped track if the project tempo changes. Beatmapper information is saved with the file so that you won't have to run it again the next time you use that file.

LET ME CHOP IT

As mentioned earlier, you aren't limited to playing complete loops in a track. Acid Pro 3.0 now includes the Chopper tool, which makes it simple to dissect, isolate, and insert parts of loops into tracks. Open the Chopper tool from the View menu or select the Chopper tab in the lower-left corner of the screen (see **Fig. 3**). Then, click on the Track List or Track View to select a loop, and its waveform display appears in the Chopper window. Now, highlight an area—a snare hit, for example—and play it using the Chopper's transport controls. Once you have captured the correct loop segment, use the Insert button to place it into the track at the cursor point as many times as you want. (There's no way to insert the segment multiple times automatically.)

If you activate the Link Arrow to Selection button and drag the arrow forward or backward, you can add space around your loop segment to create interesting stuttering effects or gaps in the audio track. You'll also find numerous tools in the Chopper that make moving a region or modifying its length simple. For example, it takes only one click to double or halve the size of the highlighted region or to shift it left or right. Best of all, you can use musical durations to define the size of a region; values from full measure to 32nd-note triplet are available. A custom setting accepts fractional durations larger than one measure (for example, 1.01 or 1.25 measures).

EDITS AND EFFECTS

One of the most useful new editing features is Ripple Editing, which closes up gaps when you cut or delete highlighted sections of track data. Ripple Editing also lets you paste data into a track and move existing data back to make room for the new data. That is an excellent tool for rearranging entire sections of a song. Acid also has a new fade-in/fade-out function that makes it easier to create track fades. Just move the Draw tool to the upper corner of a track, and a little Fade icon appears. Drag the Fade icon to the desired point, and you get a perfect fade-in or fade-out. If you right-click on the Fade icon, a drop-down box showing linear, fast, and slow fade shapes appears. Choose the type of fade you want, and it appears on the track.

FIG. 2: Acid's Mixer provides a number of flexible routing options. If you have a multichannel audio card, you can route preview material to a separate audio bus, set bus assignments for multiple outputs, and create Insert FX buses.

Sonic Foundry
Acid Pro 3.0 (Win)
loop sequencer
$499.95

FEATURES	4.0
EASE OF USE	4.5
DOCUMENTATION	4.5
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Easy to create loop-based songs. Support for MIDI and video formats. Direct CD burning.

CONS: No muting, unmuting, or soloing tracks from computer keyboard. No way to use MIDI to control program. Only one loop per track.

Manufacturer
Sonic Foundry
tel. (800) 577-6642 or (608) 204-7680
e-mail customerservice@sonicfoundry.com
Web www.sonicfoundry.com

Acid Pro 3.0 includes all three of Sonic Foundry's XFX DirectX audio effects plug-ins (18 total), which are also sold separately. Furthermore, you can now use effects as track inserts (Track Effects), through an effects send/return bus (Bus FX), or on an effects chain (Assignable FX). You can have as many as 32 effects per track, 26 buses, and 32 effects chains.

Acid Pro 3.0 has the ability to import an AVI or MOV video file into a project; that allows you to create loop-based tracks that are synced to video. Being able to see your video on a frame-by-frame basis makes that an especially useful feature. You can also add a graphic image in one of several common file types (TGA, BMP, PNG, and so on) for use as a background image.

Acid Pro can now record and play back multichannel MIDI data on one or more tracks. (Acid recognizes MID, SMF, and

FIG. 3: The Chopper tool simplifies the process of dissecting, isolating, and inserting loop parts into tracks. Just highlight the area of the loop that you want to use and insert it into the track at the cursor point.

RMI MIDI file types.) There isn't much MIDI editing capability available—you can alter the tempo or shift the pitch of an entire Event up or down in semitone increments—but it's a handy feature nonetheless. You can link directly to a sequencer (if you've specified one in the Preferences menu) and load DLS files for Acid to use when playing MIDI data. If you don't have a DLS-compatible sound card, Acid can use the DirectX soft synth for playback.

You will have no trouble finding files for use in your music. There are dozens of CDs in the Sonic Foundry Loops for Acid collection, and you can rip (or burn) CDs directly from within the program.

WISH LIST

Although Acid continues to get better and better, you still can't mute, unmute, and solo tracks on the fly using the computer keyboard. That would let you audition groups of tracks without having to assign them to a particular bus beforehand, and it would be helpful when using Acid in live performance. I'd also like the ability to control Acid with MIDI data coming from a sequencer or an external controller.

I have no other complaints about Acid Pro 3.0. I like the improvements in the program's functionality and the value added by the inclusion of Sound Forge XP 5.0 and all three XFX DirectX plug-in packages. The CD-ROM also contains Vegas Audio LE to whet your appetite for the Vegas Video program.

For those who disdain creating music using loop-based material, I advise you to give Acid Pro 3.0 a close look. I felt that way until I got my hands on the program. Now it's one of my favorite music-making tools. Although it's simple to use, it's no less challenging to create musical pieces with Acid than with other production methods. Besides, the program is just plain fun, no matter what style of music you create—and isn't having fun with music what it's all about?

Zack Price *is a digital-audio editor and a Windows digital-audio consultant in the Chicago area.*

K O R G

PXR4

A pocket-size 4-track digital recorder that's not short on features.

By Steve Broderson

With the new ToneWorks PXR4 digital recording studio, Korg seems determined to destroy two popular notions: one, size matters, and two, you can't take it with you. This diminutive 4-tracker, which records on SmartMedia cards, packs a surprisingly large feature set into a box the size of a personal digital assistant, and it can run on batteries as well as AC power. The unit records as many as four audio tracks and seven virtual tracks beneath each of those, for a total of 32 tracks (4 mixable, 28 virtual). Other features include a tuner, 99 locate points per song, 77 modeling effects, 32 metronome patterns, and 55 drum patterns. When you factor all of that in with the unit's editing capabilities, you have to wonder whether the PXR4 is the last stage of miniature recording technology before the implanted microchip.

Korg's PXR4 crams four tracks, a master fader, oodles of effects, extensive editing capabilities, and a USB interface into a box the size of a personal digital assistant.

ALL AROUND

The PXR4 is only about five inches wide by four inches long, so space, though well used, is tight. Situated along the top rear panel are the USB port, AC adapter input, power switch, and I/O section. The three-position sliding power switch lets you turn the LCD backlight off to reduce power consumption when using batteries, thus prolonging battery life. (With the backlight off, two AA batteries provide as much as ten hours of operation; with the backlight on, you get about two hours.)

The I/O section consists of a ¼-inch input jack labeled Guitar Input, a stereo miniplug input jack labeled Line Input, and a stereo miniplug line-level output labeled Output. An Input-Select switch lets you pick which input—Mic, Line, or Guitar—is active. Engaging the Mic setting activates a built-in condenser mic that's actually good at capturing sources such as acoustic guitars and even drums. The stereo miniplug input allows the PXR4 to get signals from a keyboard or even a mixing board. Another switch (also labeled Guitar Input) lets you choose between Hi and Lo settings, to accommodate either high- or low-impedance guitar pickups. The PXR4 lacks provisions for MIDI or digital I/O.

Along the bottom front panel are a stereo miniplug headphone jack; a large, numbered rotary Volume control; and a large, numbered rotary Input Trim control. The SmartMedia card slot is located on the left side of the unit.

The PXR4's top panel is where the action is. The LCD is packed with info—and it's just large enough to allow you to make sense of it all. Track names, levels, effects, time/bars, and editing information are viewable at a glance, and the backlighting comes in handy in low-light situations. Below the LCD are five faders (one per track and one master), each of which does double duty controlling monitor/playback levels as well as parameters of the effects modules. In fact, most of the top-panel buttons serve more than one function, thanks to a Shift key. For example, each track has a dedicated Select button that lets you arm it for recording, and you also use those buttons to select the four effects modules. The button above the Master fader has three functions: Mixer, which lets you control things such as track Level, Pan, and Effects Send; Display, which allows you to set levels and view them prefader or postfader; and Write, which lets you store changes to your effects patch.

The right side of the PXR4's top panel provides a large data wheel, a four-way cursor pad, the built-in condenser microphone, seven control buttons (Effect/Tuner/Assign, Locate/Bounce, System/Rhythm, Mark/Undo, Shift, Exit, and Store Mark), and, along the bottom, the five standard transport-control buttons: Play, Stop, Record, Rewind, and Fast Forward. The transport controls function as you'd expect except for one peculiar feature: the Fast Forward and Rewind buttons double as the selector buttons for songs. In other words, if you're at the zero point and hit Rewind, you jump

PRODUCT SUMMARY

Korg
PXR4
portable digital studio
$500

FEATURES	4.5
EASE OF USE	4.0
AUDIO QUALITY	3.5
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Clean, natural sound. Straightforward operation. Flexible editing. Great effects. Built-in computer interface.

CONS: Miniplug I/O.

Manufacturer
Korg USA, Inc.
tel. (516) 333-9100
e-mail product_support@korgusa.com
Web www.korg.com

The Award-Winning New Standard in Sample Libraries

SONIC CONTROL

Overall Rating: ★★★★★ (5/5)
Sound Quality: ★★★★★ (5/5)
Usability: ★★★★★ (5/5)
Programming & disc layout: ★★★★★ (5/5)
Value: ★★★★★ (5/5)

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"The Garritan Strings set a new standard of playability, dramatic range, and sound quality. The intelligent use of crossfades, multilayers, and MaestroTools give a composer many options and colors to paint with." Jeff Beal, TV/Film Composer (Pollock, Joe & Max, The Passion of Ayn Rand)

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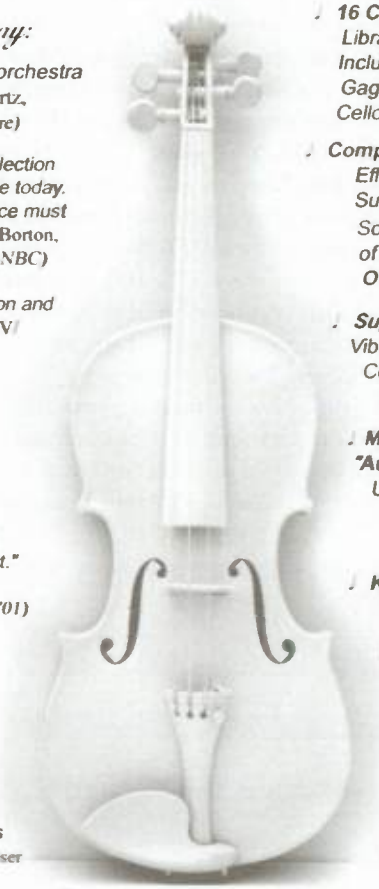

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to the song prior to the one you're working on. To get a standard rewind or fast-forward function, you have to press and hold the button.

Aside from the faders, the primary way to make data changes is with the cursor pad and data wheel. Because of the space constraints, the PXR4's deep functions are menu driven; navigating the menus is probably the most complicated thing about the unit's operation. The Effect/Tuner/Assign, Mod, and other buttons take you to menu screens where you can modify parameters. From there, a compasslike page guide in the LCD indicates the directions in which you can cursor or whether you've reached the end of the menu. Although the PXR4's manual is well written and describes all the menu pages in detail, it would be helpful if Korg provided a printed map of the menu pages and how they are organized.

The data wheel is a bit more straightforward. In general, the wheel lets you dial in precise numbers for any menu item that's blinking. I found it easier to set track levels and effects settings with the data wheel than with the minifaders.

IN THE CARDS

Using MPEG-1, Audio Layer 2 data compression, the PXR4 records 24-bit audio at 32 kHz resolution. You must select one of three recording modes—Hi Quality, Standard, and Economy—at the beginning of a project. As you'd expect, the higher quality the recording mode, the less recording time you get (see the table "PXR4 Recording Mode Times").

The unit ships with a single 16 MB SmartMedia card, all of which is taken up by the demo songs. The first order of business, then, was to free up the card by backing up the demo-song data to my Mac's hard drive. That was easily done using the built-in USB interface. Windows ME/2000 or later and Mac OS 9.0.4 or later are supported. After making the physical connection with a cable (not included), I dialed up the USB page on the PXR4 and selected Yes when prompted. A Mac volume called Unnamed popped up on my desktop, and I was in business.

The files created by this procedure are proprietary and, as of this writing, cannot be edited using PC software. If, however, you use the PXR4's Bounce feature to create a mixdown in MP2 format (not to be confused with the more common MP3 format), your computer can play the file—that is, assuming you have an audio card that supports playback at the 32 kHz sampling rate. A 16 MB card gives you only about four minutes per track, including Virtual ones—just enough memory for one song. Fortunately, SmartMedia gets cheaper every day. I'd recommend getting at least a 64 MB card right off the bat. The PXR4 supports cards up to 128 MB.

INTRODUCING YOUR GUIDE

To test the PXR4, I recorded a mix of acoustic and electric guitars and drums—an approach that required using all three ways of getting sound into the unit (¼-inch guitar jack, stereo miniplug, and built-in condenser mic). I started with a guide track from the unit's built-in rhythm generator. Beat styles range from rock, funk, and disco to jazz and Latin and time signatures from 1/4 to 8/4 and 1/8 to 8/8. Not surprisingly, most of the drum patterns employ a 4/4 time signature. The drum kit is not editable, and the patterns cannot be chained together like a sequence.

The PXR4 offers 77 types of effects arranged into four blocks: Drive/Lmt (Limiter), Cab (Cabinet)/EQ, Mod (Modulation), and Amb (Ambience). These blocks use one of eight combinations of effects called chains, any one of which can be inserted at one of four points: the guitar input, the line (stereo miniplug) input, the master bus (an effects send for the individual tracks), and at the master L/R output. The effects available simultaneously depend on the chain you're using. For instance, none of the Guitar chains include reverb—you have to add that later as an insert or a master effect.

On the whole, the effects were pleasing sonically. I had to crank the reverb returns to get enough level, but the fair selection of reverb patches was highly detailed and not metallic sounding. Tweaking any effect is fairly simple; once you are in Edit mode, the track-select buttons take you directly to parameter pages and the faders become data sliders for adjusting them. When you've dialed

PXR4 Specifications

Physical Audio Tracks	4
Virtual Audio Tracks	32
Simultaneous Record/Play Channels	2/4
Preset Rhythm Patterns	32 metronome patterns, 55 drum patterns
Sampling Rate	32 kHz
A/D/A Converters	24-bit
Internal Effects Processing	24-bit
Effects Patches	100 preset, 100 user
Analog Inputs	(1) ¼" guitar (high- and low-impedance selectable); (1) stereo miniplug
Analog Outputs	(1) ⅛" stereo miniplug
Built-In Microphone	stereo condenser
Storage Medium	SmartMedia card (4–128 MB)
Maximum Songs	99 per card
Maximum Recording Time	dependent on card size, recording grade
Supplied AC Adapter	4.5V, 500 mA
Batteries	2 AA (not included)
Battery Life	2 hours (with backlight on)
Display	backlit LCD, 0.217" (W) × 0.099" (H)
Dimensions	4.625" (W) × 4.125" (L) × 1.125" (D)
Weight	0.58 lb. (without batteries)


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in the sound you want, you can save it to one of 100 User Effects locations.

The 4-band EQ block comes in six flavors: Lo, Mid, and Hi, each of which lets you zero in on more specific frequencies; and three Wide settings that allow for broader EQ curves. Each of the six blocks lets you tweak four bands of frequencies in its dedicated range. For example, Lo allows 10 dB of boost or cut at 80, 120, 250, or 550 Hz. Unlike a conventional mixer, the EQ block must be used like an effect, either at the insert point (to EQ while tracking) or across the stereo bus (to EQ while bouncing). The PXR4 also offers a feature usually reserved for high-end standalone recorders and software: time compression and expansion. Although only casually mentioned in the manual, that's a pretty cool and useful effect.

PXR4 Recording Mode Times

With 16 MB SmartMedia Card	
Hi Quality	approximately 11 minutes on 1 track
Standard	approximately 16 minutes on 1 track
Economy	approximately 33 minutes on 1 track
With 128 MB SmartMedia Card	
Hi Quality	approximately 90 minutes on 1 track
Standard	approximately 135 minutes on 1 track
Economy	approximately 270 minutes on 1 track

PETITE MODELS

I used my Rickenbacker 360 for the guitar input test. The modeled guitar amps were very usable, though I did have to back off the levels quite a bit before using them—the default settings provide far too much gain. With its level backed off a bit, the Vox AC30 model gave me a nice, chunky rhythm guitar sound. I double tracked the part. The bass guitar had a preset designed for picking, which added some limiting and high-end EQ.

For my two acoustic-guitar tracks, I used the built-in condenser microphone, combined with the Mic'ed Acoustic preset effect, which includes a bit of gentle compression and a Vintage Condenser mic simulator. To position the mic for optimal pickup, I propped the unit up on its left side (the only one with no jacks or knobs), at an angle, so the mic faced the guitar from about a foot and a half away. That gave me a full, balanced sound.

To test the PXR4's line input, I miked a four-piece drum kit (kick, snare, two toms, and two overheads) and bused the six signals down to a stereo pair through a Mackie 1402. Using the PXR4's Vintage Comp compression preset at the input, I generated some pleasingly fat drum sounds.

FREELY EDIT

At this point, I had used seven tracks and not even done the vocal yet. Thanks to the PXR4's Virtual Track and Bounce features, that wasn't a problem. As mentioned previously, each track has seven layers of tracks beneath it, each of which can be recorded, copied, inserted, and bounced at your will. But be careful, because it's easy to eat up a 16 MB card fast. Assuming you don't erase them, you will have all of the prebounced takes for later use, in case you don't like your bounces. The only feature I missed was a track-exchange function for swapping tracks. Rather than exchange tracks, you have to copy them and then delete the old ones. That is a minor complaint, however; no other ministudio I know of lets you edit tracks this freely.

The Bounce feature allows you to mix down four tracks to one or two (stereo) tracks. (By the way, you can also overdub live tracks from the inputs as you bounce with this feature.) I'm a stereo kind of guy, so with my original tracks safely on Virtual tracks, I bounced away in stereo. In the end, I had put all four guitars and the stereo drums on tracks 3 and 4, lead vocal on 1, and bass on 2. For each bounce, I used a bit of the EQ/mastering effects on the stereo bus to keep everything hot and sweet. For the final mix, I used a preset called Final Mix. I also liked one called Tape Sim, which added some gentle limiting and tamed some of the digital edge I'd accumulated. A portion of the song I ended up with is posted at www.emusician.com.

PORTA PARTY

Every songwriter has a story about a great song that slipped away because there was no opportunity to record it—typically, because he or she was out and about somewhere, far from a recorder. In this regard, the Korg PXR4's portability and ease of use—not to mention the vast sonic improvement it makes over hissy, warbly cassette multitrackers—make it a songwriter's dream come true.

The PXR4 stands out for several reasons. It is the only recorder in its class to offer four recordable tracks and a master fader. It's also the only one to let you cut, copy, and paste audio at will. In short, the PXR4 lets you start with an infant idea and grow it into "demohood." If you don't want to wait for that implanted microchip, check out the PXR4.

Steve Broderson *is a Versailles, Kentucky–based writer and producer. He creates original music for broadcast and teaches an audio-recording course at Asbury College in Wilmore.*

MAGIX

SAMPLITUDE PRODUCER 2496 6.02 (WIN)

Host-based digital-audio software that rivals its DSP-based counterparts.

By Zack Price

I last reviewed Samplitude 2496 in April 1999 and was impressed with the program's capabilities. Version 5.12 made it possible to record multiple tracks of 24-bit digital audio at 96 kHz, apply real-time effects processing, and burn a CD of the music that you created in the program. You could even download a utility that let you use a Peavey PC 1600x as a control surface for the software. Given the state of computer power at the time, Samplitude 2496 was an outstanding host-based multitrack audio program, though it wasn't fully competitive with its digital-signal-processing (DSP)-based rivals.

What a difference a few years can make. Computer power is now sufficient for well-written, host-based audio software to compete favorably with most DSP-based workstations. Furthermore, with the advent of new hardware controllers from companies such as Radikal Technologies, Samplitude users can take better control of program tracks, effects, and editing views.

As you might expect, the renamed Samplitude Producer 2496 takes advantage of all of the improvements that the current generation of computers allows. For instance, Samplitude Producer supports WDM drivers, works with multiple processors in Windows 2000 and XP, and is optimized for use with Pentium 4 CPUs. Additionally, the program can record, play, and process 24-bit, 192 kHz audio with 32-bit floating-point resolution.

THROUGH ANY WINDOW

One thing that hasn't changed is Samplitude Producer's overall architecture. The central work space is still the Virtual Project (VIP) window, the multitrack display where you assemble data, perform mixes, and do most of your editing and processing chores. The VIP window shows graphic representations of audio clips known as Objects, which are simply pointers to the actual audio data on your hard drive.

With Samplitude you can record directly to the hard drive, in which case the recording is referred to as a Hard Disk Project (HDP). You can also record to RAM, thereby creating a RAM Project (RAP). At some point, you must save a RAP recording to disk, or you'll lose it when you turn off the computer. HDP files are always read directly from disk, whereas RAP files load into RAM before playback. Because data in RAM can be accessed so quickly, RAP files are well suited for use by short repeating sounds or loops.

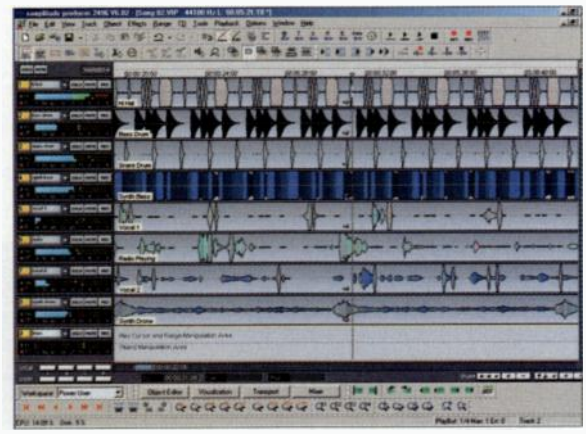

FIG. 1: Samplitude Producer's redesigned VIP window shows graphic representations of audio data known as Objects. The drop-down Workspace menu in the lower left lets you quickly reconfigure the window to show only the tools that you need.

You can create Objects by recording directly into the VIP window or by importing data from HDPs or RAPs. When you record an Object directly into the VIP window, its corresponding HDP is automatically generated. Once you have recorded an HDP or a RAP, you can simply highlight all or part of its waveform and drag it into position on the desired track in the VIP window. The main difference between the VIP window and the HDP and RAP editing windows is that VIP editing is nondestructive. Editing HDPs and RAPs is destructive, although you can create backups of each edit.

Minimum System Requirements

Samplitude Producer 2496

Pentium II/200; 64 MB RAM; Windows 95/98/ME/NT/2000/XP; DirectX 6.1; 16-bit sound card

Longtime Samplitude users will notice right away that the VIP window has a new look that makes getting around much easier (see **Fig. 1**). With the Workspace drop-down menu in the lower left of the screen, you can configure the work space to show only the tools (the buttons surrounding the Track area) that you want to see for a particular session. For example, the Power User setting displays all of Samplitude's toolbars at once, whereas the Editing work space provides a smaller toolbar configuration that is set up just for editing. By right-clicking on the Workspace menu, you can also create and store your own custom work-space settings for VIP and Waveform-Editing windows.

Next to the Workspace menu are four rectangular buttons labeled Object Editor, Visualization, Transport, and Mixer. Clicking on any of the buttons opens the corresponding window. If that window is already open, clicking on its button closes it. That's a handy way to show essential tools and hide the tools that you don't need. For example, if I'm working in the VIP window, I usually want to see the Mixer only occasionally. Likewise, I want to see the Transport just when I need to use it; otherwise, it's in the way.

Two of the cooler new features added to Samplitude since **EM**'s last review are the Visualization window and Comparisonics. The Visualization window is similar to the Oscilloscope and Phase Correlator displays but adds a Peak

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Meter, Spectroscope (see **Fig. 2**), and Spectrogram. Those are handy displays to have when you need to examine frequency response and stereo phase stability. You can run different Visualization modes simultaneously for a track or a bus routing assignment, but you can't run independent instances of the same Visualization mode to examine several tracks at once. I hope Magix includes that capability in the next update.

The Comparisonics feature enables you to quickly search for the same or similar-sounding regions in an audio file. That can be particularly useful if you're trying to identify a sound or some program material that repeats throughout a track. To use the feature, open the Waveform-Editing window that includes the material you need to search through. Highlight the area that includes the first instance of the information that you want to find and copy that area into the Clipboard. Next, select Comparisonics Audio Search from the Range menu, and Samplitude finds the other areas where the program material occurs based on the sensitivity settings that you select for the Comparisonics algorithm.

You can also set the waveform colors in the VIP window to the Comparisonics Colors option. Samplitude then uses the Comparisonics algorithm to display the sonic material in various colors based on frequency and other parameters. Low notes are given shades of blue; midrange through high frequencies appear in shades of green, yellow, and red. Noise and nonspecific audio are represented by shades of gray.

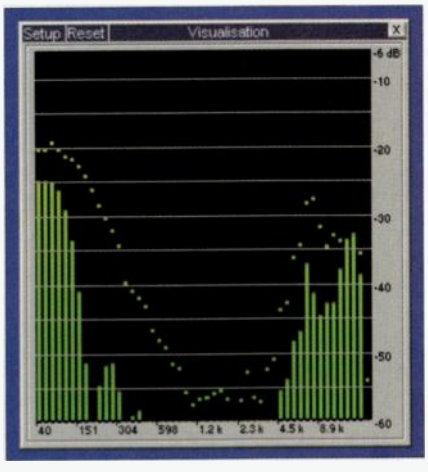

FIG. 2: The Visualization window provides an Oscilloscope, a Phase Correlator, a Peak Meter, a Spectroscope (shown above), and a Spectrogram mode. The modes let you examine frequency response and stereo phase stability for a track or a bus assignment.

Using the Comparisonics Colors option, you can search for similar sections of program material by sight alone. Depending on how distinctive a particular sound is compared to the rest of the audio in the waveform display, a quick visual inspection may be all that is needed to pinpoint what you're looking for. The Comparisonics Colors option also makes a good rough spectrogram. For example, track 7 (vocal 2) in **Fig. 1** shows a waveform display that is mostly blue, with occasional changes to green and red, and periodic sections of gray. That indicates that the vocal track consists mainly of low and midrange frequencies and that a fair amount of sibilance (as shown by the gray areas) is present as well.

MIX IT UP

Samplitude's sophisticated new Mixer (see **Fig. 3**) is a major improvement over earlier versions. It now offers gain adjustment, four auxiliary bus sends, delay/reverb, multiband dynamics processing, and 4-band EQ on all channels. In addition, you can insert DirectX plug-ins into each channel. Naturally, the number of active DirectX plug-ins that you can run is limited by your CPU's processing power.

The Master Section allows you to insert six DirectX plug-ins into the master outputs, and you can also use Samplitude's own real-time effects: Dehisser, FFT Filter, Multiband Dynamics Processor, Dynamics/Limiter, Four-Band EQ, and Stereo Enhancer. Furthermore, you can select the order in which the real-time and the DirectX processors are arranged in the Master section and in the individual channels, and you can save the routing presets for later recall.

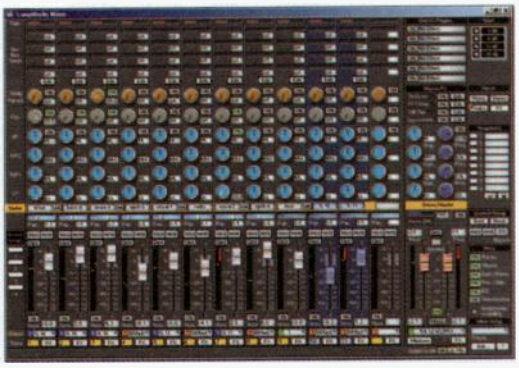

FIG. 3: Samplitude Producer's new Mixer is much more sophisticated than in previous versions, offering expanded processing features, submix and auxiliary buses, a master section, and 5.1 surround-sound mixing capabilities.

The previous version of Samplitude allowed you to switch only between a full-mixer and a small-mixer display. The newest version, however, permits you to decide which sections of the mixer you want to see at any given time. You can show or hide tracks, the Reverb/Dynamics sections, the Gain/Auxiliary bus sections, EQ, and Master section. Although it's not part of the Mixer itself, the Visualization window can also be shown or hidden from within the Mixer.

Another important addition to the Mixer is the ability to create auxiliary and submix buses that show up as new tracks in the VIP window and as new channel strips in the Mixer. Creating auxiliary buses is helpful when you want to route specific audio tracks to an outboard effects processor, especially if you're using a multichannel audio card. Creating submix buses is useful for controlling groups of tracks with one channel strip. Auxiliary and submix buses also include all of the routing and processing capabilities of regular channel strips in Samplitude's Mixer.

Although the new mixer is a vast improvement over Samplitude's earlier mixer, it still has a few shortcomings. For one thing, you can route tracks to a submix bus only if the track numbers are lower than the submix-bus channel number. For example, if you turn track 4 into a submix bus, you can submix only tracks 1, 2, and 3 into that bus. To create a submix bus that lets you assign any tracks to it, a new track

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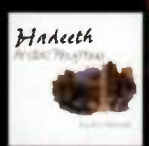

Hadeeth - Arabic Rhythms

East West:


Drum & Bass Construction Kits


Black Butta 1


Black Butta 2


Electronica

ILIO:


Big Bad Beats

Metamorphosis - 21st

Retro Funk

Trance Fusion

has to be inserted in the bottom of the VIP window. That's easy enough to do, but I'd prefer to arrange my tracks, and consequently the appearance of the Mixer channels, any way I want without that bit of inconvenience. (According to Magix, the Mixer design includes this inconvenience as a way of avoiding the potential for feedback loops.)

As mentioned earlier, in the previous version of Samplitude, you could control many parameters in the VIP window and the Mixer with a Peavey PC 1600x MIDI controller. The new Samplitude retains that feature and also includes setups for Radikal Technologies' SAC-2K, Yamaha's 01V digital mixer, and C-Mexx's C-Console (lite version) for the Yamaha DSP Factory card. Unfortunately, the controllers work only for the first eight tracks in the VIP and Mixer windows; if your project has more than eight tracks, you can't control any higher-numbered tracks. (The SAC-2K is an exception in this case; it has buttons for switching to higher-numbered banks of eight.)

Furthermore, because you must set the submix bus to higher track numbers in order to assign regular tracks to them, you won't be able to use the hardware controllers to manage those channels, either. According to Magix, the company plans to make all tracks controllable through MIDI automation in the next release. That would make it easier to control and edit program functions remotely. It would also be a natural extension of Samplitude's existing MIDI recording, editing, playback, and synchronization capabilities. Keep in mind that the MIDI features are really more of a convenience in Samplitude; the program is first and foremost a multitrack digital-audio production tool and is not intended to be used as a digital audio sequencer.

OBJECTS OF DESIRE

Samplitude Producer's Object Editor (see **Fig. 4**) is a major improvement over the previous version. As mentioned already, you can call up the Object Editor by selecting an Object in the VIP window and clicking on the Object Editor button, or you can simply right-click on a particular Object to do the same thing.

When the Object Editor opens, it presents you with a variety of options. Clicking on the Object Effects button displays all of the effects-processing capabilities that you can apply to the selected Object. For example, you can alter the input gain, boost the output to the auxiliary buses, and apply DirectX plug-ins and Samplitude's own effects.

Clicking on the Position/Fades button displays the Object's starting position, ending position, and length. From this display, you can change an Object's start or end time and you can set fade-in and fade-out curves and times. Clicking on the Pitchshifting/Timestretching button lets you alter the pitch of the selected Object, stretch its time, or combine the two processes to create real-time, pitch-shifted, time-stretched Objects. I used the tool to lower the pitch of the bass-drum track in **Fig. 1** by 1.5 semitones.

OUT OF ROOM

I could easily double the size of this review, and it still wouldn't be long enough to cover all of the things that Samplitude Producer 2496 can do. For example, the program now includes the ability to do rich-media production. Samplitude comes with Magix Video Deluxe, which allows you to import digital video into your computer if you have the proper hardware. You can then edit the video and import it into Samplitude, where you can edit the corresponding audio. Best of all, you can mix the audio down to 5.1 surround sound through using Samplitude's Mixer.

Unfortunately, Samplitude suffers from being a too-well-kept secret. That may change soon, however. Alexander University will be offering online classes for music production, using Samplitude as the primary teaching tool. Peter Alexander and Jeff Sheridan (who uses Samplitude's older sibling Sequoia in his music projects for Walt Disney Records) are forming a Samplitude Users Club in the Los Angeles area and hope to form Samplitude Users Clubs in other cities, as well.

Overall, Samplitude does a vast number of things, and for the most part, it does them all extremely well. I've used many host-based and DSP-based digital audio workstations, and in many cases, Samplitude Producer 2496 exceeds the capabilities of those systems. That's why I use the program in my own work, from the start of a project to its final CD burning. I rarely need any-

FIG. 4: The Object Editor lets you apply a variety of effects to an Object. You can also change an Object's start or end time, set fade-in and fade-out curves, and apply real-time pitch shifting and time stretching.

PRODUCT SUMMARY

Magix

Samplitude Producer 2496 6.02 (Win)
multitrack audio editor
$749

FEATURES	4.5
EASE OF USE	4.5
DOCUMENTATION	4.0
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Comprehensive digital-audio production package. Records to RAM or hard drive. Supports high-resolution audio and surround sound. Includes some MIDI capabilities. Comes with video-editing software.

CONS: MIDI hardware controller implementation limited to the first eight tracks in the Mixer window. Assigning tracks to submix buses depends too much on the order of the tracks.

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TRAK MASTER

An affordable, well-featured channel strip and outstanding mic pre.

By Richard Alan Salz

Many recordists with personal-studio-level consoles are at a distinct disadvantage when it comes to getting a signal to tape (or hard drive) unscathed because of the typically inferior quality of budget consoles' microphone preamplifiers. Even the mic preamps found in some top-notch consoles leave a lot to be desired. Add to that the frequent need to equalize and provide dynamics processing before hitting tape (or other media), and it's no wonder channel strips have been popping up from manufacturers in all price ranges.

The Trak Master, the newest member of the budget Platinum series from Focusrite in Britain, retails for only a fraction of the cost of the company's high-end—and highly respected—Red and Blue range units. But does that mean the unit offers just a fraction of the performance of its costlier cousins? To answer that question, I subjected the Trak Master to rigorous testing and real-world use in the studio.

GENTLEMEN PREFER PLATINUM

Although it is manufactured in China rather than the United Kingdom (where the Red and Blue range units are made), the Trak Master doesn't look like a budget unit, thanks in part to its snazzy brushed-aluminum front panel. The controls are nicely laid out, and the black legending is easy to read in low-light situations. Slightly sticky (in a good way) rubberlike knobs (all continuously variable) provide you with a sure grip, and there's a gaggle of LED indicators, including a blue power LED that adds a distinct touch of class. Further upping the ante, the Trak Master's microphone preamp is a discrete-transistor, Class A design.

From left to right, the unit provides four sections—labeled Discrete Preamp, Optical Compressor, 3-Band Equaliser, and Output Level—clearly separated by black dotted lines. The Discrete Preamp section provides a +48V phantom-power switch; a balanced XLR mic input; a high-impedance ¼-inch instrument input; an Input Gain knob, which ranges from –3 to +57 dBu; a Line switch (used to select a rear-panel line input); a switch that engages a 75 Hz highpass filter (this filter affects the mic input only); and two status LEDs, one for signal-present (marked SIG) and the other for overload (marked O/L). The two front-panel inputs are selected when the Line switch is disengaged; if a mic and instrument are connected simultaneously, the instrument input overrides the mic input.

The Optical Compressor section provides a Compression knob (marked simply Less and More at the extremes, with hash marks in between); a switch labeled Tight that selects a 6:1 compression ratio when engaged and a 3:1 ratio when disengaged; a switch labeled Punch that is said to restore "punch" lost to compression; a six-segment Gain Reduction meter with LEDs marked –20, –16, –12, –9, –6, and –3; a Release knob (marked Fast and Slow at the extremes, with hash marks in between); a Tube Sound knob (marked Cool and Warm at the extremes, with hash marks between); a Makeup gain knob that ranges from 0 to +20 dBu; a Post EQ switch that, when engaged, causes the compressor to process the signal after the 3-band EQ section rather than before it; and an illuminated Comp In switch (which engages the compressor section).

The 3-Band Equaliser section is itself divided into three sections: Bass, Presence, and Treble. In addition, two Inst./Vocal switches—one each for the Bass and Treble sections—let you choose among different set frequency ranges. The Bass section has a center-detented Gain knob and a Freq knob. When the Inst./Vocal switch is engaged (Inst. mode), the Bass section acts as a low-frequency shelving EQ and permits selection of values in the 25 to 400 Hz range using the Freq knob; when it's disengaged (Vocal mode), the Bass section acts as a low-mid parametric EQ and permits selection of values in the 50 to 800 Hz range.

The Presence portion of the 3-Band Equaliser consists of a center-detented Gain knob that allows cut or boost at 1.5 kHz. There is no way to adjust the bandwidth of the filter.

The Treble section comprises the aforementioned Inst./Vocal switch and a center-detented Gain knob that allows 14 dBu of cut or boost. When the Inst./Vocal switch is engaged (Inst.), the high-frequency shelving filter acts on frequencies above 3.3 kHz; when disengaged (Vocal), it acts on frequencies above 10 kHz. There's also an EQ In switch, which engages the entire equalizer section.

Last, the Output Level section provides a seven-segment meter with LEDs indicating –20, –10, –6, –4, –2, and 0 dBu levels, as well as overload (marked O/L); a Fader knob that ranges from –∞ to +6 dBu; a status LED labeled ADC Lock (for the optional digital output; more

The newest member of Focusrite's Platinum series, the Trak Master, provides a high-quality Class A mic pre and DI, an optical compressor, and a 3-band EQ in an affordable 1U channel strip.

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FIG. 1: The Trak Master's rear panel offers ¼-inch outputs at +4 dBu and –10 dBV levels. An optional 24-bit digital sound card provides S/PDIF output, external word-clock input, and multiple sampling rates as high as 96 kHz.

on that in a moment); a power switch; and the aforementioned blue power-status LED (labeled On).

Regarding the front-panel layout and design, I like that the phantom-power switch is right next to the XLR jack rather than on the back of the unit, as is often the case. It's also convenient having the instrument input on the front panel, especially when the unit is rackmounted. One thing I don't like, though, is the low resolution (3 dB) of the Gain Reduction meters in the compressor section. Even with only 6 LEDs, 2 dB steps would have been much more useful—after all, how often does one apply 20 dB of gain reduction?

BRINGING UP THE REAR

The Trak Master's rear panel (see **Fig. 1**) provides two ¼-inch output jacks, one a balanced +4 dBu TRS and the other an unbalanced –10 dBV TS. The outputs are simultaneously available, which is useful for circumventing signal latency when recording into a computer-based digital audio workstation—you can use one output to monitor the input in real time while the other sends the signal to the sound card. There is also a ¼-inch balanced +4 dBu TRS line-level input jack. I would like to have seen a +4 balanced XLR output, as well.

The optional digital output card ($250), which installs on the rear panel of the Trak Master, is 24-bit and can operate at 44.1, 48, 88.2, or 96 kHz sampling rates. It provides an S/PDIF digital output on an RCA connector, an external word-clock input on a BNC connector, and two switches for selecting the sampling rate. When installed, the card activates the Trak Master's ¼-inch ADC Ext Input (also located on the rear panel); an additional line-level signal can then be routed through the spare channel of the stereo digital output. However, the Trak Master I received for review was not fitted with the digital output card.

Under the hood, the Trak Master is equipped with a nice toroidal transformer. However, there are no user-serviceable parts inside the unit, and there's a surprising amount of unused space. Most of the components are surface-mount technology, and none of the IC chips are socketed, so forget about trying out different chips to change the sound of the unit. Moreover, I/O jacks are mounted directly to the PC board, which isn't so great but is to be expected at this price point.

BE PRE-PARED

The Trak Master's manual lists an amazing bandwidth specification of 20 Hz to 250 kHz for the mic pre section. According to conventional wisdom, that is significant overkill for audio applications. Still, certain studies indicate that some people can perceive differences in sonic quality between components that offer extended bandwidth and those that don't. Of course, when it comes to audio, specifications alone are largely meaningless. Some devices that measure terribly—the venerated Fairchild 670 limiter, for example—sound great (and vice versa).

To evaluate the sound of the preamp, I did my initial testing with the compressor and equalizer switched off. I used a variety of microphones, including a 3035, a 4051, and a 4060 from Audio-Technica; a Microtech Gefell KMT 71; and a CAD E200. I patched the Trak Master directly in to my Sony/MCI JH-24 24-track, 2-inch tape deck and recorded vocals (male and female), bass guitar, synth bass, electric guitar, and snare drum. In the end, I concluded that the Trak Master's mic pre

Trak Master Specifications

Inputs	(1) balanced XLR mic (+4 dBu); (1) ¼" high-impedance instrument; (1) ¼" line
Outputs	(1) balanced ¼" TRS (+4 dBu); (1) unbalanced ¼" TS (–10 dBV)
Maximum Input Gain	60 dB (mic); 43 dB (instrument); 20 dB (line)
Maximum Output Gain	26 dB (from +4 dB balanced output)
Frequency Response	20 Hz–250 kHz (–3 dB; mic); 20 Hz–200 kHz (–3 dB; instrument, line)
Total Harmonic Distortion + Noise (@ 1 kHz)	0.002% (mic); 0.003% (instrument); 0.007% (line)
Dimensions	1U × 10.4" (D)
Weight	9.9 lb.
COMPRESSOR	
Ratio	3:1 (soft knee); 6:1 (hard knee); (switchable)
Threshold	–22 dB to +12 dB
Attack Time	3 ms; 45 ms (switchable)
Release Time	100 ms; program dependent (switchable)
Makeup Gain	+20 dB
EQUALIZER	
Bass	25–400 Hz shelving, 50–800 Hz bell (switchable)
Midrange	1.5 kHz bell (fixed)
Treble	3.3, 10 kHz shelving (switchable)
EQ Boost/Cut	+12, –14 (bass); +12, –15 (midrange); ±14 dB (treble)

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section (including its instrument input) is the unit's strongest feature. It sounded slightly lean—a tad bass shy and not really "fat"—but good overall, especially given its cost.

I especially liked the Trak Master as a direct injection box for bass guitar; it offered a nice balance between quickness and roundness. To get a better sense of how the Trak Master fares as a dedicated bass preamp, I had Robert Graham, the bass player in my band, swap out the vintage Alembic F2-b preamp that he normally uses for the Trak Master. Considering that the Alembic is a tube-based preamp (basically a Fender Bassman input stage), I was surprised by how similar the solid-state Trak Master sounded. (That was without engaging the Trak Master's Tube Sound feature.)

THE BIG SQUEEZE

The Trak Master's optical compressor should, in theory, afford smoother results than the more common voltage-controlled-amplifier (VCA)-based units. But again, topology isn't nearly as important as implementation.

The Trak Master compressor sounds great on bass-guitar and synth-bass applications. I liked it quite a bit less on vocals, though with the right vocalist and microphone, it can yield some impressively edgy textures. Some hard-rock vocals I recorded with the CAD E200 were well within the range of acceptable, but I wouldn't look to the Trak Master for subtle gain reduction.

I rarely use compression when I track kick or snare drum, primarily because I record to 2-inch analog tape (and I therefore don't need it); however, I was curious to see how the Trak Master would perform on kick and snare tracks when recording with the compressor section engaged. I miked my 22-inch maple GMS kick drum with an AKG d12e dynamic microphone. The Trak Master preamp captured the sound of the drum nicely, though the sound was slightly thinner than what I'm accustomed to.

Switching the compressor (with the 3:1 ratio selected) into the signal path yielded some interesting tones, especially with the Punch button engaged (this lets more transients through the compressor) and a long release time. The Post EQ switch—a rare feature in units this affordable—allowed for even more variation in tone. I was able to dial in a booming kick sound with lots of overhang, for example, by engaging the Post switch, boosting the lows, and turning the Release knob to its slowest setting. But only the 3:1 ratio worked satisfactorily for recording kick: when I selected the Tight button (which changes the ratio to 6:1), I heard an unwanted clicking sound.

▼

How often does one apply 20 dB of gain reduction?

I had a bit less luck with the Trak Master on snare drum. On a 14-by-4-inch maple Ayotte snare drum miked with an Audio-Technica 4051 condenser—a good (and good-sounding) test mic for this application because it can capture transients faster than the typical dynamic mic—I achieved a fairly clean sound, and the compressor easily crushed the snare into submission. (Naturally, as with the kick drum, I generated better results with the Punch button engaged.) The resulting sound was rather one-dimensional, though, and overall the compressor seemed to make the drum sound smaller.

According to the Trak Master manual, which is well written and informative (the "Beginner's Guide to Compression" section is especially helpful), the Tube Sound circuit "simulates the warmth normally associated with tube or tape distortion." Tube Sound was slightly useful—for rounding out the sound on bass guitar, for example. But the circuit didn't really succeed in simulating tube or tape distortion; moreover, in most applications, it, too, made things sound smaller.

As a standalone insert processor—a somewhat peripheral application, perhaps, but one that is suggested by the manual—the Trak Master's compressor is pretty much without merit. The lack of fine adjustability of the attack and ratio controls, for example, made dialing in a good snare-drum sound nearly impossible. Forget about using it on a kick track—the Trak Master seemed to suck the animation out of the already-recorded track I tried it on. Processing a well-recorded vocal produced somewhat better results, but the Trak Master's compressor never made things sound bigger—and that is primarily what I want from a compressor.

HIGHS AND LOWS

The Trak Master's 3-Band Equaliser is a mixed bag. The Bass section sounded really great for bass-guitar DI duties. (I was using a Jazz-style bass with a graphite neck and a Bartolini pickup.) There was plenty of low end to be had, and it was easy to obtain a great roundness of tone.

As for the midrange band, which is somewhat misleadingly labeled "Presence" (a term more commonly referring to enhanced frequencies in the 3

PRODUCT SUMMARY

Focusrite
Trak Master
mono channel strip
$450

FEATURES	4.0
EASE OF USE	4.0
AUDIO QUALITY	3.5
VALUE	3.0

RATING PRODUCTS FROM 1 TO 5

PROS: Multiple functionality in a 1U box. Good-sounding, Class A, discrete-transistor mic preamp. Excellent DI for bass guitar. Affordable.

CONS: Poor metering for compressor. Midrange equalizer not very usable. No XLR output.

Manufacturer
Focusrite USA Inc./Digidesign (distributor)
tel. (800) 333-2137 or (650) 731-6300
e-mail sales@focusrite.com
Web www.focusrite.com

to 6 kHz region), I didn't find a single source that sounded better when the Gain control was turned in either direction. Turning it to the right (boost) yielded an annoying honk, whether on vocals, snare drum, or bass guitar. Turning it to the left (cut) sucked the life right out of vocals and guitar. The frequency of the filter and its shape seem poorly chosen.

The Treble band is quite a bit more pleasing. The Vocal setting (10 kHz shelving) can really open up a guitar track (as opposed to the Instrument setting, which adds only brightness), and it adds some presence to vocal tracks, as well. Still, a somewhat higher frequency setting—say, 12 kHz—might have sounded better.

IN FOCUS

The Focusrite Trak Master is a well-featured, easy-to-use, and quite affordable channel strip that could prove to be just the ticket for personal-studio operators seeking multiple tracking functions in a single box. The real prize is the unit's Class A mic preamp—not surprising, given the Focusrite pedigree. I also really appreciated the convenient front-panel DI input, which is especially good for bass guitar. Indeed, I wouldn't be surprised if the Trak Master found its way into dedicated bass rigs.

Unfortunately, the Trak Master's compressor and EQ sections aren't quite up to the same level as the unit's outstanding preamp. They're both useful in certain applications, of course, especially the low and high EQ filters; but those in search of more precise equalization and dynamics control might find more long-term value in putting together separate processors as their needs dictate.

Still, the Trak Master is overall a good value, thanks to its preamp and the number of functions it provides. Certainly, users on a tight budget who need the functionality of a channel strip could do worse for the same amount of money.

Richard Alan Salz *is a producer, engineer, and composer living in southern Vermont.*

AUDIO EASE

ALTIVERB 1.4 (MAC)

This real-time sampled reverb is the real deal.

By Larry the O

Dutch developer Audio Ease recently broke new ground with Altiverb, the first real-time reverb plug-in that bases its presets on sampled acoustics. As a MOTU Audio System (MAS) plug-in, the current version works only within MOTU Digital Performer. It relies on a process called *convolution,* in which samples of actual acoustic spaces are used to generate highly realistic reverb effects. (For more about convolution, see the sidebar, "Convolution Evolution.")

Altiverb uses high-powered math to reproduce acoustic spaces, which demands an upscale CPU stuffed with RAM to handle the number crunching. However, the results are unlike the digital reverbs most people are used to.

CONTROL PANEL

In spite of Altiverb's powerful capabilities, the colorful user interface is simple and clean. It is designed to emulate a rackmount unit replete with 3-D knobs, buttons, and LED readouts (see **Fig. 1**).

A single large knob, which controls the reverb time by applying a decay envelope to the impulse response, dominates the left side of the "front panel." The maximum available reverb (100 percent) is the longest that was recorded in the original acoustic space; the knob scales it down from there. A numeric display beneath the knob shows the reverb time as a percentage of the maximum. You can change the value by rotating the knob with the mouse, or by clicking on the knob's edge—the knob will snap to that position. For precise settings, you can also type a value directly in to the numeric display. Whenever you change the reverb time, Altiverb must engage in some heavy processing, so the knob has a small indicator that glows red while the program is recalculating and green when it's ready to go. That usually takes only a few seconds.

FIG. 1: The "front panel" on Audio Ease's Altiverb plug-in mimics a rackmount hardware device except for the unique Monitor display on the right, which shows photos, diagrams, and text pertaining to the current preset.

Aside from the Reverb-Decay knob, the other controls for shaping the reverb sound are the Wet, Dry, and Predelay knobs at the bottom of the center section. The Wet and Dry level knobs offer as much as +24 dB of gain, which could come in handy in some situations. The Predelay knob lets you add a delay between the dry and wet sounds, and it can be set to positive or negative values. When positive, the delay is applied to the reverb; when negative, it is applied to the dry sound, which means the dry sound can be moved right into the reverb (during or after the early reflections)if desired. In some situations, that can create a fuller, more accurate stereo image that retains important localization cues and still lets you adjust the wet/dry mix.

Impulse responses are selected from a drop-down menu in the center of the front panel. Audio Ease has presets for more than two dozen spaces, including Amsterdam's Concertgebouw, acoustically one of the world's best concert halls. Other spaces include schoolrooms, a bathroom, and a recording studio control room. At the Audio Ease Web site, you can download new impulse responses, including several created by Altiverb users. You can even create your own impulse responses in any room.

Altiverb also provides several impulse responses derived from different mic placements within each space. The mic placements often yield quite distinct sounds from one another; they are useful when the basic sound of the room is right but the exact reverb quality isn't.

When Altiverb is inserted in a mono channel, it offers mono-, stereo-, and quad-output algorithms. Stereo-channel inserts offer inputs that are stereo or a mono mix of both channels and stereo or quad outputs. It lacks quad-input algorithms, so only left-right panning is possible with Altiverb; a signal can't be panned front to back through the reverb.

If a quad-output algorithm is chosen, a mono or stereo channel can't return all four channels, so a drop-down menu (to the right of the large knob) lets you assign Altiverb's outputs 3 and 4 (surround left and right) to any odd/even pair of buses in Digital Performer. You must create another stereo strip (channel or aux) in Digital Performer and select the appropriate buses as inputs to route the surround channels.

On the right side of the front panel is an informative Monitor display that can show any kind of text or graphics pertaining to the selected impulse response. The display can include photos of the building or space from which the impulse response was derived, layout diagrams of the mic placement

PRODUCT SUMMARY

Audio Ease
Altiverb 1.4 (Mac)
reverb plug-in
$495

FEATURES	4.0
EASE OF USE	3.0
QUALITY OF SOUNDS	4.5
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Extremely smooth, realistic reverb. Simple user interface. Includes a good collection of impulse responses. Well-written documentation. Useful for applications beyond reverb.

CONS: Staggering processor demands. Lack of parameters creates dependence on impulse response collection. No quad-input algorithms.

Manufacturer
Audio Ease
tel. 31-30-243-3606
e-mail sales@audioease.com
Web www.audioease.com

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ZOOM

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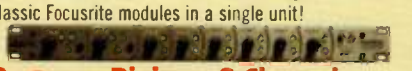

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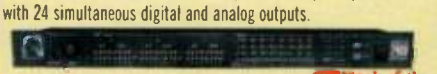

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within the space, information about the recording or the impulse response (reverb length, sampling rate, and so forth), and onscreen help. Forward and back arrows step through the available screens.

HEAVY LOAD

Above the Impulse-Response menu, radio buttons allow you to select one of Altiverb's two operating modes: No-Latency/High-Processor Load or High-Latency/Low-Processor Load. In terms of performance, that's where you must choose between the devil and the deep blue sea. When Altiverb refers to "high processor load," take it at its word. A first-generation G4/500 MHz Mac was not just brought to its knees trying to run a stereo-to-stereo algorithm in that mode—it was left facedown on the sidewalk. Although the reverb played fine, the user interface became unresponsive for perhaps 30 seconds at a time, returning for only about 10 seconds before freezing again. "Difficult to use" would be understating the situation, but I managed to get some practical work done nonetheless. However, keep in mind that I was using Altiverb on just one channel.

Even on a new G4/800 MHz dual-processor Mac, the CPU load was heavy. The quad-output algorithms (which I did not dare attempt on the G4/500 MHz) were by far the biggest offenders; a single instance of the plug-in regularly brought up an excessive-CPU-throughput error message from Digital Performer until I increased the buffer size. Even that fix was a little tricky, because the error message provides a button to access the voice-buffer setting in Digital Performer's Studio Configuration dialog, which would be convenient if that were the setting that fixed the problem. Unfortunately, it isn't; the Configure Hardware Driver buffer setting is the one that needs to be adjusted.

Remember that the sessions in which I received the error messages were not laden with tracks and plug-ins; on the contrary, they had as few as two or three tracks and no other plug-ins at all. In addition to the error messages, I battled odd behavior, such as Digital Performer consistently slamming the Performance Monitor window's Processor activity indicator all the way into overload until I restarted the Mac. The MIDI timing also went completely down the drain in some instances.

The performance issues were not consistent from one Digital Performer project to the next, but I encountered performance problems in every session that used the quad-output algorithms in No-Latency/High-Processor Load mode. The mono-to-stereo and stereo-to-stereo presets were much less problematic as long as I used no more than one of each at a time.

You should also believe Altiverb when it uses the term *high latency*. In that mode, which definitely reduces the drag on the CPU, the latency was as much as half a second or so on my G4/800 MHz and considerably more on the G4/500 MHz.

CONVOLUTION EVOLUTION

Digital reverb has always been accomplished with digital-signal-processing (DSP) algorithms (traditionally, networks of allpass and comb filters) that roughly mimic the results of the acoustical reverberation process. Convolution offers another approach, in which one signal is analyzed, and its characteristics are applied to another signal. Although there are numerous applications for convolution in audio—from the useful to the bizarre—reverb is created by applying an impulse response to a signal.

An impulse response is the response of a system, such as a room, to an impulse, such as a shot from a starter pistol, a short sine-wave sweep, or some other audio spike. When you produce an impulse in a room, you can easily hear the room's response in the form of acoustical reverberation. To use convolution to simulate a room's reverb, record an impulse and the resulting response in the room. Then, remove the impulse, leaving only the response (the reverb itself). That is the job of Audio Ease's IR PreProcessor software.

Once you isolate the room's impulse response, you can apply it to each sample of any digital-audio signal; that is the convolution process performed by Altiverb. Each sample acts like an impulse, and the cumulative effect of convolving the impulse response with all those impulses in quick succession re-creates the sound of that signal in that room.

In other words, you produce an audio spike in a nice-sounding room and run the recording through a number cruncher to isolate the impulse response. You can then crunch those numbers with your clarinet concerto to hear the clarinet concerto in that nice-sounding room.

Convolution is complicated and extremely processor intensive, so why go to so much trouble to make reverb with it? Because it produces a sound that is decidedly less artificial than traditional filter networks, and it exhibits exceptional density and smoothness.

Until recently, convolution has been available only as a non-real-time process to produce all sorts of sonic effects, not just reverb. Perhaps the first widely available non-real-time convolver was the Spectrum Multiplication function in E-mu's samplers. Several programs now have convolvers, including BIAS Peak, U&I MetaSynth, and Sonic Foundry Sound Forge. In fact, Sound Forge's convolver, called Acoustic Mirror, is designed specifically to produce reverb.

The hoopla surrounding Sony's DRE-S777 sampling reverb unit stems from the fact that it's a real-time convolution engine for generating reverb. Sony does this with what I call "big iron"—high-powered, real-time DSP chips.

Finally, there's Altiverb: the first real-time, native software convolution engine (no dedicated hardware). In a way, though, Altiverb also uses big iron—it does its heavy computing with the Mac G4's Altivec vector processor chip.

Articulations:
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Long Tones Expressivo
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Staccato Light, Martele, Marcatto, Frog
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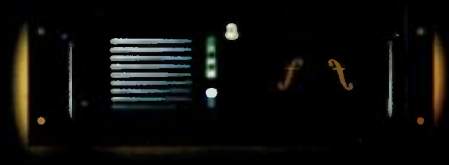

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FIG. 2: IR PreProcessor, Altiverb's companion program, lets you create impulse responses to expand your library of reverb presets.

Fortunately, Altiverb reports the amount of latency (in samples), so you can offset other tracks or edit the time lag after recording.

FOUND SOUNDS

With essentially no reverb parameters to edit, Altiverb leaves you wholly dependent on your collection of impulse responses to get a variety of reverb effects from the plug-in. For many, the provided impulse responses will be adequate, but for those who are undaunted by the prospect of gathering their own impulse responses, Audio Ease provides additional software and information (in the manual) to get the job done.

To create an impulse response, haul your recording equipment to the desired location, record the sound of a starter pistol shot within the space, edit the recording, and feed it to IR PreProcessor, the companion program that comes with Altiverb (see Fig. 2). IR PreProcessor then churns out an impulse response that you can use with the plug-in.

Instead of a pistol shot, you can use other sounds for the impulse source, including a sine-wave sweep, which Audio Ease provides as a sound file on the program CD. Once you have your new impulse response, you can add all of the photos, graphics, and text you want to the Monitor display by importing JPEG, GIF, TIFF, or PICT files.

Remember that convolution lets you impose the characteristics of any sampled and analyzed system onto another system. For example, a click through another digital reverb or a sine-wave sweep recorded through a particular mic or piece of tube gear can be used to make an impulse response for processing signals. (That is how mic-modeling systems work.) However, the results may not always be what you want or expect; it depends on how the process is carried out.

SOUND AND FURY

So, how does Altiverb sound? I tried it on voice, drums, vibes, synth, sound effects, and a few other random sources, and frankly, Altiverb is flat-out the densest and smoothest native reverb I've ever heard. Moreover, it out-performed

an assortment of hardware reverbs I had on hand for comparison.

I heard a clear qualitative difference between the sound of "traditional" digital reverbs and Altiverb. The traditional units produced reverb that sounded more "with" or "on" the source rather than "part of" it. Altiverb blended seamlessly with the source material in a way that definitely felt natural.

On the other hand, traditional digital reverbs have an unnaturally wide soundstage, which is partly how they make things sound so big. Altiverb's stereo algorithms have a narrower, if more even, soundstage than the reverbs with which I compared it. The quad algorithms have a nice sense of envelopment. Presets larger than small rooms seem to have a pronounced delay on the surrounds, which feels spacious on large-space presets, but it's sometimes a little bothersome on medium-space presets.

Altiverb's wide variety of impulse responses offer a broad selection of acoustic environments, but the plug-in retains a characteristic sound. Part of that is its density and smoothness, but almost all of the presets I listened to sounded a little tubby in the 300 to 400 Hz region. Cutting a couple of decibels in that range cleaned it up easily.

Minimum System Requirements

Altiverb

G4/400; 256 MB RAM; OS 8.6;
MOTU Digital Performer;
30 MB RAM allocated to the host program

IMPULSE BUYING

Altiverb represents a significant step forward in desktop audio. Real-time convolution is powerful, and Audio Ease has priced Altiverb on a par with other premium plug-ins. The reverb sound is strikingly good in its naturalness. However, the processor demands are more than I've ever seen in an audio product. Altiverb might be leading the market a bit: on a current top-end Mac, a quad algorithm was barely viable. As successive generations of Mac CPUs hit the streets, the processing demands should become less of an issue.

The user interface is clean and friendly, though a few controls are a bit awkward to manipulate. The 64-page manual is nicely written and well illustrated, and it clearly explains a great deal of arcane information.

Audio Ease plans to release Altiverb in other plug-in formats, and the company is also considering adding a quad-input algorithm. Those developments would be welcome improvements, but Altiverb already has everything it needs to sound great as long as you have the horsepower to do the heavy labor.

Larry the O *provides services as a musician, an engineer, and a producer, and as a sound designer for his company, Toys in the Attic. He has contributed to* **EM** *since 1986.*

PROPELLERHEAD

RECYCLE 2.0 (MAC/WIN)

The original slice-and-dice looper gets a welcome update.

By Jeff Burger

It's easy to see that looped grooves have become an enormously popular and indispensable resource for many desktop musicians. ReCycle, from Swedish company Propellerhead, continues to fuel the popularity by offering an easy-to-use program for creating and manipulating rhythmic loops (see **Fig. 1**).

In a nutshell, ReCycle is a specialized tool for altering sampled grooves, which can then be used in other programs or in hardware samplers. You import a sample, such as a drum loop, and the software automatically slices it into its rhythmic components. With the beats and other rhythmic elements separated, you can do things such as adjust the speed of the loop without altering the pitch (thereby avoiding those pesky artifacts).

After setting other parameters such as loop points, you can save the file in the native REX2 file format or export it as a single audio file. You can also send the sliced-up loop to a sampler, in which case ReCycle generates a MIDI file that enables your sequencer to fire the sample slices appropriately to re-create the loop.

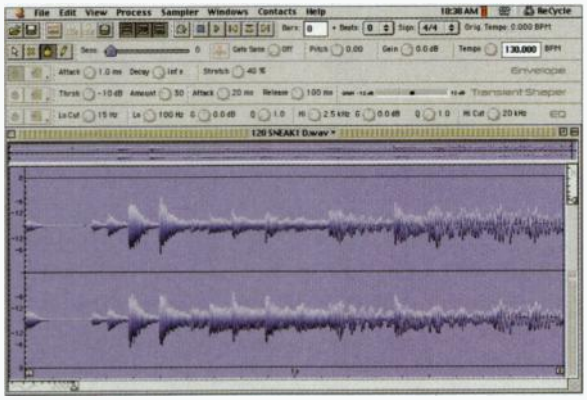

FIG. 1: The simple ReCycle 2.0 interface consists of tool strips and an audio waveform display. Additional tool strips appear when you select effects.

REWIRED FOR STEREO

ReCycle users have long lamented the fact that previous incarnations of the program have been limited to mono. (For an in-depth look at ReCycle 1.7, see the review in the December 1999 issue.) For many users, therefore, the most significant change in ReCycle 2.0 is its ability to import and process stereo samples. The Mac and Windows versions support WAV and AIFF formats with 8-, 16-, and 24-bit resolutions and can handle any sampling rate supported by your sampler or sound card. The Mac version also supports 16- and 24-bit Sound Designer II files. The maximum file length in any format is five minutes.

As mentioned earlier, ReCycle 2.0's native file format is the stereo-savvy REX2. Like its predecessor, REX, REX2 contains the source sample as well as references to the slice and loop points. Although REX files include the results of processing, such as time stretching and normalization, REX2 files include only information about how the host application will perform the processing once the file is loaded. At the time of this writing, the major products that can read REX2 files are Propellerhead Reason (reviewed in the July 2001 issue) and Steinberg Cubase VST 5.0.

ReCycle allows you to export the results of your work in a variety of formats in addition to REX2. With formats such as WAV, AIFF, and Sound Designer II, you can export a single file comprising the entire loop or you can export each slice in the loop as a separate file. The Mixman TRK format used by Mixman Studio and Mixman Studio Pro produces a single document containing the slice points, but the format supports only mono.

Digidesign's SampleCell format exports as an Instrument file along with one or more sound files. SampleCell's template feature is offered as an option. Similarly, the Akai S5000/S6000 format exports a combination of an AKP program file and its associated samples. Other export formats include SoundFonts 2 and MIDI; the MIDI files let you generate grooves when they're imported into your sequencer.

Minimum System Requirements

ReCycle

MAC: Power Mac/166; 64 MB RAM; OS 8.6; color monitor with 800 × 600 resolution

WIN: Pentium/200; 64 MB RAM; Windows 98/ME/NT 4.0/2000; color monitor with 800 × 600 resolution

RETURN TO SENDER

One of ReCycle's handy features is direct communication with a variety of samplers, and the program includes support for several new machines. You need a two-way connection through SCSI or MIDI, depending on the specific sampler and computer. For example, the SCSI drivers in Windows operate only with devices that continuously reply correctly to SCSI 2 Device Inquiry messages—something that not all samplers offer. In addition, the SCSI implementation on the Akai S1000 and S1100 is incompatible with that of some Macs. Unfortunately, those are issues beyond Propellerhead's control.

ReCycle 2 offers specific support for the Roland S-760; Kurzweil K2000, K2500, and K2600; Ensoniq EPS, EPS-16+, ASR-10, and ASR-88; E-mu Esi-32, Esi-4000, E4, e-64, e-6400, and E-Synth; Yamaha A3000; and Akai S1000, S2000, S3000, and CD3000 series. The Akai S5000/S6000 series is supported as a file format because those samplers exchange data with a computer only through Akai's ak.Sys application or through removable media.

ReCycle 2.0 offers generic support for Sample Dump Standard (SDS) through MIDI, SCSI Musical Data Interchange (SMDI), and Extended SMDI (a protocol developed by E-mu that

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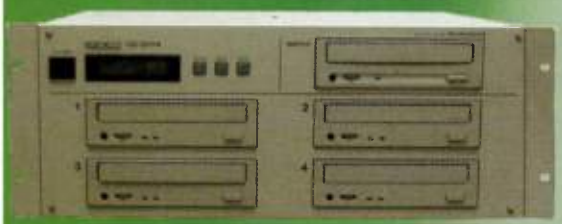

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adds keymapping to SMDI connectivity). At startup ReCycle automatically detects any specifically supported samplers, but you must manually set up the samplers that communicate through generic protocols. There are also provisions for addressing multiple units of a given sampler model. ReCycle's supplemental PDF documentation provides comprehensive information about all supported samplers and formats.

REAL-TIME EFFECTS

In another welcome update, ReCycle 2.0's Preview mode lets you audition settings and effects in real time, which means that you can check out pitch and tempo changes on the fly. EQ is provided in the form of one high-cut, one low-cut, and two parametric filters. The Transient Shaper offers controls and effects that are similar to compression, though it operates on peaks at the onset of slices as opposed to a series of peaks in a continuous audio file. The result is similar in that the individual sound slices become more even in level, which can produce more punch.

ReCycle 2.0 also adds an envelope to the Stretch feature. Stretch provides variable control over the elongation of slice tails to pad out the annoying gaps that appear when slowing a sliced loop below its original tempo. The associated attack and decay controls allow you to dramatically alter the transients and tails of the slices with little effort.

REPORT

You can use ReCycle 2.0 for a number of tasks beyond slicing samples for independent tempo and pitch manipulation. At its simplest, the program is great at finding and editing the ideal loop point in a sample and then exporting the loop as a file. The real-time effects can add some interesting spice to your grooves, though you need some other tools for a full complement of effects processing.

ReCycle is also great for extracting individual sounds, such as snare hits, and sending them to a sampler. You can even isolate and extract the sounds in a loop and send them to a sampler's different outputs. Exported MIDI files serve as groove templates that let you use a sequencer to map the timing of a sampled loop onto any MIDI track. Given the right source material, you can even use ReCycle to quantize audio passages.

For ReCycle to function optimally, the original sample must have clearly perceivable attacks and needs to have been recorded at a healthy volume. Samples with time-based effects, such as short delays, can make it hard for ReCycle to deliver accurate results. On the whole, ReCycle is excellent at finding the right slice points; you just dial in the sensitivity until it picks up all of the desired transients (see **Fig. 2**).

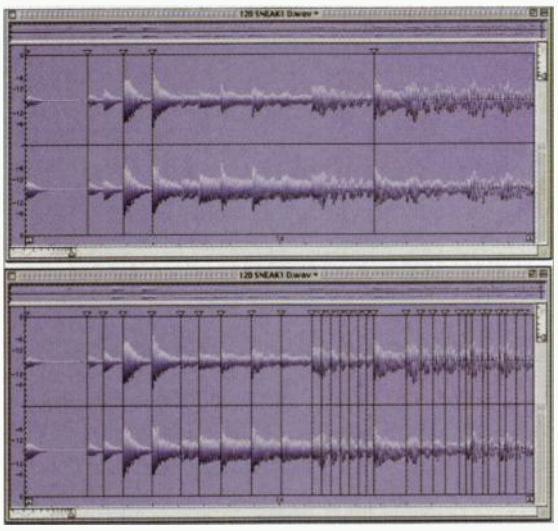

FIG. 2: The Sensitivity setting determines the number and placement of an audio file's slice points. The upper display shows the results of a low Sensitivity setting; the lower display shows the same file with a higher Sensitivity setting.

For difficult passages, you can manually add, move, lock, hide, and delete slice points. You can automatically restrict slices to zero crossings to avoid annoying clicks and pops. Processing options, such as Normalize and Remove DC, come in handy when working with poorly recorded source samples.

ReCycle 2.0 is a breeze to install and use. The printed documentation (which is duplicated in PDF) is friendly and fairly comprehensive. Issues that demand more detailed explanations, such as interoperation with specific samplers and file formats, are covered in additional PDF files on the CD. The Windows version also features online help. On both computer platforms, links to related Web sites are provided on the menu bar. In addition, the CD includes a modest assortment of loop fodder from several third-party developers.

Stereo-file support, loop previewing, and real-time effects are the icing on the cake of this venerable music-production tool. ReCycle's proven track record and its excellent support for hardware devices and other applications adds greatly to its value. If loops are your business, ReCycle 2.0 has what it takes to make your life easier and more productive.

PRODUCT SUMMARY

Propellerhead
ReCycle 2.0 (Mac/Win)
loop editor
$199

FEATURES	3.5
EASE OF USE	4.0
DOCUMENTATION	4.0
VALUE	3.5

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent control over loop slicing. Stereo support. Allows previewing of real-time effects. Intuitive operation. Support for a variety of samplers and file formats. Good documentation and Web presence.

CONS: Effects complement could be more ambitious.

Manufacturer
Propellerhead/Midiman (distributor)
tel. (626) 445-2842
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Jeff Burger *is a songwriter and producer based in Sedona, Arizona.*

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OHM FORCE

OHMBOYZ 1.20 AND PREDATOHM 1.10 (MAC/WIN/BEOS)

Multitap delay and multiband compressor effects with attitude.

By Len Sasso

Ohm Force's multiplatform effects plug-ins OhmBoyz 1.20 and Predatohm 1.10 deliver some unusual twists to two common digital signal processing (DSP) effects: multitap delay and multiband compression. For example, you can morph between presets, you can control all parameters with MIDI, and most parameters have their own dedicated LFO. That level of control makes these plug-ins capable of a variety of extreme effects. On Ohm Force's Web site, you can download demos that work for six hours before noise is introduced into the signal. While you're there, pick up a copy of the free Frohmage, an unusual multiband filter plug-in (see the sidebar, "It's the Cheese").

The Ohm Force plug-ins come in a bewildering assortment of styles and formats. For this review, I tested the VST versions on a Mac G3/300 MHz with Steinberg Cubase VST/32 5.0 as the host. With some feature limitations, I also used them in Emagic Logic Audio 4.8.1 and BIAS Peak VST 2.62. In all cases, the sound quality was excellent and the CPU drain was surprisingly low.

Minimum System Requirements

OhmBoyz and Predatohm

MAC: G3/200; 128 MB RAM; OS 8; compatible plug-in host application

PC: Pentium II/200; 128 MB RAM; Windows 95; compatible plug-in host application

THE MANY FLAVORS OF OHM

OhmBoyz and Predatohm are each available in 20 versions spanning three platforms, three plug-in formats, two levels of DSP quality, and two graphic styles (or *skins*). You can obtain most versions only as downloads, and the Ohm Force Web site is a little difficult to maneuver. On your first visit, figuring out exactly what you want and how to get it can be a daunting task.

All plug-ins are available for the Mac in VST format and for Windows in VST, DirectX, and Winamp formats. VST versions are also available for BeOS on the PC. MOTU Audio System (MAS) and Real Time AudioSuite (RTAS) versions should be available soon.

The Standard-quality versions support 16-bit audio, but they don't allow MIDI automation. Standard versions also do not include a commercial license, so you're not permitted to use them in music that you sell. The Expert-quality versions support 24-bit audio as well as MIDI automation and do include a commercial license. If you produce music mainly for free online distribution in a compressed format such as MP3, RealAudio, or QuickTime, the advantages of 24-bit processing will be lost in the compression anyway. At $9.95 the Standard versions are an incredible bargain.

FIG. 1: OhmBoyz's processing begins with a four-tap Predelay section, followed by parallel delay lines with multimode filters and distortion. Most controls have dedicated LFOs. The Classic-style control panel is shown.

The Standard and Expert versions provide one skin in one plug-in format on one platform. Bundled versions (which are called Professional) are available as well. When you buy a Pro bundle, you download whatever format you need immediately, and Ohm Force will send you a CD-ROM that includes all skins, plug-in formats, and platforms.

The Funky skins have large controls that aren't always identified clearly and have somewhat whimsical graphics and layout. The Classic skins are more compact, with smaller controls arranged in a standard fashion. I chose the Classic skins for the illustrations in this article because they're easier to understand and describe. Don't overlook the Funky skins, though; the on-screen controls respond much better to the mouse, and the goofy graphics might even inspire a little creativity. Unfortunately, you're required to make a choice before you buy; full-size illustrations of both styles are available on the Web site.

BOYZ UNDER THE HOOD

OhmBoyz begins with a stereo, four-tap predelay (see **Fig. 1**). Each tap has its own level, stereo balance, and delay-time controls. You can set delay times in milliseconds or in note increments synchronized to MIDI tempo. There is no feedback in the Predelay section; its main purpose is to feed the parallel feedback delays that follow. Because each parameter of each tap has its own LFO, you can produce interesting results with the predelay alone. For example, if you apply zero, quarter, eighth, and three-quarter beat taps to quarter-note snare or cymbal patterns and add a little random LFO for the tap levels, OhmBoyz produces a great ghost-16th feel that's much like the effect of a drummer playing light 16th notes throughout a pattern.

The Predelay section is followed by two parallel delay lines, each with multiband

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filter, distortion, and high-shelf filter stages. The predelay's left channel is fed into the first delay line, and the right channel is fed into the second. You can turn off the second delay line to save CPU cycles, in which case both predelay channels are fed into the first delay line.

The controls for the Delay section of the delay lines include level, pan, delay time, and feedback. The range of delay time is from 1/64 beat to 4 beats at tempos as low as 60 bpm, which amounts to a maximum four-second delay. (When syncing to MIDI tempo, OhmBoyz doubles tempos below 60 bpm.) Short delay times with high feedback, when applied to percussion, produce a plucked-string effect that sounds physically modeled. You can hear an example of that and other techniques in the MP3 file Ohmboyz.mp3. (I produced all of the online effects from the same four-measure percussion loop with a single pass through OhmBoyz.)

Feedback is adjusted as a percentage of the incoming level; that is, each successive echo's level is a percentage of the previous echo's level. Because no dry signal is present at the delay line output, a feedback setting of zero results in no output from the delay line. That is slightly annoying, because it makes it impossible to use the delay lines as single-tap processors. With too little feedback, you don't hear the echo, and with too much, you hear it several times.

FIG. 2: As many as four user-definable bands of compression or expansion are available in the Predatohm multiband compander. Each band also offers 13 types of distortion. The Classic-style control panel is shown.

The multimode filter section of the delay lines is one of OhmBoyz's most interesting features. Highpass, Lowpass, Bandpass, and Peak modes are available. For all modes, the frequency range is 40.00 Hz to 10.24 kHz; consequently, the first three modes always have some audible effect. For the peak filter, the resonance (Reso) control determines the amount of boost or cut, and you can eliminate the filter by setting it to zero. For the Peak and Bandpass filters, the Q control sets the bandwidth; it has no effect on the other filter modes.

The distortion and high-shelf stages hold no surprises. Distortion is a waveshaper with a variable curve that is continuously adjustable from an S shape to a sine wave. It can simulate everything from common overdrive to nearly white noise. It is most effective on pitched material, especially guitar tracks. You can set the high-shelf filter from 200 Hz to 20 kHz with maximum attenuation around 96 dB and 6 dB-per-octave rolloff.

When you use both delay lines, a balance control lets you cross their feedback. With four taps of predelay, two cross-feedback delay lines, complex filtering and distortion, and 39 LFOs, OhmBoyz can generate some pretty dense effects. When you add MIDI control and morphing between presets, things quickly get out of hand (in a

IT'S THE CHEESE

Frohmage is a free multiband resonant filter that you can download from Ohm Force's Web site. It wears a Funky skin and features the same high audio quality of Ohm Force's commercial plug-ins (see **Fig. A**). Like the other offerings, Frohmage is available in VST (Mac, Win, BeOS), DirectX (Win), and Winamp (Win) formats. MOTU Audio System (MAS) and Real Time AudioSuite (RTAS) versions are forthcoming.

Frohmage is a parallel filter array, and you can control most of its parameters with MIDI. The lowest filter is lowpass, and the rest are bandpass. The central knob sets the lowpass filter's cutoff frequency; in a nice touch, its value can be displayed in hertz or as a MIDI note name (with C5 corresponding to middle C). The arrows and knob at the upper left set the number of bands and the band spacing. The closer the spacing, the more bands needed to cover the same frequency range.

FIG. A: A multiband filter with built-in distortion and delay stages, Frohmage is also a free download with a Funky-style control panel.

At the bottom, the left knob controls the resonance of all of the filters. The Tone slider sets the output balance between the lowpass and the mix of the bandpass filters. Frohmage's most interesting twist is its Evolution feature. The Evol knob sets the lowpass filter's delay time when turned to the left or the bandpass filter's delay time when turned to the right. (What I wouldn't give for an LFO on that knob!)

The knob and switch at the top right control Frohmage's clipping-distortion stage. You can add distortion to the whole signal (left switch position) or to just the lowpass filter (right switch position). The buttons and slider at the top center allow you to load, save, and morph between Frohmage presets with a morph-time range from 1 to 99 seconds. The best way to get a good whiff of Frohmage is to plug it in, start a loop rolling, and morph the factory presets with a few seconds of morph time.

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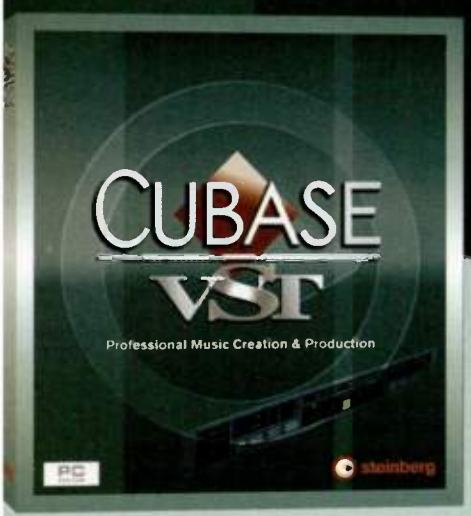

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PRODUCT SUMMARY

Ohm Force

OhmBoyz 1.20 (Mac, Win, BeOS)
multitap delay plug-in
Standard version $9.95
Expert version $79.00
Pro bundle $199.00

FEATURES	4.5
EASE OF USE	3.5
QUALITY OF SOUNDS	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Unique four-tap predelay. Very flexible dual-feedback delays. Wide range of unusual effects using preset morphing and MIDI control.

CONS: Limited implementation and touchy controls in some hosts. Can't recall presets with MIDI Program Change messages.

Manufacturer

Ohm Force
e-mail contact@ohmforce.com
Web www.ohmforce.com

good way). Fortunately, the six banks of factory presets give you a good head start, and more are available on the Ohm Force Web site.

PREDATORY PRACTICES

You can think of Predatohm as a 4-band compander or a 4-band dynamic equalizer (see **Fig. 2**). It divides the frequency spectrum into four user-defined bands and then applies distortion and independent compression or expansion to each band.

Setting up Predatohm's frequency bands is a piece of cake. Simply move the three band sliders at the top of the control panel. The Main Edit window above the sliders shows a slider's position in hertz as you move it. You can reduce the number of bands by pulling the sliders completely to the right. When all of the sliders are completely at the right, you have a single-band compressor.

Each compression band has controls for level, compression or expansion shape and threshold, and the type and amount of distortion. The Character knob controls the compression type; the left half of the knob controls compression, and the right half controls expansion. The Intensity knob determines the compression threshold. The Distortion Type knob selects among 13 types of distortion: hard clipping, 6 tube simulations, and 6 waveshaping curves.

The output of the compressors is mixed and passed through a resonant lowpass filter with variable shape and cutoff frequency. A Feedback delay stage follows the filter. The delay time is displayed in hertz to indicate the frequency at which feedback resonance occurs.

Predatohm's last stage is a stereo simulator. All Predatohm processing is in mono; stereo signals are mixed down to mono at the input. The Super Stereo stage sends the raw signal to the left output channel while sending an inverted and delayed copy to the right output channel. The Phase control sets the delay time. Super Stereo will introduce mono incompatibilities (including total cancellation when the delay is zero), so when that's a consideration, simply turn it off.

One of the most useful features of Predatohm and the other Ohm Force plug-ins is their ability to morph between presets. As many as eight presets can be stored using the preset buttons. Recalling a preset causes the controls to change to the preset values during the indicated morph time (ranging from 0 to 99 seconds). As settings morph, you can watch the controls gradually change their values. I recorded the MP3 example Predatohm.mp3 in one pass using two ten-second morphs.

As cool as the morphing feature is, it is a pity that you can't synchronize the morph time to MIDI tempo and that you can't recall (and therefore morph) presets using MIDI Program Change messages. Rather than being able to bounce a morph completely within the digital domain, you must manually click on the preset buttons while you route the audio out and back into the computer.

PRODUCT SUMMARY

Ohm Force

Predatohm 1.10 (Mac, Win, BeOS)
multiband compander plug-in
Standard version $9.95
Expert version $59.00
Pro bundle $149.00

FEATURES	3.5
EASE OF USE	3.5
QUALITY OF SOUNDS	4.0
VALUE	3.5

RATING PRODUCTS FROM 1 TO 5

PROS: Easy multiband setup for compression and expansion. Wide range of distortion types.

CONS: Limited implementation and touchy controls in some hosts. Can't recall presets with MIDI Program Change messages. Mono processing.

Manufacturer

Ohm Force
e-mail contact@ohmforce.com
Web www.ohmforce.com

OM FOR OHM

I liked both plug-ins and found them useful for a wide range of digital-signal processing needs. Of the two, OhmBoyz is the more interesting and is capable of a huge variety of effects. Predatohm's compression seems a little heavy-handed, but if you happen to be into distortion, it makes an excellent choice. The Standard versions are an incredible bargain, and the Expert versions seem fairly priced (though it would certainly be nice to get both skins).

The ease of using OhmBoyz or Predatohm depends on your choice of platform and host. On the Macintosh, in everything but Cubase VST, the Classic skin controls are touchy and difficult to use. Cubase is also the only host that supports MIDI control and tempo synchronization—two key features. Nevertheless, the sound quality is uniformly good, and both plug-ins are eminently usable in all hosts.

Len Sasso *can be contacted through his Web site at www.swiftkick.com.*

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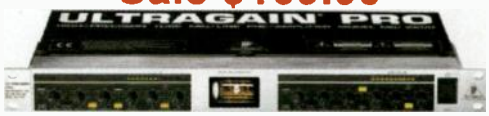

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AARDVARK

DIRECT PRO Q10 (WIN)

A flexible audio interface with plenty of I/O.

By Allan Metts

It seems that every new computer-based audio-product release offers more channels, fancier features, higher audio quality, and, on occasion, a lower price. Aardvark is a strong participant in the battle for your computer audio dollars, and with the Direct Pro Q10, your money buys more than ever before. The Q10 provides ten channels of 24-bit audio I/O, eight mic preamps, a software-controlled mixer, and a host of other goodies.

Minimum System Requirements

Direct Pro Q10

Pentium/233; 64 MB RAM; Windows 95/98/ME; PCI slot

Q IT UP

The Q10 consists of a PCI-based host adapter, a rackmountable interface box, and a cable that connects the two. The 6-foot cable was long enough to reach from my computer to the ideal mounting spot in my equipment rack (right beneath my patch bay). If your studio is more spacious, you might wish that the Q10's cable were a bit longer.

The PCI card is heavily shielded and contains only the connector for the audio interface cable. All A/D and D/A conversion takes place in the external interface box, and all of the audio, MIDI, and synchronization connections are located there. I definitely prefer the Q10's scheme to products that put audio, MIDI, or sync connections on the host card itself, an arrangement that requires me to crawl behind my computer and curse the rat's nest of cables back there.

Silver with bright purple rack ears, the Q10's interface box certainly stands out. I realize that manufacturers need to capture the attention of an increasingly fickle public, but I'm spotting a disturbing trend in my equipment rack. I thought my studio looked good when everything was composed of austere black rack units; now I have Joemeek green, PreSonus blue, JLCooper gray, and (sigh) Aardvark purple.

The Q10's front panel contains eight combo jacks that accept XLR and ¼-inch connectors (see **Fig. 1**). The line-level inputs can accommodate +4 dBu and –10 dBV signals on balanced or unbalanced connectors. Mic inputs 1 through 4 come with 48V phantom power, but a single switch puts power on all four jacks. The phantom power doesn't affect the line-level inputs. When I had only one or two condenser mics in use, I hooked line-level sources to the other powered connectors, and no inputs were wasted.

Inputs 7 and 8 can accommodate high-impedance, low-level sources such as electric guitars. By activating one of the two guitar switches on the front panel, you can plug a guitar directly in to the Q10 without using a direct box or a separate preamp.

Also on the front panel is a level control for the monitor outputs and a headphone jack with its own level control. A bright red LED glows when power is applied, and believe it or not, you have software control over the brightness of this power indicator. The interface box has no separate power connector and is powered entirely from the cable to your computer.

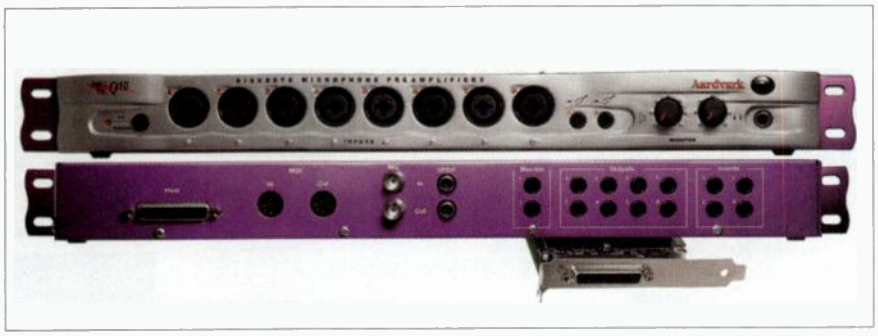

FIG. 1: The Q10 computer audio interface lets you record and play back ten channels without a mixer. Eight combo jacks with mic preamps are located on the front panel. On the back panel are ports for the host PCI card connector, MIDI I/O, word-clock I/O, S/PDIF I/O, four effects inserts, stereo monitor outputs, and eight analog audio outputs.

BEHIND THE BOX

The Q10's rear panel, in addition to the connector for the host adapter cable, sports in and out connectors for MIDI, word clock, and S/PDIF. It has eight balanced ¼-inch analog channel outputs and four channel inserts hardwired to input channels 1 through 4.

Two additional outputs, Monitor L and Monitor R, are intended to feed your control-room monitors. Typically, you run your master mix signal through those outputs. However, you can route quite a variety of signals through them. The only technical difference between the monitor outputs and the other analog outputs is that you can control their volume with the front panel's Monitor Level knob.

Aardvark offers driver support for Windows MME, ASIO2, DirectSound, and Tascam GigaStudio (GSIF), but, unfortunately, only for the Windows 95, 98, and ME platforms. MME and ASIO drivers for Windows 2000 and XP are in the beta stage; I tried them, but they weren't quite ready for prime time. Hopefully, official 2000 and XP drivers will be available by the time you read this. According to Aardvark, WDM and Macintosh drivers are in development, as well.

Installing the Q10 software is straightforward. Windows recognizes a new Plug-and-Play device as it boots up, and you simply point to the Q10 driver on the installation CD. Once the driver is in place, you install the Control Panel software from the CD.

IN COMPLETE CONTROL

The Q10 Control Panel gives you total control of the Q10 hardware, allowing you to set and check levels, route audio, and configure every aspect of your Q10 system (see **Fig. 2**). The Control Panel consists of one main screen and some ancillary dialog boxes. The window isn't resizable, but an Always on Top option lets you keep it visible when other audio software is active. You can even decide if the onscreen mixer will be blue, silver, red, or gold. As many as four Q10s (or any member of Aardvark's Direct Pro series) can be installed in your system at the same time. The Control Panel provides a menu command to switch between any installed host adapters.

The Control Panel's left side presents eight fader groups, one for each analog input, and a stereo fader for the S/PDIF input. In addition to a fader, each analog fader group has a mic/line switch, a trim control, and a level meter. The meters have peak indicators (which can be switched off); they light when an overload condition exists.

FIG. 2: The Q10 Control Panel software presents an intuitive and powerful user interface. You can access all ten input and all ten playback channels simultaneously.

Each fader group also has numerical indicators for trim, fader position, and level (or peak, if you prefer). For stereo operation, Link buttons ensure that settings for adjacent fader groups remain identical. Each fader group provides Mute and Solo buttons and a Pan control. The sole purpose of the Pan control is to place the channel's signal in the Q10 monitor bus. You'll use your audio-recording software to establish pan settings in your recordings for the input channels.

Direct Pro Q10 Specifications

Resolution	24-bit
Sampling Rates	32, 44.1, 48 kHz
Frequency Response	7 Hz–44 kHz, ±0.5 dB
Dynamic Range	110 dB (D/A); 100 dB (A/D)
Total Harmonic Distortion + Noise	0.002% @ 1 kHz
Analog Audio Inputs	(8) combo connectors: XLR mic inputs with input trim (phantom power on 1–4) and balanced ¼" line inputs (+4 dBu/–10 dBV) with input trim; (2) unbalanced ¼" hi-Z (replace line in 1 and 2) with input trim
Analog Audio Inserts	(4) ¼" TRS
Analog Audio Outputs	(8) balanced ¼" channel outputs (+4 dBu/–10 dBV); (2) balanced monitor outputs (+4 dBu/–10 dBV); (1) ¼" stereo headphone
Digital-Audio I/O	24-bit S/PDIF RCA
Sync I/O	(1) BNC word-clock in; (1) BNC word-clock out; S/PDIF clock; MTC
MIDI	In, Out
Expansion Card Type	PCI, 5" length
Onboard DSP	24-bit, 80 MIPS
Expansion Card Connector	(1) 6' shielded 25-pin connector
Dimensions (interface box)	1U × 6.5" (D)
Weight	6.5 lb.

They *LAUGHED* when I said they could have **Perfect Pitch**

. . . until I showed them the simple secret —*and they heard it for themselves!*

David Lucas Burge

Learn to recognize EXACT tones and chords—BY EAR!

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The true story behind the worldwide #1 best-selling ear training method

by **David Lucas Burge**

It all started as a sort of teenage rivalry . . .

I'd slave at the piano for five hours daily. Linda practiced far less. Yet somehow she always shined as the star performer at our school. It was frustrating. *What does she have that I don't?* I'd wonder.

Linda's best friend, Sheryl, bragged on and on to me, adding more fuel to my fire. "*You* could never be as good as *Linda*," she would taunt. "*Linda's* got *Perfect Pitch*."

"What's Perfect Pitch?" I asked.

Sheryl gloated about some of Linda's uncanny abilities: how she could name *exact tones and chords*—all BY EAR; how she could sing any tone —*from mere memory*; how she could play songs —after just *hearing* them!

My heart sank. *Her fantastic EAR is the key to her success.* How could I ever hope to compete with her?

But it bothered me. Did she *really* have Perfect Pitch? I finally asked Linda point-blank if it was true.

"Yes," she nodded to me aloofly.

But Perfect Pitch was too good to believe. I rudely pressed, "*Can I test you sometime?*"

"OK," she replied.

Now she'd eat her words . . .

My plot was ingeniously simple: When Linda least suspected, I challenged her to name tones—*by ear*.

I made her stand so she could not see the piano keyboard. I made sure other classmates could not help her. I set up everything perfectly so I could expose her Perfect Pitch claims as a ridiculous joke.

With silent apprehension, I selected a tone to play. (She'll *never* guess F♯!)

I had barely touched the key.

"F♯," she said. I was astonished.

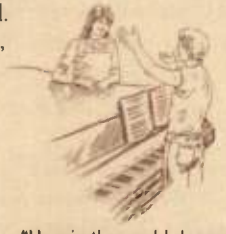
I played another tone.

"C," she announced, not stopping to think.

Frantically, I played more tones, skipping here and there all over the keyboard. But somehow she knew the pitch each time. She was *AMAZING!*

"Sing an E♭," I demanded, determined to mess her up. She sang a tone. I checked her on the keyboard—but she was right on!

Now I started to boil. I called out more tones, trying hard to make them increasingly difficult. Still she sang each note perfectly on pitch.



"How in the world do you do it?" I blurted. I was totally boggled. (age 14, 9th grade)

I was totally boggled. "*How in the world do you do it?*" I blurted.

"I don't know," she sighed. And that was all I could get out of her!

The dazzle of Perfect Pitch hit me like a ton of bricks. My head was dizzy with disbelief. Yet from then on, I knew that Perfect Pitch was real.

I couldn't figure it out . . .

"*How does she DO it?*" I kept asking myself. On the other hand, why can't *everyone* recognize tones by ear? It dawned on me: people call themselves *musicians* and yet they can't tell a C from a C♯?? Or A major from F major?! That's as strange as a portrait painter who can't name the colors of paint on his palette! It all seemed odd and contradictory.

Humiliated and puzzled, I went home to work on this problem. At age 14, this was a hard nut to crack.

You can be sure I tried it for myself. With a little sweet-talking, I would get my three brothers and two sisters to play tones for me—to name by ear. But it turned into a guessing game I just couldn't win.

Day after day I tried to learn Perfect Pitch. I would play a tone *over* and *over* to make it stick in my head. But later I couldn't remember any of them. And I couldn't recognize any of the tones by ear. Somehow they all sounded the same after awhile; how were you supposed to know which was which—just by *listening?*

I would have done anything to have an ear like Linda, but it was way beyond my reach.

So, finally, I gave up.

Then it happened . . .

It was like a miracle . . . a twist of fate . . . like finding the lost Holy Grail. Once I stopped *straining* my ear, I started to listen NATURALLY. Then the incredible secret to Perfect Pitch jumped right into my lap.

I began to notice faint "colors" within the tones. Not *visual* colors, but colors of *pitch*, colors of *sound*. They had always been there. But this was the first time I had ever really "let go"—and *listened*—to discover these subtle differences.

Soon—to my own disbelief—*I too could recognize the tones by ear!* It was simple. I could hear how F♯ sounds one way, while B♭ has a *totally different sound*—sort of like "hearing" red and blue!

The realization struck me: THIS IS PERFECT PITCH! This is how Bach, Beethoven, and Mozart could mentally envision their masterpieces—and

know tones, chords, and keys—all by ear!

It was almost childish—I felt sure that *anyone* could unlock their own Perfect Pitch by learning this simple secret of "color hearing."

Bursting with excitement, I went to tell my best friend, Ann (a flutist).

She *laughed* at me. "You have to be *born* with Perfect Pitch," she asserted. "You can't *develop* it."

"You don't understand Perfect Pitch," I countered. I showed her how to listen. Timidly, she confessed that she too could hear the pitch colors. With this jump start, Ann soon realized that she had also gained Perfect Pitch for herself.

We became instant celebrities. Classmates loved to call out tones for us to magically sing from thin air. They played chords for us to name by ear. They quizzed us on what key a song was in. Everyone was endlessly fascinated with our "supernatural" powers, yet to Ann and me, it was just normal.

Back then I never dreamt I would later cause such a stir in the academic world. But as I entered college and started to explain my discovery, many professors *laughed* at me.

"You must be *born* with Perfect Pitch," they'd say. "You can't *develop* it."

I would listen politely. Then I'd reveal the simple secret—*so they could hear it for themselves*. You'd be surprised how fast they changed their tune!

In college, my so-called "perfect ear" allowed me to *skip over* two required music courses. Perfect Pitch made *everything* easier for me—my ability to perform, compose, arrange, transpose, improvise, sight-read (because—without looking—you're sure you're playing the correct tones)—and my *enjoyment* of music skyrocketed. I learned that music is very definitely a HEARING art.

Oh, so you must be wondering what happened with Linda? Please excuse me, I'll have to backtrack . . .

It was now my senior year of high school. I was nearly 18. In these three-and-a-half years with Perfect Pitch, my piano teacher insisted I had made ten years of progress. And I had. But my youthful ambition still wasn't satisfied. I needed one more each spring, complete with judges and awards. To my horror, they scheduled me that year as the *grand finale* of the entire event.

The day arrived. Linda gave her usual sterling performance. She would be tough to match, let alone surpass. But my turn finally came, and I went for it.

Slinking to the stage, I sat down and played my heart out. The applause was overwhelming.

Later, posted on the bulletin board, I discovered my score of A+ in the most advanced performance category.

Linda got an A.

Sweet victory was music to my ears—mine at last!

Join musicians around the world who have discovered a new secret for success with the **Perfect Pitch Ear Training SuperCourse**:

● "Wow! It really worked. I feel like a new musician. I am very proud I could achieve something of this caliber." *J.M.* ● "Thanks...I developed a full Perfect Pitch in just two weeks! I don't know how it worked. It just happened out of nowhere like a miracle." *B.B.* ● "It is wonderful. I can truly hear the differences in the color of the tones." *D.P.* ● "I heard the differences on the initial playing, which did in fact surprise me. It is a breakthrough." *J.H.* ● "I'm able to play things I hear in my head a lot faster than ever before. Before the course, I could barely do it." *J.W.* ● "I hear a song on the radio and I know what they're doing. My improvisations have improved. I feel more in control." *I.B.* ● "In three short weeks I've noticed a vast difference in my listening skills." *T.E.* ● "I can now identify tones and keys just by hearing them. I can recall and sing individual tones at will. When I hear music now it has much more definition, form and substance. I don't just passively listen to music anymore, but actively listen to detail." *M.U.* ● "Although I was skeptical at first, I am now awed." *R.H.* ● "It's like hearing in a whole new dimension." *L.S.* ● "I wish I could have had this 30 years ago!" *R.B.* ● "Very necessary for someone who wants to become a pro." *L.K.* ● "This is absolutely what I have been searching for." *D.F.* ● "Mr. Burge—you've changed my life!" *T.B.* ● "Learn it or be left behind." *P.S.* . . .

Now it's YOUR turn!

For 21 years now, musicians around the globe—plus research at two leading universities (visit EarTraining.com)—have proven the simple method that David Lucas Burge stumbled upon as a teenager.

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Channels 7 and 8 lose their mic/line switches when you activate the guitar switches that are on the interface box. The S/PDIF fader group is identical to the others, except the mic/line switch and trim control are replaced with an indicator for digital signal lock. Because it represents a stereo signal, the S/PDIF fader group has no Pan control.

To the right of the input-fader groups are five stereo faders that represent the playback channels from your audio software. Accompanying each fader are peak/level meters with overload indicators for the left and right channels, a numerical indicator of the fader position, and buttons for mute and solo.

The remaining fader group controls the monitor mix bus. Separate faders and meters for the left and right channels are provided, in addition to numerical indicators for fader position and peak/level.

The Control Panel's main screen has analog-style meters for the left and right monitor signals, a phantom-power indicator, the brightness control for the interface box's power LED, and buttons to access the remaining Control Panel features. A Source-Select control lets you choose between 32, 44.1, and 48 kHz operation. (The Q10 does not support 96 kHz audio, but such support is supposed to be available in a future software release.) If you want to sync to an external source, use the Source-Select control to select S/PDIF or word clock.

TAKE THE BEST ROUTE

The Routing screen lets you choose an audio signal for each audio output that appears on the back of the Q10 interface box (see **Fig. 3**). The sources include any input channel, any playback channel, the left or right channel of the monitor mix, a 1 kHz test tone, and digital silence. The S/PDIF always outputs the signal that appears on analog outputs 7 and 8, but I can live with that restriction.

You can create any number of routing presets, representing your favorite configurations for the audio-signal outputs. That's the good news; the bad news is that it takes a whopping six mouse-clicks to recall a routing preset from the main Control Panel screen (including clicks required to close the dialogs you open in the process).

Another set of presets can be yours in only four mouse-clicks, but that's still too many clicks. Those presets let you capture all of the settings in the Control Panel, including output routings, fader positions, trim levels, and configuration parameters. The presets are handy, but what I really wanted was MIDI-based control over mixing and routing. Such capabilities would let me fully automate my mixes with a sequencer and use an external hardware control surface.

A click of the Control Panel's Advanced button brings up a multitabbed dialog box containing all sorts of settings and information. There you can fine-tune the ASIO and DirectSound performance, switch the analog outputs between –10 dBV and +4 dBu operation (each output is individually switchable), and view the in-use status for each of the Q10's five stereo channels.

The advanced-settings dialog also lets you switch the meters between prefader and postfader operation and switch the S/PDIF output between consumer and professional formats. One setting switches

PRODUCT SUMMARY

Aardvark
Direct Pro Q10 (Win)
audio interface
$999

FEATURES	4.5
EASE OF USE	4.5
AUDIO QUALITY	4.5
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Extensive routing capabilities. Intuitive Control Panel. Good-quality preamps. Clean sound.

CONS: Drivers only for Windows 95, 98, and ME. No MIDI-based controller automation.

Manufacturer
Aardvark
tel. (734) 665-8899
e-mail info@aardvark-pro.com
Web www.aardvark-pro.com

FIG. 3: Each Q10 output can carry any of the available audio signals. From the Routing screen, you can choose from sources that include any playback channel, any input channel, a mix bus, a test tone, or digital silence.

the recording source of channels 9 and 10; by default, those channels carry the signal from the S/PDIF input. When you switch them to the Monitor mix instead, you gain instant, all-digital mixdown capability from channels 1 through 8 to channels 9 and 10.

THINKING ABOUT Q

My experience with the Q10 was entirely positive. In my personal studio, I want to simply sit down and play an instrument without having to configure a bunch of software settings. With the Q10's built-in monitor mix and front-panel level control, I could do that easily (though I still had to boot up the computer). I could plug in a microphone, guitar, and keyboard and then record them with ease. I could put the instrument I was playing in a headphone mix with the playback tracks and record only the instrument. I could even mix down my tracks without ever leaving the digital realm. With a modest studio setup such as mine, it's possible to completely replace a mixer with the Q10 and a patch bay.

The Q10's audio quality is excellent. Self-noise was inaudible except at extreme settings. When I connected my Les Paul Studio, the guitar inputs sounded warm and clean. I did a very unscientific comparison between the Q10's mic preamps and a PreSonus MP20 (a 2000 **EM** Editors' Choice winner). The Q10 held up quite well. The MP20 exhibited a subtle fullness that I couldn't hear in the Q10, but I would have no qualms about using the Q10 for serious recording.

The documentation for the Q10 is adequate but not outstanding. A 44-page manual takes you through the system, and that's really all you need. There's no online help, but frankly, I didn't miss it.

The Q10 is an excellent system for the small studio, provided you run Windows 95, 98, or ME on your computer. My primary environment is Windows 2000 and Cakewalk Sonar, so I'm anxious to see solid Windows 2000 support and WDM drivers for the Q10. If you're looking for a capable system to record and mix down ten channels of audio, the Q10 is hard to beat.

Allan Metts *is an Atlanta-based musician, a software and systems designer, and a consultant.*

APHEX

MODEL 204

Musical clarity and punch in an affordable 2-channel box.

By Michael Cooper

The original Aphex Aural Exciter, first offered in 1975, came without the Big Bottom circuit, which was added in 1992. Later revisions included the Model 104 Type C and Type C2 units. Aphex has now released the Model 204 Aural Exciter and Optical Big Bottom, yet another refinement of the original unit.

For those not familiar with these proprietary Aphex processes, the Aural Exciter enhances clarity and intelligibility by adding phase shift and musically related synthesized harmonics to audio signals. The Big Bottom circuit combines a lowpass filter and dynamics processor to compress and delay incoming low-frequency information. The process is reverse amplitude dependent, meaning that more is applied as the input level drops and less as the signal gets hotter. Together the dynamics processor and time delay create sustained bass frequencies that are perceived as being louder yet don't noticeably increase peak output. According to Aphex, the sleek Model 204 updates the Aural Exciter and Big Bottom processor blocks with improved circuitry, including an optical gain-control element for the Big Bottom compressor.

SHAPE OF THINGS

Like its most recent predecessor, the single-rackspace Model 204 provides two identical channels, each with independent controls for its two processing sections. Understand that the Big Bottom and Aural Exciter circuits are wired in parallel to the dry audio signal, thus preserving the phase, spectral content, and dynamics of the original signal. The processed signals are summed with the dry signal at the 204's output jacks.

Three continuously variable rotary knobs—labeled Drive, Tune, and Mix—control the Big Bottom process per channel. Drive adjusts the amount of signal sent to the optical compressor, and an associated LED lights to alert you when the compressor's threshold is exceeded. Tune sets the corner frequency, from 49 to 190 Hz, for a lowpass filter that lies just before the compressor in the Big Bottom's circuit path. The user controls the range of the bass frequencies that the Big Bottom compresses by tweaking the Tune knob. The Mix knob determines how much of the Big Bottom's compressed bass-band signal is added back in with the dry signal.

Three other continuously variable rotary knobs—labeled Tune, Harmonics, and Mix—control the 204's Aural Exciter circuitry per channel. Tune sets the corner frequency for a highpass filter that ranges from 800 Hz to 6 kHz. All frequencies higher than the Tune knob's setting are patched through to a harmonics generator. The Harmonics knob, which controls the number of harmonics added to the highpass filter's output signal, adds higher partials as you turn the knob clockwise. The Mix knob adjusts how much of this harmonics-rich signal is added back in with the dry signal.

The Aphex Model 204 Aural Exciter and Optical Big Bottom is the latest revision of the company's popular processor. Changes include improved circuit design and an optical gain-control element for the Big Bottom compressor.

PRODUCT SUMMARY

Aphex
Model 204
spectral enhancer
$399

FEATURES	3.5
EASE OF USE	4.5
AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent for percussive instruments, remastering applications, and preconditioning cassette mixes. Affordable. Easy to use. Plug-and-play operation with balanced or unbalanced lines. Operating levels independently switchable per channel. High bandwidth and headroom.

CONS: Processes lack independent bypasses. Corner frequencies not indicated on Tune controls. No sidechain inputs. No stereo-link function. High- and low-pass filters can force compromised settings that affect frequencies adjacent to intended area of treatment.

Manufacturer
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e-mail sales@aphex.com
Web www.aphex.com

Two backlit Process In/Out switches allow for global bypass on each channel; rather than provide shunts between the input and output jacks, these simply defeat the processing for each channel so only the original signal flows through the box. Although you don't get dedicated bypass switches for the separate processes, which would be helpful for A/B comparisons, you can effectively bypass the Aural Exciter or Optical Big Bottom circuit on either channel simply by setting the processor's Mix knob hard left (off). The 204's power switch and associated LED are conveniently located on the front panel.

The 204's rear panel provides XLR and ¼-inch TRS input and output jacks for each channel (see **Fig. 1**). Although both types of jacks are balanced, they also accept unbalanced lines (albeit with the concomitant 6 dB loss in signal level).

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MORE CORE.

You can connect the 204 to your system through mixer insert jacks or by patching the unit in series with other gear.

Each channel sports a rear-panel switch that lets you select between –10 dBV and +4 dBu operating levels. The Model 204 also boasts an internal power supply and a detachable, three-prong AC cord, which connects to the IEC power receptacle on the rear panel.

NICE AND EASY

Learning to use the 204 is easy, thanks in part to the clearly written and comprehensive owner's manual. You need the manual to fully grok the 204's controls. It illustrates, for example, Tune-knob settings (for the Aural Exciter and Big Bottom functions) corresponding to seven corner frequencies. The Tune knobs on the unit, however, are marked Min and Max at the extremes with hash marks between. Clearly, the controls are meant to be set by ear, but frequency-specific marks would be helpful.

Some may wish for input- and output-level controls, but those aren't necessary on the 204 and would add degrading amplification stages to a pristine signal path. The Big Bottom's Drive control provides enough range to trigger the optical compressor, even with fairly weak input signals. If you need more gain sent to the 204, connect the unit to your mixer's insert jacks and boost the mixer channel's trim controls. Moreover, because the 204's processing causes little or no increase in signal level, at least with moderate Mix-knob settings, output level controls are largely unnecessary.

One feature I missed was a stereo-linking function, which would have allowed more accurate processing of stereo program material. But that is a minor lament—matching channel settings is simple enough, and the knob positions are easy to see, even in low light.

SESSION CAT

Using the Big Bottom and Aural Exciter processes made it a cinch to dial in a beefy bottom and snappy click to kick-drum tracks. I did, however, wish the Big Bottom's Tune control could be adjusted to a setting below 49 Hz—often on kick I want to boost only the low lows, around 30 Hz, without affecting frequencies above, say, 60 Hz.

The Aural Exciter also worked wonders on a snare-drum track. Although I had miked the drum's top head only, after processing with the Aural Exciter, it sounded like I had also used a bottom-head mic—the processed sound had much more sizzle and punch. The sole drawback was that the process, no matter how I tuned it, also hyped the hi-hat bleed in the snare mic, making the hats sound a bit harsh and too cutting. That is a good example of a case in which a bandpass filter for the Tune function would have allowed better results.

The Aural Exciter added clarity and restored nuance to electric-bass-guitar tracks. By cranking the Big Bottom's Drive control and turning down the Mix knob, I achieved increased sustain and fairly consistent levels on the track. It wasn't as good as what can be had using a conventional split-band compressor or even a broadband compressor such as the Aphex Expressor, but it was usable nonetheless. Once again, though, I wished that I could lower the Big Bottom's corner frequency to add thunder without any hint of boominess.

The Aural Exciter also lent clarity to and increased the intelligibility of vocal tracks and added wonderful sparkle to acoustic-guitar tracks. But it was on drum-overhead mics that the Aural Exciter sounded truly phenomenal. The cymbals grew richly complex from the added harmonics, and a touch of the Big Bottom gave some nice room tone to the kick drum. Drum overheads turned out to be my favorite application for the 204.

As a stereo-mix processor strapped across my Yamaha 02R's stereo-bus outputs, the 204 gave added punch and zing to 2-track mixes. The unit had no trouble handling the +26 dBu levels the 02R's analog outputs dished out. In this application, it was usually best to tune the Aural Exciter's highpass filter to its highest setting and the Big Bottom's lowpass to its lowest frequency. Still, it wasn't always possible to get the effect I wanted without also affecting frequencies at the far ends of the spectrum that I preferred to leave untouched.

For another test, I used the 204 to enhance an old 16-bit mix that was lacking in extreme highs and deep lows. After setting the controls pretty much as described in the aforementioned stereo-bus application, I was amazed by how punchy and airy the processed recording sounded. This was no substitute for a great mix, but the experiment clearly demonstrated the 204's usefulness for remastering applications.

The 204 was useful in preprocessing stereo mixes before printing them to

Model 204 Specifications

Inputs	(2) transformerless, active-balanced XLR; (2) transformerless, active-balanced ¼" TRS
Outputs	(2) transformerless, active-balanced XLR; (2) transformerless, active-balanced ¼" TRS
Maximum Input Level	+27 dBu (+4 dBu operating level); +12.5 dBV (–10 dBV operating level)
Maximum Output Level	+25 dBu (+4 dBu operating level); +12.5 dBV (–10 dBV operating level)
Operating Level	+4 dBu or –10 dBV (switchable per channel)
Frequency Response	10 Hz–38 kHz (+0.5 dB)
Dynamic Range	120 dB
Hum and Noise	–93 dBu (unweighted, 22 Hz–22 kHz)
Crosstalk	–79 dB (10 Hz–22 kHz)
Total Harmonic Distortion	0.0003% (10 Hz–22 kHz @ max. output)
Dimensions	1U × 8.25" (D)
Weight	6 lb.

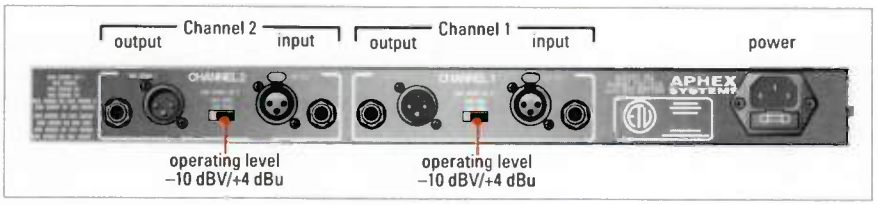


FIG. 1: In addition to balanced XLR and TRS ¼-inch I/O, the 204's rear panel also provides for +4 dBu or –10 dBV operation, switchable per channel.

cassettes. You're probably asking, "Who needs those anymore?" But recently, a major producer (whom, for privacy, I won't name) requested that demos be submitted to him on "old-fashioned cassette tapes." Compared with unprocessed cassette dubs, those preprocessed with the 204 exhibited improved clarity, intelligibility, and punch, as though a cloudy veil had been lifted from the sound.

THE GREAT ENHANCER

The Aphex Model 204 Aural Exciter and Optical Big Bottom is a unique, useful, affordable processor. For most applications, I was more impressed with the Aural Exciter function than with the Big Bottom. The Aural Exciter does what no other processor can. Especially on percussive (plucked or struck) instruments that produce inherently complex overtones, the Aural Exciter adds an airiness and richness that's hard to beat.

The Big Bottom, too, can yield great results in some applications—most notably processing drum overheads and remastering recordings that are deficient in bass frequencies—but in general, I prefer the flexibility afforded by a quality split-band compressor or an equalizer and a compressor chained in series. Then again, that signal chain costs considerably more than the 204.

My main reservation about the Model 204 is its lack of precision control: rather than lowpass and highpass filters, bandpass controls or external sidechain inputs would have allowed more exacting control over what frequencies get processed by the Big Bottom and Aural Exciter blocks. But again, such features would add considerable expense to the modestly priced 204.

One last caveat: the 204's wide-ranging controls make it easy to overprocess signals, resulting in boomy lows and harsh highs. Used judiciously, though, the Model 204 can greatly enhance individual tracks and full mixes. The 204 isn't just a valuable tool; at under $400 for two channels, it's a bargain.

Michael Cooper *is an* **EM** *contributing editor and owner of Michael Cooper Recording, located in beautiful Sisters, Oregon.*

Quick Picks

PRIMACOUSTIC

London 14 Acoustic Treatment

By Jeff Burger

One of the most overlooked aspects of putting together a project studio is acoustic treatment. A variety of factors contribute, including expense, room logistics, and the perception of acoustic design as somewhat of a black art. Primacoustic has set out to address those issues by offering comparatively inexpensive acoustic-treatment solutions in packages designed around common room sizes and audio applications. I reviewed the London 14 ($600).

All Primacoustic products are made from high-density, flame-retardant urethane foam in a charcoal gray color. According to the company, gray is least susceptible to fading when subjected to light. You can spray paint with the color of your choice, but be careful not to fill the pores. The hard paint coat will increase reflection at very high frequencies and trap more lows.

Unlike most acoustic treatments, there are no waffled molds. Instead, Primacoustic's approach is to combine prefabricated panels made from high-density, open-cell foam formed as flat panels or trapezoids. The panels are combined in kits designed for specific applications, along with various suggested assembly patterns. The company's Web site provides a good resource and some visuals that make my discussion more tangible.

London Calling

The London series is Primacoustic's package designed for the average project studio where the same space is often used for mixing and recording duties. Standard sizes include 12 feet by 9 feet, 14 feet by 10 feet, and 16 feet by 12 feet. The London kits incorporate four Primacoustic kits: Europa Flutter Wall, Orientique Washboard, Scandia Scatter Blocks, and Australis Bass Trap.

The Europa Flutter Wall is a broadband acoustic absorber designed to go on the front wall to eliminate standing waves and front-to-back room chatter. The Europa carries a low-end spec of 400 Hz, with component panels including 3-inch-thick midbass blocks, 2-inch-thick voice-range blocks, and small anechoic wedges for soft diffusion of high frequencies.

The Orientique Washboards mount on the sidewalls and are constructed to squelch side-to-side chatter and primary reflections from the sides of the monitors. The Orientique is made from the same building blocks as the Europa, except the wedges are longer and, therefore, are fewer.

The small Scandia Scatter Blocks are a series of 6-inch-by-12-inch wedges intended for distribution over the rear wall to further reduce standing waves and front-to-back chatter. The actual number and positioning you choose is a matter of taste, as those choices are determined by the amount of live or dead feeling you desire. After some experimentation, I chose a modest smattering to ensure that the room didn't become too dead—something that can lead to overly wet mixes.

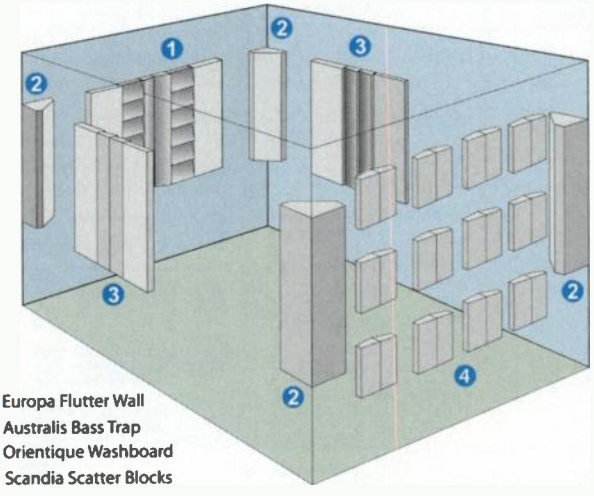
The Australis Bass Traps are 3-foot trapezoids designed to go in any corner seams to reduce the boomy and muddy qualities associated with unchecked low frequencies. You can place the four Australis wedges just about anywhere there is a corner seam, because low frequencies are mostly nondirectional. The Australis's low-end spec goes down to 45 Hz when stacked. Physics dictates that you need much larger traps (and larger rooms) for effective bass management below that range—something that is often impractical in real-world project-studio environments.

In Action

The first challenge I met when I opened the boxes was the overpowering chemical smell inherent in fresh foam. Although I admit to a modest level of chemical sensitivity, the intensity would bother just about anyone. (The manufacturer claims that it has not received any other odor complaints.) I propped them up outside around the perimeter of my house, and it still took days to bring the scent to an acceptable level. In the meantime, exposure to direct sunlight did present a fading problem.

Primacoustic kits come with plenty of Liquid Nails for attaching the foam to walls. Because my studio is in a rented space, I had some trepidation about damaging the walls. At Primacoustic's recommendation, I purchased some lightweight Coroplast (4-foot-by-8-foot corrugated plastic panels) from a local sign company and mounted the foam components on those panels. The trimmed panel assemblies were surprisingly light for their mass, allowing me to affix them to the walls with a handful of drywall screws.

The London 14 is a welcome addition to my studio. The effects in controlling unwanted acoustic artifacts have been significant. No generic kit provides the same results as acoustic treatment that's professionally custom designed for a given room, so it all comes down to how discriminating you and your wallet wish to be. Primacoustic's kit approach strikes a great balance in price and performance.



COURTESY PRIMACOUSTIC

The London 14 Acoustic Treatment from Primacoustic is a modular set of acoustic absorbers intended for use in a project studio.

Overall EM Rating (1 through 5): 3.5
Primacoustic; tel. (604) 942-1001; e-mail info@primacoustic.com; Web www.primacoustic.com

FORWARDINOUTBACK

Didgeridoo Sample CD, vol. 1

By Jeff Obee

The earthy, intriguing tone and primal musicality of the didgeridoo has made it a staple of world-fusion and trance music for more than a decade. Surprisingly, I hadn't encountered any sample CDs dedicated to this native aboriginal instrument until now. Forwardinoutback *Didgeridoo Sample CD,* vol. 1 ($49), comes from the German studios of Ralph "Stick" Hermann and offers 363 ready-to-use WAV files. Most are in mono, though some are processed with stereo effects. (For an additional $6, you can also order an audio-CD version of the disc when you purchase the CD-ROM version.)

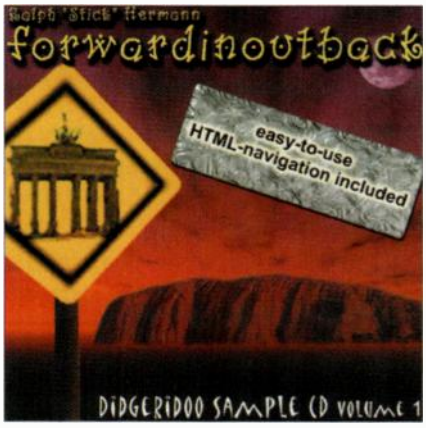
Tubular Tendencies

The samples are organized in folders according to length, type, and meter. There are short and long one-shots, loops, and loops with effects distributed among four keys: A, B, B♭, and A♭. Because the didgeridoo is a single-pitch instrument, however, I would have liked to have more key centers covered. The meters, on the other hand, are thoroughly covered, from the usual 4/4 to the less common 11/8, 3/4, 5/4, and 6/8. Tempos range from 64 to 176 bpm.

Hermann builds his own instruments, crafting them from three materials: wood, bamboo, and metal. The wooden instruments are made from Berlin birch and Canadian spruce, the bamboo instruments are selected from three types of Indonesian bamboo, and the metal instruments are fabricated from the kind of pipe you'd find at a typical construction site. All of the didgeridoos are between 62 and 67 inches in length.

Tubular Tones

The didgeridoo is difficult to record, so Hermann built a wooden box coated with a special foam to suppress unwanted breath noise. The end of the instrument was placed in one end of the box and miked through the other end and the top (with a Shure SM58, a Shure BG4.1, and an Audio-Technica AT3035). The samples were recorded and edited with Steinberg Cubase VST 5.0 and WaveLab.



Forwardinoutback *Didgeridoo Sample CD*, vol. 1, offers hundreds of didgeridoo samples of varying lengths and in different keys performed on seven handcrafted instruments.

Hermann gets a thumbs-up for capturing the didgeridoo's resonant characteristics and primitive vocal qualities. In general, the bamboo samples have a thinner, slightly more distinct tone, whereas the wooden samples have a meatier low end and a wider frequency range. The metal didgeridoo is surprisingly rich in tone but closer to the sound of the bamboo instrument.

Didgeri Does

Hermann's performances are very good and filled with all kinds of classic didgeridoo idioms, including short in-tempo percussive bursts, shifting buzzes, throaty growls, and barking. The lone offering in Bamboo 3's Play folder is an improvisation that lasts over five minutes and moves beautifully through many of the techniques.

Longer tones are provided to serve as drones. They fall mostly in the four-to-eight-second range and aren't edited to loop, though you can splice them together for longer evolving drones. Some of the long tones are deliberately "static" for just that purpose; others include specific accents and rhythmic patterns.

Of particular note are the BambusFX samples, which are juicy, rich, long tones with delay and reverb. The natural overtones ring out beautifully. Judicious use of pitch shifting and other processing is also applied to other FX samples.

The diverse short samples include normal single notes, multiple notes in quick succession, breathy tones, and natural effects such as honks, growls, and scrapes. You can splice them between longer samples or keymap them in a sampler as percussive effects.

The odd-metered phrases don't fare as well as the other loops. The 5/4 loop in the Wood 3 folder eschews the usual "Take 5" accents in a way that makes musical sense, but the 9/8 loop from the same folder needs a stronger delineation of accents to solidify the meter. The other 9/8 loop works well when you set your sequencer to loop it for a single measure.

Didgeri Docs

Didgeridoo Sample CD forgoes printed documentation in favor of a Web-based system. The disc provides a folder of HTML documents for supposedly easy-to-use navigation of the Web site, but about half of the files didn't work on my Macintosh. The site was developed on a PC, so Mac users will probably have better luck just going straight to the company's Web site for information.

A didgeridoo can often provide the perfect underpinning or centerpiece for a composition, and *Didgeridoo Sample CD* gives you a lot to work with at a great price. I would have liked to hear Hermann push the envelope a bit with even more aggressive, high-pitched squeals and yelps, but overall he uses a cool palette of guttural noises and vocal and breath techniques to punctuate the beat. If you're interested in using this unique instrument in your music, check out the MP3 demos on Forwardinoutback's Web site.

Overall EM Rating (1 through 5): 3.5
Forwardinoutback; tel. 49-30-455-84-32; e-mail info@forwardinoutback.com; Web www.forwardinoutback.com

FROSTWAVE

FatController

By Alex Artaud

If you thought the analog sequencer was dead, think again. The fatController ($350), a sturdy 16-note sequencer from Australia's Frostwave, is intuitive to use and comes with plenty of features.

Frostwave's fatController is a 16-note analog sequencer that gives you real-time control over a number of useful parameters.

Knobs and Sliders

The fatController's cream-colored chassis measures 9.5 inches wide by 8.0 inches deep. The 16 sliders are lined up in two parallel rows of eight and offer the right amount of resistance when moved. That makes it easier to zero in on notes, and jostling the unit isn't likely to move a slider and mess up your sequence. Additionally, the sliders are well spaced, so you can comfortably tune individual notes without disturbing the others.

The top and bottom rows of sliders are labeled Channel A and Channel B, respectively. Below each slider in the bottom row is an LED that lights as a note plays or when an associated parameter is being modified. The parameter names—Mode, Direction, Groove, Quantize, Clock, MIDI Channel, Load, and Save—are printed below their respective LEDs.

To the right of the sliders are six function LEDs—Speed, Length, Octave, Slide, Hold, and Rest—and a button that lets you scroll between them. Individual buttons for Start/Pause and Reset (for stopping) are also included. The four rotary pots to the right of the buttons control Tempo, Gate Time, and the slide times for channels A and B. You can also select whether slide data affects channel A or B.

On the back of the unit are two CV and two Gate outputs (one each for channels A and B), MIDI In and Out ports, a DIN Sync input, a power switch, and an input for the 16 VAC wall-wart power supply.

Moving Electrons

The fatController's two modes of operation are Serial and Parallel. In Serial mode, you get the full range of 16 notes from the CV A output: row A plays first, followed by row B. In Serial mode, the CV B output gives you only the notes set up in row B.

In Parallel mode, as you would expect, you get two parallel sequences: CV A outputs the notes set with the Channel A sliders, and CV B outputs the notes set with the Channel B sliders. Additionally, the Channel A faders determine the MIDI note numbers, and the Channel B faders control the MIDI Volume settings for the Channel A sequence, which is sent through the MIDI Out.

The Modifier buttons beneath the sliders, in Global or Function mode, give you control over a variety of aspects of the sequence. For example, in Global mode, the Direction button allows you to play the sequence forward or backward or loop it forward *and* backward. The Groove button, on the other hand, lets you add one of eight swing settings to your sequence.

To change the overall tempo, hit the Function button until the LED next to Speed lights up. Then, hit one of the eight buttons below the sliders to choose one of eight tempo ranges: button 1's range is the slowest, and button 8's is the fastest. Now you can tweak the tempo knob to taste.

In a similar way, you can easily add rests, change the octave of a note, or change the number of steps in the sequence. Such an intuitive interface is what makes this a great sequencer for real-time performance.

Although you cannot save slider and pot positions, other setup data can be held in a buffer while you continue working. The setup data can be saved and reloaded, even after power-down, using the Save and Load buttons. Unfortunately, the slider data is not sent to the MIDI output, and if I could have additional features, they would be a bpm readout and the ability to store sequences.

Sequence in Situ

I used the fatController to control my EMS VCS3 and Minimoog. It's nice that the fatController's gates can be independently switched during power-up to operate as S-trigger gates for synths, such as the Moog, that require this. That allowed me to dedicate CV/Gate A and B to different jobs: Channel A for sequencing and Channel B for signal processing. For example, I dedicated the CV B output to adjust the cutoff time on my low-pass filter.

Frostwave's fatController is a first-rate analog sequencer that is as much at home onstage as it is in the studio. It also includes a lot of great features for the price. If you're looking for tools to transform your sound, check out this small wonder from down under.

Overall EM Rating (1 through 5): 4.5

Frostwave; e-mail info@analogeffects.com; Web www.analogeffects.com

BIG FISH AUDIO

Gas Tank Orchestra

By Zack Price

What are you doing with all the junk that you and society in general have generated? Gregory John Wildes is turning civilization's discarded items into musical instruments. Wildes took abandoned automobile gas tanks, cleaned them out, and turned them into resonator cavities for a variety of homemade percussion, string, and wind instruments.

Wildes then sampled individual notes of some instruments and created loops performed on all instruments. The result is *Gas Tank Orchestra* ($99.95), a combination audio and WAV two-CD set. All of the WAV files have been "acidized" so that they can be readily used in Sonic Foundry's Acid. (However, there is no documentation for loop tempos.)

Precious Metals

The sounds are organized into familiar instrumental categories, but the sounds themselves evoke only a vague sense of their namesakes. For example, at no time would you mistake any of the 42 Drum Spots for realistic bass drums, snare drums, or tom-toms. The disc's 55 Drum Lines offer rhythmic patterns that bear little resemblance to conventional drum

loops. Nonetheless, that is part of the charm of this collection. The Drum, Kalimba, Bass, Harp, and Zither samples all sound clanky, metallic, and off-kilter. This shouldn't be surprising—the resonators and vibrators are metallic. Furthermore, because the instruments are homemade, the tuning isn't always perfect. But again, that's part of their appeal.

The Wind instruments, by virtue of their design, don't have that metallic sound. Even so, their homemade construction imbues them with a rough, detuned quality that their conventional counterparts can't begin to convey.

Gregory John Wildes "repsychles" society's detritus into musical instruments on Big Fish Audio's *Gas Tank Orchestra*.

Wildes includes photos of the instruments he has constructed in JPEG-format files on the WAV CD-ROM. In some cases, you get a straightforward picture of an instrument. Unfortunately, many of the pictures treat the instruments as if they were merely incidental to the overall composition of the photograph. That is to be expected, given that Wildes begins describing the instruments and samples only about halfway into the accompanying text on the inside CD cover jacket. The first half of the text contains a story about an incident that he experienced while bicycle riding and his philosophy of "Fundamental Repsycheling."

Even so, it is possible to piece together the look and sound of an instrument by its name and photograph. For instance, Trombone-Hose, Trumpet-Hose, and Sax-Hose all have a mouthpiece connected to a rubber hose that is connected to a gas tank. It looks and sounds as if the pitch is controlled by the embouchure. Flute looks as though it was created from PVC pipe. The Bamboo-Clarinet has a clarinet reed mouthpiece attached to the end of a bamboo flute, which is attached to a rubber hose that is connected to a gas-tank resonator at the other end. This is physical modeling gone haywire, folks!

Repsychedelic

Wildes suggests that you begin by listening to the entire CD to find the inspiration for using the sounds. I tried that approach, but quickly tired of simply listening to the sounds. I opened up Sonic Foundry's Acid to audition sounds and began to piece together compositions when the inspiration struck me. Granted, that's a more random approach, but that's how I work in Acid anyway. However, when I imported the samples into Tascam GigaStudio and Sonic Solutions Orion Pro, I focused on the Notes and Spots.

Gas Tank Orchestra is loaded with unusual sounds that you might not use every day. Nevertheless, you'd be surprised at the number of times you will want to use them. Hip-hop artists should appreciate the unusual sounds, wide variety of drum loops and drum hits, and some of the more unusual bass lines. Other artists may find that the different instrumental loops and notes add just the right spice to more conventional arrangements. It's also possible to create entire pieces of music using nothing but the loops and hits contained in this collection. Wildes includes several excellent examples in Acid song format of arrangements that he made using nothing but these sounds. I was also able to quickly create an arrangement in Acid using just these sounds. In short, this collection is a lot of fun. That's part of its charm.

Overall EM Rating (1 through 5): 4.5

Big Fish Audio; tel. (800) 717-FISH or (818) 768-6115; e-mail info@bigfishaudio.com; Web www.bigfishaudio.com.

We welcome your feedback. E-mail us at emeditorial@primediabusiness.com.

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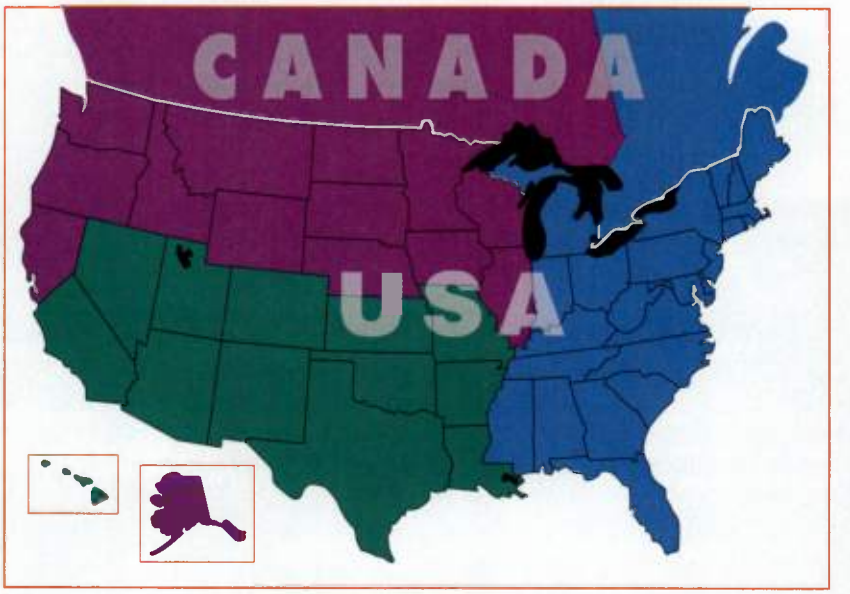

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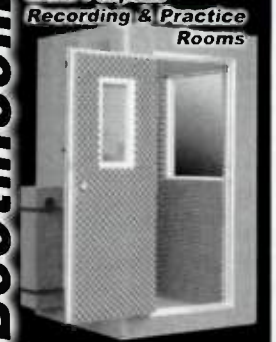

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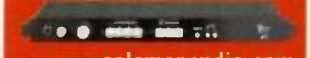

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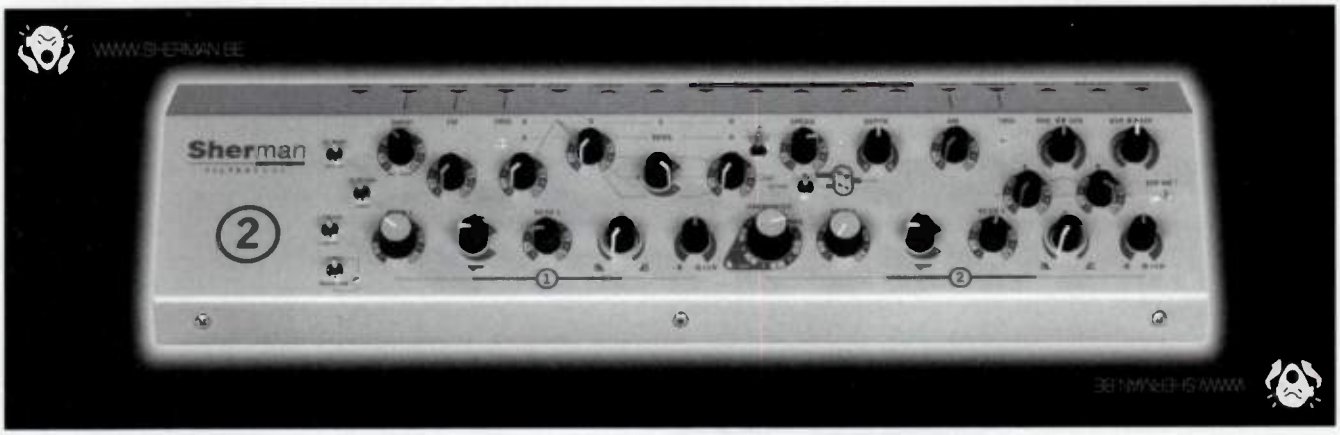

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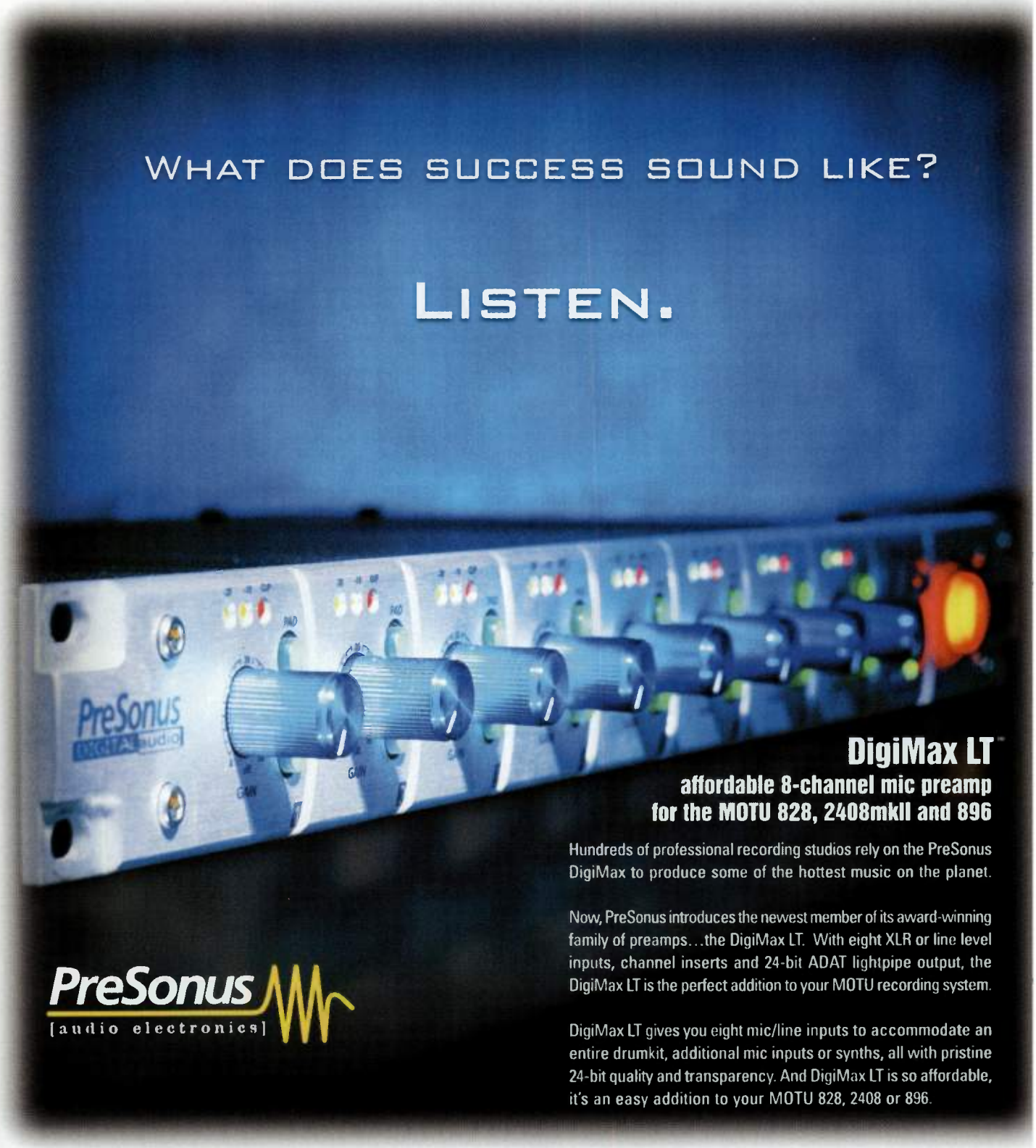

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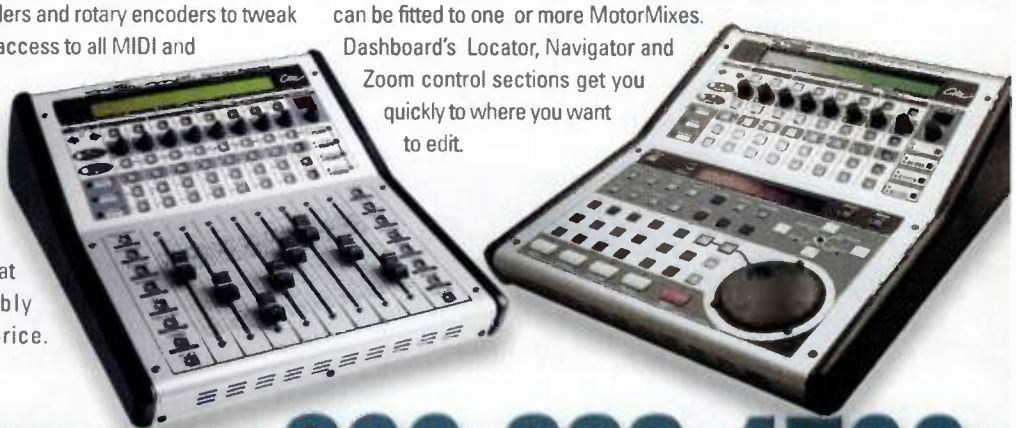

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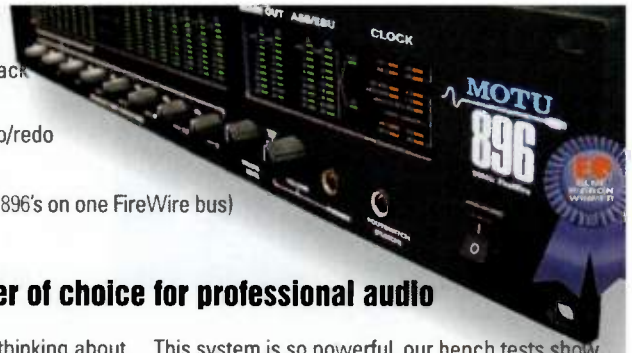

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By Larry the O

Oh Say, Have You Scene?

Whether through luck, adventurousness, or a good network of friends, many of us are fortunate enough to find ourselves, at some point in our lives, in contact with a "cool scene." The cool scenes that interest me show abundant creativity, knowledge (maybe even wisdom), cooperation, and inspiration. Such scenes focus a collective energy on an idea that may range from the specific and explicit to the vague and unknowable. In a cool scene, one is awash in a palpable, life-changing electricity that opens doors of perception and stokes the fires of imagination. It's a most amazing experience that stays with you your entire life.

A cool scene may start with a single person, but it takes a number of people to reach critical mass. What is less evident is the great importance of the number. When a couple of dozen people are involved, they tell and involve others, and things begin to coalesce. This is often the real peak of the scene—while it is still small. But word spreads, and the scene grows into a movement. Once it expands beyond a few hundred people, things change. As the scale increases, the idea's purity becomes diluted as a simple result of more points of view in the mix.

A slow decline starts from this juncture, though the scene may remain good and worthwhile for quite some time. Eventually, however, if it continues to expand, it will start to stagger under its own weight and, at some point, will simply collapse, dissipate, or shrink radically.

The life of a scene is as brief and finite as our own lives, and its decline or death is just as inevitable. If that death can't be avoided, neither should it be feared, though the passing may be mourned. As long as there are humans with hearts and minds, cool scenes will arise.

Getting the most out of a burgeoning scene comes from recognizing as early as possible that there's magic happening and appreciating and enjoying it as fully as you can. That realization came to mind recently when I dipped my toe into a small underground community of musicians who call themselves *loopers*. Loopers make music built around the idea of feeding source material (typically instruments they play) into long, repeating delay lines and then adding to, modifying, and/or playing on top of the looping sound. I've just returned from Loopstock, a one-day concert and minifestival where barely three dozen people converged on a small dance studio warehouse in San Luis Obispo, California, from points as far-flung as Oregon, Boston, and Brazil. They came to perform, to listen, and to discuss a variety of takes on looping—from a three-piece synth band to DSP Demon Richard Zvonar mangling snippets of recorded music to a solo tuba player—all working with long delay loops. Probably over half the attendees were performing.

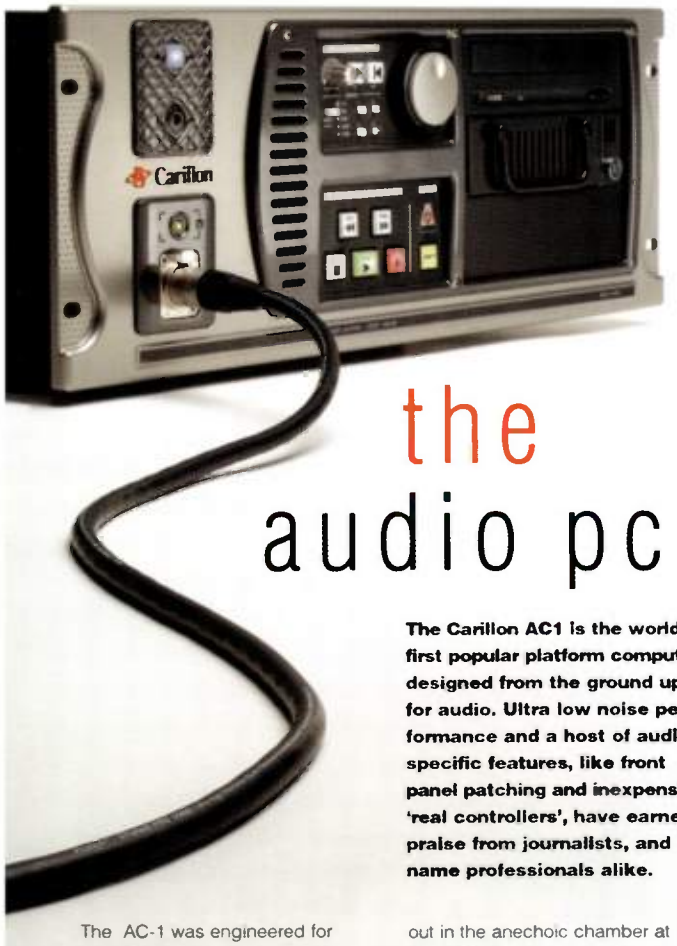
It was evident to all that something special was happening at this seminal gathering. The event was free and the performers unpaid. In other words, the attendees were there purely out of interest. This created a wholly noncompetitive atmosphere in which people were delighted to show and tell all their looping ideas and techniques. The understanding audience showed tolerance for the difficulties of a technology-dependent medium. Most of all, everyone was appreciated for what they contributed: not just the performers, who surely received hearty applause, but even a few attendees who were hailed as heroes for having created some of the loopers' favorite toys.

Each attendee left invigorated and inspired. Will Loopstock grow or happen at all next year? I certainly hope so. The looping scene should have a long time to grow before it becomes co-opted, corrupted, or otherwise brought low. But no matter whether or when that happens, that day in San Luis Obispo will remain a Golden Moment in the lives of those who were lucky enough to catch the quick left at the Mitsubishi dealership and discover the little warehouse tucked back from the street.

If you find yourself in the middle of a community brimming with life magic, don't be afraid, don't try to hold onto it too hard, and don't try to figure out what it is. Just savor the moment and cherish the memory.

We welcome your feedback. E-mail us at emeditorial@primediabusiness.com.

STEVE CURL



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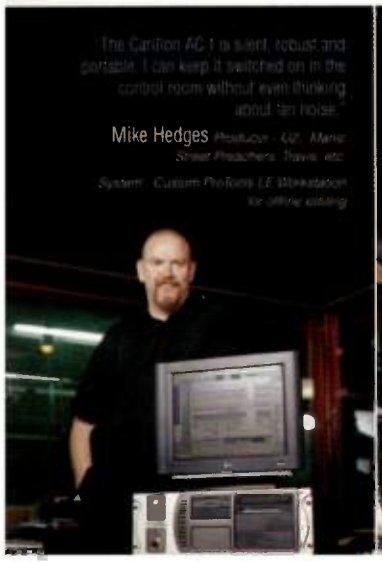

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