Mastering the Korg Karma • Zoom MRS-1044, RME Project Hammerfall DSP, and 11 more reviews

Electronic Musician®

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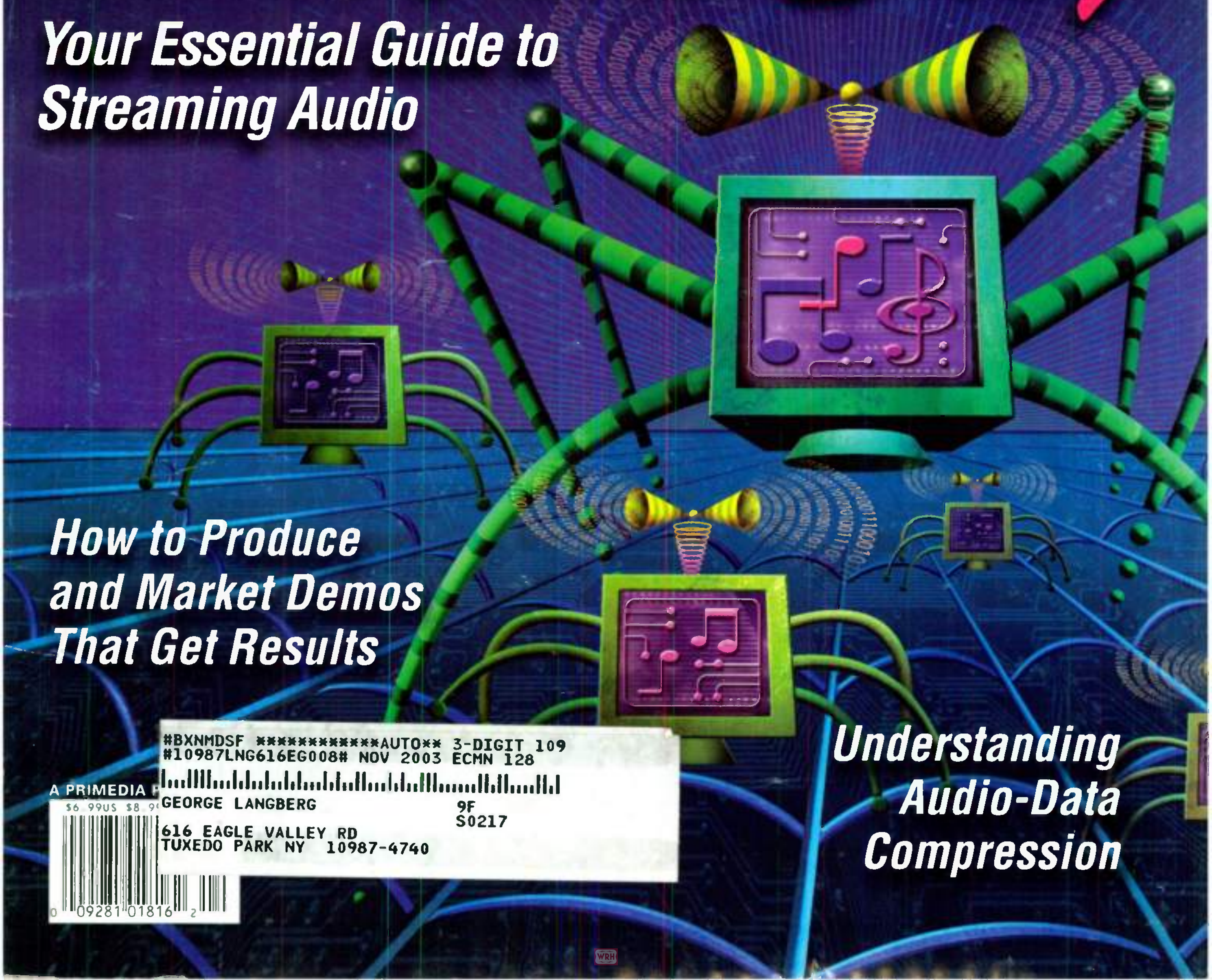
April 2002

WEBCASTING *Made Easy*

Your Essential Guide to Streaming Audio



How to Produce and Market Demos That Get Results

Understanding Audio-Data Compression

A PRIMEDIA

Introducing the HR624 active monitor and
Two new sizes. Same unflinching accuracy.

Dad and the kids. All THX pm3-approved.

Our HR824 Active monitor waited until it was well-respected and famous before deciding to sire some offspring.

First it won a worldwide reputation as the most accurate near field monitor ever made. It gained acceptance in major recording studios and mastering facilities. It earned glowing reviews (*Mix* magazine called it "*...the most accurate near field monitor we've tested.*")

Now it's part of a complete THX pm3-approved monitoring system.

The HR624. A chip off the world's most accurate block.

Our new 6.7-inch 2-way active near field monitor is essentially a slightly smaller HR824 with the same technological innovations:

- Cast zinc logarithmic wave guide that maximizes dispersion and helps merge high and low frequency output at the crossover point.
- Extended low frequency transducer with cast magnesium frame and mineral-damped polypropylene cone.
- Wave-guide-loaded 1-inch high frequency transducer with viscous, edge-damped alloy dome and ferrofluid voice coil cooling (the same design as in the HR824).
- Additional rear-firing passive transducer for tighter bass response.
- Dual FR Series™ high-current power amplifiers (100 watts LF, 40 watts HF).

The new HR624 has the same wide sweet spot as the HR824. Frequency response of both models is so flat that each one comes with an individual, signed test certificate.

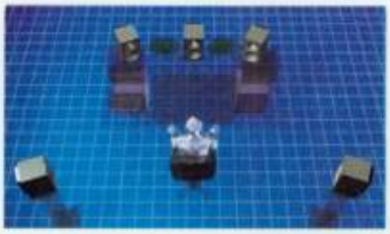
They play well with others.

HR624s make superb primary front monitors for smaller studios (and budgets) – with or without an HRS120 subwoofer. And since they're voiced identically to the HR824, they're perfect for rear channel 5.1/7.1 surround sound.

One possible combination. HR824 front speakers with HR624 rear satellites and HRS120 subwoofer.

HRS120 12-inch active subwoofer system.

Rumpus-stomporama configuration: Dual, slaved HRS120 subs and five HR824s.



The HRS120 400-watt active subwoofer. This is not a toy.

The new HRS120 12-inch 400-watt active subwoofer is designed to compliment the HR824, HR624 – and Brand X passive or active monitors that need more low frequency cajonés.

One look at the HRS120 back panel and you'll understand the difference between a true recording studio subwoofer and a mere home theater sub dressed up in pro audio cosmetics.

You can switch the HRS120 in and out *with or without* its variable crossover – allowing comparison between full range front/left/rear and high pass monitors plus sub (a.k.a. "bass management).

You can slave multiple HRS120s and run them all from one foot switch.

Rear panel control options include...

- 110Hz elliptical filter for Dolby® AC-3
- Crossover Defeat switch for THX® operation
- 0dB/+10dB Output Level switch for THX® Dot1 & AC-3
- Adjustable 60-110Hz crossover
- Polarity switch and input level control.

The HRS120's front-firing RCF Precision™ 12-inch transducer has a cast aluminum frame and four-inch voice coil, servo-coupled to a knarly FR Series high-current power amplifier with generous rear heat exchanger. Its tight, accurate output is also reinforced by an extra front-firing 12-inch passive transducer.

Hearing is believing.

If you value unflinching monitoring accuracy, you owe it to the quality of your creative product to check out our family of active monitoring systems.

Hear the new HR624, HRS120 and industry-standard HR824 at an authorized Mackie dealer today.

INSI

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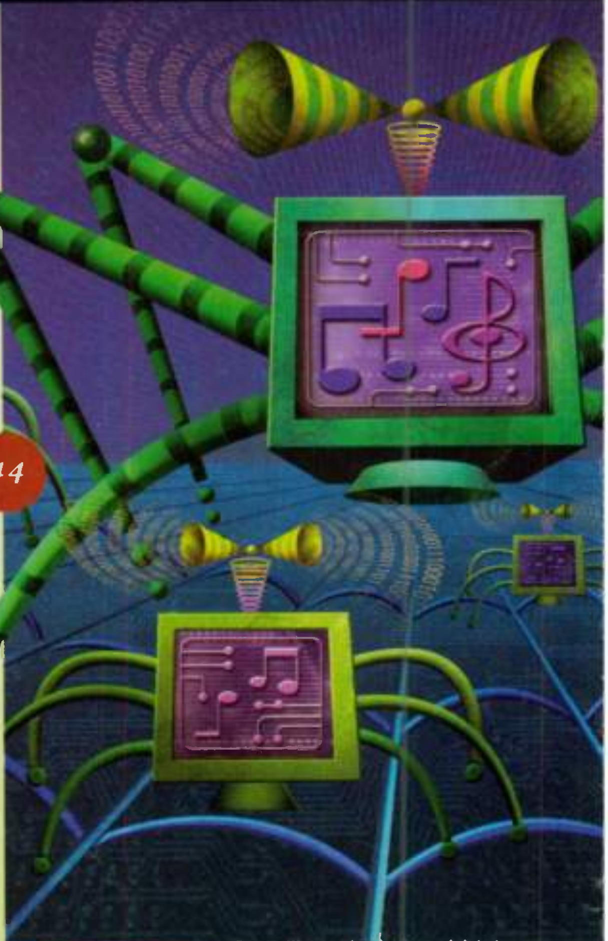

Cover illustration by www.lolabek.com

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DEPARTMENTS

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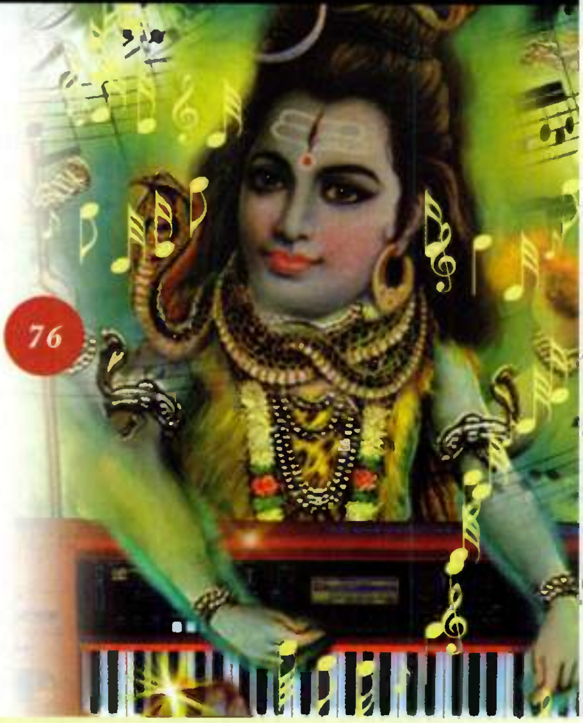

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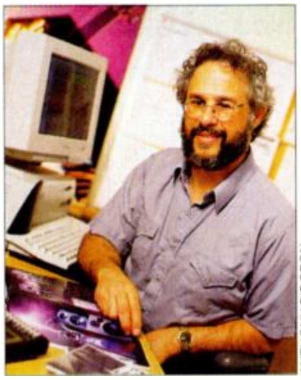
A Time for Us


ANTHONY PIDGEON

Until recently, music educators as a group have usually lagged behind when it comes to applying technology in the classroom, especially in the public schools. Public-school music teachers have tended to be conservative and uninformed about technology, and perhaps more important, school budgets generally are low even in good economic times, so getting school teachers and music-education departments to embrace technology and apply it in the classroom has been slow work.

Fortunately, the situation has improved significantly in the last three or four years. Home computing has entered the mainstream in our society, and many students and teachers use computers in their nonmusical pursuits, reducing the level of technophobia. In addition, computers have become more available to schools. One factor was the Clinton administration's attempts to wire the schools in to the Internet. Another has been corporate programs.

But there is a third, less known reason that music teachers in particular are beginning to embrace high-tech tools: support from private organizations and foundations. The primary organization that focuses on fostering the use of technology in both public and private music education is the Technology Institute for Music Educators (TI:ME; www.ti-me.org), which is mostly funded by a grant from NAMM and is cosponsored by the International Association of Electronic Keyboard Manufacturers (IAEKM; www.iaekm.org).

TI:ME's goals are to define the skills needed to understand and use technology for music education; to develop standards for in-service teacher training specific to music technology; and to develop course materials to teach sequencing, notation, computer-assisted instruction, and so on. The institute's Music Technology Certification Program tests for and certifies two skill levels, based on TI:ME's course materials and classes. (This year, the Institute will offer classes at 24 colleges in 14 states. For more information, see the Summer Study area on TI:ME's Web site.) Level 1 certification requires that a teacher demonstrate basic skills in music-notation and sequencing, computer-assisted instruction, multimedia, the Internet, and the use of electronic instruments in the classroom. Level 2 certification requires advanced skills in at least two of these areas as well as knowledge of how to integrate music technology effectively in the classroom.

In addition to its Web site, TI:ME has a newsletter and participates in conferences, notably the biannual Music Educators National Conference (MENC), which is a major trade show for public-school music teachers. At this year's national conference, held at the Nashville Convention Center from April 10 through 13, TI:ME will provide extensive technology sessions.

Teaching music teachers how to use technology in their work—and supplying them with the tools to do so—is in everyone's best interests. I strongly encourage everyone to support TI:ME and similar regional and local efforts, participate in programs for donating computers to schools, and otherwise further this noble effort.

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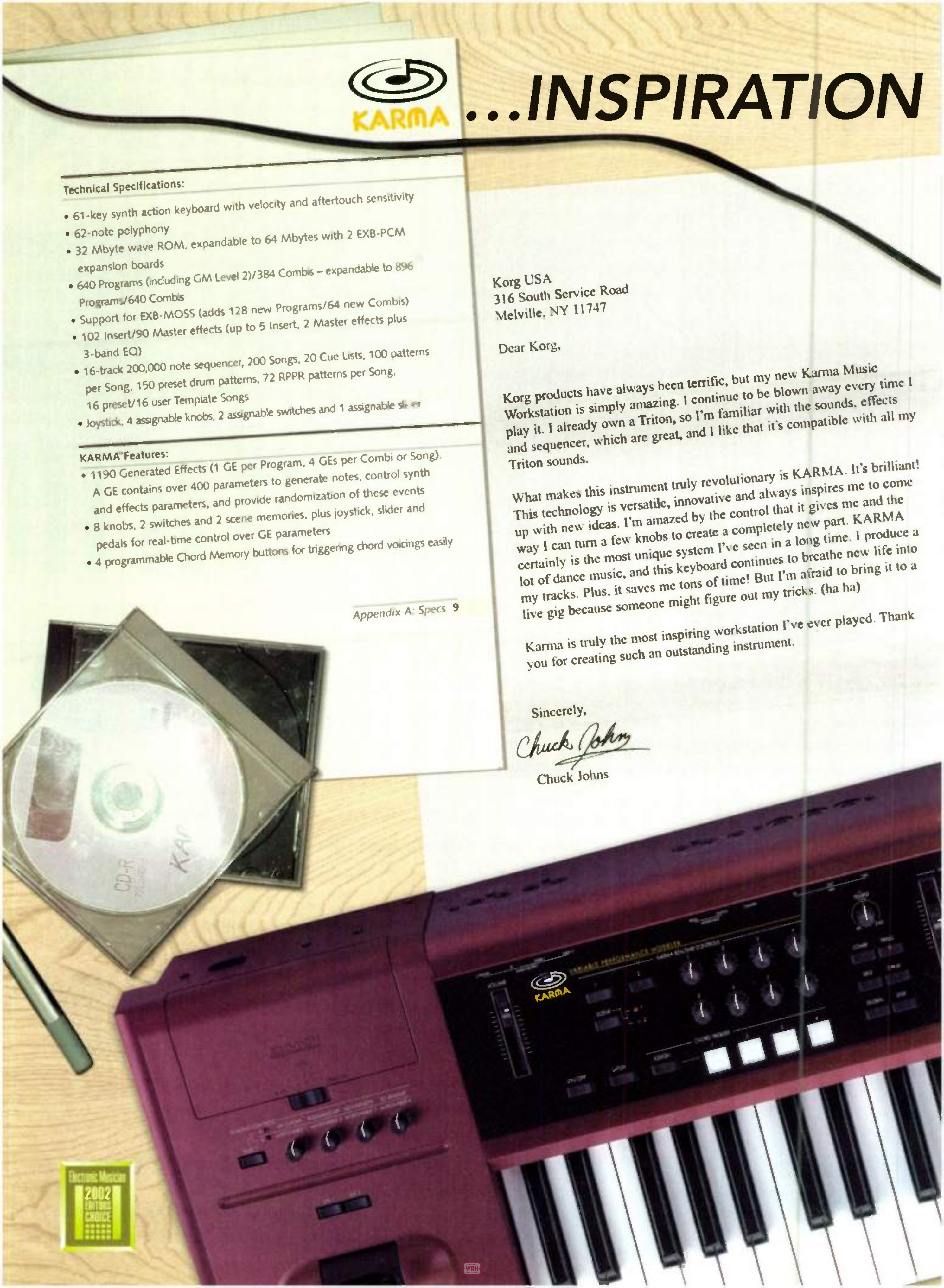
KARMA
...INSPIRATION
Technical Specifications:
• 61-key synth action keyboard with velocity and aftertouch sensitivity
• 62-note polyphony
• 32 Mbyte wave ROM, expandable to 64 Mbytes with 2 EXB-PCM expansion boards
• 640 Programs (including GM Level 2)/384 Combis – expandable to 896 Programs/640 Combis
• Support for EXB-MOSS (adds 128 new Programs/64 new Combis)
• 102 Insert/90 Master effects (up to 5 Insert, 2 Master effects plus 3-band EQ)
• 16-track 200,000 note sequencer, 200 Songs, 20 Cue Lists, 100 patterns per Song, 150 preset drum patterns, 72 RPPR patterns per Song, 16 preset/16 user Template Songs
• Joystick, 4 assignable knobs, 2 assignable switches and 1 assignable sl
KARMA Features:
• 1190 Generated Effects (1 GE per Program, 4 GEs per Combi or Song). A GE contains over 400 parameters to generate notes, control synth and effects parameters, and provide randomization of these events
• 8 knobs, 2 switches and 2 scene memories, plus joystick, slider and pedals for real-time control over GE parameters
• 4 programmable Chord Memory buttons for triggering chord voicings easily
Appendix A: Specs 9
Korg USA
316 South Service Road
Melville, NY 11747
Dear Korg,
Korg products have always been terrific, but my new Karma Music Workstation is simply amazing. I continue to be blown away every time I play it. I already own a Triton, so I'm familiar with the sounds, effects and sequencer, which are great, and I like that it's compatible with all my Triton sounds.
What makes this instrument truly revolutionary is KARMA. It's brilliant! This technology is versatile, innovative and always inspires me to come up with new ideas. I'm amazed by the control that it gives me and the way I can turn a few knobs to create a completely new part. KARMA certainly is the most unique system I've seen in a long time. I produce a lot of dance music, and this keyboard continues to breathe new life into my tracks. Plus, it saves me tons of time! But I'm afraid to bring it to a live gig because someone might figure out my tricks. (ha ha)
Karma is truly the most inspiring workstation I've ever played. Thank you for creating such an outstanding instrument.
Sincerely,
Chuck Johns
Chuck Johns
KARMA
Electronic Musician
2002
EDITORS'
CHOICE


STORY #23
Chuck Johns Queens, NY
6/21/01 DANCE TRACKS
KARMA SESSION
IT SEEMS LIKE
WE SAID GOODBYE AT
AND NOW I DREAMT WANT
AND BE RIGHT THERE WIT
I HOLD YOUR PICTURE
AND TRACE THE CURV
KORG
KARMA
MUSIC WORKSTATION
GOOD KARMA. WICKED SOUND.
KORG
KARMA
KAY ALGORITHMIC REALTIME MUSIC ARCHITECTURE
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KARMA™ (Kay Algorithmic Realtime Music Architecture) is a trademark and patented technology licensed from Stephen Kay, Karma Lab LLC, www.karma-lab.com

MAE LAROBIS

CLEVELAND ROCKS!

The article about Joe Meek ("Production Values: Meek First") in the February 2002 issue was one of the finest pieces I've read in a music tech magazine. Thanks so much for printing it. I hope it gives more people insight into the extraordinary artistry of this strange figure from modern recording's infancy. Bravo to author Barry Cleveland for making a potentially complex subject so simple.

Richard Einhorn
via e-mail

The article about Joe Meek was spellbinding. Where can I find the three songs mentioned in the article ("Johnny, Remember Me," "Telstar," and "Have I the Right?") on CD or cassette?

Frank Lewin
via e-mail

Author Barry Cleveland replies: Frank—It's Hard to Believe It: The Amazing World of Joe Meek *(BMG/Razor and Tie Entertainment, 1995) includes all three songs.*

PRAISE THE KIWI!

I tried to calculate the number of adjectives in Brian Knave's review of Blue Microphones' Kiwi (February 2002), but my coprocessor melted. I'm pleased he liked the microphone, and I like using it myself. However, the review, some five pages long, was so glossy and sugary sweet that not only did I reach for a toothbrush but I also kept reading, waiting for his conclusion to be that "the Kiwi has made me renounce God and start my own religion of worshipping this microphone."

Readers want an objective approach that weighs the pros and cons of shelling out such a chunk of change—and whether the mic will work for them—rather than knowing about Knave's torrid love affair with the Kiwi.

Mike Ingram
via e-mail

Mike—Purple prose or not, the Kiwi is an incredible microphone worthy of the highest praise. Although I would hesitate to establish a religion around the mic, I stand by my review.—Brian Knave

SHRILL OR ANGST?

A colleague just showed me the article about recording acoustic harmonica ("Recording Musician: Taming of the Shrill," January 2002). It was interesting and informative. I have recorded only one CD to date, but I wish I had seen this article a while ago.

First, author Brian Knave makes it clear that the harmonica produces a shrill, tinlike sound. I suppose from a recording engineer's point of view, that is a good lead-in—the harmonica is shrill, so it's hard to record.

Knave put too much stress on the harmonica's tone as a negative thing. If he's played for 30 years, then I hope he still experiences the same emotions when hearing a piercing first-position, high-end solo piece (for example, "Trouble in Mind," by Walter Horton with Carey Bell). That "piercing" is where the soul is—the sadness, the angst.

Players get enough bad press already, so it would have been nice if he'd said, "It's shrill and hard to record, but we just love that sound" approach as opposed to "It's shrill and difficult, but let's try to deal with it." Knave mentions that the recording engineer must work with the player and determine his or her ability. Maybe pros like him have enough skill, but rarely do I encounter a sound or recording engineer who has the first clue about how a harmonica is supposed to sound.

I think there's a national lack of understanding about the instrument as a whole. This article has done a lot, and I hope many recording engineers will read it. I plan to show it to anyone who ever records me again. I hope EM continues using its influence and presents the harmonica as a good thing to more people.

Dan Gage
via e-mail

Dan—Perhaps I overstated my case a bit, and I wish I had worked in the "but we just love that sound" angle—I do love the sound, or I wouldn't have spent all these years listening to, studying, playing, and teaching the instrument. However, I don't see how the harmonica's potentially shrill sound or how long I've been playing relate to whether I "still experience the same emotions when hearing a piercing first-position, high-end solo piece." After all, I can imagine a soulfully played first-position, upper-octave solo recorded so badly that it hurts my ears to listen to it.

As for your comment that "players get enough bad press already," I'm not sure how that relates to what I wrote, either. I sure hope nothing I said about Toots Thielemans, Norton Buffalo, Mike Stevens, or Lee Oskar came off as bad press. (By the way, all four read and approved the piece before it was printed. The only disagreement came from Lee Oskar, who feels that even bullet-type mics can be too bright sounding for recording harmonica. Hopefully it was clear, though, that the remarks I made about bullet mics were mine alone.)

In regard to your comment, "rarely do I encounter a sound or recording engineer who has the first clue about how a harmonica is supposed to sound," I know just what you mean. Indeed, that's one of the main reasons I wanted to write the piece. Still, regardless of whether the engineer has a clue, he or she should be able to distinguish between a warm, pleasant tone and a shrill, painful one.

I'd like to clarify that I did not mean to present the harmonica as a good or bad thing. If put to a choice, though, I'd pick the former.—Brian Knave

ERROR LOG

March 2002, "Burning Ambitions," p. 66: The photo credit for **Figs. A** and **B** should be "Courtesy Ritek Corp."

March 2002, "More Than Meets the Ear," p. 80: The "My Name Is True" sidebar implies that burning a Joliet-format ISO data CD on a Mac will enable other Macs to read 31-character file names on the disc. In fact, that is true only if the viewer's Mac has the Joliet Volume Access extension installed. Stock Macs will truncate the file names to 8.3-character format, though PCs running Windows 95 and up will still see the long names. A more precise version of the sidebar is available at www.emusician.com.

March 2002, "Ad Index," p. 170: The The January 2002 Ad Index was mistakenly printed instead of our March 2002 Ad Index. To see the March 2002 Ad Index, please go to our Web site at www.emusician.com.

February 2002, "Chain, Chain, Chain," p. 50: The first sentence should read, "Equalizers and compressors work best when you process the entire signal, so they are commonly used with inserts." Also, on p. 54, the first full sentence should read, "When you use an aux send, you route only a portion of the channel's dry signal to the outboard processor."

February 2002, Digidesign Pro Tools TDM 5.1, p. 160: The minimum system requirements were listed incorrectly. Mac users of Pro Tools TDM 5.1 need a Power Mac 9500, 128 MB of RAM, OS 9.04, and an open PCI slot. PC users need a Pentium III/450 MHz with 192 MB of RAM and Windows 2000/SP 1.

January 2002, "Library Science," p. 40: The word *applications* was cut off in the last sentence of the "Climbing the Decision Tree" section and inadvertently replaced with the number 40.

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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.

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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!

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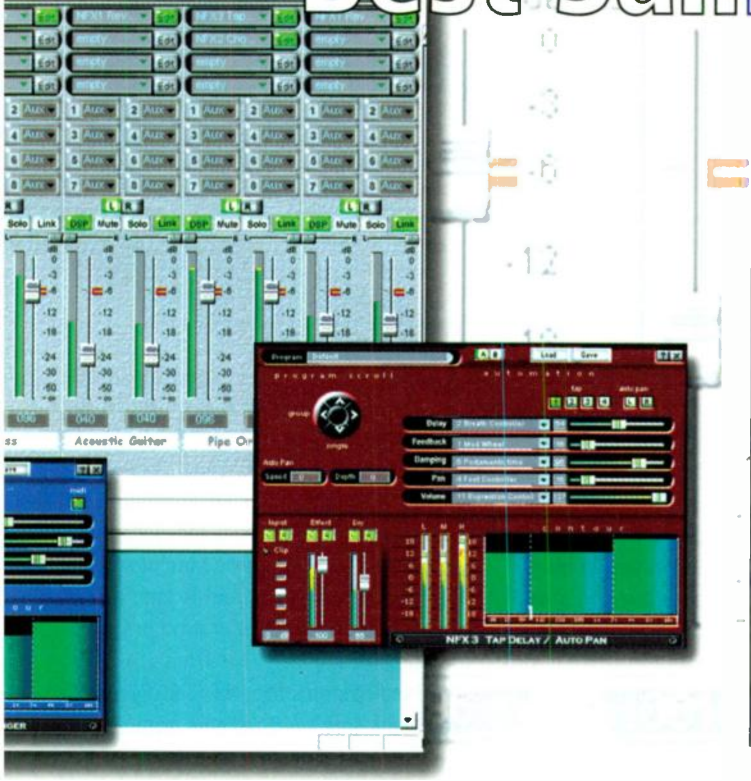


We don't like to brag, but there's no question: **TASCAM GigaStudio** offers the very best sample playback of any sampler, hardware or software, ever made. The reason is simple: it's the only sampler that employs a patented technology allowing samples to stream from your PC's hard drive instead of being limited to RAM storage. The result is amazing: you can access up to 160 voices of HUGE samples (over four gigabytes in size), with detail, realism and sonic quality blows away any other sampler. Period.

As a performance tool, GigaStudio rules. Its incredibly low latency when accessed with any GSIF computer interface allows for fast, tight musical performances that are indistinguishable from playing a "real" instrument. Plus, GigaStudio's QuickSound™ technology enables instant location and previewing of samples and instruments in real-time. Its zero-latency NFX™ effects provide professional-quality signal processing for your samples. And if you need great sound libraries, the world's finest have been created expressly for GigaStudio. Also, your Akai™ samples will automatically be read, and you can easily convert other sample files into the Giga format.

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.

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/Matt Ragan Max Strength Acoustic Guitar

The beautiful, clear tone of a massively multi-sampled Martin 000-16. More than 1,200 discreet, unlooped 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.

Bigga Giggas/ Post Harpsichords I

Two antique harpsichords captured in every detail using world-class microphones and mastered originally in 24-bit audio. This library is perfect for keyboard purists seeking to reproduce the great early keyboard compositions on the instruments for which they were written.

Q Up Arts/Symphonic Fields Forever

Beautifully evocative solo and small section orchestral instruments. Perfect for both Pop and Classical orchestration as well as acoustic textures. Features superbly recorded multisamples of celli, violins, choir, flute, bassoon, tuba, double basses, clarinet and more.

Bigga Giggas/ Harmonica Essentials

Turn your Giga system into a professional blues harmonica player! Acoustic and electric harp in 8 keys and 4 tempos, with over 1100 licks,with effect banks in each of the keys to help fill in between licks.

Q Up Arts/Psychic Horns by Jason Miles

The killer collection of brass sections of stereo trumpet, tenor sax and trombone. Includes long and short sustains, loops, riffs, swells, falls, and stabs. For Pop, R&B, Funk, Jazz...if a brass section can play it, you can too!

Sonic Implants/ Drum Series 1

From the real to the surreal, these drums sound amazing. All drums and cymbals are recorded in stereo, with no loops, and with heavily multi-velocity. Even the snares are sampled at multiple places on the drum. Includes 250 drumkits and instruments.

Bigga Giggas/Sune's L100 Hammond

Every note of this great-sounding organ's 9 drawbar settings, recorded in extremely long looped samples, with fully controllable virtual drawbars in GigaStudio.

Q Up Arts/ Heavy Guitars

A grungy, harsh, ruthless collection of guitar samples... leads, mutes, scrapes, scratches, power chords, slides, feedback, harmonics and more.

Bonus 60Hz hum sample included on CD-ROMs. Rock on!

Sonic Implants/ Amps & Pickups

The guitar and bass collection you've been waiting for! Collection includes acoustic guitars, Les Paul power rock, vintage Guild, Paul Reed Smith Electric, 12-string Rickenbacker, Spector Slap Bass, Hofner Beatle Bass, Fender Jazz Bass and more.

CONEXANT LEADING EDGE
GIGA
QS QuickSound

TASCAM
GIGASTUDIO
powered by GigaSampler Technology

WHAT'S NEW

By Marty Cutler

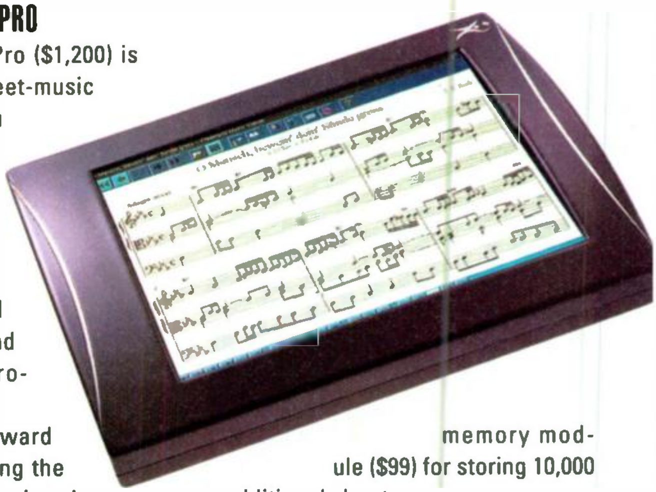
▲ SENNHEISER HD280 PRO

The HD280 Pro ($199) from Sennheiser is a closed, dynamic transducer headphone set. The headphones are collapsible and offer rotating ear cups. The over-the-ear design promises as much as 32 dB of ambient noise reduction.

Sennheiser rates the frequency response of the HD280 Pro at 8 Hz to 25 kHz and the sound-pressure level (SPL) at 113 dB. Total harmonic distortion (THD) is rated at <0.1 percent.

Many of the HD280 Pro parts are replaceable, including the ear cushions, the headband cushion, the cable, and the driver elements. Additionally, the cable features a 3.5 mm stereo minijack with a ¼-inch adapter. Sennheiser Electronic Corp.; tel. (860) 434-9190; e-mail miclit@sennheiserusa.com; Web www.sennheiserusa.com.

▶ FREEHAND MUSICPAD PRO



FreeHand's MusicPad Pro ($1,200) is more than a digital sheet-music display; the unit lets you download sheet music from your computer and edit scores. The standalone device features a flat-panel SVGA screen measuring 12.1 inches, and it can rest on a music stand or be mounted on a microphone stand.

You can turn pages—forward and backward—by touching the screen or by using the optional footpedal ($99). The MusicPad Pro comes with a stylus that allows you to annotate the sheet music onscreen. Annotations can be made in several colors.

You can bookmark score parts in order to jump back or ahead to any section. The MusicPad Pro comes with Flash Memory for as many as 300 pages, but you can purchase FreeHand's proprietary memory module ($99) for storing 10,000 additional sheets.

The MusicPad Pro has a USB connector and a 10/100Base-T Ethernet jack for downloading sheet music from your computer. The device supports Macintosh and Windows computers, and it can import music from any music-notation program. You can also import music from scanners. FreeHand Systems Inc.; tel. (650) 941-0742; Web www.freehandsystems.com.

▶ DIGIDESIGN MBOX

Developed jointly with Focusrite, Digidesign's Mbox ($495) is a 2-channel USB audio interface for Pro Tools LE that delivers 24-bit, 48 kHz recording capabilities. Sessions that you create with Mbox can be sent to other Digidesign platforms, including Pro Tools HD and Digi 001 systems.

The Mbox draws power from its USB port and includes a pair of Focusrite mic preamps with 48V phantom power. Each channel has an input-gain control and a Source button that lets you select mic, line, or instrument level. A second button centers and sums the inputs but has no effect on the playback monitoring. The front panel also includes a playback and input balance control, a headphone-level control, as well as an ¼-inch stereo headphone jack.

The Mbox's rear panel has two Neutrik combo connectors for the inputs, a pair of ¼-inch TRS insert jacks, two unbalanced ¼-inch output jacks, a ¼-inch stereo output for headphone monitoring, a USB jack, and coaxial S/PDIF jacks for digital I/O.

Mbox runs on any Power Mac with a built-in USB port that can run Pro Tools LE 5.2. You need at least 128 MB of RAM and OS 9.1. Digidesign; tel. (800) 333-2137 or (650) 731-6300; e-mail prodinfo@digidesign.com; Web www.digidesign.com.

THE NEW DIGITAL WORKSTATION FOR PEOPLE WITH MORE SENSE THAN DOLLARS.

If this all-in-one digital workstation looks familiar, that's because the new Yamaha **AW2816** strongly resembles its TEC Award-nominated big brother, the AW4416. In terms of features and performance they're remarkably similar. So, the AW2816's price – just $1,999* complete – makes excellent financial sense. Once again, Yamaha gives you more for less.

- 16+2 track, 24-bit recording with no data compression (44.1 or 48Khz)
- 28 Input channels (including digital stereo input and dual stereo effects returns)
- Dynamic automation of all mix settings, including fader movements, parameter changes and scenes
- 2 Assignable 32-bit effects processors
- Premium Yamaha CD recorder for data and audio
- I/O expansion slot for digital interface options or the new WAVES® Y56K processor card
- Large, back-lit 320x640 display screen
- MIDI control of computer based recording systems
- 4-Band parametric EQ and dynamics on all channels, all the time**
- 18 Busses, plus comprehensive digital patch bay, with easy routing

YAMAHA

CREATING 'KANDO' TOGETHER

www.AW2816.com

▼ ROLAND V-BASS

Following in the footsteps of Roland's V-guitar processors, the V-Bass ($1,345) offers access to a wide variety of bass sounds using any bass with a divided pickup and a 13-pin output. (Guitarists can also access bass sounds, albeit with a bit of tweaking.) As with the V-guitar series, MIDI is not involved in tracking your bass. However, the V-Bass is equipped with MIDI In and Out jacks so you can store and retrieve SysEx data. The unit also supports Program Change and Control Change messages as well as MIDI Clock for tempo-based effects.

Instruments modeled by the V-Bass include classic electric and acoustic basses as well as synth bass and fretless types. You can program bass sounds by combining modeled body types and pickups. Roland's proprietary COSM amp-modeling algorithms offer additional customization capabilities. An onboard expression pedal provides you with real-time control over parameters. The V-Bass has 160 presets and 100 user slots for customized programs.

A 13-pin input jack connects your bass or guitar to the V-Bass. Roland offers the GK-2B divided pickup ($275), which is compatible with four-, five-, and six-string basses. The divided output provides separate processing for each string, including polyphonic pitch shifting (for alternate tunings) and pan.

The V-Bass offers a pair of unbalanced ¼-inch outputs as well as a pair of balanced XLR outputs. An additional unbalanced ¼-inch output and an XLR jack gives you access to the unprocessed signal. An unbalanced ¼-inch input lets you plug your bass directly in to the unit so you can use the unit's COSM effects. An input for an additional expression pedal is also included. Roland Corp. U.S.; tel. (323) 890-3700; Web www.rolandus.com.

▲ LITTLE LABS IBP

Little Labs' IBP ($450) is an analog phase-alignment tool that eliminates the characteristic hollow, out-of-phase sound that can occur when audio signals are combined. The small unit also serves as an active-transformer direct box.

The front panel includes a knob that adjusts the signal from 0 to 180 degrees, a button that toggles between 90 degrees and 180 degrees, and a button that selects the phase center frequency. Additional buttons include a selector for line- or instrument-level input, a Bypass switch, a Phase-Invert button, and an Earth-Lift switch.

Inputs and outputs are balanced line-level XLR jacks. An unbalanced ¼-inch instrument-level input is included along with a buffered ¼-inch output jack for driving long guitar cables. Little Labs; tel. (323) 851-6860; e-mail littlelabs@littlelabs.com; Web www.littlelabs.com.

▶ SPECTRASONICS VIRTUAL INSTRUMENTS

Instead of supporting specific sampler formats, Spectrasonics' Virtual Instruments ship as native plug-ins for VST, MAS (MOTU Audio System), and RTAS (Real Time AudioSuite). The program is compatible with Mac and Windows platforms. Each synthesizer plug-in is fully programmable and offers a different interface and instrument type.

Stylus ($299) is a groove-oriented instrument. At 3 GB, it offers more than 30,000 sample elements, including 700 loops, in a variety of styles, including R&B, two-step, trip-hop, acid jazz, trance, funk, and rap. You also get thousands of sampled percussion elements that you can use to create your own grooves. The grooves are malleable because of the Groove Control feature, which lets you change the tempo, pitch, feel, and individual instruments in real time.

Atmosphere ($399) offers a 3 GB core library of pad sounds. Programmer Eric Persing created the pads from vintage and modern synths. The sounds include analog beds, pads, wavetable sweeps, vocoded choirs, thick string pads, and processed vocal washes. You can easily combine any layers to create your own pads.

Trilogy ($399) focuses on acoustic, electric, and synthesized bass. Spectrasonics' True Staccato sampling feature provides multiple staccato samples, which the manufacturer says yields natural-sounding repeated notes with a significantly greater level of realism than other sampled basses. Sampled articulations include glissandi, hammer-ons, pull-offs, harmonics, trills, pops, noises, and slides at various speeds. The samples cover a variety of instruments and playing styles, and the instruments are mapped with multiple dynamics.

Each Virtual Instrument offers multimode resonant filters, four LFOs, and one envelope generator each for pitch, filter, and amplitude, as well as a modulation matrix. There is also a master filter for quick tweaks. Virtual Instruments require a MAS, RTAS, or VST 2.0 host; a PPC G3/300 MHz with 256 MB of RAM; and OS 8. Windows users need a Pentium II/400 MHz with 256 MB of RAM and Windows 98. Ilio Entertainments (distributor); tel. (800) 747-4546 or (818) 707-7222; e-mail ilioinfo@ilio.com; Web www.ilio.com.

LEXICON MPX 110

Arguably the most prominent feature of Lexicon's MPX 110 ($329) is its 24-bit stereo reverb. However, this low-priced processor also offers high-quality effects such as tremolo, chorus, flange, pitch shift, detune, delay, and echo. The MPX 110 also lets you run effects independently on the left and right channels.

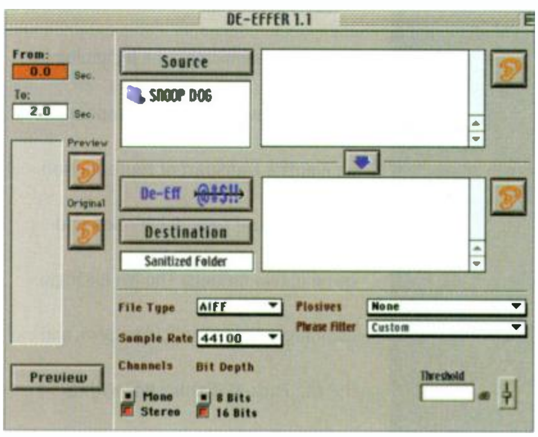
The MPX 110 offers 240 presets with 16 user locations. A front-panel knob allows you to adjust parameters, and the Effects/Balance knob adjusts effects levels or the balance between dual effects. On the front panel, dual two-stage headroom indicators let you monitor signal input.

The MPX 110 features a Learn mode, which governs MIDI control of effects level, parameter adjustments, and more. Delay effects can be locked to audio signal, MIDI Clock, or tap tempo from a front-panel button or dual footswitch. You can also modulate tempo with your choice of MIDI messages.

The rear panel has two unbalanced ¼-inch inputs; two unbalanced ¼-inch outputs; a ¼-inch stereo headphone jack; and a coaxial S/PDIF output. MIDI jacks are In and Out, with a software switch that converts Out to Thru. Lexicon, Inc.; tel. (781) 280-0300; e-mail info@lexicon.com; Web www.lexicon.com.

AUDIO-OZ AUDIO DE-EFFER

Newcomer to the software digital signal processing scene Audio-Oz Audio makes its debut with De-Effer ($129.95), a standalone program for Mac and Windows computers. The program derives its processing power from voice-recognition algorithms, letting you remove offensive language from stereo tracks in real time.



The software provides a library of target words, and you can store your own words and phrases by recording them and saving them as digital-audio files. De-Effer can read and write AIFF, AU, and WAV files at all of the commonly used sampling rates and bit depths. A software utility is provided for reading Ensoniq Mirage, E-mu SP-12, and Akai MPC-60 floppies.

As with all voice-recognition software, intelligibility of the audio material always determines the effectiveness of the process. To that end, De-Effer provides sliders for adjusting the threshold that activates the process. The manufacturer claims that a test on a recent Snoop Dogg release revealed the software's effectiveness as a vocal eliminator. You can enable look-ahead features for key audio events, including violent plosives or specific phrases. The software offers batch-file processing, permitting you to clean up as many files as your computer's RAM will allow.

De-Effer requires at least a Pentium II/200 MHz with 64 MB of RAM and Windows 95. The Macintosh version requires a PowerPC 604e/200 MHz with 64 MB of RAM and OS 8.6.1 or later. Audio-Oz Audio; tel. (800) 446-8242 or (800) IGOTCHA; Web www.audio-oz_audio.com/audio.html/~audio/de_effer.htm.

TASCAM POCKETSTUDIO 5

Tascam's Pocketstudio 5 ($599) packs four tracks of digital audio, a built-in MIDI synthesizer, an MP3 encoder and player, a USB jack for computer connectivity, a built-in condenser mic, and Compact Flash card storage into a six-by-eight-inch package. The diminutive recorder provides autopunch in and out, track bounce, and copy-and-paste capabilities for audio and MIDI tracks.

The Pocketstudio 5 can read type 0 and type 1 Standard MIDI Files. Onboard MIDI files are arranged by song sections—intro, verse, and chorus—so that you can quickly assemble song structures.

Final stereo mixes are saved as 16-bit, 44.1 kHz MP3 files. You can use the USB port to offload files to your computer for additional processing or posting to the Web. MP3 compatibility lets the Pocketstudio 5 serve as an MP3 player.

The Pocketstudio 5 has an unbalanced ¼-inch line-level input, an unbalanced ¼-inch mic-level input, and two ⅛-inch jacks for mic- and line-level inputs, respectively. You also get an ⅛-inch stereo minijack line output and an ⅛-inch stereo headphone jack. The unit has a slot that accepts Compact Flash cards up to 128 MB. The package includes a 32 MB Compact Flash card, a headset microphone, and an AC adapter. The unit runs for two hours on six AA batteries. Tascam; tel. (323) 726-0303; Web www.tascam.com.

emagic
Technology with soul

Music Production
Soft- and Hardware

successful by design | Logic Platinum 5
Logic Control

Logic Platinum 5 and Logic Control set the new standard by which success must be measured. Engineering experience, inspiration and invaluable input from professional users around the world combine to make a highly-regarded music production system even better. The superiority of the Logic Platinum 5 software becomes even more apparent when connected with Logic Control and Logic Control XT hardware. With hands-on access to hundreds of MIDI and audio functions, this is a peerless combination designed to grow along with your creative needs. New automation, 11 new plug-ins, hardware independent audio scrubbing, renowned POW-r dithering and enhanced functionality in the score and MIDI editors are just some of the innovations in Logic Platinum 5. A range of optional software instruments, including the new ES2 and EVOC20, round out a music and audio production system designed to let you work faster, achieve more success, and have more fun.

Technology with soul.

KORG TRITON STUDIO

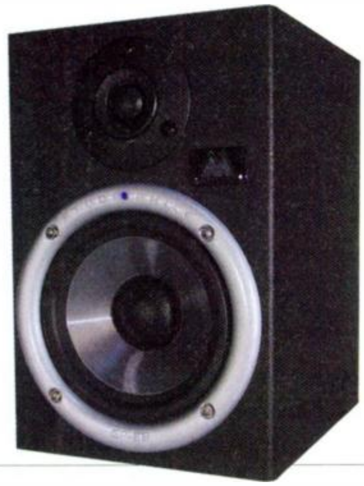
Besides the touch screen and the full capabilities of the earlier Triton synths, Korg's Triton Studio (61 keys, $3,400; 76 keys, $3,800; 88 keys, $4,200) includes the new Open Sampling System, which lets you sample or resample from Program, Combination, or Sequencer modes. The sampling is done at 16-bits, 48 kHz in mono or stereo. The unit's 16 MB of memory is expandable to 96 MB using 72-pin SIMMs.

You can sample external sources to a track while the sequence is playing; the sample is automatically assigned a trigger event that starts the sample's playback at the proper time. Korg offers the optional CDRW-1×8 ($400), an internal CD-RW drive for burning audio files of your sequences or storing patches, sequences, and samples. The drive also allows you to sample from audio CDs or load AIFF, WAV, or Akai samples. You can also use the unit's built-in 5 GB hard drive. The rear panel of the keyboard has a SCSI port so you can use an external CD drive or another device.

The unit's new processor significantly speeds up touch-screen performance. The workstation's sound set includes a 16 MB acoustic grand piano. You get 1,536 user program locations and an equal number of user combinations. You can expand sample ROM to as much as 160 MB with Korg's 16 MB EXB-PCM expansion boards ($240 each). The synth will also accommodate the EXB-MOSS expansion board ($600) for an extra six notes of digital signal processing–based synthesis.

As with other Triton instruments, you get a 16-track sequencer that can hold 200,000 MIDI events and 200 songs. The built-in effects processor offers all of the algorithms of the original Triton.

The Triton Studio has two unbalanced ¼-inch inputs, six unbalanced ¼-inch outputs, and optical S/PDIF I/O jacks. You can add the EXB-DI output connector for ADAT ($200) or the EXB-mLAN ($750) for MIDI and audio I/O via mLAN. Additionally, two control inputs accommodate a damper pedal and an assignable footswitch or pedal. MIDI connectors are In, Out, and Thru. Korg USA, Inc.; tel. (516) 333-9100; Web www.korg.com.



M-AUDIO SP-5B

M-Audio has just released the self-powered, biamped SP-5B ($399 a pair) close-field monitor. The manufacturer says that the SP-5B has a stable, balanced low-to-midrange response and well-defined middle and high frequencies.

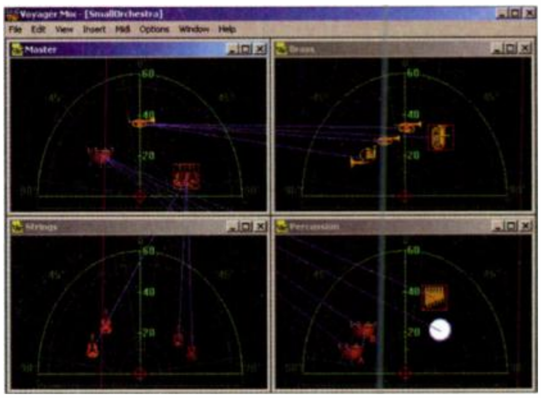
The speakers are suitable for desktop production because the drivers are magnetically shielded. Swivelmounted, ¾-inch silk-dome tweeters let you adjust the direction of the high frequencies and control the imaging. The tweeter's dome employs ferro-fluid damping, which minimizes speaker self-resonation.

You get a 5¼-inch woofer that has a mineral-filled, high-temperature voice coil. The subfrequency port channels frequencies below 30 Hz. The biamp structure delivers 40W to the bass and midrange driver and 30W to the tweeter. M-Audio promises a smooth transition in the crossover frequencies. Midiman/M-Audio; tel. (626) 445-2842 or (800) 969-6434; e-mail info@midiman.net; Web www.midiman.net.

VOYAGER SOUND GRAPHIMIX 01

GraphiMix 01 ($129) is an object-oriented graphic mixing environment for Windows computers. Its icon-driven user interface provides a programmable visual display of mix elements. You can represent tracks in a virtual sound field with text or icons; the position of icons in the display reflects mixer settings such as gain, pan, and effects. Moving the icons within the field controls your MIDI compatible mixers, sequencers, sound cards, effects processors, and more. You can control objects with the mouse, a QWERTY keyboard, or an external MIDI device.

GraphiMix 01 is set up to take advantage of Yamaha's ProMix 01 and 01V mixers; however, you can define settings and develop controls for most MIDI-capable mixers. Right-clicking on icons calls up a menu of programming options, including control assignments and constraints for icon movement. GraphiMix 01 can control 16 mixers at once; linked mixers can also appear onscreen as a single device. The program supports multiple stereo and surround mixes simultaneously.



GraphiMix 01 requires at least a Pentium II/200 MHz computer with 64 MB of RAM and Windows 95, 98, NT, 2000, and XP. Voyager Sound Inc.; tel. (781) 893-2574; e-mail sales@voyagersound.com; Web www.voyagersound.com.

EMAGIC LOGIC CONTROL

Developed as a joint effort between Emagic and Mackie Designs, Logic Control ($1,299) is an expandable hardware control surface for Logic Platinum 5 ($949; upgrade from version 4, $149). The unit provides control for hundreds of MIDI and audio features, including Logic Platinum's new automation system as well as software instruments and plug-ins.

Logic Control has motorized 1,024-step, touch-sensitive Penny and Giles optical faders for each channel, a master fader, an assignable rotary knob with an integrated push button for parameter adjustments, and Record, Solo, and Mute buttons. The controller provides access to eight audio channels at once; additional audio channels are accessible with the unit's bank-switching functions. Dedicated controls include transport, cursor keys, automation, and function mode switches. Logic Control has a backlit, multifunction display that provides you with detailed parameter and metering information.

If you have sufficient MIDI I/O, you can expand the system for simultaneous control of virtually unlimited audio channels with the Logic Control XT expander unit ($1,099). Each expander provides the same Record, Solo, and Mute buttons, motorized faders, knobs, and switches as the core unit.

Logic Platinum 5 automatically recognizes the hardware, so you have instant access to important controls right out of the box. The controller sports MIDI In and Out jacks, and you can update the firmware with MIDI System Exclusive dumps.

Logic Control presides over Logic Platinum 5's new 32-bit fader automation system. Automation data can be moved or copied independently. Track automation write modes operate independently of sequencer recording.

Logic Platinum now offers more than 50 plug-ins, including Adaptive Limiter, SubBass, Phase Distortion, and DeEsser. New mastering tools include StereoSpread, Denoiser, Limiter, and Multiband Compressor. A new mixer sidechain permits you to use external audio as a control signal. Logic Platinum 5 adds support for Propellerhead's REX 2.0 files and integrates support for the Logic Control hardware system.

To use Logic Control, you need Logic Platinum 5. For the Mac, that requires a PowerPC 604e/250 MHz with 128 MB of RAM running OS 9.1 or higher and a USB port. Windows users need an Athlon, Duron, or Pentium II processor running at 300 MHz or faster; 128 MB of RAM; Windows 98SE, ME, 2000, or XP; and a USB port. Emagic USA; tel. (530) 477-1051; e-mail emagic@emagicusa.com; Web www.emagic.de.

PRIMERA COMPOSERPLUS

The ComposerPlus Optical Disc Duplicator ($2,795) from Primera Technology uses a 24× CD-R with a FireWire interface to duplicate as many as 100 discs per session without reloading blank discs. With the optional Signature IV CD Color Printer ($1,495), you can simultaneously print 1,200 dpi disc-surface designs.

A robotic arm shunts blank discs from an input bin to the recorder; when a disc is finalized, the arm transports the disc to the printer. For higher-resolution graphics, you can purchase the SignaturePro CD Color Printer ($1,895) for 2,400 dpi printing or the Inscripta ($2,995), which offers thermal printing.

System requirements for the unit are a Pentium II/450 MHz computer, 128 MB of RAM, and Windows 98, ME, 2000, or XP. The package includes Prassi's PrimoCD-Pro duplication software. Primera Technology, Inc.; tel. (800) 797-2772; Web www.primeratechnology.com.

AKAI PD16

Tired of sequencing drum and percussion tracks from your MIDI keyboard? Akai's PD16 ($349) provides you with 16 drum machine–style, Velocity- and pressure-sensitive pads and an assignable slider in the tradition of the company's MPC series of instruments.

The controller can be powered from an AC adapter or from a computer's USB port. You can send MIDI data either through the USB connection or from the unit's MIDI Out jack.

The PD16 has two Velocity modes. Full Level sends a fixed Velocity of 127, and 16 Levels divides Note-On Velocity into 16 values. The Active button engages the slider, which sends your choice of Control Change messages.

You can program MIDI channels, Pad Note Numbers, and Control Change message assignments from your computer. Akai provides setup utility software for the PD16 and USB MIDI drivers for Mac and Windows platforms. Akai Musical Instrument Corp.; tel. (800) 433-5627 or (817) 831-9203; e-mail info@akaipro.com; Web www.akaipro.com. ❁

Introducing the **PM-1 Powered Monitors**

Listening doesn't need to be expensive. To understand how truely important accurate and clear sound is, you only have to go as far as the new Fostex PM-1 Powered Monitors. With 120 watts of biamped power, a bass response that's full and dynamic, and a high end that sizzles, you'll wonder how you can pay so little for so much. Priced at $499 MSRP, you can now have accurate and clear sound that is affordable as well. The PM-1 Powered Monitors from Fostex: just listen.

FOSTEX

www.fostex.com

Fostex America, 15431 Blackburn Avenue, Norwalk, CA 90650, Tel: 582-921-1112 Fax: 562-802-1964

TECH PAGE

By Scott Wilkinson

Imagine

> *Imagine all the people sharing all the world.*

Few who were alive on September 11, 2001, will forget the date. For many it was a call to arms, but others are concerned that waging war will only perpetuate the cycle of violence, leading to more terrorist attacks, followed by further retaliation, and so on until civilization crumbles into dust.

Among those concerned is Dr. Spanky N. R. Ganglia of Callosum Corp. U.S. As recounted in this column two years ago, Ganglia has been working on an ingenious invention called the Mindophone (see "Tech Page: Music on the Brain," in the April 2000 issue). The device includes a headpiece embedded with sensitive electromagnetic detectors that monitor brain activity associated with musical thoughts and send those impulses to a computer, which converts them into audio signals that can be played through any sound system.

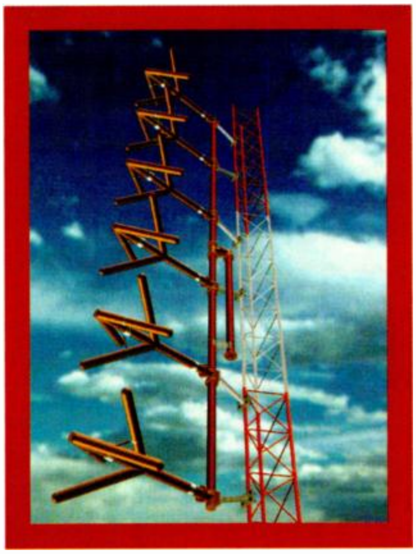
After watching the plane bombs explode on TV that terrible Tuesday, Ganglia wanted to find a way to inhibit the hate that impels people to brutalize each other—not only al-Qaeda terrorists and Americans but also Arabs and Israelis, Irish Catholics and Protestants, goth teens and their parents, and countless others. He figured that violent tendencies could be replaced by harmonious ones, and the vicious thoughts might dissipate as a result. The drive to annihilate in anger would be reduced or eliminated, thus breaking the seemingly endless cycle that grips the world.

Simply playing music won't work, even if it could be reliably received by everyone involved. But the events of 9/11 turned many beliefs upside down, leading Ganglia to a creative breakthrough: his Mindophone might become a potent tool against terrorism and violence if it could be applied in reverse. Drawing on the latest brain and mind research and the old adage, "Music soothes the savage beast," he decided to try broadcasting radio signals not in the audio range but in the brain-activity range to stimulate the sensation of hearing music.

Ganglia started by recording a musical selection into the computer, which he programmed with an inverse algorithm to convert the signal into impulses analogous to the brain activity associated with hearing or thinking about that music. Then, he broadcast the impulses from a specialized radio antenna (see **Fig. 1**), inducing the corresponding electrical activity in nearby brains. As a result, those within range appear to hear the music "in their head."

Initial experiments involved volunteers with adversaries who agreed to participate, hoping to resolve their differences peacefully. Bickering couples, hostile neighbors, and *Real World* roommates were gathered together within range of the antenna, which broadcast brain-wave impulses derived from selections such as "Give Peace a Chance," "Dona Nobis Pacem," "All You Need Is Love," "Let There Be Peace on Earth," and, of course, "Imagine." In addition, Ganglia mixed in more layers of subliminal pacifism than Enya has reverb, including sacred chants and hypnotic trance loops that help synchronize the brain's electrical activity.

The results were astounding. Instead of fighting, the subjects calmed down as the music penetrated deep into their psyches. Perhaps for the first time, they actually heard the messages of the songs that had been selected. One participant, a musician, was particularly moved by these lines from Malvina Reynolds's "Singing Jesus": "Beat your swords into plowshares /and your guns into steel guitars."



COURTESY ANTENNA CONCEPTS, INC.

FIG. 1: Dr. Spanky N. R. Ganglia of Callosum Corp. U.S. uses a specialized radio antenna to broadcast brain-wave impulses derived from musical signals to calm violent tendencies.

The next step is broadcasting the impulses over a large area from satellites. (To prevent messages of hate from being transmitted, Ganglia uses an artificial-intelligence algorithm that analyzes lyrics and encodes only songs of peace and love.) If it's successful, perhaps John Lennon's vision might finally become a reality: "Imagine all the people / living life in peace." ❂

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

Demos That Deliver

Music pros reveal how to produce a winning demo. By Michael Cooper

The best musicians spend years mastering their instruments and perfecting their composition chops. But that's often not enough to guarantee success in the music business. As in any other business, you need some sort of promotional device to get the recognition and earnings you deserve in return for your skills and services. In many instances, your most important promo is your demo recording.

Demos can be used effectively in the pursuit of many goals, including securing club and concert gigs, a music-publishing contract, and a record-label deal and producing music for film or television. But depending on which objectives you're shooting for, you might need to take a different approach in producing your demo. How much instrumentation do you need for your demo to be effective? Is the recording quality crucial to your demo getting favorable notice? Should you mix your demo to sound like a commercially distributed record or soundtrack, or should you goose the levels on important tracks, such as the vocal, to get your point across? Can you produce a winning demo on a tight budget, or must you outspend the competition? Where can you cut corners to save money without killing your chances of success?

To learn the answers to those and other critical questions, I asked top decision makers—producers, record-label A&R reps, and music publishers—what they want to hear in a demo. I also queried successful soundtrack composers for film and TV about how they produce demos to find work in those mediums. In addition, I drew upon my decades of experience as a studio owner and engineer to offer my own perspective on producing an attention-getting yet cost-effective demo for securing club and concert gigs. (For additional background about the pros I interviewed, see the sidebar, "To Their Credit.")

Illustration by Jack Desrocher

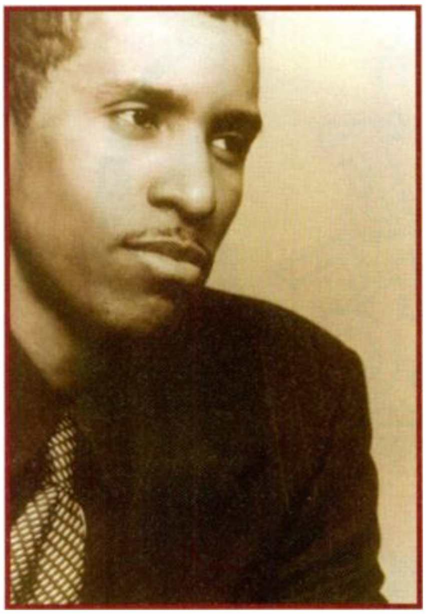
I won't discuss how to effectively pitch your demo after it's been produced; that's a subject worthy of its own article (see "Working Musician: Here Comes the Pitch" on p. 108). My sole focus is on what you need to do to produce a demo that will get noticed and propel your career forward. I'll start with how to record a demo to get gigs and then examine how to produce song, artist, and soundtrack demos.

CLUB AND CONCERT DEMOS

Many musicians mistakenly assume that even a roughshod demo will get them work performing in local nightclubs. Although that's sometimes the case, the bar was raised by the digital-recording revolution that began about a decade ago. It's not hard to produce a great-sounding demo these days, and if you don't produce a quality demo, the next band will. With every kind of demo (including song and artist types), it pays to always put your best foot forward.

That doesn't necessarily mean you should produce an ultraslick studio recording for your gigging demo. Most nightclub owners want to hear something that sounds slightly live and raw. If the recording sounds too polished, they'll probably assume (in most cases, rightfully so) that you won't be able to pull off the same sound live. That leaves the nightclub manager back at square one, wondering how you will perform in a live setting.

But another misconception is that a raw, live sound equates to sloppy performances. You will never be penalized for being a tight band or flawless solo performer. On the other hand, derelict grooves and out-of-tune guitars might cost you the gig.



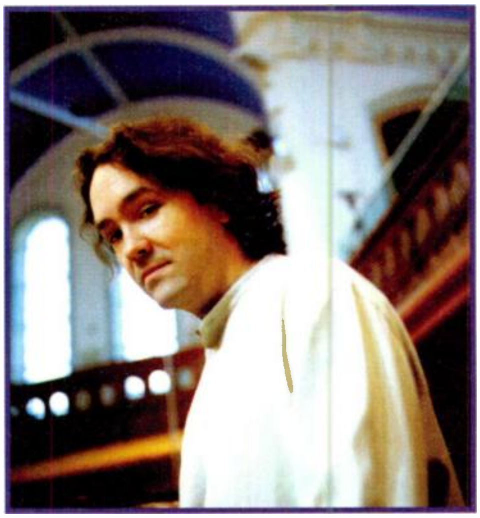
COURTESY DOUBLE XXPOSURE

"If you have the savvy to put together an entire album," says Jojo Brim of Def Jam/Def Soul Records, "it definitely gives you a sweeter deal, because you've already saved me half a million dollars in recording."

LET IT BLEED

The key to getting a rougher sound while maintaining professionalism lies in your microphone-placement choices and the types and amounts of reverbs you use in the final mix. For a full-band recording, consider having some instruments playing in the same room (as opposed to playing in separate isolation booths) to add a little mic bleed to tracks. If you're recording in a pleasant-sounding room, placing one or two additional room microphones several feet back from the drums will add some natural ambience to the recording. Just make sure that the amount of mic bleed and ambience you add are subtle, or your recording might sound as though it was made in someone's unfinished basement.

To add a sense of realism to your productions, restrict your choice of reverbs to plates, springs, small chambers, and small rooms (whether digital emulations or the real things). Unless you're already performing concerts, large concert-hall settings tend to sound fabricated. Add enough reverb at mixdown to create a sense of depth, but not enough to be readily noticed by nonmusicians. With a moderate amount of natural ambience and signal processing added to your tracks, you can fool most nightclub owners into thinking they're listening to a well-made live recording, in spite of the numerous clean punch-ins you might have done. That will make them confident that you can pull off a live show in their club.



TODD KASOW

When you pitch ideas for a film-scoring project, composer Carter Burwell says that you should "create one brilliantly produced synth suite to convince everyone of your genius, followed by more pedestrian sketches of the film cues."

KEEP IT SIMPLE

Never forget your purpose for recording your gigging demo. Unless you aim to release the recording for general distribution and sale, skip the temptation to triple-track the background vocals and add a tambourine overdub during the choruses. Such frills will have no material effect on whether you get the gig at a local nightclub, and most likely, the club owner won't even notice them.

Limit the number of songs on your demo to no more than three or four. Those who listen to your demo will probably make up their minds whether to hire you by the time they've heard the first couple of songs.

Plan your budget and your time in the studio to allow for getting great mixes. If possible, reserve 30 to 50 percent of your budget for mixdown. Trust me: you need that much time to get great mixes, especially if you're mixing more than 16 tracks. Most club owners cannot discern the difference between

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horrible mixes and poor performances. A wretched mix of great performances won't impress anyone. Great mixes of killer performances will get you enough gigs that you'll recoup your mixdown expenses in no time.

If you are trying to win concerts, consider producing some clean recordings of your live shows. Wild audience reaction garners instant credibility for your performance abilities. Just make sure any microphones intended to capture the crowd are far out into the seating area. If the mics are too close to the stage, your performance might overpower the audience reaction, making those mics worthless or even detrimental to the overall sound.

You might need to add a short delay to stage mics to get their signals in phase with the audience mics. If you don't align the phase of the stage and audience mics, the overall sound will likely suffer comb-filtering effects that thin out your sound as well as confusing slapback echoes that ruin the music's groove.

SONG DEMOS

Unlike demos that purposely promote the abilities of the musicians involved, the sole goal of a song demo is to sell the song itself. However, that doesn't mean that the musicians' performances aren't critical to the song demo's success. Always seek out the best singers and musicians to perform on your demo so that the song is represented in the best possible light.

Country-music producer Byron Gallimore advises most songwriters not to sing their own demos unless they are really great singers. "If the vocal isn't up to snuff," he says, "there's a danger that [the person you're pitching the demo to] may focus on the singing and miss the song."

Jim Vellutato, creative director for Sony/ATV Music, agrees with Gallimore. "I believe having the best vocal you can get increases the chance of you getting a record deal," he says. "On every song, but especially on ballads, the vocal has to be exceptional."

That advice runs contrary to an urban legend that a great vocal is the kiss of death for a song demo. The theory behind the myth is that an artist might shy away from singing a song that the demo singer does better. "I haven't found that to be true," Gallimore says. "Most artists believe in themselves. Nobody I've ever worked with has been afraid to sing a song because a great demo singer sang it, and we have some great demo singers in Nashville, too."

Vellutato sheds further light on the subject: "If you're sending your demo to Celine Dion, she's definitely going to want to hear a great vocal. If, on the other hand, you're sending a song to a weak vocal artist, sure, a great vocal could scare him or her away. But it's probably not the type of song that person would do anyway."

Before beginning production on a demo, a songwriter must decide whether to use a male or female vocalist to sing the song. That can be a difficult choice if you know you'll be pitching the song to artists of both genders. According to Vellutato, the dilemma is that "if you're pitching to Celine Dion, for example, it's really difficult to use a male vocalist and have her 'hear' that song."


J. M. GUYAUX

To succeed as a film and television composer, Douglas Cuomo says, "You pretty much have to have your own recording gear. It's not really feasible to go back to an outside studio and spend money every time a change is requested."


BETH GWINN

Grammy Award–winning producer Byron Gallimore works with some of the hottest artists in country music, including Tim McGraw, Faith Hill, Jessica Andrews, and Phil Vassar.

Vellutato suggests making two versions, male and female, of the same song so that you can pitch it to both sexes. That approach can be expensive, but Vellutato maintains that it's a justifiable strategy for producing a special song that you think has great potential. "One song can make your career," he says. "A lot of writers are making a ton of money based on one hit." That one hit often leads to other deals down the road.

STYLE CONSCIOUSNESS

If you're planning to pitch your song to a specific artist, you might be tempted to have your demo singer mimic that artist. That approach can help artists hear themselves singing a song, but it has one potentially huge pitfall.

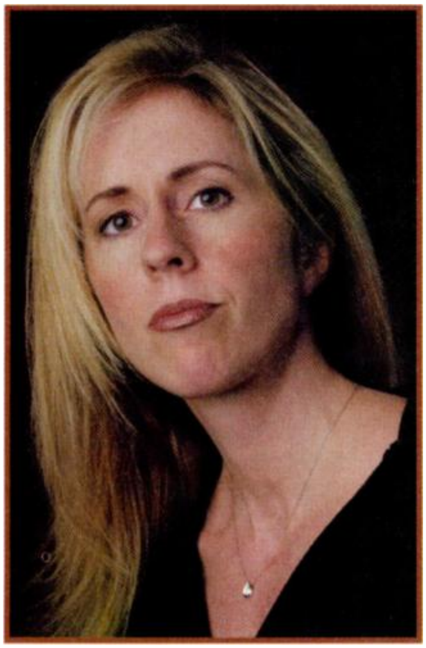
"Michael Jackson likes to hear everything demoed by someone who sounds like him," Vellutato says. "But the problem is that if you do a demo that sounds like Michael Jackson, it's really difficult to get anyone else interested in the song. I think you should realistically think about who would do the song and lean the vocal toward that artist, but I wouldn't try to make it sound identical."

Mike Whelan, director of Creative Services for Acuff-Rose, says the stylized-vocal approach can be expensive. "If

you stylize the vocal too much for one artist," Whelan says, "once you have pitched your demo to them and they've passed, then you have to put another vocalist on it. Unless we're going for a certain pitch, we try to have demo singers put the vocal right down the middle of the road. [That lets us make] a broad pitch on a song, hitting a bunch of artists with it."

If pigeonholing the vocal's style is often not advised, nailing the song's style for a particular artist is most definitely recommended. "In the majority of cases," Vellutato says, "I try to send something that's almost identical to what the artist recorded on his or her last album." The reason is that artists tend to stick with a winning formula, because it's in the style that won fans over.

Gallimore concurs, advising songwriters to "listen to the last two or three albums, and try to see the thread that runs through those."


PETER NASH

Carole Ann Mobley, the director of A&R for RCA Records Nashville, has blunt advice for anyone seeking a recording career in country music: move to Nashville.

Adhering too closely to what an artist has already done can put you behind the curve when an artist breaks new ground. But how can a songwriter know beforehand that an artist is about to change his or her style? In many cases, publishers tip their staff writers off to the upcoming change in direction.

"I think this is where a publisher is very important to songwriters," Whelan says. "Part of a publisher's job is to communicate daily, weekly, and monthly with the producers and record label of an artist to find out what the artist is looking for. Then we relay that information back to our songwriters." Having current information on what types of songs are needed is an advantage that staff songwriters have over the unsigned songwriter who makes broad, uninformed pitches.

HOW MUCH IS ENOUGH?

Compared with the guitar-and-vocal or piano-and-vocal offerings that were common 10 or 15 years ago, many recent song demos feature full productions. Still, opinions vary as to how much production is necessary for a song demo to be effective.

"As long as the production does an adequate job of presenting the song," says Gallimore, "I don't think [the amount of instrumentation] really matters. You don't have to have 5,000 parts on these things. What you want to sell on a song is a vibe. If you are selling the vibe of the song with adequate instrumentation and a great singer, that pretty much does it." On an up-tempo song demo, Gallimore feels strongly that "adequate instrumentation" includes drums.

Gallimore's credits attest to his stature as someone with the ability to recognize great songs, even in stripped-down form. "But those guys are few and far between," says Whelan. "Usually, we tell our staff writers that they're going to have to make a record in order for producers and artists to hear it."

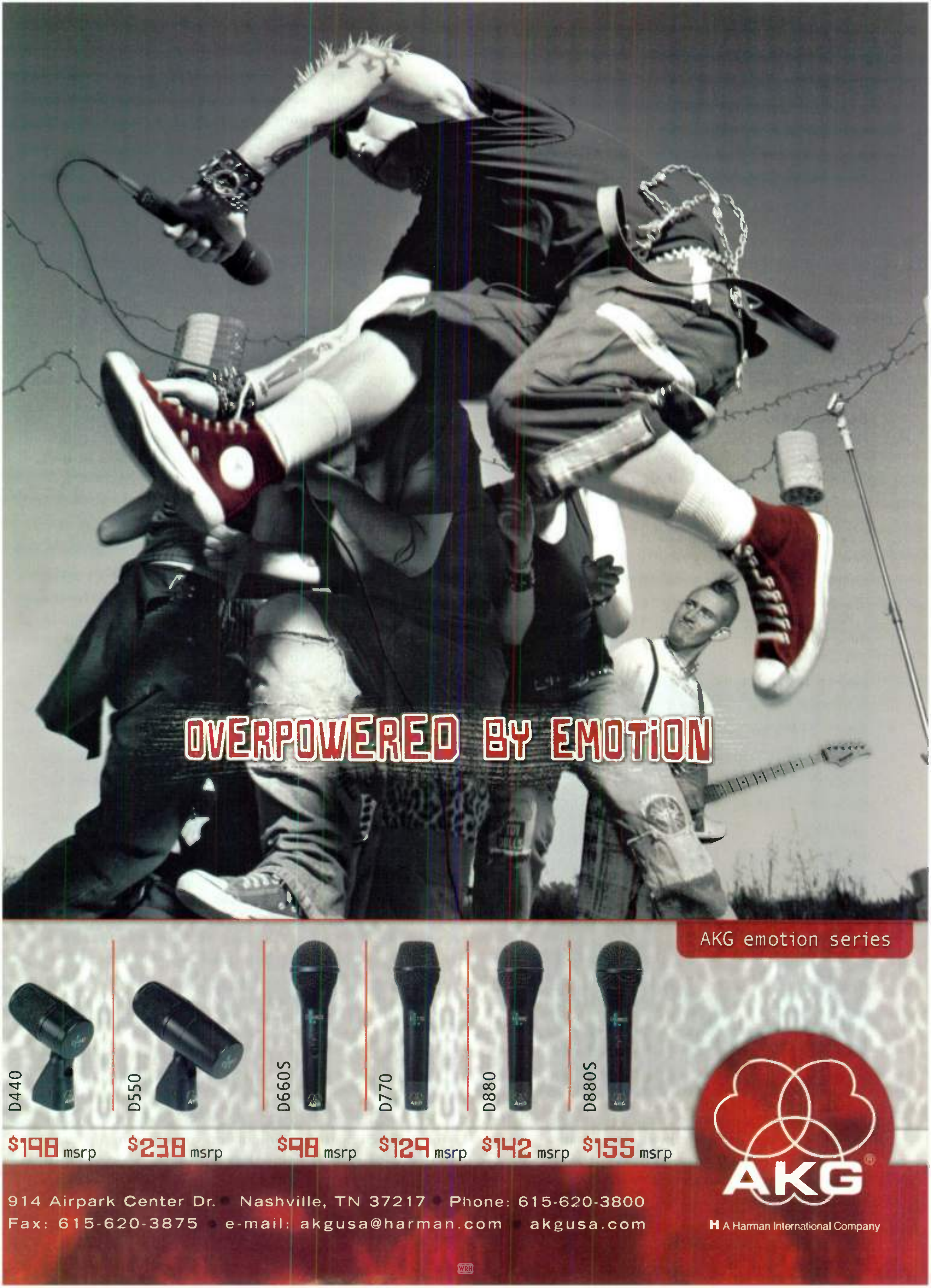

COURTESY JIM VELLUTATO

Regarding mixdown levels, Sony/ATV Music Creative Director Jim Vellutato suggests, "Anything that you want to attract attention should be prominent in the demo.

What if you're not pitching your song demo directly to a producer or an artist? Songwriters who present material to publishers need not be so thorough in their productions, according to Whelan. He notes that publishers typically have more time to dissect an undeveloped song demo than artists and producers do. As a result, he says that publishers can more readily hear a potential hit behind an undeveloped demo. "If you have a good guitar-vocal demo, most publishing companies can hear the talent," Whelan says. "If we hear a writer that we really like, we'll probably spend our money to do a [more fully developed] demo here in Nashville."

Whelan cautions songwriters not to submit outdated productions of their unplaced songs. "You probably need to update anything that sounds dated production-wise," he says. "You want to sound like you're on the cutting edge. When we get a song that we know was demoed in the '80s, we conclude that the songwriter probably hasn't written anything since the '80s and is still pitching back-catalog songs that he or she wrote 20 years ago. That sends up a red flag."

Vellutato says that the level of production a song demo needs depends on the style of music you're producing. "If you're writing a ballad," Vellutato says, "you can have a simple demo, perhaps consisting of only keyboard, drum machine, and vocal. R&B music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is dictated by the track; if you don't have a great musical track, a lot of people aren't interested in listening to the song. A rock demo, on the other hand, doesn't have to be incredible, but it has to be a great song that the band can't write itself. But rarely do I have a rock song cut by a band, because bands are mostly self-contained."

SOUNDS GOOD?

Opinions differ concerning the level of recording quality needed to have your song demo succeed. Gallimore indicates that he often picks up poorly recorded songs; he considers the song's vibe to be far more important to a demo's success. Whelan's view, from a publisher's perspective, is that recording quality is more important.

Whelan recommends that you produce your song demo in a quality studio, even if the studio's rates and your budget constrain you to keeping the production simple. He explains that an unprofessional recording will not favorably distinguish itself from the huge volume of other song demos that the publisher hears. "You have to present your demo in a professional way," he says. "The music business is a music *business*."

Vellutato concurs. He recommends hiring an engineer or going to a quality studio if you can't get great results in your own studio. "It would be really difficult for me to rerecord a song," Vellutato says. "It would be up to the writer. Part of getting a publishing deal is being able to present sellable material. It's just like any other business. There's a certain level of quality that you have to attain in order to sell your product."

MIX IT UP!

Okay, you've recorded all of your tracks, and now is the time to mix your song demo. Considering that the vocal track delivers the two most essential ingredients of a song, melody and lyrics, how loud should it be in the mix? Should you goose the vocal's level more than usual or marry it to the instrumental tracks as you would if you were mixing a record bound for commercial distribution and airplay?

"You want people to be able to hear the vocal," Vellutato says, "but it shouldn't be blatantly out front." That's good advice, but determining the right level for the vocal is sometimes difficult when studio time is whizzing by and your budget is being drained like water through a sieve. "It's difficult to perfectly set a demo's vocal against backing tracks like you would on a record," says Gallimore. "If I were going to err on one side or the other, I'd give it a touch more vocal. Try to make sure [the vocal is loud enough that] you can understand the lyrics. I don't like to look at lyric sheets when I'm listening, because it takes me out of the listening zone. I really need to be in a zone to see if a song is doing much for me."

Whelan takes no chances when it comes to setting vocal levels. "For country music," he says, "we encourage our writers to mix the vocal out front on the demo. For the most part, vocals should be mixed higher on demos than they are on records. We also encourage our writers to bring up any instrumental lick that's going to help get the song cut."

Vellutato agrees that instrumental hooks should be goosed. "Anything that you want to attract attention should be prominent in the demo," he says.

Gallimore also concedes that key licks should be easily heard but cautions songwriters not to go overboard. "I don't think [instrumental] hooks have to be stupid loud," he says.

ARTIST DEMOS

The primary objective of an artist demo is to obtain a record deal for the artist performing on the demo. In many instances, the artist will also be the songwriter whose songs are featured on the demo. Artist and Repertoire representatives (A&R reps) are typically the people who screen artist demos at record labels.

When producing an artist demo, it's generally more important to make the completed product sound like a finished record than with a song demo. Presenting an A&R rep with a demo containing fully fleshed-out arrangements will often place you squarely ahead of the competition as long as your performances and songs are hot.

"The more music you include," says Jojo Brim, producer and senior director of A&R for Def Jam/Def Soul Records, "the more it increases your chances of someone like myself being interested in you."

Not everyone feels that full productions are always necessary, however. For Carole Ann Mobley, director of A&R for RCA Records Nashville, the amount of production you need on a country demo depends on the style of music you're doing. "For an Alan Jackson–style song," Mobley says, "a guitar-vocal

COURTESY ACUFF-ROSE MUSIC PUBLISHING

According to Mike Whelan of music publisher Acuff-Rose, a rough demo might fail to effectively communicate a song's potential. "We tell our staff writers that they're going to have to make a record in order for producers and artists to hear it," he says.

demo is great. It's definitely not necessary for you to produce [that type of song] like a full record." On the other hand, Mobley notes that a more contemporary country song might need more production.

Brim and Mobley propose that an artist should submit at least three or four songs in a demo. "Three is a good number," Mobley says, suggesting "a ballad, an up-tempo song, and something else."

For those people who have the time and budget to produce an entire album, presenting such a package to an A&R rep is sure to raise some eyebrows. "If you have the savvy to put together an entire album," says Brim, "by all means, do that. It definitely gives you a sweeter [record] deal because you've already saved me half a million dollars in recording."

Brim recognizes that not everyone can deliver master-quality recordings, however. Although he appreciates a high-quality recording, he's also quick to say that a poorly recorded demo is not a deal breaker.

More important to Brim is that the artist's demo is fresh and real. He advises new talent not to mimic established artists and records. "Your work should be an extension of the deepest part of your personality," Brim says. "If it's not honest and fresh and who you really are, then I'm not going to be interested in it. If you're not helping to push the culture forward, then why are you participating?"

If Mobley is relatively short on advice on producing an artist demo, it's for good reason. To her, country music is mostly about live performance. "The demo package per se is not really that important," Mobley says. "There are lots of producers in town who can make a great-sounding recording. You can make anyone sound good nowadays. That doesn't tell me it's an artist we should sign. When we are presented with a new act that doesn't have a deal, we always have [the performer] come in and sing for us live. That's how we decide if it's worth moving forward."

If a musician impresses Mobley and the other members of the RCA staff at the meeting, the record company will then cover the costs for that person to make a demo. RCA will decide whether to pass on the artist or give him or her a record deal based in part on how the label-sponsored demo turns out. Personal contacts and working relationships are absolutely essential for a new artist to go the distance from initial meeting to demo to record deal. Recognizing that, Mobley has blunt advice for anyone seeking a recording career in country music: move to Nashville.

FILM AND TV DEMOS

If you want to produce music for film or TV, your demos had better sound great. Full productions and high-quality recordings are critical to a demo's success in securing that type of work.

"You should do as much as you can to make it sound as good as possible," says TV and film composer Douglas Cuomo. "You have to realize that other people's stuff sounds really good, so yours had better, too. That sometimes means using live players [instead of using only sequenced synths]."

Film composer Carter Burwell agrees that quality is paramount in all aspects of demo production. "Let's face it," Burwell says. "Film is a medium where form often triumphs over content. Your demos have to be as impressive as they can possibly be. The people listening to them can't necessarily distinguish between composition, arrangement, and production."

THE RIGHT STUFF

Cuomo advises that you should provide a variety of material on your general demo reel (an anachronistic term, considering that demos are now usually presented on CD) to showcase your versatility. For example, the music might include a mix of underscoring (which can serve as background music) and thematic pieces.

Once you're considered for a project, Cuomo suggests that you talk with the producer or director to find out what kind of music he or she is looking for. Then create some music you think would be appropriate for the project

TO THEIR CREDIT

The people interviewed for this article constitute some of the leading names in the music and entertainment industries.

When he's not in the studio producing leading R&B and hip-hop artists such as Mary J. Blige and Montell Jordan, **Jojo Brim** is discovering and developing talent as senior director of A&R at Def Jam/Def Soul Records in New York.

Carter Burwell has composed music for a multitude of feature films, including *Being John Malkovich, Conspiracy Theory, Fargo, The General's Daughter, Gods and Monsters, Raising Arizona*, and *Rob Roy*.

Douglas Cuomo's music credits include the hit TV series *Sex and the City* (HBO), *Homicide* (NBC), and *Now and Again* (CBS), as well as various film and theater projects.

At the time of this writing, **Byron Gallimore** has production credits for 8 of the top 60 singles on *Billboard*'s "Hot Country Singles and Tracks" chart. Gallimore also claims several ACM and CMA awards for his productions.

Carole Ann Mobley is director of A&R for RCA Records in Nashville.

Jim Vellutato is the creative director for Sony/ATV Music Publishing in Santa Monica, California.

Mike Whelan is the director of Creative Services for Acuff-Rose, one of the most prominent publishing companies for country music.

and include it on your demo reel. But here's a hot tip: don't tell the producer or director that the resulting demo was made specifically for the project.

"Part of the danger of doing a demo is that you're saying, 'This is my take on your project; this is what I would do,'" Cuomo says. "If they really like it, that's great. If they don't, then they might say, 'Oh, well, that's what *he* would do; forget it.' The hope is that you can engage them in some dialogue. If you do something and they say, 'Oh, that's not right,' you can then ask, 'What is it about this that you don't like? What do you want it to do that it's not doing?'"

CH-CH-CH-CH-CHANGES

If you get the nod to score music for film or TV, expect the producer or director to ask you to make a lot of changes to the music throughout the life of the project. "You pretty much have to have your own recording gear," Cuomo says, "because you'll regularly be going back and changing things. It's not really feasible to go back to an outside studio and spend money every time a change is requested."

In a sense, the demo process never ends until the underlying project is finished. "Most composers will still be creating demos—synth sketches—once they're hired to illustrate their musical ideas to the director prior to recording the final score," says Burwell. "My own view is that those demos don't need to be as well produced as the ones created to get a job, partly because you can count on more good will at this point. Also, time is of the essence [because of tight deadlines], and I don't want to waste it perfecting a synth demo [for instance, writing continuous controller data for every MIDI track] if it's all going to be replaced by real players. On the other hand, there is often a sales-pitch aspect to these demos. You may find that it pays to create one brilliantly produced synth suite to convince everyone of your genius, followed by more pedestrian sketches of the film cues."

ONLY THE BEGINNING

No matter whether you're producing a demo to get nightclub gigs, to score music for film or TV, or to win a publishing or recording contract, you need to make a commitment to stay in the game for the long run if you want to succeed in the music industry.

"Be realistic with your goals," Brim says, "and understand that the first few demos you make may not succeed." Practice makes perfect, but persistence makes a career.

Michael Cooper *has placed third in the Music City Song Festival and won first place in the Portland Music Association Songwriter's Contest.*

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

Webcasting Made Easy

A musician's introduction to the exciting and sometimes terrifying world of Internet broadcasting.

By Todd Souvignier

Webcasting, also known as Netcasting, allows you to send audio and video over the World Wide Web. Webcasting utilizes two key technologies: data compression and streaming.

Webcasters use a variety of media formats. Popular audio formats include M3U (streaming MP3), QuickTime, RealAudio, and Windows Media. In order to receive a Webcast, each audience member must have a media player installed on his or her computer, and that player must be compatible with the format of the Webcast. In this article, I will examine each of those media formats and discuss how to work with them after I cover a few basics. (Additional information about many of the products and services mentioned in this article is available online; see the sidebar, "Online Resources.")

SIGNIFICANT BIT

Prior to the advent of streaming, audio or video files had to be downloaded in their entirety to the end user's computer before the files could be played. Such an arrangement works—indeed most file-sharing networks continue to rely on downloading—but it means you always have to wait.

Streaming provides near-instant gratification. Rather than downloading a complete file, the user receives the audio or video as a bitstream. The bits are played in the order they are received, and then they are discarded. Streaming puts the content in front of the end user much faster than downloading, and it doesn't consume a lot of disk space on the user's computer. Streaming also provides the content owner with an additional level of security because disk files are not created on the recipient's computer: data is simply buffered until it's played and is then thrown away.

Illustration by www.lolabek.com

Webcasting Made Easy

Audio and video files can be huge. One minute of CD-quality audio uses about 10 MB of disk space. To distribute such large files over the Internet, the purveyors of streaming technologies utilize data compression.

Unlike the kind of compression used to reduce the dynamic range of an audio signal, data compression reduces the size of a file so that it can be transmitted over the Internet more efficiently. There are two stages to data compression: encoding and decoding.

The encoding stage requires software called an *encoder*. The encoding algorithm analyzes the original file, determines which portions can be omitted or represented with fewer bits, and then creates a new, smaller version of the file.

Unlike CD audio or WAV files, encoded audio and video files cannot be played back in their raw form. They must be decoded using a software application called a *player*. Some popular players are QuickTime, RealPlayer, and Windows Media Player.

STREAMING BASICS

Now that you understand the fundamental technologies of compression and streaming and are familiar with the roles of the encoder and the player, here's a look at a few of the concepts peculiar to streaming and Webcasting.

Client/server. A *server* is any computer that hosts files on a network and makes them available to other computers. There are special software packages, also called servers, that facilitate file serving, but in general use, the term *server* refers to the host computer and its file serving software as a whole.

The *client* is the other end of the equation. A client is a computer that receives files from a server across a network. The term *client* can also refer to the software that the recipient computer uses to play or access files.

Bit rate. The sound and picture quality of a compressed media file or stream is measured in terms of *bit rate*, which is expressed as a numeric value in kilobits—or thousands of bits—per second (kbps). Bit rate is a throughput measurement: it tells you how much data must be shoved across the Internet or pushed through a player in order for the content to be heard or seen in an uninterrupted fashion. Bit rate is the single most important setting in any encoder software because it determines the playback quality, the size of the output file, and the transmission requirements.

Bit rate also happens to be the way that modem speed is expressed. Because most 56 kbps modems rarely connect at speeds faster than 33 kbps, a lot of streaming media is available at 28 or 32 kbps bit rates.

Bit rate also gives you an indication of the amount of compression that has been performed on a file and a general sense of the resulting sound or picture quality. For example, 128 kbps has become a de facto standard for encoding stereo music files into the MP3 format. At the rate of 128 kbps, the compressed audio file is about 1/10 the size of the original uncompressed file; it sounds pretty good and is roughly equivalent to the sound quality of FM radio. At 64 kbps, an MP3 is about 1/20 the size of the original CD track, with sound quality roughly comparable to AM radio. At 32 kbps, an MP3 file is around 1/40 the size of the original and noticeable distortion or aliasing can be heard.

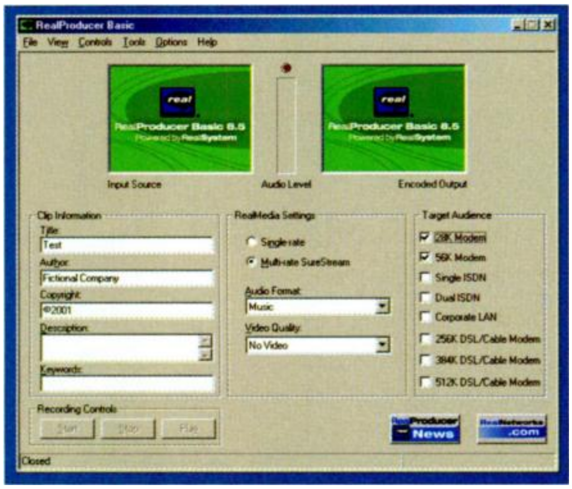

FIG. 1: RealNetworks' free encoder RealProducer Basic allows you to Webcast using RealAudio.

Unicast. *Unicasting* means sending one stream, point to point, from a server to another computer. The computer receiving the stream may be viewing or listening to the Webcast. Alternately, the receiving computer may be relaying the Webcast to other computers (see *reflector*). Unicasting is simple and may be all that is required if you do not anticipate a large number of concurrent visitors.

Multicast. As the name implies, *multicasting* means sending out multiple copies of a stream to many people at once. A multicast server (sometimes called a reflector) hands a copy of the stream to each client that requests it. Multicasting uses special server software that is equipped for the task and requires significant upload bandwidth for it to handle multiple concurrent users. Most multicasting utilizes Content Distribution Networks (CDNs), although peer-to-peer multicasting schemes are starting to come to the fore. I will go deeper into those subjects in a moment.

Reflector. Reflectors are simply servers, typically located within a CDN, that facilitate multicasting. The content originator sends a single stream—a unicast—to the reflector, which, in turn, relays the stream to multiple end user recipients to create a multicast.

URL. The Uniform Resource Locator, or URL, is the familiar Internet address scheme (for example, http://www.whatever.com/stream.asx). In the world of streaming and Webcasting, each stream is located at a unique URL. A Webcaster gives the URL to people who might wish to receive the Webcast and places links to this URL on Web sites. For unicasting purposes, the stream URL could simply be the location of the original media file on your Web server. In multicasting, the URL is typically an address at a reflector to which the source stream is directed and

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from which the end user receives the relayed stream.

Archive. In streaming terminology, the term *archive* refers to audio or video that exists as a disk file on a server, for the purposes of on-demand or time-delayed delivery. This is the opposite of a *broadcast*, which exists only as a bit stream that is played through a server and is not saved as a disk file.

On demand. In an *on-demand* situation, the stream is played whenever the end user requests it. Most streaming and Webcasting occurs on demand: the archive media files are always available and can be played from the start of the file at any time. Many **EM** readers have Web pages at MP3.com; the type of streaming done there is on demand.

Live event. For a live event, the stream begins and ends at predefined times, and audience members must click in to the stream during the scheduled Webcast in order to receive it. People who arrive late miss the start of the show and join the Webcast already in progress. Many radio stations retransmit their programming as live events over the Internet.

Simulated live. A simulated live event is one that has been prerecorded (and presumably edited) and is Webcast subsequently at a specific time that the Webcaster determines. As with live events, audience members must click in during the scheduled cast or they will miss the show.

Metafile. *Metafiles* are files about files: they point to other media files on the Internet. On-demand streaming relies heavily on metafiles. An M3U is an example of a metafile: it is a text file that contains a URL that is the location of an MP3 file. ASX and RAM (for Windows Media and RealAudio, respectively) are other common metafile types. I will explain metafiles in greater depth later.

Metadata. *Metadata* is data about data; it describes or annotates the content of a media file. ID3 tags are a familiar example of metadata. These text fields are included in the file header portion of an MP3 file and provide a convenient way to attach the author's name, song title, album title, musical genre, and other information to the media file. Most media players will display the contents of metadata fields.

USING STREAMING FORMATS

When approaching a Webcasting project, one of the first hurdles is deciding which media format to use. There is no universal format. However, during the past six years, the field has boiled down to a handful of popular choices. Because this is *Electronic Musician* magazine, I will focus primarily on audio, rather than video, streaming.

RealAudio. Arguably the most widely installed media player, RealNetworks' RealPlayer and its proprietary RealAudio format have the distinction of being the first participants in the streaming-media space. The sound quality associated with RealAudio has steadily improved over the years. RealPlayer comes preinstalled on many computers, and the Mac and Windows versions can be downloaded for free.

Webcasting in RealAudio requires a RealProducer encoder (also referred to as RealSystem Producer). RealProducer Basic (see **Fig. 1**) is distributed free of charge. The professional edition, RealProducer Plus, currently costs $199. These encoders compress audio files into the RealMedia (RM) format and simultaneously create a metafile, called a RAM file, that references the location of the RM file.

For multicasting applications, RealNetworks has RealSystem Server and a proprietary technology called SureStream. SureStream-encoded files contain several copies of the audio at different bit rates. RealSystem Server identifies each listener's connection speed and transmits at the appropriate bit rate. The selection is made automatically so that listeners receive the best-quality transmission possible at their current connection speed.

RealSystem Server Basic supports as many as 25 simultaneous listeners and is available free for a one-year trial period. RealSystem Server Plus, which handles 60 concurrent users, sells for $1,995. If money is no object, you can get the Professional edition of RealSystem Server: for as many as 100 simultaneous users, it costs $5,995; for 400 simultaneous users, the fee is $21,000.

All RealSystem Server packages require Windows 2000 or NT4, Linux, or FreeBSD; there is no Mac equivalent. RealSystem Server can be run on a computer at your Internet service provider's (ISP's) hosting facility. RealNetworks offers its own hosting and distribution service, which is priced and packaged for high-end corporate users (as opposed to cash-strapped musicians).

Windows Media. When RealAudio and MP3 became hot, Microsoft decided that it needed to come up with a competing

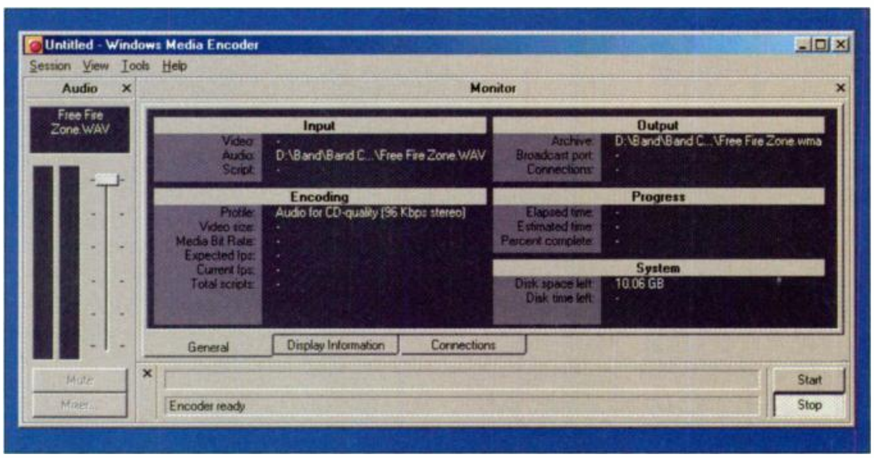

FIG. 2: The Microsoft Windows Media Encoder runs on Windows only and is distributed free.

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technology. Windows Media is a good-sounding format; its quality is roughly equivalent to RealAudio's and noticeably better than MP3's at comparable bit rates. Windows Media Player is bundled with Microsoft's Internet Explorer Web browser, much to the consternation of certain well-known competitors and ambitious antitrust attorneys.

Windows Media Player is fairly ubiquitous on the PC platform but is not widely installed on Macs, in part because of general cultural bias. Both versions are distributed free of charge.

You must have Windows Media Encoder (WME) to create content in the Windows Media format (see **Fig. 2**). The application is distributed free of charge but runs only on the Windows platform; no Mac version is offered. Moreover, WME seems to run best under Windows 2000. There are ongoing problems and some well-documented crashes that occur under Windows ME, and in my opinion, Windows XP is too new and untested to be relied upon at this time.

WME accepts an input source, typically a WAV file, and compresses the audio into a WMA file. (It will also convert MP3 files into WMA, but please resist the temptation: you don't want to compress a file that has already been compressed, for reasons of sound quality.)

To stream a WMA file, you need to create a metafile called an ASX. An ASX file is simply a text file that points to the location of a WMA file. To make an ASX, open a word-processing program such as Notepad and create a new document. This document will reference the location of the WMA file and should look something like this:

```
<ASX Version="3.0">
<ENTRY>
<REF href="http://www.hostfacility
.com/user/audiofile.wma"/>
</ENTRY>
</ASX>
```

The parts of that example in bold text must appear as shown for your ASX to work. Obviously, the URL will vary in every case because it must point to the unique location of your WMA file. Save this new text file with a name that ends in the file extension .asx (for example, songtitle.asx).

One of the most interesting things about Windows Media and ASX streaming is its support for URL scripts. Adding a URL to the file metadata during the encoding process allows you to create audio files and streams that open other Web pages. That lets you show lyrics, photos, band information, and even advertisements to the end user every time the audio is played. When the URL script is executed, a new Web browser window pops up and the specified Web page is displayed. This technique is not well documented and is somewhat involved. My "Desktop Musician" column on WME, in the February 2002 issue (available at www.emusician.com), offers complete step-by-step instructions on URL scripting.

MP3. The closest thing to a universal file format for music on the Internet is MPEG-1, Audio Layer 3, better known as MP3. MP3 constitutes the audio portion of a much larger digital-video specification that was established many years ago by the Moving Picture Experts Group. A number of MP3 players are available for the Macintosh and Windows platforms, many of them for free. Nullsoft Winamp (see **Fig. 3**), a product of AOL, is the most popular MP3 player for Windows PCs and is bundled with AOL's Netscape Navigator Web browser. Apple Computer's iTunes is the happening Mac player and successor to the venerable SoundJam MP.

FIG. 3: On the Windows platform, Nullsoft Winamp MP3 player is most prevalent.

As with all other compressed audio formats, encoding software is required to make MP3 files. Sometimes called *rippers* (as in *rip-off*), a variety of encoders are available for Mac and PC. Because of the restrictions that surround the licensing of MP3 encoding, most encoders cost money (after their trial periods time out). An exception is iTunes; Apple paid a hefty license fee and can give away its encoder for free.

MusicMatch Jukebox is a well-known encoder for the Windows platform. However, it times out and demands payment after 20 rips. Drop by the software section at MP3.com for a comprehensive selection of encoders and players.

When encoding audio into the MP3 format, don't overlook the ID3 tags. Those metadata fields allow you to include basic information such as artist name, album title, musical genre, and even track notes and Web site URLs. Most MP3 encoders support some form of ID3 tags, and nearly all MP3 players will display their contents.

To stream an MP3 file, you must create a metafile called an M3U. An abbreviation of MP3 URL, the M3U is simply a text file that contains the Internet address of an MP3 file. Open any word processor and create a new text document. The document must contain the complete URL of the file that is to be streamed and will look like this:

```
http://www.hostfacility.com/user/
audiofile.mp3
```

That's all there is to it! An M3U file can be much simpler than an ASX file; it need only contain the MP3 URL. As with an ASX file, the URL of an MP3 will vary in every case because it refers to the unique location of your MP3 file. Save this text file with a name that ends in the extension .m3u (for example, songtitle.m3u). Additional commands (called *tags*) can be incorporated into

an M3U or ASX file to modify its behavior. Use your text editor to open up metafiles you encounter on the Internet and take a peek at what other Web designers are doing.

Because most Internet music listeners have one or more MP3 players, M3U is a good format for on-demand streaming. Unfortunately, most MP3 encoding software is not designed for live Webcasting (unlike WME and RealProducer). You can find hardware solutions that perform on-the-fly MP3 encoding, such as the Audioactive MPEG Real Time Encoder. At a suggested retail price of $2,800, that product is clearly geared toward professional markets. There is, however, a dirt-cheap solution for live MP3 streaming: Shoutcast (more on that in a moment).

QuickTime. Developed by Apple Computer back in 1991, QuickTime was one of the first widespread digital-video technologies. Strictly speaking, QuickTime is not a format unto itself, but a *wrapper* or platform that can contain any of dozens of different audio, video, and graphics file types, including MPEG video or MP3 audio. QuickTime Player (see **Fig. 4**) is given away free of charge and is bundled with all new Macintosh computers and some Windows PCs. As one would expect, QuickTime Player is pervasive on the Mac platform but not quite as popular with Windows users.

If you want to author QuickTime media, you need to purchase QuickTime Pro, which sells for $29.95 and runs on Mac and PC. Unlike WMA, RealAudio, and MP3 streaming, QuickTime does not require the creation of a metafile. Instead, QuickTime begins playing any supported file type as it is downloading, even if the file is a garden-variety MP3.

For live-event streaming with QuickTime, you must have third-party live-streaming software, such as Sorenson Broadcaster (Mac, $199; Windows, $249). In order to multicast using QuickTime, you need to set up a Macintosh G3 or G4 computer as a reflector and run QuickTime Streaming Server, an application that comes bundled with Mac OS X Server. Alternately, an open-source version, called Darwin Streaming Server, is available for Windows, Solaris, Linux, and FreeBSD. QuickTime Streaming Server and Darwin Streaming Server are distributed for free.

ON-DEMAND STREAMING

The most primitive form of Webcasting would be to offer on-demand streams of individual songs or videos that can be unicast from a Web server. That kind of setup assumes that you already have a Web site and that that site is hosted on a Web server, either at a colocation facility or on a computer that's under your own control. Moreover, it assumes that you will be experiencing fairly low traffic levels—no more than one or two concurrent users at any time.

To make that work, simply upload the audio files and their associated metafiles to the Web server. Then, create a Web page that includes links to the metafiles and upload that to the server as well. If the metafiles contain the correct Internet addresses of the audio files, everything should run smoothly. Users clicking on the links will hear the media files begin streaming almost immediately.

If you're going to have only one or two concurrent users at a time, that type of setup should work fine and can be hosted at any inexpensive ISP's Web hosting facility. The downside is that this type of arrangement cannot scale to handle large audiences: if you get more than a handful of simultaneous listeners, the performance will begin to seriously degrade.

TURNKEY WEBCASTING

The simplest way to get your own Webcast online is to use one of the turnkey Internet radio services. Live365.com is the best-known and most user-friendly offering in that class. Broadcasters pay

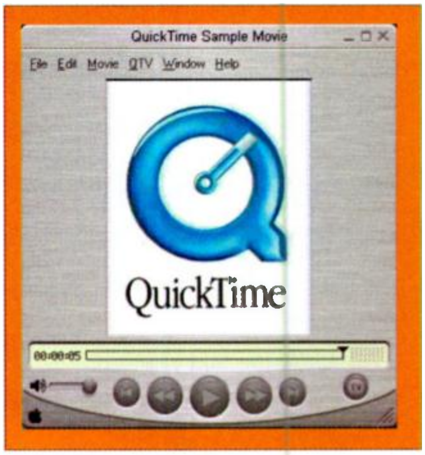

FIG. 4: Apple's QuickTime Player is available free for both platforms. This is an example of the Windows version.

a small initial setup fee as well as a nominal monthly charge to maintain their shows. Then they can upload MP3 files and a Playlist, which specifies the order in which the songs are to be played, to Live365's server.

Live365 handles media storage as well as streaming and adds a link to your show in its program guide. Live365 offers a software application that automates the process to some degree. At the end of the process, you'll have a simulated live event that loops continuously; listeners who click in to your Webcast will join it in progress. (For more details on how to Webcast using Live365, read Ron Simpson's article "World Wide Wadio" in the 2002 issue of **EM**'s *Desktop Music Production Guide*.)

A slightly more involved, but absolutely free, route is offered by Shoutcast, an MP3 Webcasting technology developed by the Nullsoft Winamp team. With Shoutcast, you simply drag audio files into a Winamp playlist or select sound-card input for live broadcasting, and then blast the program out to the world.

There are three primary components to a Shoutcast setup: the Winamp player software, the Shoutcast Source DSP plug-in, and the Shoutcast Server software. Download and install all three software elements. You can run the broadcasting software and the server on the same computer.

Once installed, launch Winamp and

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type Control + K to open the Preferences dialog; click on DSP/Effect in the left-hand panel; then, double-click on Nullsoft SHOUTcast Source DSP in the right-hand panel to open the configuration dialog. Select the Bit Rate for your stream in the Encoder tab. Choose Winamp Playlist or Sound Card Microphone Source in the Input tab. Most importantly, go to the Output tab (see **Fig. 5**) and click on the Yellowpages button to reveal the Description, URL, and Genre fields, where you will name and categorize your broadcast. Assuming you leave Make This Server Public checked on, your broadcast will be added to the Shoutcast program guide, where members of the general public can easily find you. Finally, launch the Shoutcast server software and click on the Winamp Play button.

Shoutcast Server basically configures itself and adds new broadcasts to the Shoutcast online directory automatically. With no prior Shoutcast experience, I was able to download, install, configure, and commence broadcasting in a mere ten minutes, just using the default Shoutcast DSP and Server settings. Any MP3 player can listen to Shoutcast streams, but a Windows PC with a high-speed connection, such as DSL or a T1 line, is required for Shoutcast broadcasting.

LIVE WEBCASTING

For real Webcasting excitement, try presenting a live event. With the proper gear and preparation, any live gig or studio performance can be encoded on the fly and broadcast to fans around the world.

First, you need a computer at the performance site running an encoder. Shoutcast is a viable option, though professionals tend to gravitate to RealProducer or WME. Next, you need a high-speed Internet connection such as DSL or a T1 line. The encoder will stream the broadcast audio to a Web server or reflector for relay to your listening audience.

To configure the encoder for broadcasting, you need to know the name or IP address of the destination server, the server port from which the stream will be transmitted (a four-digit number), and the file name users will hit to hear the stream. RealProducer and WME offer the option of archiving the broadcast to a disk file for later use.

In order for people to find and hear your Webcast, you can add links on your Web site to the broadcast file name on the Web server or reflector. You can also send the URL to your fans as a link in an e-mail message. Finally, patch the mixing board output to the sound-card input on the encoding computer, start the encoder software, and commence with the Webcast.

As with radio or TV live broadcasts, lots of potential problems can crop up. I've seen audio cables fail because they were stepped on, sound cards become unplugged or unseated, encoding software freeze and crash, typographical errors in links, inexplicable reflector failures, deliberate screwups by ISP employees, crummy audio mixes by negligent student interns, and unforeseeable bottlenecks on the Internet.

All of this potential for disaster makes live Webcasting exciting, sometimes to the point of being terrifying. Make sure to keep an archive program from a previous broadcast, or some long musical selections, stored on the encoding computer's hard disk, in case of a control-room hardware or wiring failure. You can quickly switch the encoder source to the backup audio and keep your listeners engaged while you locate and correct the point of failure.

Before you attempt any live Webcast, you should thoroughly test each part of the signal chain. Give yourself ample time for correcting any technical difficulties. It's also a good idea to communicate your Webcasting intentions to everybody downstream: let your ISP or CDN know about your live events well in advance of the actual Webcasts. Otherwise, they might do something like bring down your server for routine maintenance right in the middle of your broadcast.

CDNS

Running your own multicast server or reflector is generally beyond the scope of what most individuals and businesses are willing or able to do. With the increasing popularity of streaming media, a new business category has arisen: the CDN.

CDNs are in the business of getting content from its creator to the end user quickly and efficiently. CDN topologies vary; typically, they store data (such as Web pages, download files, or streams) in multiple places, usually making use of server farms or caches located near large ISPs or Internet backbones. In all cases, a CDN will assign you a URL that the public can link to, take a unicast from your encoder, and then multicast the stream to the listening audience. The URL may link to a single computer in one data center, or it may route traffic to any number of servers located at various points across the Internet.

Like many sectors of the Internet, the stream delivery business was overbuilt in the late 1990s and has been going

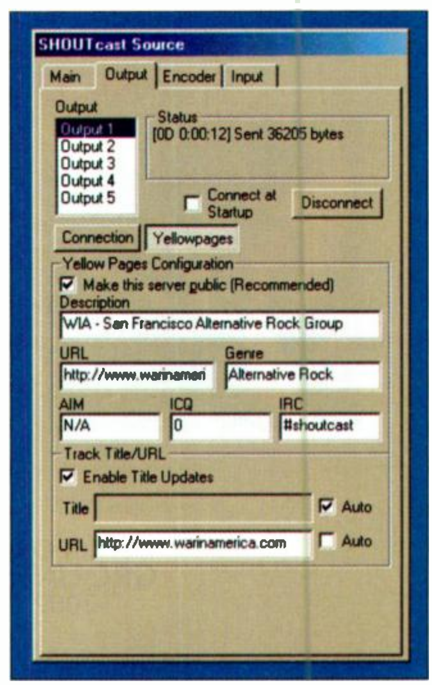

FIG. 5: This example of the Nullsoft Shoutcast DSP Plug-in shows the Yellowpages options under the Output tab.

Webcasting Made Easy

through a shake-out and consolidation of late. One of the most prominent stream relayers, Intel's Internet Media Services (IMS), shut down entirely after investing an estimated $200 million in its network. Another big player, iBeam Broadcasting, was recently purchased by Williams Communications after declaring bankruptcy.

Well-known CDNs that serve the high-end streaming media business and that are still alive at the time of this writing include Digital Island, RealNetworks, and sector leader Akamai. Additionally, there are smaller CDNs dedicated to medium- and low-volume multicasting. The Stream Guys is just one of many providers in the down-market end of this sector, providing stream deployment for as little as $1 per concurrent stream per month. If you need the capacity to have 100 simultaneous users, it would cost you $100 per month at the Stream Guys.

PEER-TO-PEER WEBCASTING

Along with the general dot-com shake-out, brutal price competition, and the worldwide downturn in advertising revenue, another factor hammering the CDN business model is the ascendance of peer-to-peer or *distributed* networks. Taking a page from the Napster playbook, peer networks rely on the bandwidth and computing resources of individual network users to distribute content. Instead of running massive server infrastructures that handle all user traffic, the peer network directs new users to other users that are already hosting a particular piece of content. Kontiki, a startup founded by some Netscape veterans, is a prominent participant in the enterprise side of this sector.

Of particular interest to musicians is AllCast, a New York–based startup that holds a core patent in peer-to-peer stream distribution. AllCast's technology, which is currently in a free beta release, allows any Windows PC with Windows Media Player to become a broadcast encoder. The AllCast Broadcaster software (see **Fig. 6**) allows users to select a Windows Media Player playlist, the computer's CD player, or the sound card's microphone input as the signal source.

AllCast Broadcaster encodes audio into the Windows Media format on the fly and adds the broadcast to the AllCast program guide. Users who wish to listen to an AllCast stream download a small plug-in and are directed to other users who are already receiving the stream. Users tend to come and go on peer networks, and the AllCast server handles that quite well by redirecting users to new sources with little or no interruption.

Because they make use of the bandwidth and resources of the network users, peer networks (in theory) don't require the expensive infrastructure that traditional client/server CDNs rely upon. As a result, they hold the promise of being far less expensive, particularly for small-audience Webcasting, than solutions such as Akamai's or even the Stream Guys'. Indeed, AllCast's solution is entirely free at the time of this writing, though it has indicated that it intends to charge for the production release when it becomes available.

Unfortunately, peer streaming is in its infancy. The technology is explicitly in beta (which means that bugs are known to exist), and the sound quality needs some work before it can be truly competitive with traditional CDN service.

DMCA COMPLIANCE

The Digital Millennium Copyright Act (DMCA) and the Digital Performance Right in Sound Recordings Act (DPRA) were passed by Congress in the late 1990s. Those laws place certain restrictions on Webcasters that affect what, and how, they may broadcast over the Internet (for a closer look at the ins and outs of the DMCA, see "Working Musician: Do the Right Thing" in the April 2001 issue).

FIG. 6: AllCast Broadcaster requires a Windows PC with Windows Media Player.

Most significantly, Webcasters are getting hit with a special license fee in order to "perform" music over the Internet. In addition to paying ASCAP, BMI, and SESAC for the right to broadcast musical compositions (like any radio station or nightclub), Webcasters will also be required to pay a new and unique license fee to the owners of the copyrights on the sound recordings themselves.

Those sound recording fees are administered by an organization called SoundExchange, which, not coincidentally, is a division of the Recording Industry Association of America (RIAA), the record industry's lobbying and litigating trade organization. No longer content with handing out Gold record certifications and suing the Internet music firms, the RIAA has set itself up in the admirable business of collecting and disbursing this new Internet tax.

SoundExchange is establishing a statutory license rate for Webcasting of sound recordings. What the exact rate is, or will be, remains under negotiation at the time of this writing. The Copyright Office is administering arbitration that is expected to conclude by the time you read this.

There is a remote possibility that the statutory license will be thrown out, though that seems unlikely considering the influence wielded by the participants in this arbitration. In the meantime, Webcasters are urged to notify SoundExchange of their intent to comply with the licensing requirements and to send along a good-faith prepayment.

In addition, the ways in which music can be presented in a Webcast are severely restricted by the DMCA and the DPRA. Under the terms of the statutory license, Webcasters can provide only noninteractive programming that is not on demand or personalized. Moreover, Webcasters are not permitted to play more than three songs from any one album during a three-hour period or more than two songs from an album consecutively. Webcasters cannot preannounce upcoming songs or post their playlists in advance. Archive programs may not be less than five hours in duration, and looped programs may not not be less than three hours long.

WEB OF POSSIBILITIES

Does that sound like a headache for Webcasters? It's meant to be, and that's just the first page of the rule book. Drop by the RIAA Web site and view its Webcasting FAQ for the complete list of regulations. Note that if you're thinking about offering on-demand or interactive music services, the statutory license will not apply: interactive services still need to negotiate separate licenses with each record label or copyright holder.

Naturally, the reason for all this legislation and regulation is that Webcasting is expected to become a big business. It poses a potential threat to portions of the traditional music business, and the major record labels expect to be in control of the game. Upstart Internet music services, from MP3.com and Napster on down to kids Shoutcasting from their dorm rooms, are considered competition for the record labels' own online initiatives. Preemptive hampering or outright elimination of these competitors just makes good business sense.

My recommendation is that if you want to play other people's music, particularly music that is commercially released by major labels, then play by the rules and render unto Caesar. On the other hand, you can easily set up a Webcast that plays only your own music or secure agreements from your friends in unsigned or independent bands that will let you play their music with little or no restriction. Providing airtime to artists that do not (or cannot) get mainstream commercial exposure is where Webcasting's real value is realized.

Todd Souvignier *wrote* The Musician's Guide to the Internet, *2nd ed., (Hal Leonard Publishing) and is president of Exploit Systems, Inc. (www.exploitsystems.com).*

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RIAA Webcasting FAQ
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CRAMPED QUARTERS

Audio- (as well as video-) data reduction, also known as data compression, is one of the most important media technologies to come along in recent years. Many capabilities that you take for granted—streaming audio, fast music downloads, and DVD surround sound, to name a few—simply would not exist without the ability to reduce audio data to a fraction of its size while retaining most of its fidelity.

But many people have only a vague idea of how those key technologies operate. Other articles in EM have covered the how-tos of compression in various formats. This one examines the principles that underlie audio-data compression in order to help you get the most from the technology. When you know what's under the hood, you're in a better position to understand when using audio-data reduction is appropriate, what the impact will be on fidelity, and how to select the right data-reduction scheme for the application.

Before moving ahead, a disclaimer is in order: it won't be possible to discuss every single audio-data-reduction scheme that has been developed for commercial and noncommercial use, because too many exist. Fortunately, the principles covered apply to nearly all the schemes.

WHY AUDIO-DATA COMPRESSION?

The purpose of reducing audio data is to get a free lunch, so to speak. By collapsing the data in a song or track to a fraction of its original size, you can get more out of any transmission channel or storage medium. Here are just a few benefits audio-data compression has brought:

1. fast downloading and streaming of songs and albums from the Internet;
2. discrete surround sound in DVD;
3. compact optical media such as MiniDisc;
4. more channels of recording and playback in multitrack digital audio workstations (DAWs); and
5. vastly increased audio storage in CD-ROM and hard disk.

Audio-data compression is mostly used to fit some number of audio channels into a space in which they would never fit as linear pulse-code modulation (PCM) while preserving something more or less resembling CD quality.

ILLUSTRATION BY MIKE CRUZ

The technology of audio-data compression.

By Gary S. Hall

There's another way of looking at it, though. When you compress audio, you're actually enhancing the quality of what can be carried by a given channel. For example, games and other computer applications have long used monaural, 8-bit audio at a sampling rate of 22 kHz (equivalent to a bit rate of 176 Kbps, which is frequently used for MP3 downloads) for sound effects and music. The difference is that 8-bit mono at 22 kHz sounds awful; stereo MP3 at the same bit rate, however, is considered highly listenable by most folks. That is, you could have transmitted terrible audio at the same rate, but you didn't. Rather, you used data reduction technology to improve fidelity for this particular channel-data rate. Although the emphasis today is on fitting audio into small spaces and through thin pipes, the identical notions can be applied to deliver much higher fidelity audio from standard media such as CD.

One point to take home from the study of audio compression is that linear PCM is actually an inefficient way to encode audio. **Fig. 1** illustrates the differences between the frequency response of PCM and that of a typical audio signal. Even though most real-world audio signals have markedly less energy in the high frequencies than, say, completely flat white noise, PCM encodes all signals as though they were flat, which wastes a lot of bits.

LOSSY AND LOSSLESS

Any method that reduces the size of audio data can be referred to as data compression, but there is an important distinction to be made between methods that reduce data in such a way that it can be restored bit for bit—so-called *lossless compression*—and methods, known collectively as *lossy compression*, that allow the essence of the sound to be restored but don't preserve the precise bits. The latter includes most familiar forms of audio-data compression, including MP3, Dolby's Audio Coding 3 (AC-3), and Sony's Adaptive Transform Audio Coding (ATRAC) compression (used in MiniDisc). Lossy compression schemes are useful because they can reduce data size quite a bit more than lossless techniques. Experience shows that the results can be more than acceptable for most listeners—not perfect, but good enough.

By definition, lossless compression schemes, including Meridian Lossless Packing (MLP) used in DVD-Audio, must restore every bit of the original uncompressed audio data—the audio equivalent of a Zip or StuffIt archive. No one in his or her right mind would compress a file of important text or numeric data in such a manner that the data would come back approximately right. It's either all correct or useless. Lossless audio compression is the same. If a compression method is truly lossless, you can be confident it will not corrupt a single bit.

Questions about fidelity that apply to lossy methods such as MP3 do not apply to lossless compression methods. The trade-off is that lossless techniques cannot achieve nearly as much compression as those that intentionally eliminate information.

Because the word *compression* is usually applied to lossy and lossless techniques, confusion can occur. In this article, I will use the term *data packing* to describe lossless audio-data compression schemes; the familiar *compression* will be used only in reference to lossy methods.

A third class of compression techniques sits midway between data packing and lossy compression. It can be described as *nearly lossless* because the audio data is not necessarily returned with perfect bit accuracy upon decompression; only slight deviation is tolerated, however, and fidelity remains transparent, even for trained listeners in controlled listening tests. Notable among compression standards that claim that status is Digital Theater Systems' Coherent Acoustics (more commonly called DTS Digital Surround or simply DTS), which is used increasingly in 5.1- and 6.1-surround tracks on DVD. The DTS stream occupies about four times the bandwidth of Dolby AC-3, and many listeners prefer its audio quality.

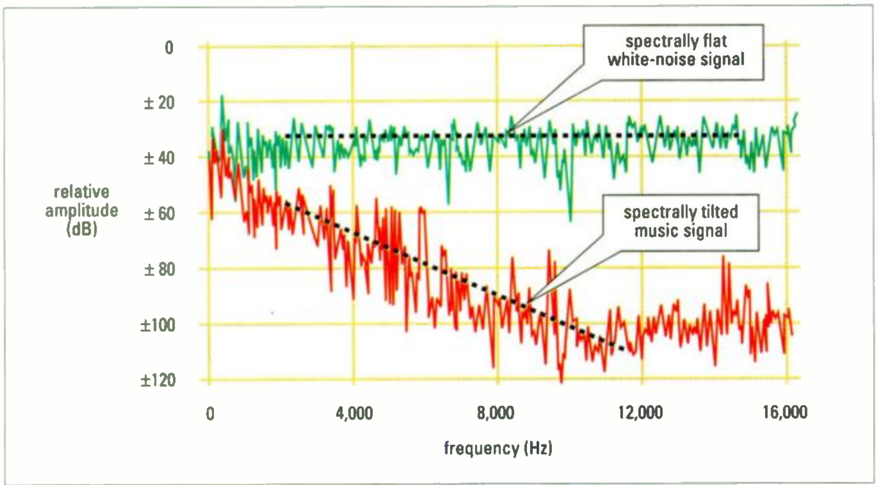

FIG. 1: Linear PCM is ideal for coding signals at high levels across the frequency band. But because music is not evenly distributed across the frequency spectrum, PCM is inherently inefficient at coding real-world audio.

GENERAL TECHNIQUES

Data packing, near-lossless compression, and lossy compression share many of the same techniques. The exception is masking-based perceptual coding, which is inherently lossy and used to secure the much higher compression ratios characteristic of lossy schemes such as MP3.

Fig. 2 shows a block diagram of a

somewhat generic coder that relies on techniques typical of lossless coding schemes, without the use of perceptual coding. The process illustrated can be made truly lossless, provided that individual elements in the chain are designed appropriately.

FRAMING AND FILTERING

In any audio coder, incoming PCM data is initially stored in a data buffer. The buffer allows for analysis of data across time. Generally speaking, audio coders process data in blocks, or frames, of a few hundred to a few thousand samples. The exact size of each block depends on the particular compression scheme as well as the sampling rate of incoming data, the target bit rate, and the characteristics of the incoming signal.

Larger block sizes permit more efficient coding of steady-state tonal signals but can result in audio artifacts when transients are encountered. Recognizing that different frame sizes are optimal for different types of signals, some compression schemes change the frame size on the fly, using smaller frames when transients are present and reverting to larger frames when the audio is holding more constant. Frame sizes of 256 to 4,096 samples are employed in common coding schemes, with 1,024 something of a standard default.

In most coding schemes, the first major step in processing is the division of each block of incoming data into some number of subbands using a digital filter bank. The number of bands can vary. Some well-known compression schemes use 32 subbands, but others use fewer.

Different coding schemes employ different filter designs. The most common designs are polyphase, Modified Discrete Cosine Transform (MDCT), and hybrid filters, which are a combination of polyphase and MDCT. Filter selection is based on trade-offs of processing efficiency, sharpness (which affects the amount of data reduction that can be achieved), and accuracy of reconstruction. If true lossless coding is required, the filter designs must be accurate.

The effect of this filtering is to divide the samples in the original source frame into smaller numbers of samples in "bins" representing each subband. Thus, if the frame size is 1,024 samples and 32 subbands are used, 32 samples will be used to represent audio in each subband (see **Fig. 3**). If the filters are designed appropriately, the 32 bins of 32 samples can be used to reconstruct the original block of 1,024 samples with perfect accuracy.

After it has been divided into subbands, audio can be processed with a much higher degree of sophistication—that is, with accurate gain ranging, prediction, and psychoacoustic analysis for each band.

SCALING

The amount of total energy in each subband varies. Typically, audio energy is centered in a few bands, with other bands showing much less signal. For each band, a scaling factor can be derived. The signal in that band can then be normalized to full scale for processing and then scaled back to the original level in decoding. Over time these block-by-block scale factors describe the amplitude envelope of the signal in all of the subbands, much in the way a classic vocoder extracts the changing levels of different bands in a program signal.

A RIGOROUS TEST

For a fascinating view of how the process of testing audio-data compression schemes is pursued at the high end, check out www.tnt.uni-hannover.de/project/mpeg/audio/public/w2006.pdf. The PDF is a detailed account of the formal evaluation of MPEG AAC (Advance Audio Coding) versus MPEG-1, Audio Layer 2 and MPEG-1, Audio Layer 3 (MP3) by the International Standards Organization (ISO).

The ISO is not known for doing things halfway, and the process of qualifying AAC as a standard proved no exception. This compression scheme, after all, represents the collaborative effort of nearly everyone in the business to create the be-all and end-all of lossy compression schemes, with a goal of delivering audio from the original at bit rates of 128 kbps.

The tests were conducted by the BBC, NHK, and MIT Media Labs, with participation by a number of other interested ISO members, including Deutsche Telekom, AT&T, and Scientific Atlanta. A number of audio samples were selected for the test, including stereo music clips and individual, "difficult" sources such as castanets, harpsichord, and pitch pipe. These were compressed using the three profiles of AAC coding (Main, Low Complexity, and SST) as well as with MPEG Layer 2 and MPEG Layer 3, at various bit rates. Care was taken to make sure the compressed samples represented the actual compression scheme, with third parties called on to verify that the test samples matched bit for bit the results of compression using standalone software.

For the test sessions, the selected samples were assembled onto DAT tapes following triple stimulus, double-blind, hidden reference methodology. That means that, for each test, three versions of the sample were presented to the listeners. The first was always the original, uncompressed PCM signal. Of the two following signals, one was a compressed version and the other was the original reference again, with no one but the preparers of the tapes (not

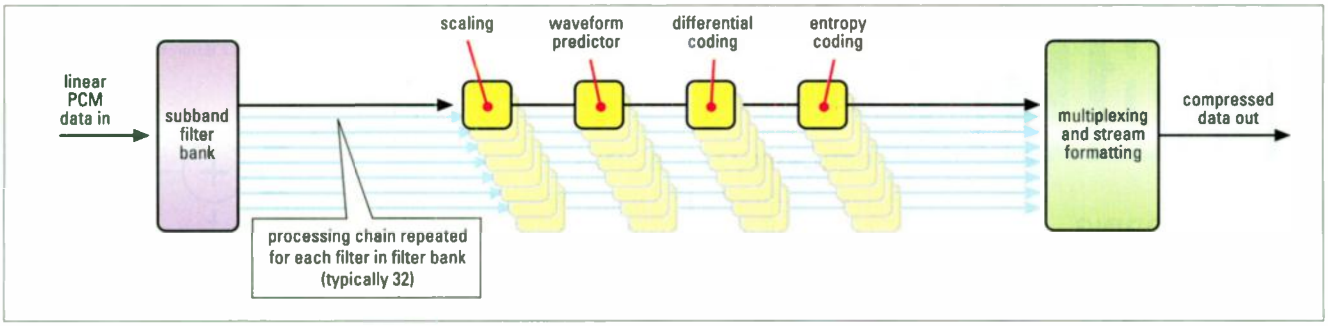

FIG. 2: Most lossless encoding schemes divide the spectrum into some number of subbands, which are then encoded using waveform prediction with differential coding. The results can be further reduced in size using table-based entropy coding and the outputs for all subbands multiplexed to create a single data stream.

WAVEFORM PREDICTION

In the next step of processing, the data in each band are analyzed for correlations, which are used to predict upcoming samples. The notion of waveform prediction is that, based on analysis of the current and preceding audio, one can make a reasonable prediction of the shape of the audio wave to come.

With a good prediction filter, the core characteristics of a correlated audio signal can be reduced to a few coefficients. When applied to band-limited signals, such as the output of the subband filter bank, the process can be effective.

Most of the time, the prediction process will not describe the exact waveform that occurs. To benefit from the prediction process, however, it's only necessary to approximate the signal so that the true signal can be described by its difference from the predicted values.

As part of the prediction process, the amount of correlation in the source signal is measured. That becomes important in the next phase of processing.

DIFFERENTIAL CODING

In the next coding stage, adaptive differential coding, the representation of the waveform in a given band is converted from absolute level to the difference in level between an expected value and the value that actually occurs. That is a process of simple subtraction between the predicted and the actual value (see **Fig. 4**).

Adaptive differential coding of that kind is only as good as the prediction used. As noted previously, prediction works just with correlated audio signals. Uncorrelated noise and transients do not benefit from the process at all. This is where the measure of correlation comes in. If the degree of correlation in the signal is too low, then a different signal is measured between the current sample and the prior sample. That yields some advantage in coding efficiency, though not nearly as much as adaptive coding with a well-correlated input.

the folks running the test session) knowing which was which or what compression scheme had been used. The inclusion of the hidden reference made it possible to determine whether the listeners could hear actual differences or not.

A test group of 31 individuals, all of whom were audio professionals, was assembled. Over a period of days, the test group was taken into a carefully prepared listening room and asked to evaluate a series of test samples. In each test, the listeners were asked to identify which signal was the hidden reference and to evaluate the other signal to a single decimal point on a 1-to-5 scale, with 5 representing absolute fidelity, 4 representing perceptible but not annoying, and so forth.

The process yielded thousands of responses, which later were subjected to careful analysis. First, it was necessary to determine whether the respondents had been able to distinguish the compressed signal from the hidden reference—particularly tricky, because at least some settings were expected to yield indistinguishable results. It was determined, however, that the pattern of response would indicate the validity of the tests. The evaluations of quality were then assessed for averages and spreads (range of responses by different individuals) and for distinct signal types and compression schemes.

When the final results (which are included in the ISO report) were in, the ISO concluded that AAC at 128 kbps was not completely transparent for all sources. Interestingly, a majority of respondents felt that Suzanne Vega sounded better after being coded with AAC at 128 kbps. More importantly, the test indicated that AAC at 128 kbps did, in fact, significantly outperform MP3 at the same rate and delivered performance equivalent with MPEG Layer 2 audio at 192 kbps. Those are impressive results that demonstrate that MPEG AAC is the state of the art in data compression for stereo signals.

CRAMPED QUARTERS

ENTROPY CODING

The principle of entropy coding is simple. In any data stream, some values appear more frequently than do others. For every possible value, an alternative value is extracted from a lookup table. Data values (called symbols in this context) that occur frequently are represented by shorter strings of bits, whereas those that happen only occasionally are represented by longer strings. The result is a stream of variable-length words rather than the original fixed-length PCM data. The average word length in this stream will be smaller than that of the original, so there is a net savings. Decoding is performed by the reverse method: looking up shorter codes to extract the exact 16-bit (or 20-bit, 24-bit, or whatever) value that occurred in the original stream.

The key is in selecting a lookup table that works well for the data at hand. Audio data has some characteristics (for example, more samples around zero than at the extremes) that make it amenable to this kind of coding. There are many possible symbol tables, and a "dictionary" of tables is defined by the particular compression scheme. During encoding, a table from the dictionary is selected to provide the largest amount of data reduction.

PERCEPTUAL CODING

Lossless and lossy coding methods can employ all of the techniques described in the previous section. The degree of precision and rigor applied may be quite different depending on the goal, but the principles remain the same.

Lossy compression methods also use a variety of techniques collectively known as *perceptual coding*. Those exploit the demonstrable principle that human beings cannot hear everything in an audio signal. Specifically, there are "masking" phenomena, found to be consistent for virtually all listeners, in which a signal of given frequency, perfectly audible by itself, cannot be heard when another, significantly louder component that is close in frequency is present.

In some descriptions of perceptual coding, the process is described as one of "identifying the components that cannot be heard and removing them," but that is a bit misleading. What actually occurs in the common processes is a band-specific reduction in bit resolution, based on a calculation of the amount of quantization noise that can be masked by a signal in that band. Bit resolution for a given band can be reduced to as low as 0 bits, effectively removing that band (though generally that occurs only when no signal is present in that band).

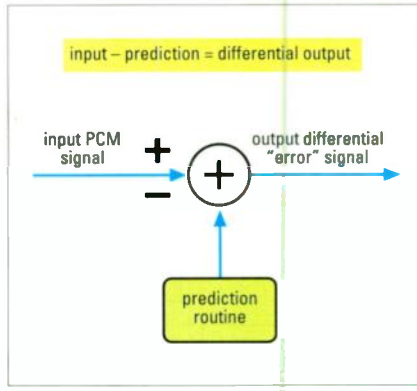

FIG. 4: In differential coding, a time-varying signal is represented by the difference between one sample and the next. In adaptive differential coding, the difference is taken between the actual sample that occurs and a sample predicted by analysis of the overall waveform.

MATRIXING AND GAIN

Stereo or multichannel-surround audio inevitably contains a lot of redundancy between channels. By identifying the content in common between channels, a substantial reduction in the amount of information to be coded for each channel can be achieved. Sum-and-difference matrices that are at the head of the audio-processing chain extract redundant information for more efficient coding.

Furthermore, every audio signal has an amplitude envelope. By extracting the envelope of the signal, the audio waveform's level can be normalized before processing. The envelope is preserved with the coded audio data and restored in decoding.

MASKING

The key to successful lossy compression is in analyzing the incoming signal to determine where resolution can be reduced. **Fig. 5** illustrates the essential principles of audio masking. **Fig. 6** shows a block diagram for an audio coder that incorporates bit allocation based on psychoacoustic analysis. Generally, that is done using a fast Fourier transform (FFT) that provides an accurate representation of tonal and nontonal energy by band.

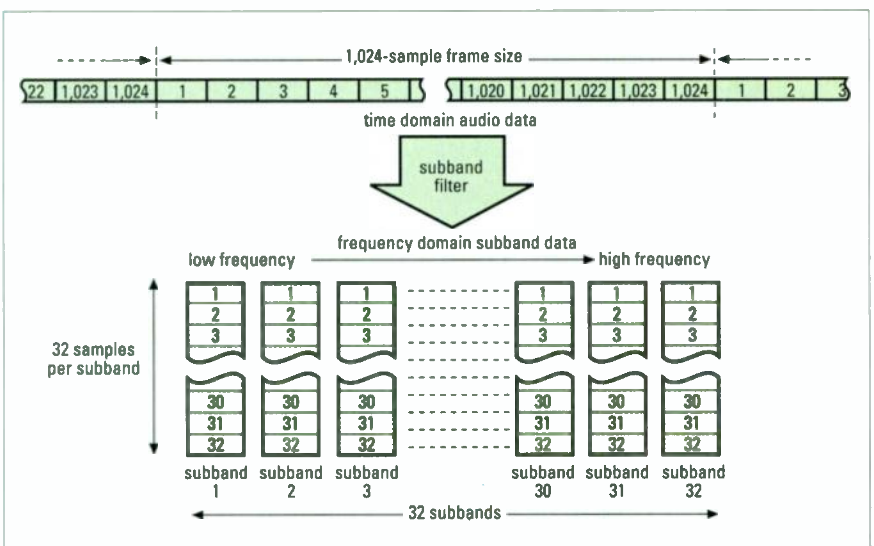

FIG. 3: This diagram illustrates framing and subband filtering of incoming uncompressed PCM audio.

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The results of the analysis are applied to a psychoacoustic model to determine the amount of masking in each band. In general, the amount of energy in a band defines a spreading function that determines masking. The more energy in a given band, the broader the range of frequencies it will mask. The spreads of the various bands overlap to create a composite masking curve (see **Fig. 7**).

The psychoacoustic models are based on large amounts of data gathered by researchers over many years. Most of that material is in the public domain, and the models used for all lossy methods draw from essentially the same pool of data. However, the models derived from this data can be more or less elaborate, and that is another area in which the quality of results can be affected by the complexity of the process. In MPEG audio, for example, two models are available. MP3 employs the second, more sophisticated of the two models. The exact ones that are used for proprietary processes such as AC-3 are confidential.

While researching this article, I found the following explanation of audio masking (reproduced here courtesy of Mattnet; www.mattnet.freeserve.co.uk): "Imagine you are in a room with some mice. If the room were completely silent and one of the mice were to fart, you may just about hear it. Now imagine there is a stick of dynamite in the room, and it explodes just as the mouse farts. The chances of you still hearing the fart are practically zero, owing to the fact that it has been drowned out by the . . . explosion." That's as good an explanation as any I've heard.

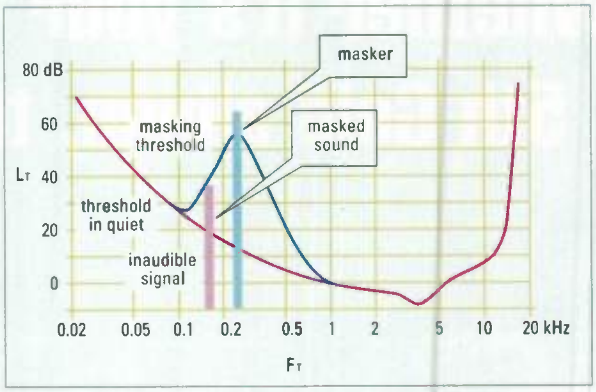

FIG. 5: Any strong audio component produces a masking "shadow" that renders weaker audio components, as well as quantization noise that is close in frequency to the masking component, inaudible.

VARIABLE QUANTIZATION

The results of psychoacoustic analysis are applied to select the bit resolution to be used for each subband. By looking at the energy and audibility threshold for each band, the encoder can determine the lowest bit resolution that can be applied in that band. If total energy in a band falls below what is determined to be audible, then that band may be deleted.

Thus, subband coding adds additional content in the form of quantization noise and removes bands that fall below audibility. The combination of those processes results in the overall compression ratio.

The highest performance in lossy compression is realized by combining the available techniques of lossless and lossy methods. **Fig. 8** shows a block diagram of a coder using all of the methods discussed so far.

JOINT STEREO

Localization of a signal falls off markedly as frequency increases, and that, too, can be exploited in lossy compression. If you've ever tried to pinpoint the source of a high-pitched whine, you may have an idea how this works—it's hard to tell where the whine is coming from.

By taking advantage of that phenomenon, you can mix the top end of both channels of a stereo pair together and code them as a single signal to be distributed between both channels on decoding. In some cases, the amplitude envelope of each channel is preserved, even though the signal beneath the envelope is the same for both channels.

CODING PERFORMANCE

You might wonder how developers test audio compression schemes. In the case of lossless packing, the testing is relatively straightforward: either the source signal is restored bit for bit or it's not.

For the lossy compression schemes that dominate the field, it's more difficult, especially as consumers gain experience and the stakes for performance are raised. By definition, all schemes increase noise and distortion compared with the original signal, making comparison by conventional audio measurements meaningless. Somehow one has to determine how successfully the changes from the original signal are being hidden from the listener.

FURTHER READING

Quite a few books on data compression are available; most discuss image and video compression, usually with shorter sections on audio. That is more useful than it appears, as there is substantial overlap in the techniques used.

The Data Compression Book, by Mark Nelson and Jean-Luc Gailly (Hungry Minds, Inc., 1995)

Data Compression: The Complete Reference, by David Salomon (Springer Verlag, 2000)

Digital Video and Audio Compression, by Stephen J. Solari (McGraw-Hill Profession Publishing, 1997)

Introduction to Data Compression, by Khalid Sayood (Morgan Kaufmann Publishers, 2000)

The MPEG Handbook: MPEG-1, MPEG-2, MPEG-4, by John Watkinson (Butterworth-Heinemann, 2001)

MPEG Video: Compression Standard, Joan L. Mitchell, editor (Kluwer Academic Publishers, 1996)

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In the end, the evaluation of a compression scheme (which typically includes comparing it with different, competing schemes) comes down to an elaborate game of "If it sounds good, do it." Listening tests are the only method that have been agreed upon as being useful for the evaluation of lossy coding schemes. (See the sidebar "A Rigorous Test" for an account of a series of tests that were conducted by the International Standards Organization.) Many questions remain, however, about the details of the methodology that should be used and about the validity of such tests. (**Fig. 9** shows a system that can be used for triple-blind listening tests.)

The fact is that the success or failure of audio-data reduction schemes in the market seems to proceed quite independently of any formal testing. MP3 became a major phenomenon not because it was shown to work well through testing, but because millions of users enjoyed it.

I have mentioned a number of data packing and data compression schemes. Here's a closer look at some of the most common possibilities. This is not an exhaustive list by any means, but these should be of interest to musicians. I've also included notes on their technology and application.

MERIDIAN AUDIO MLP

MLP has the distinction of being the only lossless packing scheme that has qualified as an official standard to date, having been selected by DVD Forum as the standard form of data reduction for use in DVD-Audio. It's no wonder.

It was one of those moments of brilliance.

Meridian has done an excellent job not only in developing the data packing algorithm but also in addressing needs of production and delivery in a real-world medium.

Meridian has been careful in qualifying MLP; the company recognizes that the amount of data reduction that can be achieved without loss will vary with the characteristics of the program. Data reduction ratios are specified as either "typical" or "minimum." Some variation in those figures is based on resolution, sampling rate, and channel configuration (generally speaking, greater reduction can be accomplished for multichannel surround because of redundancies between channels), but the figures for typical ratios hover in the area of 50 percent (2:1) data reduction. During compression, warnings are generated if the required compression ratio cannot be achieved losslessly, and it is then up to the operator to make any needed changes in the program.

The table, "High-Density Stereo and Multichannel Options," illustrates the application of MLP compression in DVD-Audio. DVD-A supports many options for the number of audio channels, bit resolution, and sampling rates. Some combinations of those exceed the total data transfer rate available for a standard DVD player, which is 9.8 Mbps. In those cases, MLP lossless coding is used to reduce the data rate to conform to the DVD standard.

MERGING TECHNOLOGIES LRC

A number of lossless audio-data packing schemes have been developed, but only a few have made it to the market. Merging Technologies' Lossless Realtime Coding (LRC) has been fully readied for license to provide compression and decompression programs for Mac, PC, and common digital signal processing (DSP) chips. Merging Technologies is focusing on offering the LRC technology for license to manufacturers of DAWs and other high-end audio systems that can benefit from lossless packing.

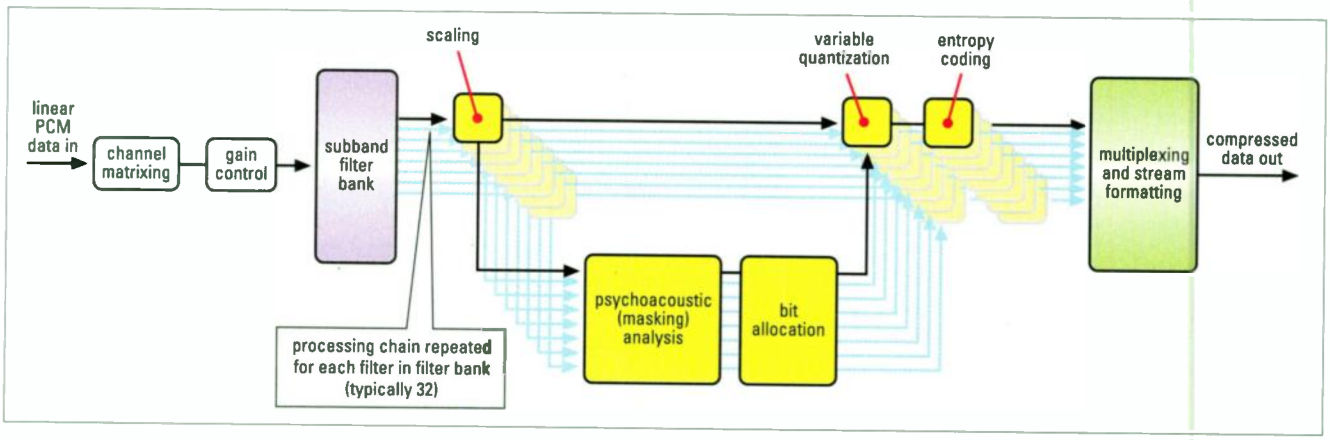

FIG. 6: Common lossy compression schemes, including MP3 and Dolby AC-3, rely on analysis of masking within and between the various bands' output from the initial subband filter bank. Channel matrixing, gain control, and entropy coding provide additional data reduction, but the largest leverage comes from perceptual coding.

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Merging Technologies states that LRC can reach a compression ratio of 3:1, depending on input signal. The charts provided in online documentation indicate that most real-world audio signals achieve a lossless ratio in the range of 2:1 to 2.5:1.

ROLAND R-DAC

Many higher-end integrated audio workstations specify that audio remain uncompressed to maintain optimal audio quality. Although it's true that linear PCM guarantees that what's recorded on disc will be reproduced accurately, it's misleading to claim that the exclusive use of unreduced audio data will always give the best performance or even the best audio quality. Lossless, or "virtually" lossless, data packing offers the opportunity to use higher-resolution, higher-sampling-rate audio in a multitrack hard-disk recorder. If the data packing technology holds up, there is no reason to presume that this approach will be in any way inferior to linear, uncompressed PCM.

So far Roland's VS-series workstations are the only systems in widespread use to take this approach. The high-end VS-2480 and VS-1880 workstations offer linear PCM recording as well as recording modes that employ the company's proprietary Roland Digital Audio Coding (R-DAC) technology. Using R-DAC, as many as 16 channels of audio at 24 bits and a 96 kHz sampling rate can be recorded in real time, with three times the recording time available from linear PCM at the same resolution.

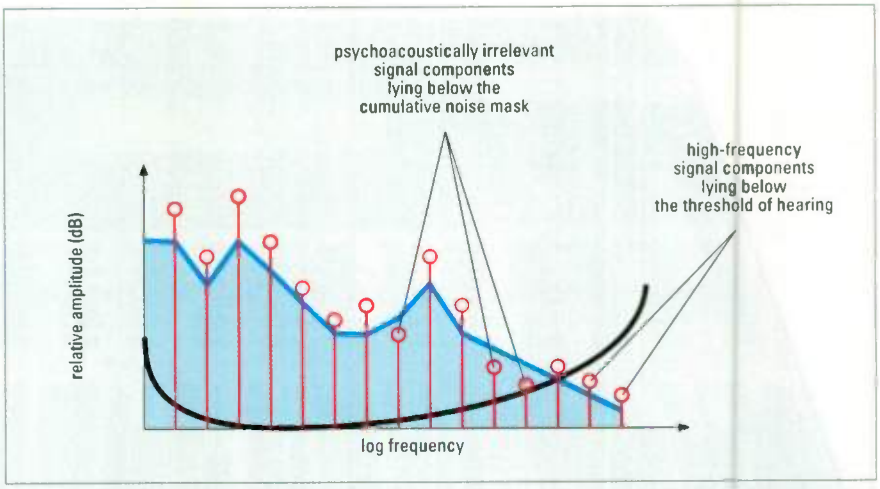


FIG. 7: The masking "shadows" of components present in each subband combine to form an overall masking curve that is used to determine the allocation of bits between the various bands.

MPEG AUDIO LAYERS

MPEG-1, Audio Layer 3 audio compression, commonly called MP3, is certainly the most famous audio compression scheme in the world, thanks to the phenomenon of Internet downloads and the near-apoplectic legal response from the recording industry. MPEG Layer 1 and Layer 2 are not as well known.

The scheme of audio layers came about when MPEG-1 for video was originally defined. At that time, there was a recognized need for audio coding to meet individual needs and to address the technology available at different points in time. The three layers basically represent a hierarchy of complexity and performance. Layer 1 required the least complexity (and had the lowest latency) but has effectively passed out of use. Layer 2, which is substantially more complex, has seen widespread use in Video CD, DVD, and broadcasting.

Layer 3 (MP3) took a much bigger step. The basic framework of the encoding algorithm for Layers 1 and 2 is preserved, but additional elements are added to allow for compression that is substantially more efficient. Likewise, MP3 takes significantly more digital processing to implement than Layer 2. It emerged in the late 1990s as the preferred method for audio coding of music because it could deliver quality acceptable to a broad range of listeners at considerably lower bit rates.

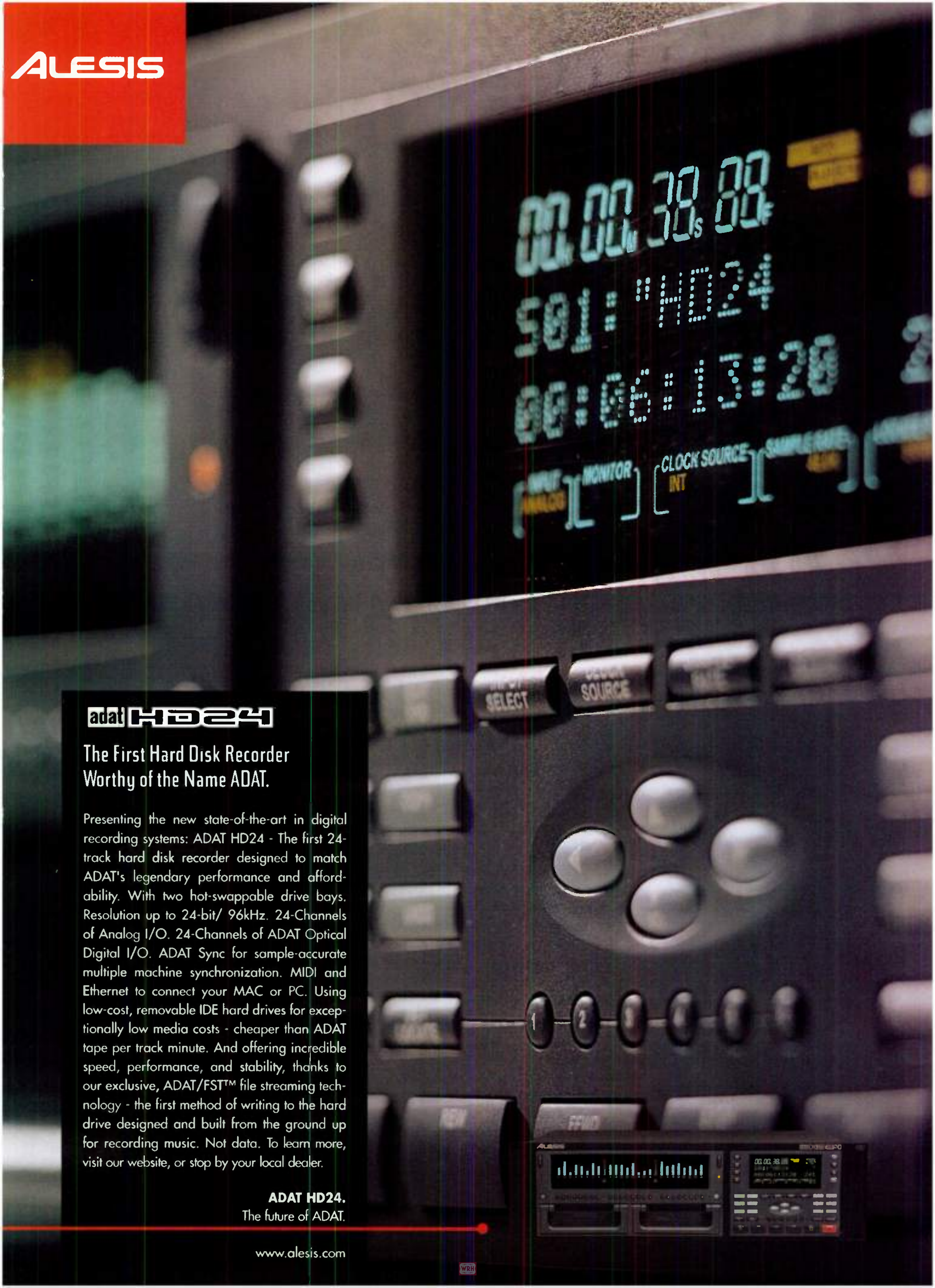
MPEG-2 AAC AND MPEG-4

Like Theodore Bikel's devil in the film *200 Motels*, Advanced Audio Coding (AAC) is known by many names. Originally, it was called MPEG-2 NBC Audio, with "NBC" standing for "nonbackward compatible." AAC is a collaborative effort aimed at creating a definitive standard for lossy audio coding—one that can deliver a lot better quality for a given

High-Density Stereo and Multichannel Options

DVD-Audio supports a broad range of channel configurations, bit depths, and sampling rates. MLP data packing is required for delivery of multichannel surround at higher densities. Even so, the highest sampling rates available in DVD-A (176.4 and 192 kHz) cannot be supported for multiple channels.

Audio Format	Data Transfer Rate (Mbps)	Supported in DVD-A
2 channels, 96 kHz, 24 bits	4.6	as uncompressed PCM
2 channels, 192 kHz, 24 bits	9.2	as uncompressed PCM
6 channels, 48 kHz, 24 bits	6.9	as uncompressed PCM
6 channels, 96 kHz, 24 bits	13.8	as MLP
6 channels, 192 kHz, 24 bits	27.6	no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00:06:13:20
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SOURCE
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bit rate than MP3 and, conversely, can provide acceptable audio performance at rates much lower than are required for the same perceived quality with MPEG-1 Layer 2 or Layer 3. Carefully controlled listening tests have demonstrated that those goals were achieved.

SONY ATRAC AND ATRAC3

Sony got an early start on mass-market application of audio-data compression when it released the MiniDisc format in 1992. As such, Sony was exposed to a fair amount of heat, in that consumer acceptance of audio perceptual coding had not yet been established. Also, the technology of lossy compression had not advanced to anything like its current state of development, particularly in the encoders that were built into early MiniDisc devices.

In the MiniDisc, ATRAC achieves a compression ratio of 5:1 over CD. This translates to about 280 kbps and is much higher than ratios routinely used with MP3, AAC, and others. Sony has since released an updated version of ATRAC, known as ATRAC3, that claims to deliver the same level of audio quality as MiniDisc ATRAC at bit rates of 128 kbps, making it approximately equivalent to AAC.

DOLBY AC-3 (DOLBY DIGITAL)

After MP3, Dolby's AC-3 scheme is probably the best known audio compression scheme, thanks to its position as the de facto standard for DVD production. Whereas AC-3 can be (and is) applied to 2-channel audio, its roots lie in multichannel surround for theatrical playback and HDTV. The predecessors of AC-3 (AC-1 and AC-2) were methods for coding two channels of audio. For HDTV transmissions with surround audio, it was assumed that the source audio would be matrix coded in the manner of Dolby Pro Surround before being digitally compressed. The decoder would then decode two audio channels and dematrix them to yield surround sound with the same kind of performance you had before DVD.

Somewhere along the way, it became clear that audio compression technology offered an opportunity to do something much better by coding all of the channels of a 5.1 field. By taking advantage of interchannel redundancies, the bit rate for complete discrete surround, with total separation and full bandwidth on every channel, would not be a lot more than that needed for stereo. (The original target for 5.1 coding was 320 kbps, corresponding to the amount of bandwidth available within HDTV.)

It was one of those moments of brilliance you will have reason to be thankful for for many years to come. One can find fault with the fidelity of 5.1 AC-3, but there is no doubt that

WEB LINKS

There are a large number of sites and links for audio-data compression. Many of the best are maintained by companies and institutes directly involved in developing and marketing the various competing schemes. The following links take you past the home page to the parts most directly concerned with audio compression technology.

Dolby Laboratories
www.dolby.com/digital

DTS Professional Audio
www.dtsonline.com/proaudio/index.html

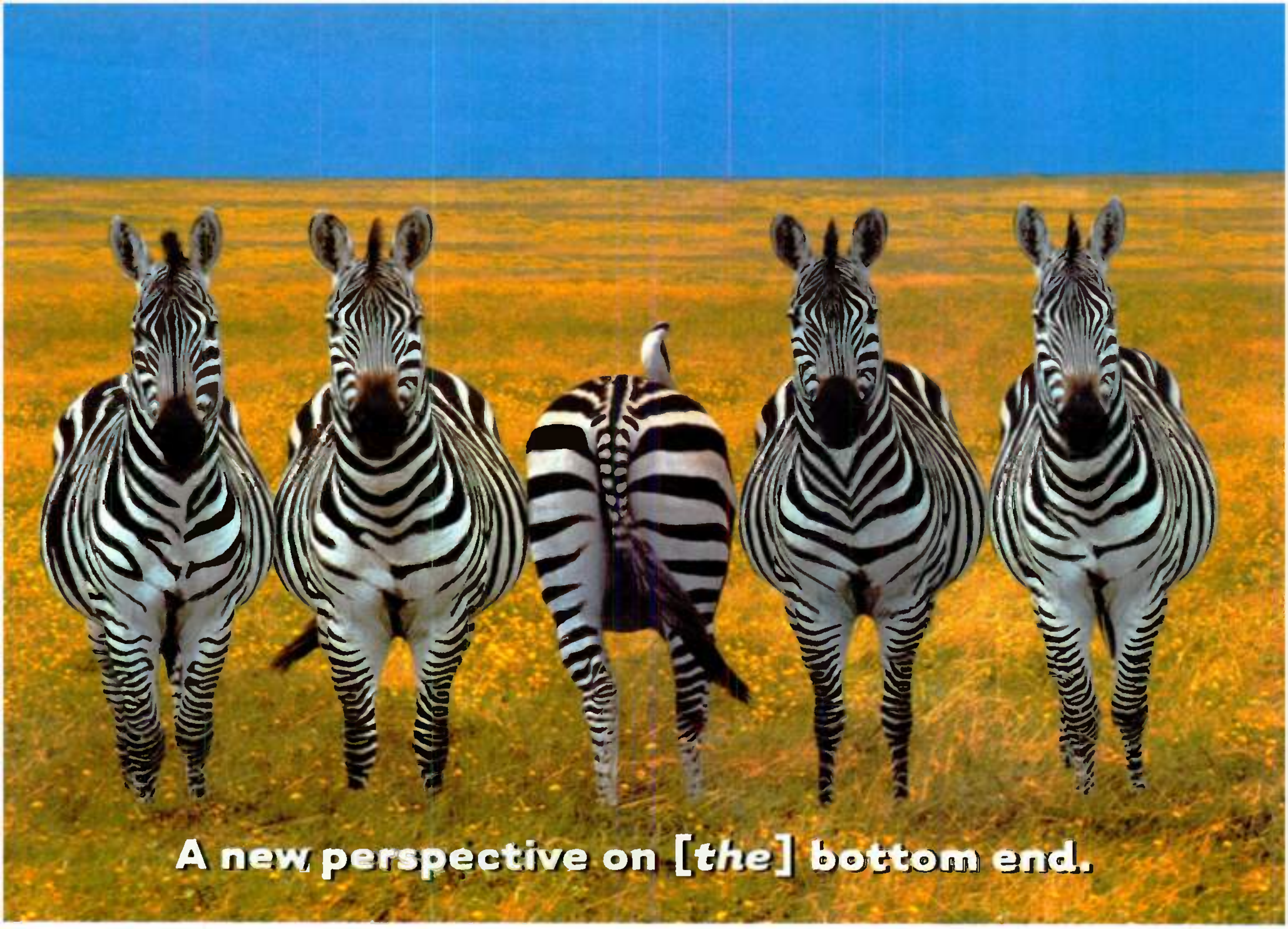
Fraunhofer Institute
www.iis.fhg.de/amm/techinf/index.html

Merging Technologies, Inc.
www.merging.com/products/lrc.htm

Meridian Audio
www.meridian-audio.com/m_mlp_in.htm

MPEG.org
www.mpeg.org/MPEG/audio.html and
www.mpeg.org/MPEG/mp3.html

Sony
www.sony.co.jp/en/Products/ATRAC3



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the aural impact far exceeds anything previously available; it opened new worlds of pleasure for movie viewers and exciting opportunities for composers and music producers.

DTS COHERENT ACOUSTICS

DTS, with roots in movie-theater sound that started with *Jurassic Park* in 1993, actually markets two audio-data reduction technologies. The apt-X100 scheme, used exclusively in the company's theatrical systems business, employs a combination of linear prediction and adaptive quantization to deliver data reduction ratios of 4:1 with effectively lossless results, according to the company's literature.

DTS's offering for use in consumer DVD, CD, and Laserdisc is officially called Coherent Acoustics; it is represented on playback devices and discs as DTS Digital Surround. That is a more flexible algorithm than apt-X100 (say that three times quickly), which is designed for scalability for data rates from 32 kbps up and input sources as high as 24-bit resolution at a 96 kHz sampling rate.

Coherent Acoustics uses a combination of technologies; whether it qualifies as a perceptual coding system depends on the data and ratio as compared to the original source. At the bit rates recommended for use in DVD, the company states that perceptual coding techniques are not used, but it doesn't claim complete bit-for-bit restoration of the audio source. In this application, then, Coherent Acoustics qualifies for the select category of virtually lossless data reduction schemes, warranted to be fully transparent to the listener, but not fully lossless at the data level.

Coherent Acoustics is best known for its use in high-end surround for DVD. The company states that the original impetus for developments was a drive to deliver superior sound within the bandwidth of CD and Laserdisc, potentially packing higher-resolution, higher-sampling-rate data within the limits of the established media. The combination of market opportunity in surround sound, united with a relative absence of consumer interest in ultrafidelity stereo, has led the company to emphasize its surround-sound applications.

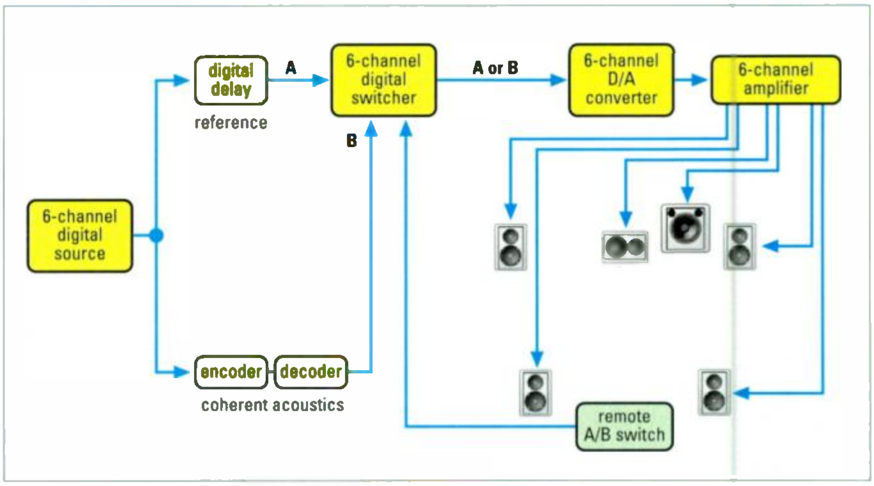

FIG. 9: Shown is a system for subjective evaluation of a multichannel audio encode-decode chain. This setup may be used for triple-blind testing or under control of the listener.

Gary S. Hall *lives and works in Alameda, California. His current interests include effects design for surround audio, DVD technology, meditation, travel, and recording. He's working with collaborators in Brazil, Switzerland, and upstate New York on a surround techno-tribal album for release on DVD.*

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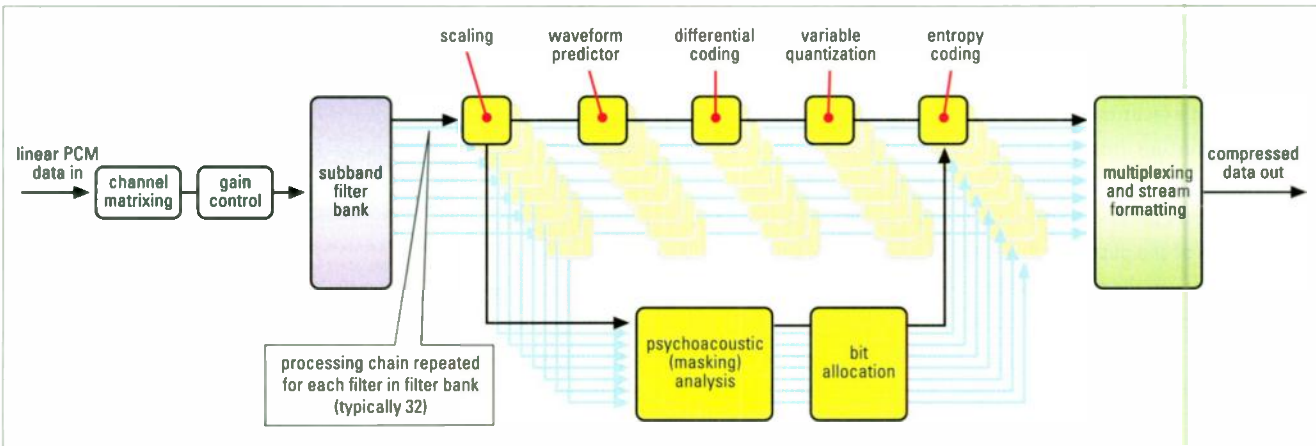


FIG. 8: The most sophisticated audio-coding algorithms rely on a combination of techniques to derive the most efficient compression of signal. Different elements of processing may be switched on or off depending on the target compression ratio and the character of the source signal.

ILLUSTRATION BY ROBERT SAUBER

Generating Good KARMA

Master your Karma with instruction from the master Karma mechanic.

BY STEPHEN KAY

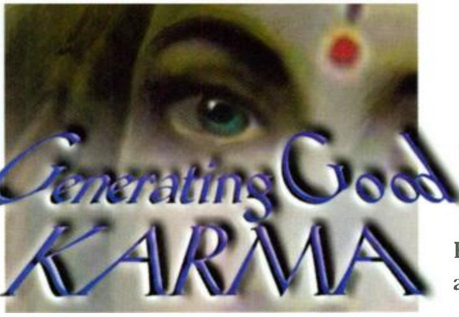
Korg Karma owners are well aware of the risks of possessing a keyboard that mates the Korg Triton's sound engine with an advanced algorithmic music-generation system. Press a Chord Trigger button one last time on your way to bed, and you could end up in an unexpected all-night jam session inspired by cascades of interactively generated notes and grooves.

The Karma's onboard Variable Performance Modeler will modulate, arpeggiate, interpolate, improvise, randomize, repeat, echo, transpose, strum, pick, slice, dice, and nearly make you coffee from the notes you play. Getting a handle on all of that power and complexity, though, can be daunting.

Because the Karma has the same sound engine and sequencer as the Korg Triton, you can apply most of the tips and tricks discussed in EM's "Master Class: Taming the Triton" (June 2001; text available online at www.emusician.com). So rather than discuss sequencer basics and program editing, I'll concentrate on using (and perhaps abusing) the Kay Algorithmic Real-time Music Architecture (KARMA) function (see **Fig. 1**).

THANKS IN ADVANCE

Manual Advance is a powerful KARMA feature that wasn't used much in the factory voicing, though it received prominent exposure in some online demos and marketing materials. If you're not sure what I'm talking about, try Program E004 Spanish Gtr C6->, Scene 1. Play a chord with one hand near middle C (an Input Chord) and advance the guitar picking pattern by striking Trigger Notes in the top or bottom octave with the other hand. You can control the rhythm, Velocity, and number of simultaneous notes (strums) that are generated by the way you play the Trigger Notes. However, the chord you play and the internal processing of the algorithms will determine the actual pitches you hear. You can use your note-triggering technique in Combination mode to control multiple KARMA Modules at the same time, as demonstrated by Factory Combis B007 and A039.



Manual Advance is an additive feature that's fun to play, but except in those three examples, it remains relatively hidden. Nonetheless, you can apply Manual Advance to nearly any Generated Effect (GE) in the Karma; it just requires several steps to set it up correctly.

Try setting up Manual Advance with another guitar program, B069 Strato-Chime. When you first select the program and press one of the Chord Trigger buttons, you hear a clean guitar with a picking pattern. To turn Strato-Chime into a Manual Advance–controlled program, it is necessary that you provide a zone (or zones) on the keyboard for the Trigger Notes. Assign the keyboard's bottom and top octaves to serve as Trigger Notes so that you can control the Manual Advance comfortably with either hand.

Go to Edit Page 6.1-2a [KARM] KeyZ/T and set Key Zone Bottom to C3 and Key Zone Top to B5 (see **Fig. 2**, top). If you use the cursor buttons to highlight the field, hold down the Enter key, and play the note on the keyboard, you can quickly assign the note to the field. That technique also works in a lot of other places on the various edit screens—anywhere you want to enter a note number or Velocity value.

With KARMA turned on, play some notes in the top and bottom octaves and notice that nothing happens. Play a chord in the middle, though, and the picking pattern begins.

Disconnect the internal clock and set it to respond to Trigger Notes. On Page 6.2-2b [KMdl] Parm2, set Clock Advance mode to Dynamic MIDI (Dyn; see **Fig. 2**, middle). That disconnects the internal clock; you can check it by playing a chord in the center area, and the picking pattern will no longer play.

Use Dynamic MIDI to route the top and bottom octaves to the Trigger Notes. On Page 6.4-3 [K RT] DynMIDI, use row 4, which is set to Off; the other rows are already being used. Set Dyn4 Source to Notes Outside of Zone (Notes Out Z), meaning all of the notes that are not in the zone you set up previously—the bottom and top octaves. Set Dyn4 Destination to Clock Advance (see **Fig. 2**, bottom), play a chord in the middle of the keyboard, and then try the notes in the top and bottom octaves.

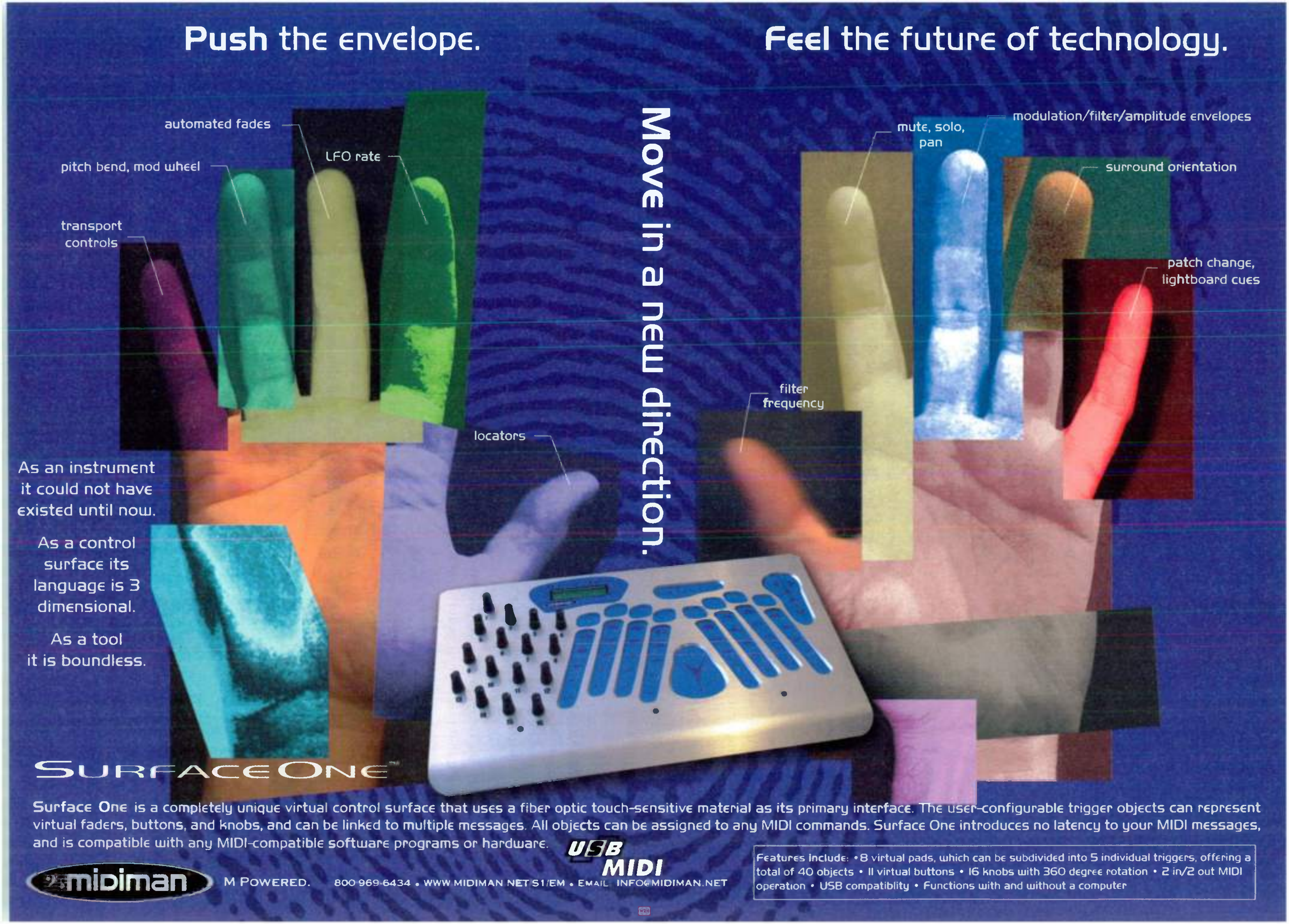
The basic Manual Advance effect should now be working, but experiment with a few other adjustments. Notice that when you play the Input Chord, no notes sound until you play some Trigger Notes in the top or bottom octaves. (The situation is similar to a guitarist changing the position of the fretting hand to prepare a chord for the picking action of the strumming hand.) Wouldn't it be more useful, though, if the chord were strummed when you played it?

FIG. 2: You can configure the Karma to perform the addictive Manual Advance feature with almost any Program.

Go back to 6.2-2b, set Clock Advance Chord mode (ChdMode) to 1st, and the whole chord will be strummed when you first play it. (Choosing 1st indicates that you want the first event of the Cluster Pattern to play, and in this case, a full chord is the first event. Other GEs might play only the first note, which is why other options are available for playing full chords.)

Play some Trigger Notes at different Velocities, hard and soft. You will hear no difference in the output volume of the notes. That's because Clock Advance Velocity Sensitivity Bottom (VelSensBtm) is set to 127. By varying that parameter, you can set the Velocity response to whatever value you like. Setting it to 001 results in maximum Velocity Sensitivity (1 to 127), and setting it to 064 results in half Velocity Sensitivity (64 to 127).

FIG. 1: A wealth of programming possibilities is waiting within the Korg Karma's algorithmic architecture.



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After that step, experiment with the KARMA Real-Time Controls (K.RTC). Twisting knob 5 all the way to the right greatly expands the picking pattern's octave range. Knob 7 controls the tightness of the initial strum you hear when you play the Input Chords, and switch 1 adds an interesting Melodic Repeat effect. Knob 3 can take the pattern from long sustained notes to short plucky muted notes, and knob 6 makes it possible to occasionally hear full strums with each single Trigger Note.

For further experimentation, set Clock Advance mode to Auto + Dyn1, which provides automatic advancement at the same time as Trigger Note advancement. Auto + Dyn2 lets you trigger automatic advancement with the Input Chord; the first Trigger Note stops it, and you continue advancing manually.

DOING IT MANUALLY

You can also use Manual Advance to do some interesting things with filter and effects control. Program E057 Power Saw uses GE 0919 Dr. Chopper 3/E57, which is a Generated-Gated GE type, meaning that rather than generating repeated notes, it generates Control Change (CC) values to "chop up" a sustained sound. When you turn KARMA on and press one of the Chord

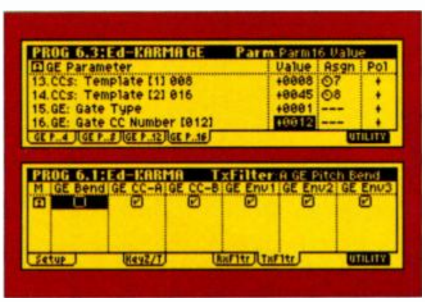


FIG. 3: You can remove the "chop" from a pad chopper (top). If you want to uncheck GE Bend, filter stepping is done manually.

Trigger buttons, you'll hear a slice-and-dice gated effect on a sustained sawtooth pad. Notice the sample-and-hold-style filter modulation that's also generated by the GE—specifically, by CC-B, which is assigned to CC 16 (ribbon), which in turn is assigned internally in the Program to control filter cutoff frequency.

To control the filter stepping with Trigger Notes, apply the same steps as in the previous example. Set the KARMA Module's Key Zone to provide a trigger area, but this time use only the bottom octave by setting the Key Zone Bottom to C3 and leaving the top unchanged. Set Clock Advance mode to Dyn and Chord mode to 1st, and then configure Dynamic MIDI to route Notes Outside of Zone to Clock Advance. Leave the VelSensBtm at 127 so that you can concentrate on just the filter stepping.

You could end up in an unexpected all-night jam session.

Now you can play a chord with your right hand and control the filter stepping and chopping by playing the Trigger Notes with your left hand. Use KARMA knob 3 to dial in a length for each slice, and use knob 8 to change the sequence of steps that's applied to the filter (it's actually changing the CC 16 Pattern).

At this point, the GE is still chopping the sound each time you trigger it. To make it a sustaining pad and control only the filter stepping, go to Page 6.3-4 [K GE] GE P..16 and set GE Parameter 16 (Gate CC Number) to the value 12 instead of 11 (see **Fig. 3**, top). Controller 11 (Expression) has been doing the chopping, but because CC 12 is not attached to anything in the Program, changing the parameter value effectively disables it.

The pad will now sustain when you advance the filter, which also uncovers a cool Pitch-Bend effect that wasn't noticeable before. I like it, but if you want to get rid of it, go to Page 6.1-4 [KARM] TxFltr, where you can filter out various data that KARMA is generating. If you uncheck GE Bend, you have a straight pad with manually advanced filter stepping (see **Fig. 3**, bottom).

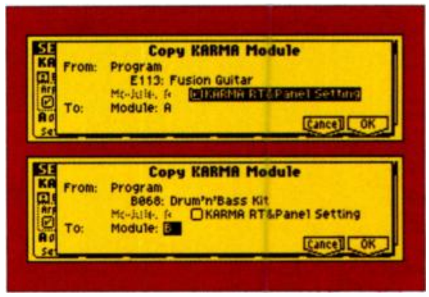
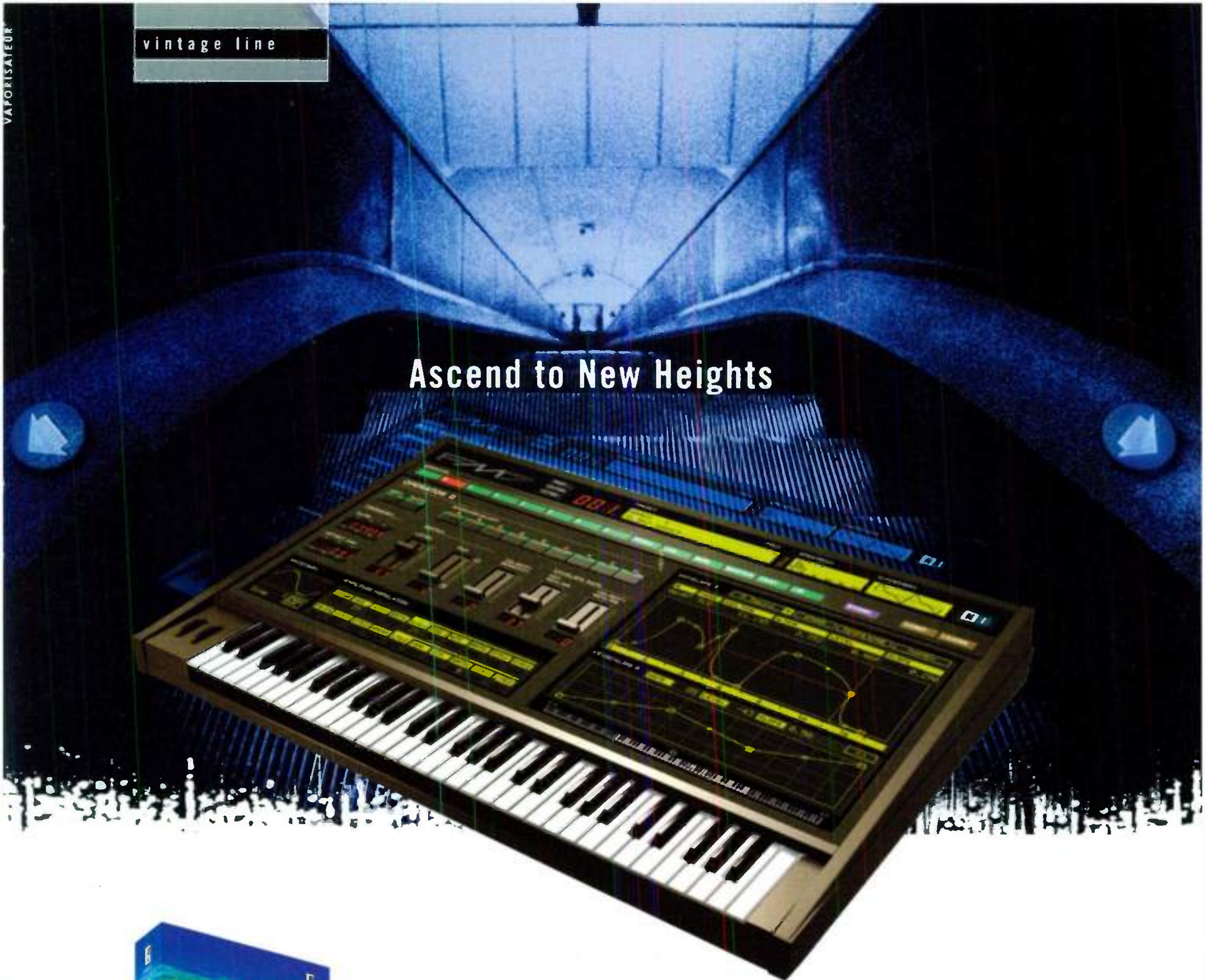


FIG. 4: This illustrates copying KARMA Module settings from a Program into Module A with RT&Panel Settings (top) and into Module B without RT&Panel Settings (bottom).

GETTING BENT

Program E113 Fusion Guitar produces convincing lead-guitar hammer-on bending techniques with the KARMA function. The Program uses GE 0250 RT Bender/E113, a real-time GE type, which means that the actual notes played on the keyboard are the starting point for various musical effects—in this case, an automatic Pitch-Bend effect in Scene 1 and a repeated-note effect in Scene 2. Because that Program is the only one in the factory voicing that uses GE 0250, you might think that's all there is to it. However, the GE was written to be a general-purpose automatic bender that's useful for a number of effects that aren't immediately obvious.

To use GE 0250 to create realistic ethnic bending effects, first select Program B095 Koto and copy the KARMA Function from E113. Go to Page 6.1-1 [KARM] Setup, press Utility, and select Copy KARMA Module. In the dialog that pops up, set the Program to E113; to save time, use the cursor buttons to get to the selection field and use the E bank key on the far right of the keyboard, followed by 1-1-3 and Enter on the ten-key pad. Then, use the cursor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buttons to checkmark the KARMA RT&Panel Setting checkbox to ensure that you copy all related settings such as Dynamic MIDI and the actual configuration of the real-time knobs and switches (see **Fig. 4**). Press OK.

Now adjust the GE to produce something other than a hammer-on, which doesn't really sound appropriate on the Koto. Press Exit F5 to jump to Page 1.4 [K.RTC], where you can see what the KARMA Controls are assigned to. Knob 2 is Bend Shape; turn it all the way to the left to select a Ramp shape for the bend rather than a Hammer. Now when you play the keyboard, instead of bending from the current pitch to the previous pitch and back again, it simply bends from the current pitch to the previous pitch and stays there. When you play a simple pentatonic scale, the bend will definitely sound more ethnic.

With the current settings, the GE bends from the note you play to the previous note. Switch 1 lets you change the Bend Direction; turn it off to bend from the previous note to the one that you're playing, which sounds a little more natural. The effect is almost like portamento, but more controllable. Use knob 1 to shorten or lengthen the Bend Length; about nine o'clock works well. You can use knob 5 (Bend Start %) to change the point at which the bend starts. If it's turned far left, the bend starts immediately; turning the knob to the right delays the bend so that you hear more of the original pitch before the bend starts. You can use knob 3, Bend Alternation, so that each note you play alternates between bending to or from the previous pitch.

This autobending technique works well on A123 Sitar, B083 Indian Stars, B099 Santur, and other programs. For sounds that have a long release (such as B083 Indian Stars), it will sound more realistic if you shorten the release using real-time knob 4A above the joystick, so that the release doesn't overlap the bending. That type of bending effect sounds most realistic when the intervals between the notes are a fifth or less.

CHAIN GANG

Another powerful feature of KARMA that received relatively little use in the factory programming is the ability to "chain" KARMA Modules so that one module triggers the start of another. You can set up cyclic triggering of different riffs on different timbres, drum grooves that alternate, instrumental phrases that trade off—you get the picture. Some Combis that employ this chaining feature in the factory presets include A022 4 Arp Cycle-Tsig, A107 4 Gate Cycle SW1, B012 4 Arp Cycle-Note, and E007 TheHarpist LH/RH.

FIG. 5: When copying IFX and MFX, make sure that you set the Bus Select for a Program in Sequencer mode the same way that it's set in Program mode.

In the first three examples, a single GE is used four times on four timbres, resulting in a cyclic tone-color change. In the case of TheHarpist, the same riff in a separate inversion allows some overlapping of the previous one, simulating overlapping hands moving up and down the harp.

You can use any of these four Combis as a study tool to determine ways to set up and use the ability to chain KARMA modules, a technique that I call *trigger by percentage*. It can be tricky to set up correctly, because it involves parameter changes on several screens, including the triggering options. You want to trigger only the first GE from the keyboard; the others are then triggered by various percentages of completion of the other GEs. To set up four GEs that trigger each other in succession and loop continuously, you need to make the settings listed in the table, "Trigger by Percentage."

Using that setup, the keyboard triggers Module A's notes and envelopes. In

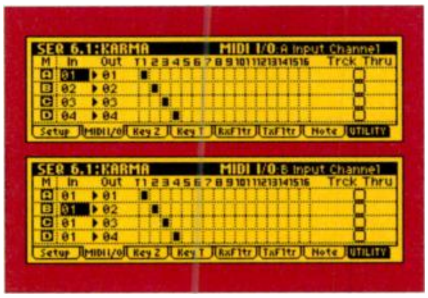

FIG. 6: Set the MIDI I/O matrix to allow the separate use of different GEs on each track (top) or layered triggering from a single track (bottom).

KARMA RESOURCES

Stephen Kay's KARMA Technology site (Karma Lab)
www.karma-lab.com
The Karma Lab site contains useful documents and resources, including name definitions for various software sequencers; text format lists of all Programs, Combis, GEs, Drum Mappings, and more; audio demos, articles, and reviews; a KARMA FAQ and Tips and Tricks (updated regularly); and additional new sounds.

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P4 1.7GHz Windows 98SE (XP[†])	256Mb 133MHz SDRAM	40GB Seagate Softsonic ATA/100 7200rpm Drive (Partitioned for System & Audio)	Black 40x Speed CD-ROM	Cubase VST 5.1 Halion VST	M-Audio AudioPhile	32Mb ATI Rage Pro Card	✓
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P4 1.7GHz Windows 98SE (XP[†])	512Mb 400MHz RDRAM	20GB System 7200rpm 60GB Audio 7200rpm (Seagate Softsonic ATA/100)	Black 16/10/40 Speed CDRW & 5 blank CD's	Cubase VST 5.1	MOTU 2408 MKII Steinberg Midex 8	32Mb ATI Rage Pro Card	✓
P4 2.0GHz Windows 2000 (XP[†])	512Mb 400MHz RDRAM	80GB System 7200rpm 80GB Audio 7200rpm (Seagate Softsonic ATA/100)	Dual Plextor 24x10x40 CDR	Sonic Foundry Sound Forge	Lynx Two ISDN Adapter Ethernet Adapter	32Mb ATI Rage Pro Card	✓
P4 2.0GHz Windows 2000 (XP[†])	1024Mb 400MHz RDRAM	20GB System 7200rpm 80GB Audio 7200rpm (Seagate Softsonic ATA/100)	Plextor 24x10x40 & 5 blank CD's	Steinberg Nuendo	Nuendo 8 I/O + MIDI Nuendo 9652 PCI Carillon RTM1 Midex 3	Matrox G550 32Mb Dual Head	✓
P4 1.7GHz Windows 2000 (XP[†])	512Mb 400MHz RDRAM	18GB System Barracuda SCSI Dual 36GB Audio Barracuda (SCSI with Hot Swap Caddies)	Plextor 24x10x40 & 5 blank CD's	Digidesign Pro Tools TDM	Digidesign Pro Tools Mix24 System Digidesign 888/24	Matrox G550 32Mb Dual Head	✓
P4 1.7GHz Windows 98SE (XP[†])	512Mb 400MHz RDRAM	20GB Seagate Softsonic ATA/100 7200rpm Drive (Partitioned for System & Audio)	Black 16/10/40 Speed CDRW & 5 blank CD's	Sibelius GigaStudio 96 5x Siedlaczek Orchestral CD's	M-Audio AudioPhile	32Mb ATI Rage Pro Card	✓
P4 1.7GHz Windows 98SE (XP[†])	512Mb 400MHz RDRAM	20GB System 7200rpm 60GB Samples 7200rpm (Seagate Softsonic ATA/100)	Black 16/10/40 Speed CDRW & 5 blank CD's	GigaStudio 160 Sound Forge XP	M-Audio Delta 1010	32Mb ATI Rage Pro Card	✓
P4 1.7GHz Windows 98SE (XP[†])	512Mb 400MHz RDRAM	20GB Seagate Softsonic ATA/100 7200rpm Drive (Partitioned for System & Audio)	Black 40x Speed CD & ZIP250	Reaktor B4 Pro 52, Retro AS1, Tassman, Rebirth, Reason, Unity DS1	Mixtreme I/O, Midisport 8x8, Carillon RK8	Matrox G550 32Mb Dual Head	✓
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P4 2.0GHz Windows 98SE (XP[†])	512Mb 400MHz RDRAM	20GB System 7200rpm 80GB Audio 7200rpm (Seagate Softsonic ATA/100)	Plextor 24x10x40 & 5 blank CD's	N/A	N/A	Matrox G550 32Mb Dual Head	✓

† xp XP Pro available as a standard upgrade March 2002, add $79

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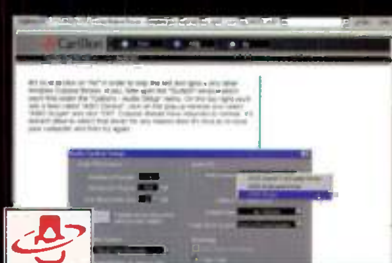

help

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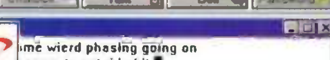

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Modules B, C, and D, the note and envelope triggers are set to Dyn and nothing is assigned in Dynamic MIDI, essentially disconnecting the keyboard. Module B's Trigger by Module (Trig By Mod) is set to A, with a Module percentage of 50. When Module A has completed half of its riff or pattern (as determined by the GE's time signature and Phase Pattern), it will trigger the start of Module B.

The Cutoff parameter determines whether the newly starting Module will cut off any other Modules that are playing at the time. If you want Modules to overlap, leave the Cutoff parameter turned off. In the preceding example, when Module B starts, you want it to shut off Module A so that the two Modules perform a handoff. Likewise, Module B triggers Module C and cuts off B, and Module C triggers Module D and cuts off C. To loop all four Modules repeatedly, Module A is triggered by Module D in addition to the keyboard and cuts off B, C, and D when it starts (the keyboard can trigger it at any time).

In the case of the Harp Combi, I didn't want it to loop continuously, so Trig by Mod for Module A is set to Off. Also, in the Harp Combi, only the first three GEs trigger each other. The fourth GE in Module D is a Melodic Repeat effect in the right hand, set to cut off Modules A, B, and C so that playing with the right hand silences what was started by the left.

ALTERNATE REALITY

You have a bit of the theory behind how Trigger by Percentage works, so now

FIG. 7: Set Module A (top) and Module B (bottom) Trigger Parameters to make them alternate with each other.

Trigger by Percentage

Set percentage to whatever works; you can assign the percentages to a real-time control knob and vary them in real time. Try knob 8 in the first three Combi examples or knob 2 in TheHarpist.

Page	6.2-2a	6.2-2c	6.2-2c	6.2-2c
Parameter	note and envelope triggers	Trig by Mod	Module percentage	cutoff module
Module A	Any, AKR, 1st	D	80	B, C, D
Module B	Dyn	A	50	A
Module C	Dyn	B	40	B
Module D	Dyn	C	75	C

I'll show you how to create your own real-world application. I want you to set up two drum kits (played by two KARMA GEs) that alternate with each other, with the alternation time controlled in real time by one of the KARMA knobs.

Begin by copying a Drum Program and all of its associated effects into the Sequencer. Karma OS version 2.0 (which should be available soon) allows you to do the following in a single step, using a new Copy from Program utility, but for now do it the old-fashioned way. Go to Sequencer mode and start with a new Song. Press F2 Prog..8 under the LCD and then set the Program for Track 1 to A052 Psycho Kit.

You need to copy the Insert and Master Effects settings, so go to Page 7.2-1 [IFX] Setup. Use the Utility menu to select Copy Insert Effects, and in the resulting dialog, select Program A052, checkmark All, and press OK.

Check the setting for the Program's Bus Select so that you can manually set it the same way. Switch to Program mode and Program A052, and go to Page 7.1-1 Bus. You can see that Bus Select is set to IFX1, so go back to Seq mode 7.1-1 [BUS] BUS8 and set Bus Select for Track 1 to IFX1 (see **Fig. 5**). Most Drum Programs use Use Drum Kit Setting (DKit), and most other programs use IFX1, but it doesn't hurt to check, as the example illustrates—I almost missed it.

Next, go to Page 7.3-1 [MFX] Setup. Use Utility to select Copy Master Effects and then select Program A052 again, checkmark All, and press OK.

You need to copy in the KARMA settings, so go to Page 6.1-1 [KARM] Setup. Use the Utility menu to select Copy KARMA Module, and in the resulting dialog, select Program A052 Psycho Kit. Make sure that KARMA RT&Panel Setting is checked as in the earlier example. Press OK. Set the Tempo to 134 (the same as the original Program) using the Tempo knob or the Tempo field on any of the upper-level Sequencer Pages. Turn KARMA on and give it a test.

Now add a second drum program and KARMA GE. You don't want to copy all of the IFX and MFX settings again; that would overwrite the current settings. The second drum kit should use the same FX settings as the first one.

Return to 1.1-2 Prog. 8 and set the Program for track 2 to B068 Drum'n'Bass Kit. Move to 7.1-1 BUS.8 and set Bus Select for track 2 to IFX1. Go to 6.1-1 [KARM] Setup and use the Utility menu to select Copy KARMA Module. This time set it to Program B068 Drum'n'Bass Kit and turn off KARMA RT&Panel Setting (you don't want to overwrite the settings you already copied for the first Module). Then, set the Destination to Module B and hit OK.

The result is that you now have a different drum GE on track 2, which you can play by setting Track Select 1.1-1a to track 2. If you turn KARMA on, you will hear a chopped, gated drumbeat. Then switch back to track 1 to hear the other drum groove. If you want to make them alternate with each other,

both need to be triggered by the same track.

You can find KARMA's MIDI I/O Routing matrix on page 6.1-2 [KARM] MIDI I/O. There you can see that Module A's Input Channel is 1 and Module B's is 2 (see **Fig. 6**, top). Change Module B to channel 1 (see **Fig. 6**, bottom) and set your Track Select back to Track 01 (if it isn't already). When you play the keyboard, you trigger both drum GEs at the same time, on top of each other. It sounds pretty cool, but file away that idea for another time; the goal is to make the drum grooves alternate.

Head for Page 6.2-2 [K.Mdl] Parm2 and set Module A Trig by Mod to B. For now set Module percentage to 50. Because you want Module B to stop when Module A is triggered, checkmark B for Cutoff (see **Fig. 7**, top).

Using F7, switch the display to Module B and enter the same settings in a complementary fashion. Set Module B Trig by Mod to A and Module percentage to 50, and checkmark A for Cutoff. Unlike Module B, set note and envelope triggers to Dyn, because you don't want the keyboard to trigger Module A (see **Fig. 7**, bottom).

Now turn KARMA On and hit a Chord Trigger button. Groove A will play Psycho Kit for two bars, switch to Groove B playing Drum'n'Bass Kit for two bars, and then loop back to Psycho Kit—cool!

FIG. 8: This illustrates removing a GE Parameter from KARMA knob 8 (top) and setting the knob to vary the alternation time between two GEs (bottom).

LOOP THE LOOP

Now that you have the grooves looping and alternating, set up a KARMA knob to control the alternation time. Because you previously copied in the KARMA RT&Panel settings, though, all of the knobs are already assigned to do something; you need to free up knob 8.

Go to Page 6.3-1 [K GE] GE P..4 and make sure you see Module A (use F6/F7). Nothing is assigned to knob 8 there, so go to the next tab, GE P..8. GE Parameter 7 is assigned to knob 8, so set it to Off (see **Fig. 8**, top). Check the other two tabs to ensure that nothing else is set to knob 8. Use F7 to switch the display to Module B and perform the same steps; locate anything assigned to knob 8 and turn it off (GE Parameter 11).

You now have a free knob, so next go to Page 6.4-1 [K RT] RT P..4. In row 1, set Group (Grp) to Trigger (Trig) and set Parameter to Module percentage; Min will automatically set to 0, and Max will set to 100. Checkmark Modules A and B so that you can vary the loop time for both at the same time. Value will automatically set itself to 50, the setting that you entered for Module percentage a few steps earlier (see **Fig. 8**, bottom). It's important that you understand that setting, because it represents the center of the knob. Whatever you enter will be the setting at the center position, and the rest of the range between Min and Max will be scaled accordingly. You can set up nonlinear scaling if you desire, but for now leave it at 50 to produce an even, linear scaling.

The last step is to set Assign (Asgn) to knob 8. Put knob 8 in the center position, and as before, the groove should trade off every two bars. Twist it all the way to the right, and the two grooves will alternate every four bars. Twist it all the way to the left, and they'll alternate in a machine-gun fashion every 16th note, with all sorts of weird time signatures between those settings.

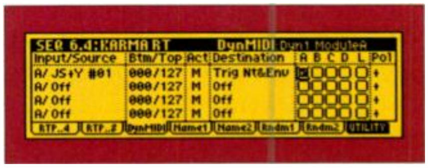

FIG. 9: Using the joystick to retrigger a drum groove is an effective and easy setting to make.

If the knob is slightly off center in its calibration (as mine was), you might need to adjust the value; a setting of 56 works fine. Also, try setting Min to 0, Max to 50, and Value to 12. Now center is a two-beat alternation and right is an eight-beat (two-bar) alternation, providing an example of a nonlinear scaling of the knob.

TRIGGER-HAPPY

For a handy and fun effect, assign the joystick to retrigger the drum groove. Go to Page 6.4-3 [K RT] DynMIDI and set Dyn1 Source to JS+Y #01. The destination should be Trigger Notes and Envelopes (Trig Nt&Env). Checkmark Module A and set Action (Act) to Momentary (M; see **Fig. 9**).

Now you can push the joystick forward as the drumbeat plays for stuttering, samplelike retriggerings of the whole groove, but playing on the keyboard does not retrigger the groove. Such a setup is especially useful if the keyboard is controlling another sound, such as other KARMA melodic GEs or a pad. Combi A000 =Voice Of KARMA= is a perfect example (though it uses the joystick in the downward direction, accomplished by setting Dyn Source to JS–Y #02).

I've only scratched the surface in this article, but hopefully you've grown more comfortable with configuring the KARMA function, copying modules and effects, and exploring some of the Korg Karma's more esoteric features. Just be careful with those Chord Trigger buttons around bedtime.

Stephen Kay *is the inventor of KARMA (Kay Algorithmic Real-time Music Architecture). He spends his free time dreaming up acronyms and the technology to go with them.*

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Virtual Cables

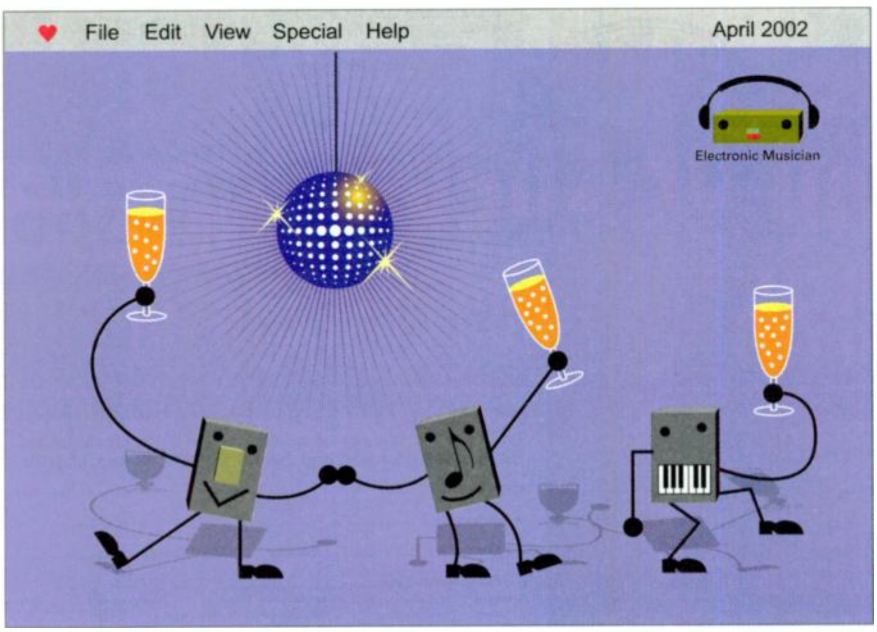
Learn how to connect programs by using software connections.

By Brian Smithers

At the heart of any good audio education is the study of signal flow and wiring: get your inputs and outputs straight, and you can adapt to just about any situation. Moreover, knowing how to trace the routes traveled by MIDI data and audio signals is critical to effective troubleshooting.

So what happens when most of your "gear" is actually software? What good is that stack of freshly soldered MIDI cables when you're trying to send timing signals from your digital audio workstation (DAW) to your virtual drum machine? What if you want to route the audio output of your software synthesizer to your sequencer so that you can apply a virtual gate to it? In that case, neither a balanced nor an unbalanced cable will do you much good.

What you need are *virtual cables:* those clever inventions that function just like real hardware cables, except that they're actually software that connects one program to another. Here's a closer look at how they work.



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VIRTUAL I/O

To get MIDI Timing Clock (or any MIDI message) from your sequencer to your software drum machine, you must use a utility commonly known as a *virtual MIDI cable.* The cable may appear as a function integrated into both programs, or it may be a separate application that shows up in the list of MIDI ports available to the two programs. Either way, it's a software connection—an *internal pathway*—that passes the beat-clock signal between the two programs in a manner similar to the way a physical MIDI cable passes a signal between two hardware devices.

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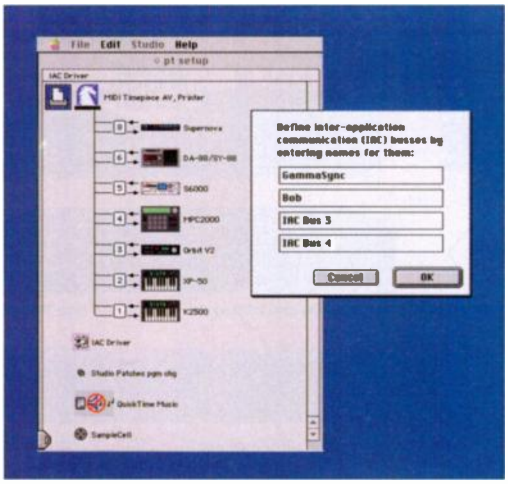

FIG. 1: OMS Setup lets you assign useful names to the four IAC buses. The buses appear as MIDI connections between applications.

in like fashion, except you use a *virtual audio cable* to make the connection. The concept resembles that of an audio *bus*, except that a bus is ordinarily restricted to making connections within a DAW's mixer. Like a physical patch cable, a virtual audio cable connects one software application's audio output with another's audio input.

Increasingly, virtual audio and MIDI connections are being integrated into host programs. When a soft synth is implemented as a plug-in or a virtual instrument, it appears as a prewired audio insert within your sequencer, and it then shows up as an available MIDI output device on a MIDI track. That's an ideal arrangement; however, not all software synths are available as plug-ins, and the synth you want may not be available in the plug-in format you need. In either case, you'll need a virtual cable.

Generally speaking, virtual cables come in two types: *interconnection standards* and *ancillary applications*. When two programs support the same interconnection standard, they see each other as I/O options. That's the next best thing to operating as a plug-in, because it's extremely convenient. If the "gear" you want to hook up doesn't work as a plug-in and doesn't support a common interconnection standard, you must use a separate application to emulate a physical cable. Although that is a bit less convenient, it is nevertheless a potentially powerful and useful tool for your virtual toolkit.

BUS RIDE

Let's take a closer look at the example I mentioned earlier and see how to lock a software drum machine to a DAW using MIDI Timing Clock. For this example, I'll use a Macintosh running Digidesign's Pro Tools LE and Koblo's Gamma9000. Because I'll be using Opcode's Open Music System (OMS) to organize the MIDI setup, I will make the MIDI connections through a virtual pathway that is called the *IAC bus*.

The IAC (Interapplication Communication) bus is a set of virtual MIDI cables built into OMS and therefore available to any application that supports OMS. (Mark of the Unicorn's [MOTU's] FreeMIDI offers similar functionality through its Interapplication MIDI.) It allows as many as four internal MIDI ports, each with the customary 16 MIDI channels.

If you double-click on the IAC bus icon in OMS Setup, you can rename the IAC ports (see **Fig. 1**). Take advantage of this feature to name your IAC ports something useful and meaningful. For this example, I'll use the mundane but informative name "GammaSync." Now the virtual cable has a label.

In order to get Gamma9000 listening to the IAC bus for MIDI Clock, you must first set its MIDI driver (found in the File menu under Select Drivers) to OMS, and then set its Doc Bus (found in the Options menu under MIDI Setup) to GammaSync as shown in **Fig. 2**. Simply click on the Sync button, and Gamma9000 waits for MIDI Timing Clock.

Sending clock pulses down the IAC bus from Pro Tools LE is as simple as checking "Enable MIDI Beat Clock" and then choosing GammaSync as the desired port. When you start playback in Pro Tools LE, Gamma9000 will be right in step. I will address virtual audio routing next.

AUDIO EGRESS

DirectConnect acts as a virtual audio cable between Pro Tools and other audio applications, such as software synths and drum machines. Other programs that support Digidesign hardware, such as MOTU Digital Performer and Emagic Logic Audio Platinum, can also use DirectConnect. As many as 32 independent audio channels are available, though in a host-based system such as Pro Tools LE, the computer's performance may be a limiting factor. In the window where you selected OMS as Gamma9000's MIDI driver, you must also select DirectConnect as Gamma-9000's audio driver so its audio outputs will show up in DirectConnect.

To set up a direct audio connection between Gamma9000 and Pro Tools LE, create an aux track in Pro Tools and assign the DirectConnect plug-in to an insert on the aux track. Don't look for DirectConnect to be listed alphabetically (under *D*) among the other plug-ins, though; it takes the name of the compatible program, so in this case, it will appear as Tokyo. (Gamma9000 is one of five synthesizers you can select from within Koblo's Tokyo, the parent application.) In the DirectConnect plug-in window (see **Fig. 3**), choose "Tokyo

FIG. 2: Enabling Koblo's Gamma9000 to receive MIDI Clock via the IAC bus is as simple as selecting the IAC bus as Gamma9000's Doc Bus. Check the Sync button (not shown), and Gamma9000 is ready to lock up.

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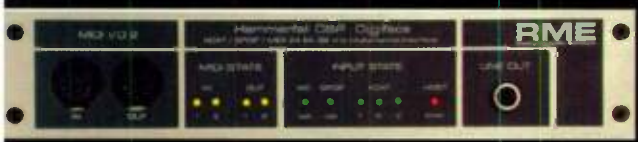

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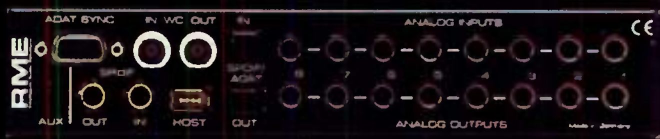

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Stereo Out 1-2" as the plug-in's output. If that seems backward, think of it as choosing the output (Gamma9000's) you're patching in.

At that point, when you start playback in Pro Tools LE, it sends MIDI Clock to Gamma9000 through the IAC bus; Gamma9000 follows Pro Tools LE's tempo to the letter and sends its audio output back to Pro Tools LE through DirectConnect. Pro Tools LE receives Gamma9000's audio like any other audio input assigned to an aux track and sends it to whichever output you have assigned.

You can now process Gamma9000's output with any audio plug-ins you have available. You can automate its volume, pan, and send levels as you would any other audio. You can even bounce the audio to disk with or without other audio tracks. In short, it now functions like any properly patched piece of outboard gear.

If you have trouble setting up DirectConnect, check to be certain that the DirectConnect plug-in and the external synth's or drum-machine's "plug-in description" file are in the Plug-Ins folder inside the DAE folder. Another file called Digidesign StreamManager must be in the Extensions folder. Those files should appear automatically during a normal installation, but they're worth knowing about if you need to do any troubleshooting. You may also run into DirectConnect-compatible soft synths that operate only in mono or only in stereo. If you have difficulty firing up a certain soft synth, you may simply be trying to use it on the wrong type of track. It's also a good idea to enable Active in Background from Pro Tools LE's Operations menu to keep the program from stopping whenever you bring Gamma9000 to the foreground.

Numerous soft synths, samplers, drum machines, and other applications support DirectConnect, including Native Instruments B4 and Reaktor, BitHeadz Unity DS-1 and Phrazer, and TC Works Spark XL. DirectConnect doesn't do Windows.

FIG. 3: Pro Tools' DirectConnect plug-in allows the audio output of another program to be routed directly into Pro Tools to be mixed, processed, and bounced like any piece of outboard gear.

GET WIRED

So how about running a virtual synth's output into your sequencer to apply

effects to it and to record it? One great solution is ReWire from Propellerhead. ReWire is a virtual audio pathway of the interconnection-standard variety. It provides up to 64 virtual pathways for real-time streaming of audio between supported applications. It also provides sample-accurate synchronization and transport integration between the programs.

With that many virtual audio cables at your fingertips, you can run the outputs of a program such as Propellerhead's ReBirth into separate audio inputs of Steinberg's Cubase VST and apply different effects along with mix automation. The setup process is essentially identical on Mac and PC. ReWire-enabled applications detect each other automatically, so all you have to do is route the audio exactly where you want it to go.

Fig. 4 shows Cubase VST's ReWire control panel with ReBirth's Mix outputs and individual instrument outputs available. Activate one and it pops up in Cubase VST's Mixer window. Any output can be enabled independently of the rest, though little is gained in system resources by leaving them disabled.

ReWire is supported by other host programs such as Steinberg Nuendo, Emagic Logic Audio, MOTU Digital Performer, and Propellerhead's own Reason. Other ReWire-compatible synth applications include BitHeadz Unity DS-1 and Retro AS-1, Cycling '74 Max/MSP, as well as Koblo Vibra9000, Gamma9000, and Stella9000. In addition, because ReWire supports multiple outputs, it could, for example, handle Reason and ReBirth simultaneously. Check at www.propellerheads.se/products/rewire/products.html for the latest information regarding ReWire support.

HELP ME, HUBI

Trying to get MIDI communication enabled between Windows applications often requires an extra step: the installation of a third-party virtual MIDI cable. The Windows platform lacks a unified

> **ReWire-enabled applications detect each other automatically.**

MIDI management tool, such as OMS or FreeMIDI, so interapplication functions must either be built into a program or facilitated by applications such as Hubert Winkler's Hubi's Loopback Device or Jamie O'Connell's MIDI Yoke, both of which are available as freeware. (Check www.synthzone.com for links.)

Interestingly, both programs are installed using the Windows Add New Hardware function. It makes sense when you consider that the operating system regards them as hardware devices. What may not make sense as quickly is that whatever data enters the "cable's" Output port gets sent to its Input port. If that seems backward to

you, you're in good company. But if you compare virtual MIDI cables to a real hardware setup, in which data flows out of a controller to the input of a synth, it becomes clearer.

In other words, choose Hubi's as the destination on one or more tracks in your sequencer (the "controller"), and Hubi's takes over the role of controller. Hubi's Output port, which is receiving the data from your sequencer, will then send that data to whatever is connected to its Input port, which will be your soft synth or drum machine. Simple, right? (**Fig. 5** shows the virtual cables as they appear in a program's MIDI setup menu.) Hubi's supports 4 loopback ports; MIDI Yoke offers 16. Both support multiple inputs and outputs to each port.

However, as in the Gamma9000 example, you're only halfway there at this point. To hear the output of your soft device, you may need a second sound card, a multichannel sound card, or a virtual audio cable, such as the appropriately named Virtual Audio Cable from Ntonyx. Offering as many as 64 multiclient cables with resolution as high as 32 bits, Virtual Audio Cable installs and operates much like Hubi's Loopback Device, but it has a couple of quirks worth noting.

First, the demo version uses a fixed interrupt period that, on my system, resulted in a significant amount of clicking in the audio stream. The full version lets you adjust the interrupt period, which should result in smoother performance. Second, Virtual Audio Cable requires a helper application so you can monitor the audio output. When I connected the audio output of VAZ Modular into Cakewalk Pro Audio using Virtual Audio Cable, I was able to record but not monitor the sound. A tiny applet called Audio Repeater is included with the Virtual Audio Cable distribution file. It allows you to monitor the signal while recording.

To put it all together, here's what's happening: the output of a Pro Audio MIDI track is assigned to Hubi's Loopback Device, which carries the data into VAZ Modular and triggers the selected synthesizer. The audio output of VAZ Modular is sent through Virtual Audio Cable simultaneously to Pro Audio for recording and to Audio Repeater for monitoring through the audio interface. As before, think of each element as a hardware device; you'd wire it exactly the same way and expect the same results.

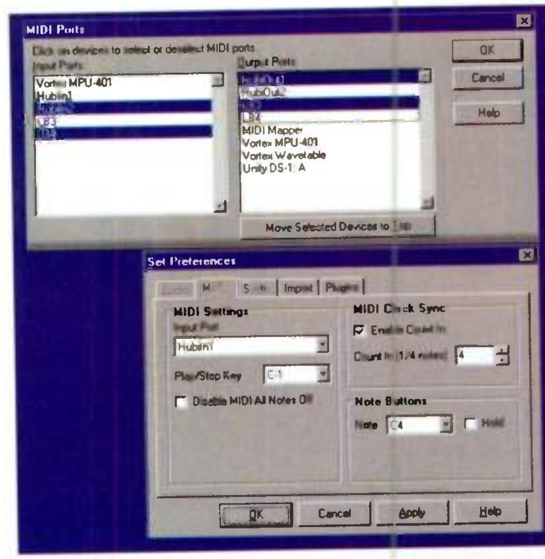

FIG. 5: This example shows Cakewalk Pro Audio's MIDI Ports dialog box (top) with input and output ports from Hubi's Loopback Device enabled. The default names (such as LB3) have been changed to more useful names, such as HubiIn1, by editing entries in the win.ini file. VAZ Modular's Set Preferences dialog box (bottom) shows HubiIn1 designated as the input device.

WRAPPING CABLES

Virtual cables have a number of other uses. MIDI Yoke can route its output to a related program called MIDI-OX that allows you to filter or remap MIDI data. You can also combine the two to provide multiclient functionality to your hardware MIDI inputs and outputs. Dozens of shareware MIDI utilities available for PCs let you do things such as analyze the output of a MIDI control surface routed through a virtual cable, so you can adapt it for use with unsupported programs. If your favorite editor/librarian isn't tightly integrated with your sequencer, a virtual cable lets you connect the two to record your edits as SysEx data.

Although plug-in technology continues to simplify the interconnection of MIDI and audio software, virtual cables still have a place in a desktop studio. Whether you use an interconnection standard, an interapplication bus, or a shareware cable program, the principles are the same. Follow the same logic as with physical connections, and it should all come together.

FIG. 4: In Cubase VST's ReWire panel, all of ReBirth's audio outputs are available to be enabled and named. All enabled ReWire connections show up as channels in the Mixer window (right).

Brian Smithers *is associate course director of MIDI at Full Sail Real World Education in Winter Park, Florida.*

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

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We love hearing stories about the creative ways people use our gear. And when we found out just how clever up-and-coming alt rockers Ottoman were in making EZbus the brains of their entire rehearsal and live show rigs, we just had to share it with you. Thanks to the EZbus's wide variety of analog and digital I/O—and its any-input-to-any-output signal routing—the band is able to plug in all of their audio sources, including vocal mics, keyboards, virtual instruments (resident on the now-ubiquitous laptop), electro-acoustic guitars, and effects devices (to name just a few), and generate multiple customized monitor mixes, DI-level mixes for the front-of-house console, and even a separate mix for recording their performance.

But that's just half the story. Keyboardist Marianna Hetrick also controls her synths, sound modules, and virtual instruments via MIDI using the EZbus's rather deep (not that we want to brag) software control surface functions. She's able to adjust virtually any parameter on any instrument (hardware or software) in real-time from the EZbus's front panel—and she can do it all while simultaneously operating the group's recording software with the EZbus's transport controls. And all the while the drummer is triggering samples via the EZbus's second MIDI input. (The samples, of course, play back through the EZbus over USB.)

We could go on about how the rhythm guitar tone is created through judicious use of the EZbus's on-board EQ and compression. Or how fully user-programmable mixes—recalled via footswitch—make it super easy for the band to use custom settings for each song in their set. Or how . . .

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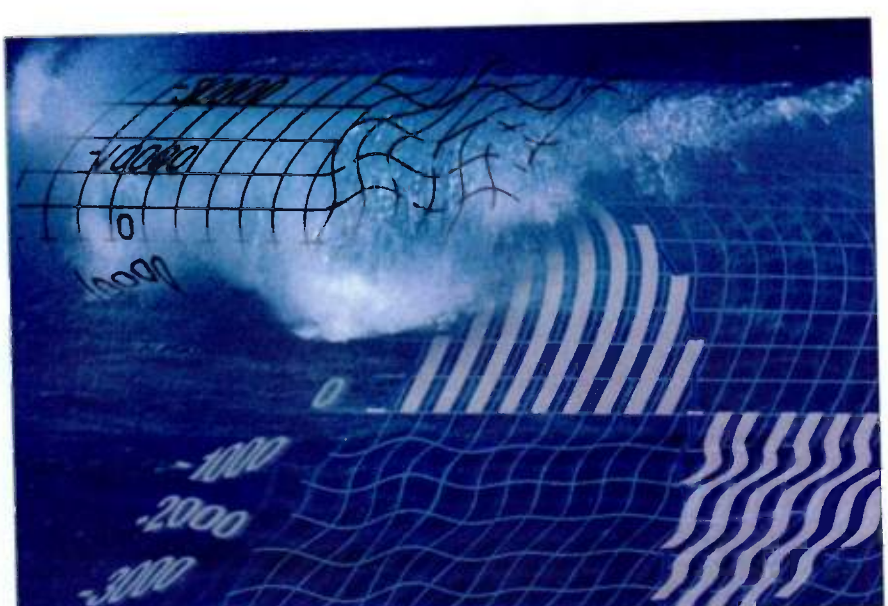
Making Waves

Get up close and personal with sound-file formats.

By Peter Hamlin

Everyone who works with digital audio soon encounters a wide variety of sound-file formats: WAV, AIFF, SND, Sound Designer I and II, and MP3, to name just a few. In most cases, the different formats present few problems. Software simply opens, plays, edits, and saves the audio files, sparing you from knowing the details of exactly how each format is constructed. But how does a program know what type of data the file contains? And what exactly is in an audio file?



A digital sound file is basically a long list of numbers representing the momentary values of an analog waveform measured (sampled) at a periodic rate. A file containing just those numbers is called a *raw-data* sound file. Usually, a lot more information must be embedded in a file for it to be read and played back properly. Aside from the *sampling rate,* the necessary information includes the *resolution* (which is the number of binary digits, or *bits,* that represent each sample). Other information indicates whether the file is monaural or stereo and whether the file creator has included looping information and cue points, a title, the name of the engineer or composer, a copyright notice, or other similar text.

That kind of information is included in a *header,* typically found at the beginning of a file. Different types of files have headers that are configured in distinct ways. (A raw-data sound file is also known as a *headerless* file.) For an application to read or write sound in a particular format, it must understand how the data is organized in that format.

PLAYING DIGITAL RIFFS

As an example, here's a close look at the familiar WAV sound-file format. A WAV file is a type of Resource Interchange

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File Format (RIFF) file, a format developed by Microsoft and IBM for multimedia files. (The familiar AVI video format is another type of RIFF file.) WAV files have been in use since Windows 3.1 and so are very widespread.

A WAV file is divided into sections, or *chunks*, that contain certain prescribed information. It's a more flexible arrangement than having just a single header. At the beginning of the file, a *RIFF* chunk defines the data as a WAV file and also reports its total length. Embedded within the RIFF chunk are two other chunks: a *format* chunk with information about sampling rate, resolution, number of channels, type of coding, and so on; and a *data* chunk, in which the actual sample values are stored.

To examine the format, I created a simple WAV file with my audio editor. The file contains a single cycle of a mono 2,205 Hz sine wave, synthesized at a 44,100 Hz sampling rate with 16-bit resolution. After saving the file, I displayed the data in the standard binary file-viewing format shown in **Fig. 1**. (There are many binary file-viewing utilities, including one called *debug* that is part of DOS. For the display in **Fig. 1**, I used Helios Software Solutions' *TextPad*, which lets you view files in many formats, including binary.) Color coding is added to differentiate each chunk.

GOOD TO THE LAST BYTE

Notice in **Fig. 1** that the RIFF-chunk header information is found in the first 12 bytes of the file (highlighted in pink). Every pair of numbers represents a unique byte; in the table "Interpretation of Data in a WAV File," a space between each byte shows how the data is organized. You can see the meaning of each byte in the table. Note also that the file data is in *hexadecimal (hex)* format. (If you're not familiar with hex, see the sidebar, "All About Numbering Systems.")

Interpretation of Data in a WAV File

NUMBER OF BYTES	FILE DATA (IN HEX)	INTERPRETATION	VALUE
4	52 49 46 46	ASCII characters identifying file as a RIFF file	"RIFF"
4	4E 00 00 00	total size of file minus 8 bytes of header	0x4E (hex) or 78 (decimal)
4	57 41 56 45	ASCII characters identifying file as WAV file	"WAVE"

The first four bytes in the file (the hexadecimal numbers 52, 49, 46, and 46) represent the *ASCII* characters for the acronym "RIFF," which denotes the format type. (ASCII characters are numbers that represent letters of the alphabet. See the Value column at the far right of the table.) The next four bytes (4E 00 00 00) indicate the total number of bytes of data in the file after the first eight bytes of the header. This four-byte integer is in a format called *little-endian*, which means that the least significant bytes come first when the computer lists them byte by byte. That takes some getting used to, because the string of bytes actually appears in the opposite order than you'd expect. In other words, the four bytes, 4E 00 00 00, signify the hexadecimal number 0x0000004E, which can be shortened to 4E or 0x4E. (The *0x* prefix is often used to indicate that the number is in hexadecimal format.)

```
 0: 52494646 4E000000 57415645 666D7420  RIFFN...WAVEfmt
10: 10000000 01000100 44AC0000 88580100  ........D¬..|X..
20: 02001000 64517461 2A000000 00008E27  ....data*.....|'
30: 3C4B8D67 BB79FF7F BB798D67 3C4B8E27  <K|g»yÿ|»y|g<K|'
40: 000072D8 C4B47398 45860180 45867398  ..rØA's|E|.|E|s|
50: C4B472D8 0000                        Ä'rØ..
```

= RIFF chunk   = format chunk   = data chunk

FIG. 1: This is a binary-file view of a simple WAV file containing a single cycle of a sine wave. The color coding indicates the different chunks of data in the file. All of the numbers on the left are in hexadecimal, and each 8-bit byte of data is represented as two hexadecimal digits. On the far right, the same data is reproduced in ASCII code, so you can see any text embedded in the file. When the data is not text, you just see gibberish in that column.

(The term *little-endian* in computer lingo is taken from Jonathan Swift's *Gulliver's Travels*. At one point in the story, the Lilliputians are divided into two warring political camps: the Little-Endians, who believe you should first crack a soft-boiled egg on the little end; and the Big-Endians, who believe the opposite. The computer term *big-endian*, as you would guess, means numbers are listed with the most significant digits first.)

The next four bytes (57 41 56 45) are the ASCII characters "WAVE"; they tell any application reading the file that this is WAV-audio format and not one of the other possible RIFF multimedia file types.

The next 24 bytes of data in **Fig. 1** (shown in blue) represent the format chunk, where several of the file's important characteristics are coded. This segment begins with the bytes 66 6D 74 20. The first three bytes of this string are the ASCII symbols for "fmt," and the "20" indicates a space, which just fills out this segment so that it takes up a full four bytes. The next four bytes (10 00 00 00) indicate the length of the format chunk. That value is hex 0x10, or decimal 16. The table "Format-Chunk Data" shows how the format data is arranged. As before, the second column shows the exact sequence of numbers in hex as they appear in the file.

Notice that the Type of Coding is PCM (Pulse Code Modulation). PCM is a common uncompressed-audio data

format. Other possibilities for coding include µ-law (pronounced *mu-law,* designated by the number 0x0101) and a-law (0x0102). Both are methods of scaling the sample data to try to minimize the audible quantization noise. (Quantization noise is a rounding error that occurs when you translate analog audio information into the more limited realm of digital numbers. If you use large enough digital data words, the quantization noise can be made so small that it does not cause audible problems.)

Another coding technique you may encounter is ADPCM (Adaptive Delta Pulse Code Modulation), which is designated by the number 0x0103. Interested programmers can find the exact formulas for those coding techniques on the Internet. (I'll deal only with uncoded PCM data here.)

The format chunk also indicates that the number of channels is 1 (monaural). The sampling rate is coded as 44 AC 00 00, or 0xAC44 (44,100 in decimal). The next four bytes (88 58 01 00; or 0x15888 in hex and 88,200 in decimal) define the number of bytes per second. This value is two times the

ALL ABOUT NUMBERING SYSTEMS

The numbering system people are most familiar with is called *base-10,* or *decimal.* No doubt it developed because you have ten fingers to count with. A decimal number has a ones column, a tens column, a hundreds column, and so on, depending on the size of the number. The decimal number 158 really means this:

100 (1 × 100)
50 (5 × 10)
\+ 8
158

But base-10 isn't the only possibility. When you tell time with minutes and seconds, you use a *base-60* system. That is, 1 hour, 5 minutes, 30 seconds, which might be represented as 1:05:30, really means this:

3,600 (1 × 60 × 60)
300 (5 × 60)
\+ 30
3,930 seconds

Binary. Computers use *base-2,* or *binary,* numbers because at the most basic electrical level, computers have two states: off (0) and on (1). A one-digit binary (or 1-bit) number has two values, 0 and 1. A 2-bit number can have four values: 00, 01, 10, or 11 (decimal values 0 to 3). An 8-bit number (called a *byte*) can have 256 values (0 to 255), and a 16-bit number can have 65,536 values. When referring to audio files, the size of a binary number is called the *resolution,* and the larger the number, the more accurately a sound can be represented.

Hexadecimal. Binary numbers are difficult to read, so they are typically represented with a *base-16* (hexadecimal) system. Hexadecimal numbers use the digits 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, and F (0 to 15 in decimal), and each hexadecimal digit can represent four binary bits. That means an 8-bit byte of data such as 1001 1110 can easily be represented by two hexadecimal digits, which, in this case, are 9E. (To be clear, the prefix 0x is often added to specify that the number is in hexadecimal format—for example, 0x9E.)

Hexadecimal numbers, like decimal numbers, are written in columns. But in place of the hundreds, tens, and ones columns, a hexadecimal number has columns based on the number 16. In the number 9E, for example, the left column is for multiples of 16^1, so the number 9 is multiplied by 16 to give a decimal value of 144. The right column is for multiples of 16^0 (that is, 1), and there you find the letter *E,* which is equivalent to the decimal number 14. So in total, 9E in hex is the same as 158 in decimal:

144 ($9 \times 16^1 = 9 \times 16$)
\+ 14 (hex E $\times 16^1 = 14 \times 1$)
158 (decimal)

If you had a larger hexadecimal number, you would have more columns, each representing a further power of 16 (16^2, or 16 × 16; 16^3, or 16 × 16 × 16; and so forth) depending on the size of the number.

In summary, the decimal number 158 can be represented in hexadecimal notation as 9E or in binary notation as 1001 1110. It's the same value in each case, but the system of showing that value is different.

Two's complement. A common way to represent negative numbers in binary arithmetic is with *two's complement* numbers. You get the two's complement by changing all 1s to 0s and all 0s to 1s (a process called *complementing*) and adding 1. The number 10,126 in the table "Amplitude Values for a Sine Wave" is shown in hexadecimal format as 0x278E. In binary, 0x278E is 0010 0111 1000 1110. The complement of that number is 1101 1000 0111 0001, and when you add 1, you get 1101 1000 0111 0010. This number can be represented in hexadecimal numbers as 0xD872, and in the table, the value 0xD872 is used to represent –10,126 when the waveform is negative.

ASCII. Computers understand only numbers, so when you want to use letters, you need to represent them in numeric code. American Standard Code for Information Interchange (ASCII) is a standard code in which the lowercase letters *a* through *z* are represented by the numbers 0x61 through 0x7A, and *A* through *Z* (the capital letters) are represented by 0x41 through 0x5A. Commas, periods, spaces, and other commonly used language symbols are given their own number codes, as well.

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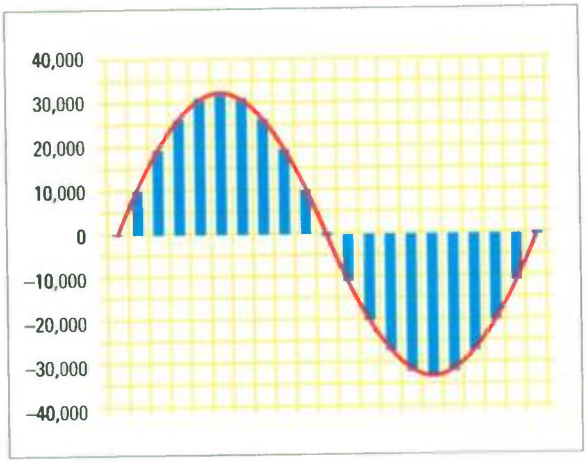

FIG. 2: This figure is a graphical representation of the amplitude values found in the data chunk of the sound file in Fig. 1. The graph plots the digitized sine wave encoded by the Pulse Code Modulation data in the sound file.

sampling rate (44,100), because each sample contains two bytes of data. Next is the number of bytes per sample: 02 00, or 0x2 (2 in decimal). Finally, 10 00, or 0x10 (16 in decimal), shows the number of bits in each sample.

The data chunk (highlighted in yellow) follows the format chunk and is announced in **Fig. 1** by the ASCII characters for the word *data* (64 61 74 61). The next four bytes, 2A 00 00 00, or 0x2A, indicate that the chunk length is 42 bytes. The table "Amplitude Values for a Sine Wave" lists the remaining 21, 16-bit integers (stored in little-endian format) and their decimal values. These are the bytes that represent the actual amplitude values for the sine wave.

The decimal numbers in the "Amplitude Values for a Sine Wave" table are *two's complement* numbers in the range –32,768 to 32,767. Two's complement is a way of representing both positive and negative numbers. If you look at the graph in **Fig. 2**, you can easily see how these numbers trace the shape of the sine wave.

Several additional chunks are optional in a WAV file. They are a *fact* chunk, used to store information about the file contents; a *cue* chunk, for indicating cue or marker points; a *playlist* chunk, to establish play order of cue points and looping information; and an *associated-data-list* chunk for attaching annotations to parts of the sample data. The RIFF format also supports an *info* chunk where you can place a title, copyright notice, creation date, and other similar text information.

SOMETHING SIMILAR

Another common sound-file format is AIF or AIFF (Audio Interchange File Format), developed for the Macintosh platform. AIFF files are similar to WAV files and have chunks with similar functions. The chunks are identified by

Format-Chunk Data

NUMBER OF BYTES	FILE DATA (IN HEX)	INTERPRETATION	VALUE
4	66 6D 74 20	ASCII characters identifying the format chunk	ASCII characters "fmt" plus a space
4	10 00 00 00	length of format chunk (in bytes) after this point	0x10 (hex) or 16 (decimal)
2	01 00	Type of Coding	PCM
2	01 00	Number of Channels	1 = mono
4	44 AC 00 00	Sampling Rate (per second)	0xAC44 (hex) or 44,100 (decimal)
4	88 58 01 00	Byte Rate (per second)	0x015888 (hex) or 88,200 (decimal)
2	02 00	Bytes per Sample	0x02 (hex) or 2 (decimal)
2	10 00	Bits per Sample	0x10 (hex) or 16 (decimal)

different names, however. For example, "FORM" is used to identify the sound-file format, "COMM" is the chunk with the format information, and "SSND" is the chunk that contains the sound data. Like WAV files, AIFF files require that these three chunks be present in all files, and again as in the WAV format, these words are followed by the length of the chunk. However, AIFF files use big-endian format for multibyte data. So unlike WAV data, a sequence of bytes for a 16-bit integer such as AC 04 would be the number 0xAC04 (as opposed to 0x04AC, the little-endian interpretation). There are other differences—the layout of the format chunk is not exactly identical, for example—but the basic idea is the same.

AIFF also allows optional chunks such as a *marker* chunk, where cue points can be stored; an *instrument* chunk, containing playback data (such as looping information) for sampling keyboards; and a *MIDI-data* chunk, for MIDI System Exclusive messages or any other type of MIDI data. It also permits an *audio-recording* chunk that has information used by some audio-recording devices; an *application-specific* chunk that could be used by a specific application

Amplitude Values for a Sine Wave

File Data	Hexadecimal Value	Decimal Value
0000	0x0000	0
8E27	0x278E	10,126
3C4B	0x4B3C	19,260
8D67	0x678D	26,509
BB79	0x79BB	31,163
FF7F	0x7FFF	32,767
BB79	0x79BB	31,163
8D67	0x678D	26,509
3C4B	0x4B3C	19,260
8E27	0x278E	10,126
0000	0x0000	0
72D8	0xD872	−10,126
C4B4	0xB4C4	−19,260
7398	0x9873	−26,509
4586	0x8645	−31,163
0180	0x8001	−32,767
4586	0x8645	−31,163
7398	0x9873	−26,509
C4B4	0xB4C4	−19,260
72D8	0xD872	−10,126
0000	0x0000	0

for any desired use; a *comments* chunk; and additional *text* chunks for name, author, copyright, and annotation.

HANDY TOOLS

Other sound-file formats have different specifics, but you will recognize similar principles. The Sound Designer I and II formats were created for Digidesign's Sound Tools and Pro Tools software. The AU and SND formats are associated with Sun computers, the NeXT, and UNIX machines. If you need more information about these or other formats, a good place to start is Chris Bagwell's Audio File Format FAQ (http://home.sprynet.com/~cbagwell/audio.html).

So what happens if you need to use a sound-file format that is not supported by the hardware or software you are using? If you use Windows, you should consider a sound-file format-conversion program such as Lance Norskog's SoX (short for Sound Exchange; available at www.spies.com/Sox/). SoX is a DOS command-line program and has a simple command syntax. You should also look at FMJ-Software's Awave (www.fmjsoft.com), which can read more than 60 formats and convert them into nearly 30 flavors.

Mac users should check out Tom Erbe's Soundhack (www.soundhack.com). It can read almost any sound-file format you'll encounter on a Mac and save it in many other formats. Soundhack also includes a wondrous array of powerful processing options.

It's not often that you'll have to dig into an audio file, but knowing what's inside can often be a help when problems arise. The information these files contain helps ensure that your digital sound files sound the way they're supposed to and allows you to go about your business of making music.

Peter Hamlin *is a composer who teaches at St. Olaf College. He is also a member of the live electronic-music improv band Data Stream. He is currently developing a set of electronic pieces based on the paintings of George Todd.*

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

I'M BIASED

David Torn aka splattercell
Guitarist/Texturalist/Producer

Collaborators include: david bowie • ryuichi sakamoto • carter burwell • kd lang • david sylvian • cliff martinez • me'shell ndegéocello • chute • page hamilton • b.l.u.e.

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.

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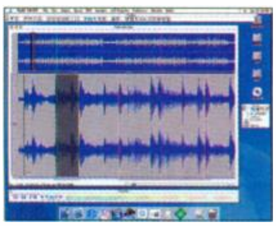

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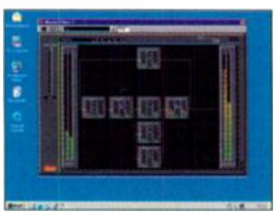

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WORKING MUSICIAN

Here Comes the Pitch

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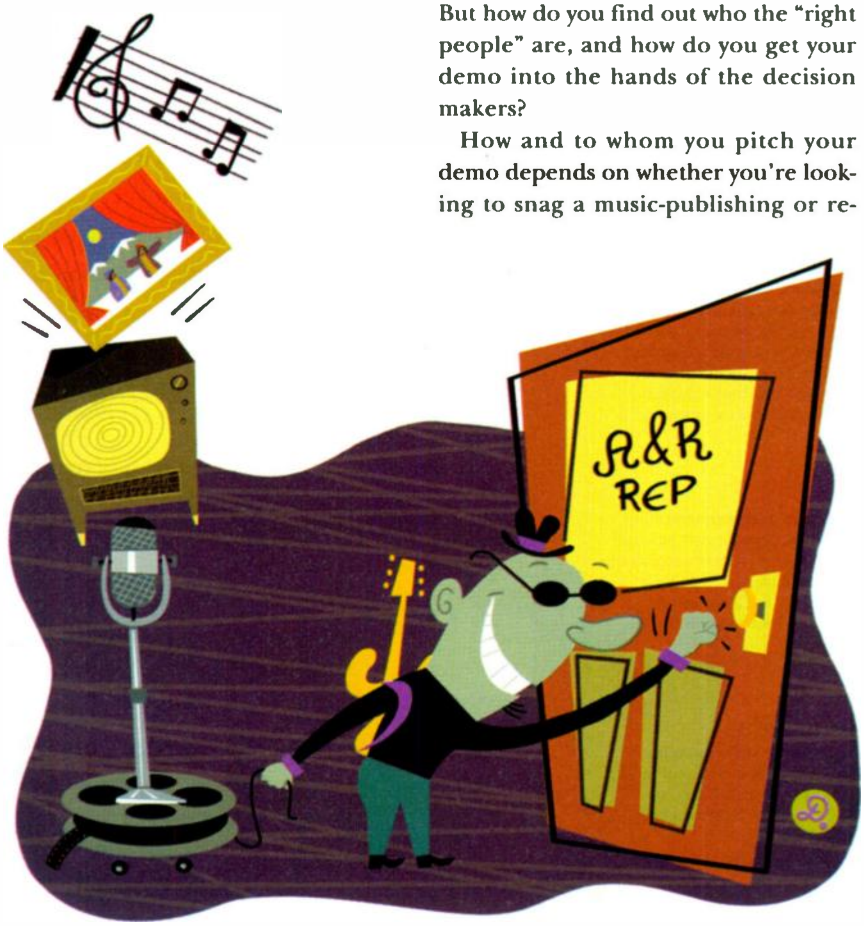
By Michael Cooper



JACK DESROCHER

You've just finished producing a killer demo. You're certain it will kick start your career, if only you can get the right people to hear it. But how do you find out who the "right people" are, and how do you get your demo into the hands of the decision makers?

How and to whom you pitch your demo depends on whether you're looking to snag a music-publishing or recording contract or hoping to score music for film or TV. Industry movers and shakers have strong preferences about what they want to receive in your demo package. You will have to decide what kind of deal you want and tailor your pitch and your promotional package to match.

PITCHING SONG DEMOS

The goal of a song demo is to obtain a record deal or music-publishing contract for the song (for more about demos, see "Producing Types of Demos" on p. 32). For most songwriters, winning a recording deal for a song is an almost impossible task to accomplish alone. The necessary time commitment, industry connections, and knowledge of contract law are so daunting that most songwriters enlist the help of a music publisher in shopping their songs for a record deal.

The best music publishers know, on a day-to-day basis, what types of songs industry decision makers such as producers and A&R reps want, and they have working relationships with and ready access to those people. When you sign a music-publishing contract for your song, you're essentially giving the publisher a cut of your royalties in return for its promise to pitch your song

and, hopefully, get it recorded by a major-label artist.

How do you find a music publisher to pitch your songs to? You can find the names, addresses, and telephone numbers of leading music-publishing contacts in readily available directories (see the sidebar "Music Business Directories"). Another way to make contacts is through one of the performing-rights societies (see the sidebar "Performing-Rights Societies"). Make an appointment to see a writer/publisher-relations person at one of the agencies to play them three or four of your best songs. That may require you to travel to one of the major music centers. If you can impress the writer/publisher-relations person with your songwriting abilities, he or she may set up an appointment for you with a publisher who would be hard to meet without a referral.

Do it yourself. If you have the time and know-how, you can bypass music publishers and pitch your song demo directly to record producers and A&R reps, the latter being the record label staff who often choose songs for artists on their label's roster. To be successful at pitching your song demo to producers and A&R reps, you'll need to know which artists are looking for songs to record, what kind of material (for example, up-tempo or ballad) is being sought, and who the artists' producer(s) and A&R reps are and how to contact them. Thankfully, pitch sheets (aka tip sheets) can provide that information in condensed form, saving you many hours of research (see the sidebar "Industry Pitch Sheets").

Some pitch sheets provide more current information than others. Also, many pitch sheets are geared toward industry insiders who already have established connections; they may list the artist, type of material sought, and names of contact people to pitch your song to, but not the contacts' phone numbers or addresses. For that reason, it is important to have an up-to-date directory of A&R reps on hand if you will be pitching to them directly. Contact information for record producers is often harder to come by. Some producers double as A&R reps or music publishers; you might find their names and contact information in the index of directories for those types of entities.

Pitch sheets and directories are helpful, but making contact with a producer or an A&R rep will not guarantee that your song will be heard. Mike Whelan, director of creative services for country-music publisher Acuff-Rose, says songwriters "have a better chance of winning the lottery than succeeding at a cold pitch to a producer of a major artist in Nashville." Whelan notes that producers might receive thousands of songs for one artist. "If they get a CD from a major publisher, that often goes to the top of the pile," Whelan says.

▼

Many pitch sheets are geared toward industry insiders who already have established connections.

Jim Vellutato, creative director for Sony/ATV Music Publishing, credits publishers' generally superior research and catalog depth for their preferential treatment. "Most songwriters don't do their homework very well," Vellutato says. Many songwriters often pitch inappropriate songs, and unlike publishers, who have a wealth of alternate songs to choose from, songwriters usually don't have more suitable songs to offer if their first pitch is off the mark.

First contact. Before you contact a music publisher, a producer, or an A&R rep for the first time, make sure you know what kind of material he or she is seeking. Sending a hip-hop tune to a country-music publisher will ruin any credibility you may have had with that

MUSIC BUSINESS DIRECTORIES

Whether you're pitching song, artist, or music-scoring demos, you'll have an easier time finding the right contacts to pitch your music to if you let your fingers do the walking. Here is a sampling of helpful music-industry directories.

A&R Registry
The most comprehensive domestic and international listings of names, titles, phone and fax numbers, and e-mail addresses for A&R reps ($350 for a six-issue annual subscription, $65 for a single issue; also available on disc). The Music Business Registry; tel. (818) 769-2722; e-mail info@musicregistry.com; Web www.musicregistry.com.

Billboard International Buyer's Guide
Domestic and international listings of record labels, music publishers, entertainment attorneys, performing-rights societies, and much more. Contacts for each organization are not nearly as comprehensive as in the three Music Business Registry offerings listed here, but many more business categories are included ($155). Billboard Music Group; tel. (800) 344-7119; e-mail ndavis@bpicomm.com; Web www.billboard.com/directories.

Film and Television Music Guide
More than 200 pages of listings organized by record labels, music publishers, film and TV music departments, music supervisors, film composers, composer agents, music editors, scoring stages, and much more ($95; also available on disc). The Music Business Registry (see A&R Registry listing for contact information).

Music Publishers Registry
Comprehensive domestic and international listing of names, job titles, phone and fax numbers, and e-mail addresses for leading music-publishing contacts. A must-have for songwriters ($125 for a two-issue annual subscription, $75 for a single issue; also available on disc). The Music Business Registry (see A&R Registry listing for contact information).

Musician's Guide to Touring and Promotion
Listings for A&R reps, personal agents, entertainment attorneys, booking managers, and more ($15.95, published twice a year). BPI Communications; tel. (800)407-6874; Web www.billboard.com.

"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.

Can TAXI get you into the inner circle? They'd be the first to tell you they can't promise anything. But four A&R people watching my show was all the proof I needed to know that TAXI can really deliver, if your music is right on target.

TAXI®

The Independent A&R Leader

1-800-458-2111

And if your music is a little bit off-the-mark, TAXI is probably the best thing you can do to whip it into shape. The written feedback you'll get from their A&R department is incredible.

You'll also get to meet top industry executives face-to-face at TAXI's annual convention, the Road Rally. As a member, you'll get FREE passes for you and a couple of guests.

This private convention is renowned for being the best in the business. Just one pass is worth far more than your TAXI membership fee, but you'll get three for FREE.

Whether you're pitching yourself as an artist, pitching your songs, or going for Film and TV placements, TAXI is definitely the place you need to call.

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.

publisher, closing the door to your future demo submissions.

Most industry contacts prefer—and many require—that you submit your song demo on a CD. Include a lyric sheet with any songs that contain vocals. You don't need to include your photo or bio with a song demo. However, it's important to include a discography if you've already had major records cut.

You can mail your demo package to your contact person, but setting up a face-to-face appointment to play your demo will increase your odds of success. Whelan says country-music songwriters who live in Nashville have the best chance of winning deals because of the daily contacts they make.

PITCHING ARTIST DEMOS

The purpose of an artist demo is to secure a recording contract for the musicians performing on the demo. The best people to shop your artist demo to are A&R reps, producers, and music-business attorneys. The best way to find an attorney is by getting a referral from your contact at a performing-rights society. Generally speaking, it's more important to have an attorney than it is to have a manager when seeking a record deal.

"You don't need a manager," says Jojo Brim, producer and senior director of A&R for Def Jam/Def Soul Records. "We're more inclined to listen to something that an attorney brings to us."

Your odds of getting a record deal increase if you wrote the songs that appear on your artist demo. Many artists break into the business by first getting music-publishing deals. The leading publishers often have major-label affiliations. A songwriter who consistently serves up awesome performances on song demos will probably be noticed for his or her potential as a recording artist and may be offered a record deal.

Some pitch sheets list record labels looking for new acts, but most new acts come to the attention of record labels through industry buzz surrounding their live performances. You're especially likely to grab major-label attention if your concerts get rave reviews and you have a self-produced album that receives regional airplay (though the latter is by no means required to get a record deal). On the other hand, an act with a great studio demo but little touring experience will usually get little consideration from a record label.

In fact, demos carry little weight with some country-music labels. Many Nashville labels require unsigned acts to come in and sing for the label executives. Nashville's insistence on live performance and personal contact makes it difficult for unsigned artists to succeed if they don't live in Music City.

Most A&R reps want song demos to be submitted on CD. Because good looks are so important in the country-music industry, Carole Ann Mobley, director of A&R for RCA Records in Nashville, advises country artists to include a photo in their demo packages. "A head shot and a body shot are always good," Mobley says. "The bio and press releases, I don't even look at those."

PITCHING FILM AND TV DEMOS

In talking with prominent film and TV composers Carter Burwell (whose credits include *Fargo* and *Being John Malkovich*) and Douglas Cuomo (HBO's

PERFORMING-RIGHTS SOCIETIES

Performing rights societies collect performance (such as radio airplay) royalties for songwriters and music publishers. The writer/publisher-relations staff at performing-rights societies will sometimes assist promising songwriters with meeting music publishers. Here are the main headquarters for the three most prominent performing-rights societies in the United States.

ASCAP
tel. (212) 621-6000
e-mail info@ascap.com
Web www.ascap.com

BMI
tel. (212) 586-2000
e-mail newyork@bmi.com
Web www.bmi.com.

SESAC
tel. (615) 320-0055
Web www.sesac.com

INDUSTRY PITCH SHEETS

Pitch sheets (aka tip sheets) provide current information on songs sought by producers and A&R reps for the recording artists they work with. Some pitch sheets also list record companies and music publishers looking for new talent.

Row Fax
This country-music pitch sheet is sent out once a week to subscribers and includes many major-label listings. It lists the artist, record label, producer, studio dates, and type of song requested, but usually does not provide the address to send submissions to. However, Music Row Publications offers a directory through its Web site for $34.95 plus postage. (Subscriptions are $129 per year by e-mail and $155 by fax.) Music Row Publications; tel. (615) 321-361; fax (615) 329-0852; e-mail news@musicrow.com; Web www.musicrow.com.

Taxi
Unlike pitch sheets that provide direct links to industry contacts for named projects, Taxi screens demos from bands, artists, and songwriters and forwards the cream of the crop to "major record labels, top music publishers, and music supervisors working on film and TV projects." Listings are for all styles of music and include top-tier companies ($299.95 for the first-year subscription, $199.95 for renewal, plus $5-per-song submission fee). Taxi; tel. (800) 458-2111; e-mail memberservices@taxi.com; Web www.taxi.com.

Sex and the City, NBC's *Homicide*), it quickly became apparent that no clear formula exists for pitching demos to secure work in those mediums.

"As far as pitching your demo goes, it's as much of a mystery to me as it is to anybody," Burwell says. "Most composers I know fall into [music scoring] by accident."

Burwell suggests that budding composers consider going to film school, where they can write music for thesis projects. If and when the directors of those projects go on to making feature films, your established relationships with them could lead to you scoring those films. You're not likely to score Hollywood feature films without first establishing a track record on smaller projects. (See "Working Musician: You Ought to Be in Pictures," in the January 2002 issue, for a primer on breaking into scoring music for film and television.)

Cuomo says that making cold pitches to producers and directors is typically a futile strategy. "Most of the cold-calling stuff doesn't directly pay off," Cuomo says. "The thing that really helped me the most was to have done things that people have heard of a little bit." Getting mentioned in a prominent review of an off-off-Broadway theater production helped kick start Cuomo's career.

Making cold pitches to producers and directors is typically a futile strategy.

He used the favorable press to win work on projects he heard about through a network of friends and associates in the TV industry.

Once you succeed in making personal contact with the producer or director of a film or TV project, both Burwell and Cuomo recommend you submit your demo on CD. They also suggest you include in your package a list of any completed film and TV projects you've done. It's not necessary to include your photo in your demo package.

PLEASE FORWARD

No matter what type of demo you pitch, be sure to always include your name, telephone number, and mailing address on every item (CD, lyric sheet, and so on) you submit with your package. Those items sometimes get separated from one another—and sometimes even lost—in the process of listening to the music. Nothing would be sadder than losing a major contract for your knockout song because the person you pitched it to couldn't find a way to contact you. ⊕

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

REVIEWS

ZOOM

MRS-1044

Extras and ease of use set this multitracker apart.

By Steve Broderson

The explosion of affordable and sophisticated digital multitrackers has produced its share of frustrated musicians. Plunging headlong into the waters of affordable digital recording, they soon discover that they're in over their heads with gear that takes weeks to learn and months to master. Thankfully, the Zoom MRS-1044 has a shallow end for beginners and a deep end for experienced divers.

The MRS-1044 gives you ten tracks of digital audio at 16-bit, 44.1 kHz resolution (onboard effects are processed at 24 bits). Tracks 7/8 and 9/10 are treated as stereo pairs but can be edited separately. Each track has 10 associated virtual tracks, called V-Takes, for a total of 100 tracks per project. Two inputs offer a variety of ways to get sound into the MRS-1044 (see **Fig. 1**). Both have unbalanced ¼-inch jacks and balanced, phantom-powered XLR inputs; input 1 adds a dedicated guitar and bass input. The MRS-1044 lacks a digital input.

Each input has a trim pot for adjusting gain, a clipping-indicator LED, and a lighted on/off button. Output options include a stereo headphone jack (Zoom thoughtfully oriented the Volume knob

The Zoom MRS-1044 provides ten tracks of digital audio at 16-bit, 44.1 kHz resolution and includes built-in effects and drum and bass synthesizers. The unit is well suited for beginners as well as more advanced users.

upside down, knowing users would likely reach over the unit to adjust it), a stereo pair of RCA outs, and an S/PDIF optical digital out. Auxiliary sends and returns are notably absent, but the makers of this unit assume you need only the effects and accompaniment it can provide.

MIDI In and Out ports let the MRS-1044 send and receive information for its drum and bass generators. It also outputs MIDI Clock, though it cannot receive MIDI Clock or be set to slave to any external signals. Zoom also warns that transmitting MIDI Clock and drum or bass information from the MRS-1044 can interfere with synchronization. Rounding out the back panel are jacks for a punch-in pedal and an expression pedal (neither pedal is included) and a phantom-power switch.

RED-LIGHT DISTRICT

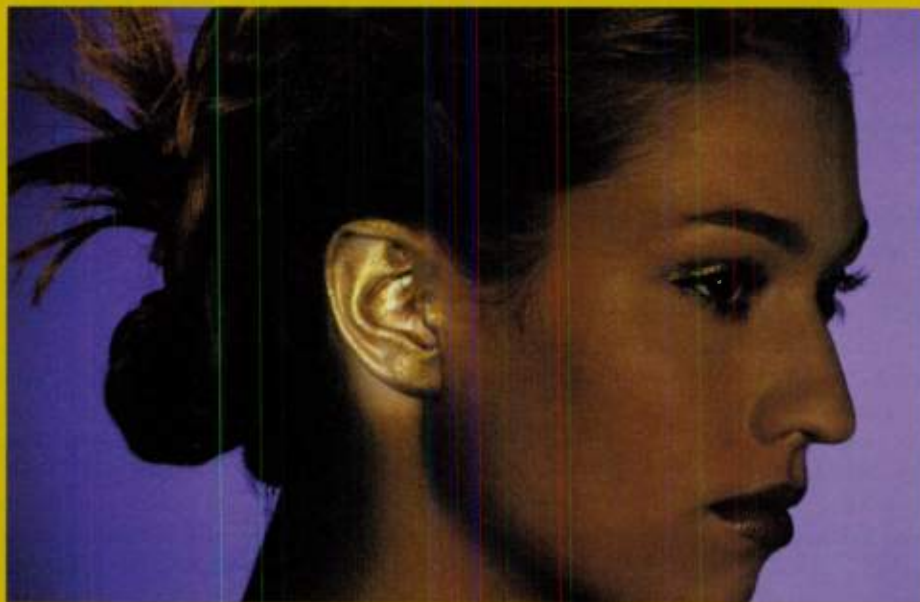
Functions and settings for all-in-one multitrackers are often buried deep within menus. In contrast, the MRS-1044 has an abundance of dedicated controls that, as a bonus, have red LED status indicators. Instant access and visual feedback are major factors in the MRS-1044's ease of use, and its layout is well designed. For instance, the five Track-Parameter buttons are laid out vertically in the same top-to-bottom order as a conventional mixer channel strip: EQ High, EQ Low, two effects sends, and Pan. Also, the large data-input wheel is near the control buttons you'll use it with most often: the cursor keys, Enter, Exit, Undo, and Edit. The effects section is similarly helpful, showing you at a glance which effects are active in any given preset. In addition, the drum/bass pads light as their sounds are played.

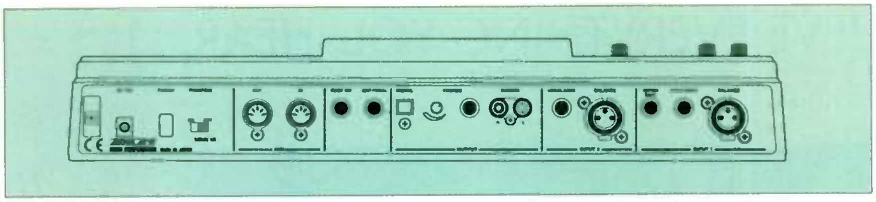
The backlit LCD, though not huge, provides track-level information, edit parameters, and time bars and beats information. The alphanumerics are nice and readable, but I would have preferred more display real estate dedicated to mixer levels. The meters also double as indicators to let you know if you've recorded any V-Takes; for that use, the meters are pretty tiny. An audio Scrub feature lets you dial in precise points in your sounds, but there's no waveform display. You can switch your Time Base display from minutes and seconds to bars and beats with the push of a button. In short, the MRS-1044 has a well-thought-out control surface other manufacturers would do well to study.

DRUM 'N' BASS

The MRS-1044 can record only two tracks at a time—not ideal for bands wanting to lay down eight tracks of drums. But the unit seems particularly tailored to the songwriter who's building an arrangement track by track. Besides, who needs eight tracks of drums when you have an onboard drum machine? Many competing multitracks include a rhythm generator, but their limited flexibility and wimpy sounds make it clear that they're only for playing to a beat, not creating a final demo. Zoom upped the ante by including a full-featured, great-sounding drum machine.

Eight rubberized Velocity-sensitive pads control three banks of sounds each





COURTESY ZOOM CORP

FIG. 1: The MRS-1044's back panel provides two inputs, each with an unbalanced ¼-inch jack and a balanced, phantom-powered XLR jack. Input 1 also offers a ¼-inch high-impedance guitar and bass input. Guitarists will appreciate the unit's expression-pedal input.

(that's a hefty 24 sounds per pattern, more than either of my standalone drum machines). Those sounds are grouped together in 30 drum kits and a stunning 733 preset patterns—with plenty of room for user patterns, too. Like most drum machines, the MRS-1044's Patterns can be programmed and arranged into Songs that lock to your project's tempo. The variety of terrific-sounding samples says that this feature wasn't an afterthought.

The pads also double as input triggers for the onboard bass generator, which includes 15 acoustic, electric, and synth-bass sounds. Each sound covers a five-octave range. Each drum pattern has an associated bass pattern (based on the pattern's musical style); you can modify them or create your own. A keyboard MIDI controller is handy, as tapping on the keys for a bass line is a little tedious. Because you have only eight pads to work with, there's a lot of root-chord and scale switching involved using the data-input wheel, and the end result can be pretty mechanical. For some reason, the process reminded me of playing the handheld Merlin game in the '80s. Having said that, the procedure is more time-consuming than difficult and would be a welcome feature for songwriters without access to a keyboard or bass.

DIRECT ROUTE

When it's time to plug in and get a signal to the recorder, the MRS-1044 is far from intimidating. Complex digital-mixer operations, such as routing and scene automation, are simplified to push-button operations. For instance, you don't have to tell the mixer to route input 1 to track 9 so you can record on it; the MRS-1044 assumes that because you plugged in to input 1 and hit the Record status button above track 9, that's what you want to do. Bouncing tracks is simple, too. Just press Bounce and set the destination track or tracks to Rec.

Routing is also easy for effects. The Input Source button lets you assign patches to any track, including the drum and bass generators—very cool for whacking out the drum sounds. At the input stage, the only thing that could be confusing is the level-setting procedure. Your inputs have three level controls: a Recording Level knob, an Overall Gain knob, and the Destination Channel fader (used for monitoring your signal). In addition, some insert effects also add or subtract gain, so the signal path requires extra attention.

As mentioned previously, track parameters such as EQ, sends, and pan are controlled by a single row of channel-strip buttons, also with status lights. If you've switched off any of those for a selected track, you can tell at a glance. It bears repeating that such thoughtful engineering makes tracking and mixing much less tedious for beginners and veterans alike.

Mixer and effects settings can be saved as 1 of 100 scenes (per song) and associated with marks you've set, which lets you create a sequence of scenes that automates your changes as the song plays. Again, the procedure is much less complicated than on similar recorders; I just stepped through my marks to the point where I wanted the change to occur, pressed Mark, and dialed in the scene I wanted.

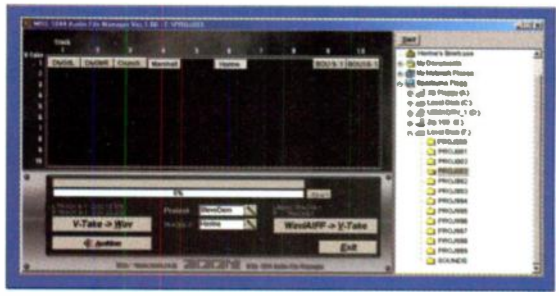


FIG. 2: Audio File Manager (PC only) lets you visualize your tracks as blocks on a grid. Being able to rename them with your keyboard is a great timesaving feature, as well.

ON YOUR MARK

Marking sections of your song inside the MRS-1044 couldn't be more intuitive, thanks to the six buttons dedicated to the task. The MRS-1044's procedure for setting Auto Punch In/Out points is easier than that of similar recorders. One button sets mark points, one clears them, and two let you step forward or backward through them. The remaining two buttons are dedicated to Auto Punch mode and A-B Repeat mode. To engage either mode, go to your start mark, press Punch or Repeat, step to your end mark, and press it again. You zero in on points in your song in three ways: by specifying bar:beat:tick or Hours/Minutes/Seconds/Frames, or by using the audio Scrub feature.

If you track to the drum machine, Bars and Beats is the fastest time base to work with. Just dial in the first bar of your chorus, verse, or whatever, and press Mark. Zoom could improve on one thing, though: the MRS-1044 lets you scroll through measures and beats using the data-input wheel, but you can't round off the clock ticks to zero. That requires some extra wheel turns that could be eliminated if the locations were always round numbers.

CHAIN OF TOOLS

As in traditional mixers, signals can be processed through inserts or sends. The MRS-1044 has four insert-effects types: Guitar/Bass (100 patches), Line (50 patches), Mic (50 patches), and Mastering (20 patches). Each type has its own status button, and each is constructed of six effects modules in series. Those modules include compression/limiting, an amp or preamp, Zoom's noise reduction (ZNR), a volume-pedal control, EQ, and a delay/modulation effect. Four modules have dedicated status buttons, as well, and the noise reduction, volume pedal, and overall patch level are all grouped under the Total button.

You call up patches by simply pressing the large Effects button. As you scroll through the preset effects with the data-input wheel, each module's button lights if it's used in the patch, providing instant visual feedback of the active effects. To edit any module or algorithm, just press it and then hit the Edit button and cursor through your settings. The two send/return effects, chorus/delay and reverb, are engaged in similar fashion. There are 20 of each type, and they have many editable parameters. If you're keeping score, that's three independent 24-bit stereo effects generators at once.

User settings for send/return and insert effects can be saved, but not in a dedicated user location; you just write over the factory presets. The good news is that once you create an effect you like, it can be shared among projects. No more re-creating that great vocal or guitar setting for every song. Sweet!

CUT . . . V-TAKE TWO

The MRS-1044 employs the standard editing features found in most portable

PRODUCT SUMMARY

Zoom
MRS-1044
portable digital studio
$1,199.99

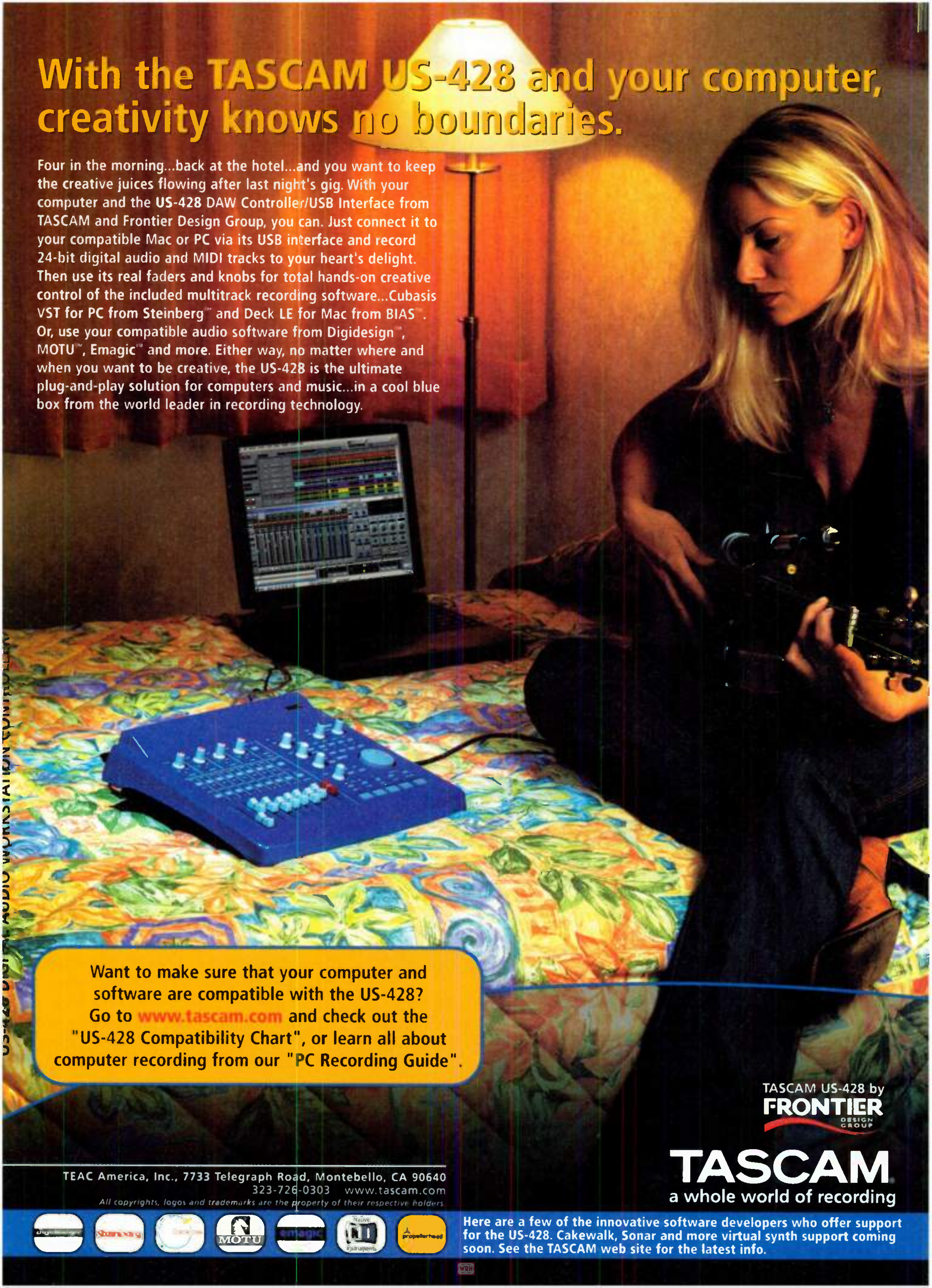
FEATURES	4.0
EASE OF USE	5.0
DOCUMENTATION	4.0
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Easy to learn. Intuitive control surface. Great effects. Many extras, such as drum machine and bass sequencer.

CONS: Records only two tracks at a time. Hard drive not upgradable. Macs not currently supported.

Manufacturer
Zoom Corp./Samson Technologies (distributor)
tel. (800) 372-6766 or (516) 364-2244
e-mail support@samsontech.com
Web www.samsontech.com


With the TASCAM US-428 and your computer, creativity knows no boundaries.
Four in the morning...back at the hotel...and you want to keep the creative juices flowing after last night's gig. With your computer and the US-428 DAW Controller/USB Interface from TASCAM and Frontier Design Group, you can. Just connect it to your compatible Mac or PC via its USB interface and record 24-bit digital audio and MIDI tracks to your heart's delight. Then use its real faders and knobs for total hands-on creative control of the included multitrack recording software...Cubasis VST for PC from Steinberg™ and Deck LE for Mac from BIAS™. Or, use your compatible audio software from Digidesign™, MOTU™, Emagic™ and more. Either way, no matter where and when you want to be creative, the US-428 is the ultimate plug-and-play solution for computers and music...in a cool blue box from the world leader in recording technology.
Want to make sure that your computer and software are compatible with the US-428? Go to www.tascam.com and check out the "US-428 Compatibility Chart", or learn all about computer recording from our "PC Recording Guide".
TASCAM US-428 by
FRONTIER
DESIGN GROUP
TASCAM
a whole world of recording
TEAC America, Inc., 7733 Telegraph Road, Montebello, CA 90640
323-726-0303 www.tascam.com
All copyrights, logos and trademarks are the property of their respective holders
MOTU
Here are a few of the innovative software developers who offer support for the US-428. Cakewalk, Sonar and more virtual synth support coming soon. See the TASCAM web site for the latest info.

multitrackers: Cut, Copy, Paste, Move, and Erase. Those are implemented in standard ways, with one exception: undoing any edit requires that you first capture a track in its original form. In similar recorders, that's usually taken care of by the operating system, which keeps track of changes to your data—not so on the MRS-1044. If you don't capture your guitar lead before you record another take, you can't go back with Undo. The same is true for any edit function that copies over or erases data. With all the other elegant things about this unit, I don't understand the reason for that clumsy extra step (maybe I'm just grousing because I always forgot to do it).

Thankfully, recording alternate versions of a track with V-Takes is much easier. A dedicated V-Take button takes you directly to a screen where you can dial in one of ten layers of tracks beneath your primary track. The MRS-1044 even lets you name the virtual tracks, which is a feature I've often wished for, even in higher-priced units. Audio can be copied from any V-Take to any other, providing lots of arrangement flexibility. You can copy, move, or erase tracks individually or in pairs of adjacent tracks (no more than two at a time, though). The MRS-1044 takes an unusually long time to execute those functions, however, and copying pairs of tracks takes even longer.

PC CONNECTION

The MRS-1044 is equipped with a generous 15 GB hard drive. (The hard drive will be increased to 20 GB by the time you read this. A new model, the MRS-1044CD, will have a 40 GB hard drive and an internal CD burner.) No matter how big the drive is, one day you'll manage to fill it up. When that happens, the MRS-1044 has optional user-installable USB or SCSI interface cards ($99 each). The more I thought about it, the wiser Zoom's decision seemed. Because SCSI and USB machines are common, you can buy the card your machine needs. If you don't want such capability, you don't have to pay for it.

The boards come with a CD-ROM of software that lets you back up your projects, convert tracks and V-Takes to editable WAV files, and update your unit's operating system. The Audio File Manager software also offers some great benefits that aren't so obvious. One is the ability to see your tracks and V-Takes in a graphical grid format (see **Fig. 2**). Clicking on Audition lets you hear your tracks (as many as two at a time).

Another big time-saver is the track-renaming feature. That lets you enter names for projects, tracks, and V-Takes with your computer keyboard, which is much less tedious than entering letters one at a time using the data wheel. As cool as Audio File Manager is, as of this writing, the Macintosh is not supported. Zoom hints that Mac support is on its way through software downloads from the Zoom Web site.

IN ACTION

During the review period, I laid down some song ideas using only the MRS-1044's internal effects, onboard bass, and drum generator. The basic sound quality was pristine and detailed. Using a Røde NT1 microphone, I dialed in an acoustic-guitar preset and generated a pleasing sound with just a little tweaking. The reverbs and spatial effects are particularly attractive because of their 24-bit resolution. In addition, the great sounds and wide assortment of onboard drums and basses are much more than guide tracks. The biggest disappointment was the electric-guitar presets. For the most part, distortion effects sounded thin and transistory, and few had the personality of a miked amplifier. On the plus side, I created some passable presets from scratch, and there are lots of parameters to tweak. The expression-pedal input is a nice touch, too.

Overall, the MRS-1044's strengths far outweigh any gripes I have. Songwriters—particularly those without access to a drum machine, a sequencer, or a bass guitar—will get a huge bang for their multitracker bucks. If you have a PC with SCSI or USB, definitely spring for the interface card. Newcomers to recording and production can dive in without fear of a steep, frustrating learning curve.

If you've been putting off buying a multitracker because you thought you'd never figure out how to use one, check out the MRS-1044. It sounds and works like a hard-disk recorder should.

Steve Broderson *is an adjunct professor at Asbury College in Wilmore, Kentucky, where he teaches audio for media. His recently formed Studio 246 creates original music for broadcast.*

MRS-1044 Specifications

Physical Tracks	(10) mono; (1) stereo drum; (1) mono bass
Virtual Tracks	100 (10 per track)
Simultaneous Record Tracks	2
Sampling Rate	44.1 kHz
Sampling Resolution	16-bit
Analog Inputs	(2) unbalanced ¼"; (2) balanced XLR; (1) ¼" high-impedance guitar/bass
Analog Outputs	(2) RCA; (1) ¼" stereo headphone
Digital Output	(1) S/PDIF optical
Additional Connections	MIDI In, Out; SCSI or USB (optional); (2) ¼" footpedals
Internal Storage	15 GB EIDE hard drive
Effects Processors	(3) 24-bit
Sound Engine	sample playback
Frequency Response	20 Hz–20 kHz (±1 dB)
Dynamic Range	>97 dB (IHF-A)
Total Harmonic Distortion	<0.02% (@ 400 kHZ)
Display	2.38" (W) × 1.63" (H) backlit LCD
Dimensions	14.10" (W) × 2.53" (H) × 8.53" (D)
Weight	9.5 lb.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STEINBERG

HALION 1.1 (MAC/WIN)

Stream your heart out with this multitimbral sampler and 1.7 GB of content.

By Len Sasso

Steinberg's Halion is a multitimbral, multioutput software sampler that features direct-from-disk streaming of samples. Disk streaming lets you use large acoustic-instrument samples, and Halion comes with a ton of them. Halion supports several common sample formats, including Akai, E-mu, SoundFont, Giga, and REX, so you should have no trouble finding additional content suitable for your needs. As expected, it supports AIFF and WAV formats, as well.

Halion is a VST2 Instrument capable of playing a separate program on each of 16 MIDI channels. Its 12 virtual outputs (four stereo pairs and four mono) make it possible to apply separate digital signal-processing effects to different programs. If you use a VST host that supports multiple outputs (which, at this time, means Steinberg's Cubase or Nuendo), you can get a lot of music out of a single Halion instance.

I tested Halion 1.1 using Cubase VST/32 5.1 on a Mac G3/300 MHz with 384 MB of RAM running OS 9.1 and on a Pentium III/700 MHz PC with 128 MB of RAM under Windows 98SE. The Mac had an Emagic AW8 audio card, and the PC used an Emagic EMI 2/6 USB interface. Although both systems are well above the manufacturer's stated minimum requirements, the speed of the Mac and the limited RAM on the PC were barely adequate. To take full advantage of Halion's multitimbral, multioutput, and streaming features, you need some serious CPU muscle, a huge amount of RAM, and a fast hard drive, preferably one that's dedicated to Halion streaming.

OUT OF THE BOX

Shortly after its initial release, Steinberg introduced an update, Halion 1.1, which should be available when you read this. Much of the audio content has been revised for version 1.1, and for obvious reasons (size being chief among them), it is not downloadable. You can download the latest version of Halion from Steinberg's Web site, and version 1.0 users can contact Steinberg to receive the new content.

Like other synthesizers and samplers, Halion's programs are organized into banks of 128. In typical fashion, you can select programs on individual MIDI channels using MIDI Program Change messages. Halion calls its programs Instruments, and each Instrument consists of samples, a keymap, and various parameter settings, such as those for filters and envelopes. You can load and save individual Instruments or entire banks.

Although Halion can play samples directly from your hard drive, it must load a portion of the file into RAM in order to achieve low-latency playback response. Banks that contain many programs can take a long time to load and can absorb huge chunks of memory. Therefore, tailoring banks to individual songs is generally the most efficient approach. (Halion's audio files include individual Instruments but no banks.)

Minimum System Requirements

Halion

MAC: PPC 604e/250; 128 MB RAM; OS 9.0

PC: Pentium II/266; 128 MB RAM; Windows 95/98/ME/2000

FIG. 1: In one compact window, Halion's Macro Page View gives you access to the most common parameters. It's an efficient place to work.

THE HALION VIEW

When you first open Halion, it displays the Macro Page View, providing access to some of the most frequently used parameters from Halion's seven other Page Views (see **Fig. 1**). The Macro Page View is half the size of the others and, therefore, a more convenient place to work; no doubt you'll spend most of your time there. In the Macro Page View, you can assign any Instrument to any MIDI channel and to any of Halion's 12 audio output channels. You can also adjust filter, filter envelope, amplifier envelope, tuning, and LFO settings.

The other Page Views—Keyzone, Chan/Prog, Env/Filter, Mod/Tune, Waveloop, and Options—offer more settings, graphic-editing features, and, in many cases, the ability to apply settings on an individual-sample basis. When you edit settings in the Macro Page View, the changes are global but can be applied absolutely or relatively to the individual sample settings. For example, if you edit the filter cutoff on the Macro Page View in Absolute mode, the cutoff setting for each sample will jump to the new value. In Relative mode, each sample's filter cutoff will be offset from its existing value by the amount you specify.

Halion's other Page Views are typical of multitimbral samplers but offer some unique editing features. Each view shares a common display called

the Program List, which does much more than its name implies (see Fig. 2). The Program List is hierarchical and lets you open a sublist for each Instrument to show the samples it contains. Samples can be dragged from the list to Keyzones, and multiple samples can be selected, allowing some of their parameters (such as filter, envelope, and LFO settings) to be edited simultaneously. The Program List is where you choose whether edits apply to the whole program or to selected samples and, in the latter case, whether the edits apply absolutely or relatively.

The Chan/Prog Page View is a map of Halion's 16 MIDI-channel assignments; it has drop-down menus for selecting the program and audio output for each channel. The Keyzone Page View shows the arrangement of an Instrument's samples across MIDI note and Velocity zones. You can drag samples there from the Program List and import samples in various formats from your hard drive. Samples can be freely dragged and sized in the Keyzone Page View, and overlapping regions can be layered or crossfaded.

The Waveloop Page View is Halion's mini sample editor. You can't actually

PRODUCT SUMMARY

Steinberg
Halion 1.1 (Mac/Win)
software sampler
$399

FEATURES	4.0
EASE OF USE	3.5
AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Multiple MIDI channels and audio outputs. Flexible per-sample modulation routings. Large Instrument collection.

CONS: Requires very fast computer and hard drive. Features limited in VST hosts other than Cubase and Nuendo.

Manufacturer
Waldorf/Steinberg North America (distributor)
tel. (818) 678-5100
e-mail info@steinberg.net
Web www.steinberg.net

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The Env/Filter Page View is for adjusting filter, amplifier, and envelope settings (see **Fig. 3**). There you can edit settings for individual samples, groups of samples, or all samples simultaneously. That means you can have separate filters, amplifiers, and envelopes for each sample. In a multisampled Instrument—a piano or stringed instrument, for example—you might use slightly different settings for upper and lower sections of the keyboard.

The ability to specify settings for individual samples or groups of samples becomes even more valuable with mapped samples such as drums, percussion instruments, and sound effects. With a different sound on each key, you frequently want unique filter settings—a high-pass filter to take a little of the bottom off a kick drum and a resonant bandpass filter to punch up a crash cymbal, for instance.

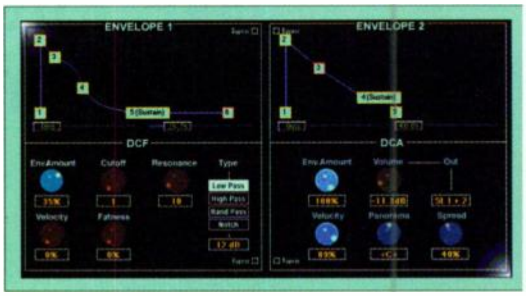

FIG. 3: The Env/Filter Page View features eight break-poin envelopes for the multimode filter and the output amplifie Separate settings are available for each of an Instrument' samples.

Halion's envelopes are flexible and permit as many as eight break points with individual segment shapes scalable from convex to concave. You can graphically adjust all envelope break points and segment shapes.

The Mod/Tune Page View provides 12 modulation routings, and again, you can assign them to individual samples, groups of samples, or all samples. Modulation sources include most types of MIDI messages, two built-in LFOs, a noise source, a glide, either envelope generator, and 12 user-definable constants. Any modulation source can be scaled by any other modulation source. Destinations are filter cutoff, resonance, volume, pan, pitch, and any of the 12 constants just mentioned. The LFOs offer sine, saw, and pulse waveforms and can be delayed and synchronized to MIDI tempo.

The Mod/Tune Page View is also where you assign sample groups, set the amount of glide (aka glissando or portamento), and set Raw and Drum sample-playback modes. Assigning samples to the same sample group lets you restrict the polyphony of that group. Typically, the polyphony is restricted to one note (a function often called Hi-Hat mode). When a sample is set to Raw, all of its parameter settings are ignored except the amplifier-related settings: Velocity, pan, envelope, and spread. In Drum mode (which automatically turns on Raw mode), the loops and sustain are also ignored, so the sample plays from beginning to end in one shot.

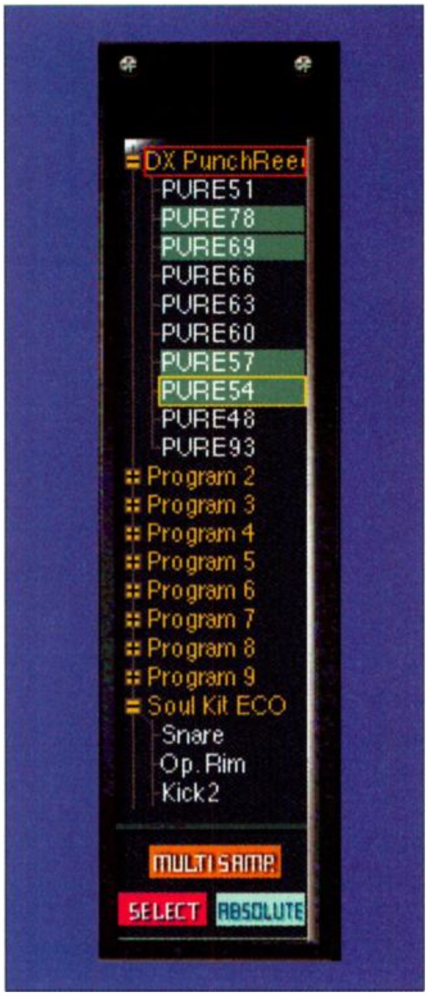

FIG. 2: Halion's Program List shows Instruments and their samples for convenient selection and drag-and-drop Keyzone positioning.

The last Page View, Options, is for global settings that affect the overall performance of Halion. Those include the amount of each sample to preloac into RAM (which applies to all sample in all Instruments), the number of note (which sets the disk buffer size), anc the playback quality (which can be usec to limit bandwidth and reduce CPL load). Settings you make in the Option Page View, together with audio-drive settings (particularly the driver's buffe size), have a profound effect on Halion' performance. Options is also the page where you can import third-party samples and audio files.

With any software sampler or synthesizer, there are always trade-offs betweer real-time performance (latency), audic quality, and the number of notes o tracks. On my relatively slow Mac, the highest audio-quality setting with a one second RAM buffer and a minima ASIO buffer provided only five or si notes before sounds started breaking up and dropping out. On the othe hand, medium quality (which stil sounded fine), four-second buffering and a larger ASIO buffer supported 3 notes along with a couple of audic tracks. (The latency caused by the large ASIO buffer made real-time playing impossible, however.) In a nice touch you can override the quality setting when bouncing to audio files, whicl means you can have more notes during playback but still retain the highest resolution output when you do you final audio rendering.

THE CONTENT

Halion comes with four CDs containing more than 1.7 GB of Instrument and samples. The program devote


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MEDIA MATTERS
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more than 1 GB to four instruments from Wizoo: an acoustic piano, a nylon guitar, drums and percussion, and a six-string electric bass. Presumably, the reason for such large Instruments is to show off Halion's streaming technology. The primary advantage of streaming is that you can make use of a sound's natural decay rather than loop a portion of the sample, which always sounds somewhat unnatural. The sounds Steinberg has chosen illustrate that well, and if you do a lot of work with acoustic music, you will find them quite useful, especially the bass and guitar Instruments.

Because the individual samples in those four Instruments are huge, one way to save memory is to use fewer samples in each Instrument. To that end, the piano, guitar, drum, and bass Instruments come in three sizes: XXL, MID, and ECO. (You can hear an example of the differences in the mp3 file Quartet, which is a piano, bass, drum, and guitar blues starting with the ECO versions, crossfading into the MID versions, and crossfading again into the XXL versions.) Four-second RAM buffering for the four-Instrument bank required 54 MB for the ECO version, 93 MB for the MID version, and a whopping 220 MB for the XXL version.

The rest of the Halion content supplies 200 MB of electronic-instrument samples from Wizoo and 500 MB of loops and tools from eLab. The collection features Magnetic Instruments (electronic pianos, Clavinets, guitars, and organs), Electronic Instruments (including synth basses, electronic drums, pads, and leads), and selections of drum loops and music loops ranging in tempo from 65 to 170 bpm. Most of the Instruments sound good, but the collection is not extensive, and you'll quickly wind up looking for additional content in those areas.

IS IT FOR YOU?

Halion is clearly a formidable product, and under the right circumstances, you get a lot of performance for your sampling dollar. But the decision to choose Halion comes with many caveats. You need a very fast computer with a lot of RAM for maximum performance. In addition, you must have a large, fast hard drive with a fast bus; for the best results, you'll need to dedicate a drive to Halion streaming. Finally, to take full advantage of Halion's multitimbral, multioutput features, you need to use Cubase or Nuendo as your VST host.

Halion's 1.7 GB of content is good and should be a factor in your purchase decision. The big instruments from Wizoo are well recorded, and even the economical versions sound great in most acoustic projects. The Wizoo collection of electronic instruments and sound effects is less generous but does cover the bases. The eLab collection of loops and tools is also outstanding. Although some eLab material has been overused, you'll still find plenty to like if dance music is your thing.

Len Sasso *can be contacted through his Web site at www.swiftkick.com.*

TC WORKS

POWERCORE 1.5 (MAC/WIN)

Take a load off your CPU with this powerful DSP card and plug-in bundle.

By Len Sasso

TC Works' PowerCore is a PCI card incorporating a Motorola PowerPC CPU and four Motorola 56362 digital signal processors. The PowerCore hardware comes bundled with PowerCore 1.5, a suite of six digital signal processing (DSP) effects plug-ins, and a mono software synthesizer modeled after the famous Roland SH-101. The included effects are primarily aimed at pro-quality mixing and mastering, freeing your CPU to run more specialized, native effects. Several third-party providers—including Antares, Sony, and Waldorf—have announced upcoming PowerCore versions of their effects.

You need a reasonably modern computer that accommodates full-size PCI cards to use PowerCore, but you don't need a tremendously fast CPU—the whole point of PowerCore is to extend the audio-processing capacity of your system. PowerCore is not an audio interface; it simply passes audio back and forth to your plug-in host application, which can be any software that supports VST or MAS plug-ins. (MAS support is Mac only. The Macintosh version includes TC Works' Spark LE sample-editing software, which can be used as a host.) Audio quality depends on your system and audio interface, but PowerCore's processing remains entirely in the digital domain and supports full 24-bit, 96 kHz systems.

For this review, I installed the PowerCore card in a Macintosh G3/300 MHz and used Emagic's Logic Audio Platinum 4.8.1 and TC Works' Spark XL 2.02 as VST hosts. Although the minimum requirements specify OS 9.0, I had no problems using PowerCore with versions 8.6 and 9.1.

HOW IT WORKS

Once you have the PowerCore card and its software drivers installed, your only interaction with PowerCore is through the PowerCore plug-ins, which are installed like any other VST or MAS plug-ins. Also, like other plug-ins, you can copy the PowerCore plug-ins for use in different host applications.

FIG. 1: The MegaReverb control panel provides graphic setup for most reverb parameters. The left strip handles mixing and stereo balance; the middle strip controls room shape, size, and diffusion; and the right strip controls filtering, predelay, and reverb-tail characteristics.

After a cold start of your computer, launching any host application and inserting a PowerCore plug-in initializes the PowerCore card, as evidenced by a fuel gauge that indicates the 20-second lag during which the PowerCore starts up. Thereafter, until you power down your computer, the operation of the card is mostly transparent, and the plug-ins operate just like native plug-ins that run on your computer's CPU. I say "mostly" because there are several differences of which you need to be aware.

Even though the PowerCore hardware is doing the heavy lifting, there is some drain on your computer's CPU as it performs the processing necessary to manage the audio transfer to and from the PowerCore hardware. The PowerCore 1.5 drivers, which run on your computer's CPU, have been greatly improved to minimize that drain, but it is still apparent. One thing that affects CPU drain is the audio-buffer size. That is usually set in your host application's I/O preferences, and larger buffer sizes result in lower CPU drain.

Shifting data to and from the PowerCore hardware also takes time, as does the PowerCore DSP. That can cause audio tracks using plug-ins to be slightly delayed relative to unprocessed tracks, and unfortunately, the delay gets worse as the audio-buffer size is increased. Because latency can also be a problem with native effects, many host applications have a built-in delay-compensation option. For cases in which that is not available or satisfactory, TC Works provides a native plug-in called TC Compensator that can be used to delay all unprocessed tracks. Being native, TC Compensator uses no PowerCore resources but adds slightly to your computer's CPU load.

Finally, all PowerCore plug-ins have a No Latency mode. In No Latency mode, data passed from the audio buffer is processed immediately, forcing your CPU to communicate continuously with the PowerCore hardware. Needless to say, that adds considerably to CPU drain, but it is ideal for situations, such as live performance, in which latency is less tolerable. In short, the latency problem can be addressed in several ways, each with its own advantages and disadvantages. PowerCore does a lot of the work, but there is no free lunch.

The way your system determines how

Minimum System Requirements

PowerCore

MAC: G3/233 (with PCI slots); 128 MB RAM; OS 9.0; compatible VST or MAS host application

PC: Pentium III/300; 128 MB RAM; Windows 98/2000/ME/XP; compatible VST host application

many plug-ins can be used is also different for PowerCore than for native plug-ins. In the native environment, you add plug-ins until your CPU chokes. On the PowerCore card, each plug-in takes a fixed amount of one of PowerCore's four DSP chips. For example, a single DSP chip can run two MegaReverbs, three EQsats, or four stereo or six mono Vintage CL compressor/limiters. (You can mix plug-ins on a single DSP chip.) The allocation of plug-ins to the four DSP chips is managed automatically, but it is necessary to be aware of the total plug-in count and to monitor native CPU usage (for which most host applications provide visual gauges).

To arrive at a rough benchmark, I compared the CPU usage for eight tracks of audio with no plug-ins (15 percent) with that for eight PowerCore plug-ins (25 percent). The eight PowerCore plug-ins used up the four DSP chips on the PowerCore card. No native plug-ins exactly match the quality and functionality of the PowerCore plug-ins, but eight native plug-ins of similar function required a 75 percent CPU load.

POWERCORE 1.5 PLUG-INS

PowerCore's software bundle of six effects and a mono synthesizer includes a trio of mixing and mastering effects known collectively as TC Tools. This set contains a high-end reverb (MegaReverb), a multipurpose delay (Chorus/Delay), and an EQ (EQsat) with high and low shelving and three parametric bands. The other three effects are dynamics processors with individual features designed for specific tasks: VoiceStrip for vocal processing, Vintage CL for emulating analog compression, and MasterX3, a 3-band compressor/limiter/expander designed primarily for mastering.

TC Tools of the trade. TC Tools was the original PowerCore bundle, and it covers the standard bread-and-butter mixing jobs. One DSP chip can handle two reverbs, two delays, or three EQs. Using all four chips, you can have four complete TC Tool kits. The obvious omission is compression, and PowerCore 1.5 includes three dynamics processors.

FIG. 3: MasterX3 is a 3-band compressor. Optional dither, limiter look-ahead, and digital-ceiling controls make it ideal for mastering and finalizing. Its Target Curve scaling system greatly simplifies compressor, expander, and limiter setup.

MegaReverb (see **Fig. 1**) is a true stereo reverb with flexible room-design, time, filtering, and mixing controls. It sounds as good as any I've used, and given the amount of flexibility it provides, it's surprisingly easy to set up. By tweaking the presets (all of the usual candidates are offered), I was always able to achieve the sound I wanted very quickly.

FIG. 2: VoiceStrip is a complete voice-processing channel strip in one plug-in. It includes noise gating, de-essing, 3-band equalization, and compression. The effects can be switched in and out independently, and the order of EQ and compression can be reversed.

Room characteristics are managed using graphic windows labeled Shape, Size, and Wall Diffusion. Six shapes are provided: Hall, Horseshoe, Prism, Fan, Club, and Small—each incorporating the mathematical characteristics of a specific room. (Fan, for example, is based on the structure of La Scala, the famous Italian opera house.) Each room can be scaled from 0.04 to 4 times in size. The room's reflective surfaces (diffusion) can be scaled from –50 percent (tile and brick) to +50 percent (cottage cheese) with 0 representing the original room's diffusion characteristics. The room design can be heard most clearly in the early-reflection portion of the reverb.

Prefiltering, pre- and postdelay, and the reverb tail are controlled by the three graphic windows at the right in **Fig. 1**. You use the Highcut Filter window to set the cutoff and slope of a stereo lowpass filter that comes first in the signal path. The Decay/Frequency window controls the decay times of the reverb tail in three frequency bands. There is independent control of the bandwidth and decay time for each frequency band.

The Predelays window sets both the time (ranging from 0 to 160 ms) before the early reflections begin and the time (from 0 to 100 ms) between that and the beginning of the reverb tail. That is also where the early-reflections and reverb-tail levels are set (over a range of 0 to –∞ dB).

The mixing controls include input and output level, wet/dry mix, and separate left-right balance for the early reflections and the reverb tail. You can also set the stereo width from mono to full stereo, and for each channel, you can separately control whether the early reflections are crossfed into the other channel.

Chorus/Delay is another good-sounding, easy-to-set-up plug-in. Its graphics have the same look and features as MegaReverb, and starting from the presets, you can tweak your way to victory in short order. It is somewhat

limited in its off-the-wall-effects potential, but for general feedback-delay purposes (chorus, flanging, echo, and so forth), it is fully capable.

Like MegaReverb, Chorus/Delay starts with a lowpass filter, but in this case, the raw signal can be routed around it. That is what you want when you use the chorus as an insert effect, because it allows the source signal to be heard without any processing. In the context of the delay effect, you may or may not want the raw signal in the mix. The chorus section is a short delay line (40 ms maximum) with an LFO for modulating delay time. In addition to delay time, the Chorus/Delay has controls for LFO speed and depth, LFO phase between the right and left stereo channels, and feedback amount and filtering. The chorus section can produce anything from standard chorus and flanging to highly resonant feedback effects. Its one limitation is that only a sine LFO waveform is provided, and that can't be synchronized to MIDI tempo.

The delay section features longer delays (as long as 500 ms), and conveniently, the delay time can be set in beats per minute (120 bpm, for example, is the equivalent of 500 ms, or one beat). Unfortunately, like the chorus, the delay time cannot be synchronized to MIDI tempo.

The delay parameters are as simple as it gets—delay time, feedback-filter shape, and feedback amount. The feedback filter is the same for the delay and chorus sections. It is a bandpass filter for the signal being fed back to the input. You can set the filter's upper and lower band limits independently. (In other words, you can control the bandwidth.) Chorus/Delay will fulfill all the requirements of a basic feedback-delay line, but the absence of separate right- and left-channel delay times with cross feedback does limit its range of effects.

FIG. 4: PowerCore 01 is a lead synth patterned after the Roland SH-101. It is monophonic, but the PowerCore card can run as many as 12 of them. You can choose from four front-panel designs; the most distressed, called Ashes, is shown here.

One of the more interesting uses for Chorus/Delay is as a dual resonator. Because both the chorus and delay sections are capable of very short delay times and the feedback filters can pick out different frequency bands for feedback, you can resonate two separate bands of the input signal. MP3 example_1, at www.emusician.com, illustrates that effect on a bell-like sound effect.

The third TC Tool, EQsat, is a 5-band equalizer with shelving filters on each end and three bands of parametric equalization in the middle. Actually, each section's frequency is free ranging, which lets any of the bands overlap. The parametric bands are bell shaped with a gain range of ±18 dB and selectable bandwidth from 0.10 to 4 octaves. The shelves can slope from 3 to 12 dB per octave, up or down. A soft-saturation (nonlinear distortion) stage after each filter segment adds warmth by simulating analog-tube distortion.

Voice mastering. PowerCore 1.5 comes with three dynamics-processor plug-ins. CL, the most basic, emulates an analog compressor/limiter and features soft- or hard-knee compression and full-range limiting with optional soft saturation. That is PowerCore's workhorse compressor, and you can get six mono or four stereo CLs on one DSP chip.

VoiceStrip is, as its name implies, an all-in-one voice-processing plug-in (see **Fig. 2**). It starts with a low-cut filter followed by a noise gate, which is in turn followed by equalization and compression stages. The switch that is provided for reversing the order of EQ and compression is a nice touch. VoiceStrip's last stage is a de-esser. VoiceStrip is truly the Swiss Army knife of voice processing, and there are lots of knobs and switches to deal with. But the controls for each section are straightforward, and the provided presets cover all the bases.

PRODUCT SUMMARY

TC Works
PowerCore 1.5 (Mac/Win)
plug-in accelerator card and software
$1,299

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

RATING PRODUCTS FROM 1 TO 5

PROS: Top-quality effects plug-ins in all the basic categories. Greatly reduces CPU load.

CONS: Limited range of compatible plug-ins currently available.

Manufacturer
TC Works
tel. (805) 373-1828
e-mail info@tcworks.de
Web www.tcworks.de

Having multiple VoiceStrip modules in your kit—you can fit two of them on each DSP chip—gives you a lot of flexibility in multitrack voice mixes. For example, slightly different EQ and compression settings are often all you need to bring several mismatched background tracks into line. MP3 example_2 illustrates another approach to using multiple VoiceStrips, exaggerating different characteristics in a multipart laugh track.

The last plug-in, MasterX3, is a 3-band expander/compressor/limiter based on TC Works' renowned Finalizer hardware device (see **Fig. 3**). It was a temporary part of the PowerCore 1.5 bundle and, as of March 2002, is sold separately for $249. (A 5-band version is also available for $499.) So by the time you read this, MasterX3 may not be in the packages you find on your music dealer's shelf.

MasterX3 is intended primarily as a mastering tool and includes optional output dithering at anywhere from 8 to 22 bits. The limiter circuit features look-ahead delay and the option to lower the output ceiling by as much as

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0.1 dB. Using the Frequency/Levels graphic at the top center of **Fig. 3**, you can make adjustments to the crossover frequency and gain for the three bands extremely quickly.

Although MasterX3 operates on three frequency bands, it uses a clever system of Target Curves to simplify the compressor, limiter, and expander settings. Each of those modules has a single set of parameters that are then scaled to fit the Target Curve for the three frequency bands. Four Target Curves are provided: Linear, Pink (less high-band processing), Hyped (more high-band processing), and Smiley (more low- and high-band processing). The amount of scaling for each process (compression, limiting, and expansion) is controlled by a Target Factor setting for that process. It sounds complicated, but is a lot simpler than setting three complete sets of compression, limiting, and expansion parameters. MP3 example_3 applies three mastering settings during the course of a short dance mix.

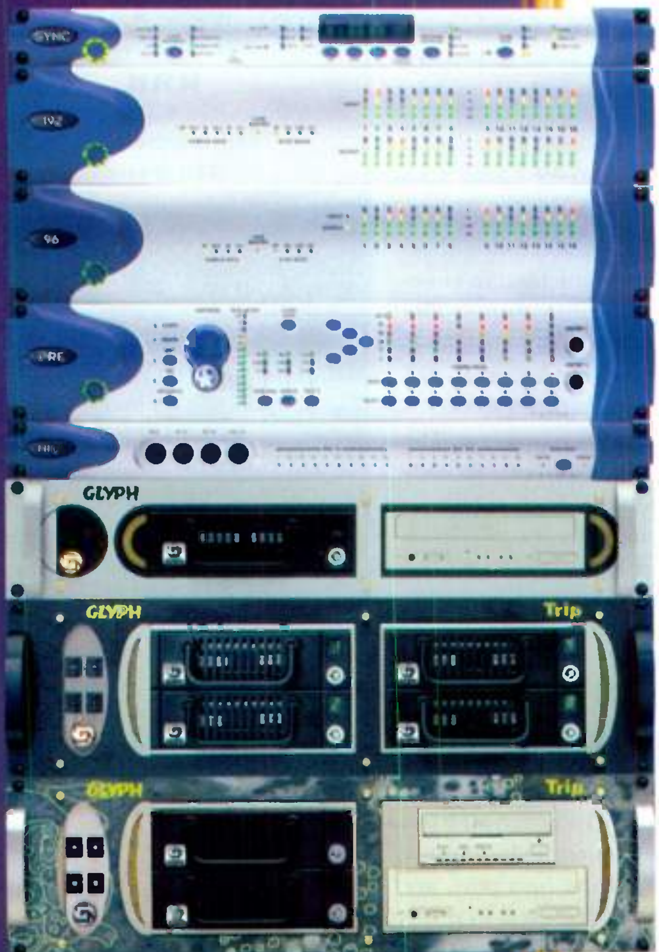
Taking the lead. For good measure and, presumably, to amuse you when you get tired of tweaking effects settings, TC Works has added a lead synth to the PowerCore 1.5 bundle. PowerCore 01 is modeled after the Roland SH-101 and is TC Works first VST Instrument. **Fig. 4** shows PowerCore 01 in one of its four skins.

PowerCore 01 is a monophonic synth with a single mixed-waveform oscillator, a resonant lowpass filter, an LFO with square or triangle waveshape, and a single ADSR envelope generator for amplitude control. The filter cutoff frequency can be modulated by a mix of the LFO, envelope generator, and MIDI note pitch. For leads and basses, PowerCore 01 offers a lot of bite, a classic analog-synth sound, and very low latency, especially when run in No Latency mode. (It was playable without No Latency mode enabled.) You can get 3 synths on a DSP chip, so if you want to dedicate your PowerCore card to it, you can have 12 synths running at once.

GET PLUGGED IN

The PowerCore hardware will substantially expand the audio-processing capacity of your computer, and the PowerCore 1.5 software bundle contains great-sounding, top-quality effects in all of the standard categories. With the TC Works offerings growing and third-party PowerCore plug-ins on the way, it is likely that more esoteric processing applications will appear in the future. But even without that, getting the load of everyday processing off your CPU frees up processing power for your favorite native plug-ins.

At $1,299, the PowerCore will put a dent in your budget, but considering that several professional-quality native plug-in bundles cost as much, the price seems well justified. If you do a lot of multitrack processing with plug-ins in the categories included in PowerCore 1.5, it is definitely worthy of serious consideration. ⊕



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KRK

V-4

Surprisingly big, accurate sound from a pint-size active monitor.

By Michael Cooper

Size matters, especially when you're short on space—hence the appeal of compact close-field monitors. The problem, though, is that small monitors tend to have severely limited bass-frequency response. That was one of my main concerns when I agreed to review the new KRK V-4 monitors, which have woofers only 4 inches in diameter. But a quick glance at the specifications quelled my apprehensions somewhat: the V-4's frequency response is rated down to 65 Hz—almost as low as some models using 6-inch woofers. Those who require an accurate assessment of what's going on below 65 Hz can always add a subwoofer to a V-4 setup. KRK offers three models: the S8 ($749.99), S10 ($1,199.99), and S12 ($1,499.99).

The KRK V-4 offers a surprisingly big sound in a cabinet with a small footprint.

ABOUT-FACE

The V-4 is a two-way, biamplified, active monitor that can be used in close-field or midfield applications. The monitor is magnetically shielded, making it especially suitable for small, digital audio workstation–based studios.

Construction quality is first-rate. The V-4 cabinet feels solid and reassuringly heavy at nine pounds. Each cabinet is constructed of internally braced, ½-inch medium-density fiberboard (MDF) and finished in an attractive gray fingerprint-proof material called Zolatone.

Two custom-made drivers are used in each cabinet. The 4-inch woven-Kevlar woofer features a high-sensitivity voice coil. A 1-inch fabric-dome high-frequency driver is inset in a smoothly curved wave guide. Moreover, all front and side cabinet edges, including the front-firing bass port, are heavily radiused to minimize diffractive effects that would otherwise compromise imaging and high-frequency linearity.

AROUND BACK

Hooking up the V-4s is a cinch. A Neutrik combo connector on the rear panel accepts balanced as well as unbalanced signals. That means that you can patch the control-room outputs of your mixer to the V-4s using cables fitted with ¼-inch TS or TRS connectors or with XLR connectors (pin 2 is hot).

The rear panel has a continuously variable sensitivity trim pot that allows for as much as 6 dB of speaker gain or as much as 30 dB of attenuation. That trim control, which comes after the fixed 28.7 dB of amplifier gain, can be adjusted with a small slot-head screwdriver to optimize output levels for the control room. Owners of surround-sound studios will especially appreciate being able to fine-tune each V-4's output level.

The included power cable, roughly seven-and-a-half feet long, attaches to an IEC receptacle also located on the rear panel. When you switch the V-4's rear-panel power switch to the on position, a yellow LED on the front cabinet face lights to remind you the juice is flowing.

The V-4's active circuitry is a crucial part of this speaker's success story. Separate 15W and 30W amplifiers power the V-4's high- and low-frequency drivers, respectively. Three active filters are employed in the design, including 12 dB/octave highpass and lowpass Butterworth filters tuned to a 1.7 kHz crossover. The third filter is a

PRODUCT SUMMARY

KRK
V-4
compact active monitors
$399.99 each

FEATURES	4.0
EASE OF USE	5.0
AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Warm, nonfatiguing sound. Superior reproduction of transients. Very good imaging and depth of soundstage. Compact. Affordable. Magnetically shielded. Neutrik combo connectors allow quick, hassle-free setup. Sensitivity trim pot permits output-level calibration for each monitor.

CONS: Upper-bass and low-mid frequencies sound slightly boomy and full, resulting in a somewhat veiled sound. No high-frequency "tilt" adjustment. Weak response in low-bass region.

Manufacturer
KRK Systems/Stanton Magnetics
tel. (714) 373-4600
e-mail sales@krksys.com
Web www.krksys.com

WRH

12 dB/octave subsonic filter that provides a –3 dB down point at 32 Hz. Thankfully, no signal compression is used in the electronic circuitry.

The V-4 employs a toroidal power transformer to minimize hum, and the monitors are some of the quietest active monitors I've heard. They're also quite efficient—the drivers are rated 90 to 91 dB SPL at 1W and 1 meter.

TRACKING DUTIES

I began testing the V-4s in a stereo monitoring setup by tracking drums and electric bass guitar. For this application, I had the V-4s positioned on top of Acoustic Sciences Corp. (ASC) Monitor Traps. The Monitor Traps are part of ASC's Attack Wall acoustical system, which I use at the front of my control room.

The V-4s were plenty loud with my Yamaha 02R's control-room monitor pot turned up only about a third of the way. I was amazed by how much bass the little speakers pump out. Although they are just half the size of Yamaha NS10Ms, the V-4s sound quite a bit bigger.

Drums sounded articulate and tight except for a little blurriness in the upper-bass and low-midrange regions. The bass guitar and drums sounded a tad flabby in the upper-bass region but otherwise natural. Although I could clearly distinguish the pitches of bass notes down to low E, the V-4s could not reproduce the bottom-most octave in the audio spectrum (20 to 40 Hz). Note, however, that most other two-way, close-field monitors are also deficient in this range, so this is not a criticism specifically of the V-4. Again, should you need a truer picture of what's going on down in the thunder zone, you can always add a subwoofer.

Overall, I was impressed that I heard no holes in the midrange response while tracking with the V-4s. Also, the V-4's transient response proved outstanding. I wish that the monitors had a tad more zing in the top octave of the audible spectrum, but that is more a personal preference than an indictment.

MIXING IT UP

I also mixed some rock and country projects using the V-4s, and I listened critically to several mixes I'd already finished. Initially, I mixed with the V-4s sitting on my ASC Monitor Traps; later I moved the monitors to the top shelves of my Omnirax 02R MixStation—a positioning more akin to meter-bridge placement.

Placed on the Monitor Traps, the V-4s sounded a tad boomy and veiled in the 150 to 300 Hz zone. But imaging, depth of soundstage, and especially the reproduction of transients were good. The V-4s are not even slightly fatiguing to listen to during the course of long sessions.

As expected, moving the V-4s to shelftop placement hyped the monitors' reproduction of upper-bass and low-mid frequencies. Placing any monitors on lightweight shelves produces less-than-optimal coupling, which, in turn, can lead to flabby bass and blurry mids. Overly bright monitors, such as Yamaha NS10Ms, can endure some added low-end muddiness and still be quite revealing; the V-4s, however, sound much warmer than NS10Ms and thus offer less latitude in regard to how placement causes them to couple to furnishings and to your room.

For the tightest bass, clearest mids, and best imaging possible with the monitors, place the V-4s on substantial monitor stands. With the monitors situated in that fashion and with the addition of a subwoofer, you should be able to get terrific results with the V-4s, once you've learned their subtle colorations.

IMPRESSIVE RUNTS

The KRK V-4s are a good choice for anyone looking for warm, nonfatiguing, big-sounding active monitors with small footprints. The pint-size units transcend the usual shortcoming of small cabinets—insufficient low end—to produce an amazingly extended frequency response. In fact, I've never heard monitors this small sound so good. At $400 apiece, the V-4s are a good value, as well.

V-4 Specifications

AUDIO	
Frequency Response	65 Hz–20 kHz (±2 dB)
Peak Output	104 dB (@ 1 meter)
AMPLIFIER	
Power Rating	15W (high frequency); 30W (low frequency)
Signal-to-Noise Ratio	>90 dB
Input Impedance	10 kΩ balanced input
Total Harmonic Distortion	<0.05% (@ full output, 1 kHz)
Gain	28.7 dB (fixed)
Input Sensitivity Range	+6 dB to –30 dB
Power Consumption	60 VA
SPEAKERS	
High-Frequency Driver	1" fabric dome
Low-Frequency Driver	4" woven Kevlar
Input Connectors	Neutrik combo
CROSSOVER	
Crossover Frequency	1.7 kHz
Crossover Slope	12 dB/octave Butterworth (LF and HF)
Subsonic Filter	–3 dB @ 32 Hz, 12 dB/octave
ENCLOSURE	
Material	½" MDF
Dimensions	6.00" (W) x 9.25" (H) x 7.75" (D)
Weight	9 lb. (each)

RME

HAMMERFALL DSP

A flexible, low-latency multichannel audio-interface system.

By Brian Smithers

German manufacturer RME attracted a lot of attention a couple of years ago when it released the Project Hammerfall audio interface, a PCI card that combined boatloads of digital I/O with exceptionally low latency. Its follow-up, the Hammerfall DSP system, takes a modular approach to interface design while building on the strengths of the original.

The Hammerfall DSP system comprises two computer cards and two half-rackspace breakout I/O boxes, letting you configure the system as you see fit by combining the appropriate components (see **Fig. 1**). The PCI card works with any desktop Mac or PC with PCI 2.1 slots. A CardBus interface that provides the same functions as the PCI card is available for laptops. Whether you choose the PCI or the CardBus interface, you extend a FireWire cable to your choice of I/O boxes. (The CardBus cable uses a proprietary pin-out assignment.)

The Digiface I/O box provides two ports (32 channels) of MIDI I/O, three ports (24 channels) of ADAT Lightpipe I/O, one port (2 channels) of AES/EBU-compatible S/PDIF coaxial I/O, word-clock I/O, ADAT 9-pin sync, and a front-panel ¼-inch headphone output. With that much connectivity, I'm hard-pressed to find any fault, but I still wish the headphone output had a dedicated volume knob. The other I/O box, the Multiface, replaces two banks of Lightpipe with eight 24-bit, 96 kHz A/D inputs and D/A outputs on ¼-inch TRS connectors. It also eliminates one MIDI port.

You may be wondering why you give up two ADAT ports to get 8 analog channels. It's because 16 channels at Lightpipe's maximum 24-bit, 48 kHz resolution requires the same bandwidth as 8 channels at 24-bit, 96 kHz resolution. Like a number of current 96 kHz devices, the Hammerfall DSP series provides half as many channels at 96 kHz as it does at 48 kHz. The ADAT ports on the Digiface and Multiface perform "sample splitting" to squeeze four 24-bit, 96 kHz channels through a pipeline originally meant for eight 24-bit, 48 kHz channels. (Besides, the Multiface has no room for any more connections!)

TURN THIS DRIVER OUT

RME shows its concern for driver implementation in an unusual way. A portion of the Hammerfall DSP's ASIO driver is actually embedded into an EPROM chip on the card itself. (That's true of the original Hammerfall, as well.) Because the purpose of a driver is to handle communication between software and hardware, this arrangement takes some strain off the CPU and reduces latency. An ASIO system can often achieve latencies as low as 6 to 8 ms. However, RME's resourceful design can reduce latency to 1.5 ms—close to the theoretical minimum.

If you're looking for an audio interface to use with the many Mac and PC ASIO 2.0–compatible applications, the Hammerfall DSP is very attractive. RME has not left Windows MME applications

PRODUCT SUMMARY

RME
Hammerfall DSP
digital-audio interface
PCI interface card $315
CardBus interface card $355
Digiface digital I/O box $650
Multiface A/D I/O box $860

FEATURES	5.0
EASE OF USE	4.5
AUDIO QUALITY	4.5
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Flexible modular design. Support for high-resolution audio. Low-latency operation. Word-clock I/O. ADAT sync. Can cascade multiple units. Powerful software mixer applet. Good notebook support.

CONS: Lowest latencies are highly host-dependent. No front-panel headphone volume knob.

Manufacturer
RME Intelligent Audio Solutions/
X-Vision AudioUS (distributor)
tel. (330) 747-3857
e-mail info@xvisionaudio.com
Web www.rme-audio.com

FIG. 1: The Hammerfall DSP system offers a choice of two I/O boxes: the Digiface and the Multiface. Both provide significant connectivity, with the Multiface replacing a couple of Lightpipe ports with ¼-inch TRS analog I/O.

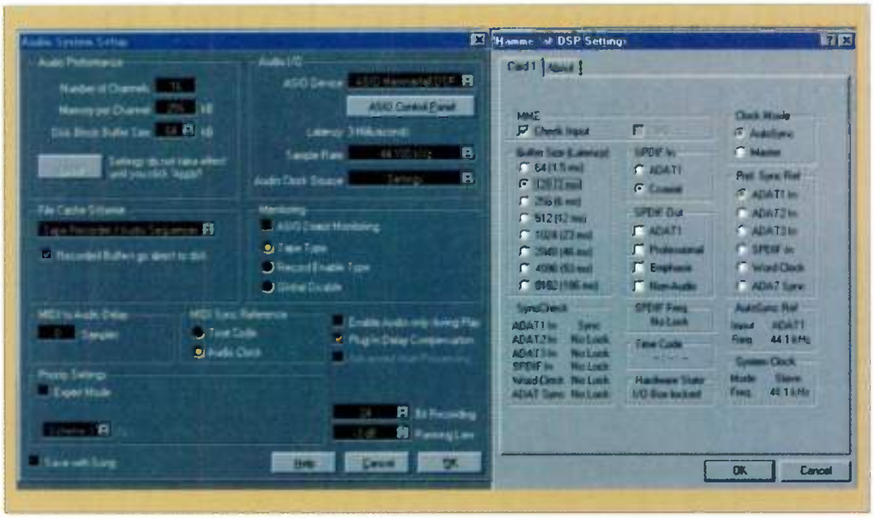


FIG. 2: The Audio System Setup window in Steinberg Cubase VST/32 5.0 (left) provides access to the Hammerfall DSP's settings (right).

out in the cold, though. The Hammerfall DSP's MME driver can reach the same low latencies with compatible applications running under Windows ME/2000/XP. Note that *can* is the operative word, and RME disavows any guarantee of extremely low MME latencies. RME plans to develop WDM streaming drivers in the future, though no target date has been announced as of this writing. Those drivers are intended to make low-latency operation much more achievable under Windows ME/2000/XP.

Low-latency GSIF drivers for use with Tascam's GigaStudio are included, as well. In addition, Linux drivers are under development for the Hammerfall DSP system and may be available later this year.

GET IT STARTED

I installed the Hammerfall DSP CardBus interface in a 1 GHz Celeron notebook with 256 MB of RAM and running Windows XP. I installed the Hammerfall DSP PCI interface in a Pentium III/450 MHz desktop computer with 128 MB of RAM and running Windows 98SE. I swapped a Digiface between the two systems, using RME's ADI-8 Pro A/D/A converter as a front end. I begged and pleaded for a Multiface, but at review time, there was a significant waiting list. (RME expects that to be resolved by the time you read this.)

Installation was a breeze on the notebook computer and a minor annoyance on the desktop. However, the Hammerfall DSP's U.S. distributor, X-Vision Audio, patiently and accurately guided me to a resolution. Score one for customer support.

Once I had the hardware properly installed, Steinberg Cubase VST/32 5.0 and Emagic Logic Audio Platinum 4.8 recognized it and let me configure it in the normal ways (see **Fig. 2**). Cakewalk Sonar 1.3, however, was a bit more of a challenge, because at first it didn't recognize the card. An e-mail to technical support quickly brought a suggestion to tell Sonar to always use MME drivers, which immediately solved the problem.

Another occasional annoyance was that Sonar kept telling me that the Hammerfall DSP was "not compatible with current settings or in use by another device" when I started the program or made changes to the audio setup. That response proved to be

GETTING CONVERTED

Adrift in a veritable sea of digital inputs and outputs, I needed a good set of converters to connect my analog world. RME was kind enough to send me its ADI-8 Pro ($1,750), which is an 8-channel, 24-bit, 48 kHz analog/digital interface. The ADI-8 Pro features eight analog inputs and outputs on balanced ¼-inch TRS connectors, two ADAT Optical I/O ports, two TDIF-1 ports, and word-clock I/O. The analog inputs are also available through 25-pin D-sub connectors.

A lot of thought went into the design of the ADI-8 Pro. The analog, ADAT, and TDIF-1 outputs are all active simultaneously, and in normal operating mode, the secondary ADAT and TDIF-1 outputs mirror the primaries, which lets you use the ADI-8 Pro to feed as many as four digital devices at the same time.

The real purpose of the extra digital ports, however, is to support *bit splitting*, a way of remapping each channel's 24 bits across the space available on two channels of a 16-bit device. On playback the ADI-8 Pro recombines the split signal into a single 24-bit output. Four input channels split that way would use all eight channels of a single ADAT/TDIF port, so the extra ports are needed to transfer channels 5 through 8 to a second bank of eight 16-bit channels.

All of the unit's functions are selectable using front-panel switches that have clearly labeled status LEDs. Two-stage input and output level meters are situated on the front panel, as well. Input and output levels are independently selectable between –10 dBV and +4 dBu.

The ADI-8 Pro is easy to use and sounds great. Its feature set might be overkill for some users, but for others its flexible digital I/O is just what the doctor ordered.

erroneous, because all I had to do was pat the computer reassuringly and click on Use Anyway, and all was well. One of the few ways in which RME's support disappointed me was that its configuration tips for Cakewalk were for Pro Audio 7—three versions back.

With a desktop computer, the Digiface is powered through the FireWire cable, but when used with a notebook, it requires external power. A CardBus connection simply doesn't provide enough voltage, so that interface ships with three power supplies. First is the traditional lump in the line, which is always preferable to the dreaded wall wart. Kudos to RME for making the transformer light and small, giving it a power-present LED, and putting it in the middle of a 12-foot cord.

Showing real attention to the needs of mobile musicians, RME also includes a 12 VDC adapter, which plugs into an automobile cigarette lighter, and another cable to be used with—believe it or not—a rechargeable battery. Each cable is a healthy 10 feet long. Go to RME's Web site and check out the article "HDSP System: Notebook Basics—The Audio Notebook in Practice" to see some thoughtful design at work.

HAMMER TIME

So how does it sound? The Hammerfall DSP's all-digital I/O doesn't "sound" like anything. Okay, the headphone output sounded perfectly fine, clear, and quiet. Although it wasn't officially part of the review, I was pleased with the sound of the ADI-8. Its front-panel controls couldn't be easier to use, and I like the fact that it has a schematic on the top panel. (For more on the ADI-8, see the sidebar "Getting Converted.") I can't vouch for the Multiface, but if it lives up to its published specs, it should be a worthy contender.

The question, then, is how about that latency? The long and short of it is that the Hammerfall DSP can indeed achieve negligible latency on a properly configured machine under the right circumstances. During my testing, I was sometimes able to operate at the system's lowest buffer size, and though I didn't do any bench testing to confirm the 1.5 ms claim, it was as close to real time as it needs to be.

Maintaining those ideal circumstances, however, proved to be difficult. Even though I run a pretty lean configuration, the audio performance at the lowest latencies was not consistently clean.

The FireWire cables packed with the PCI and CardBus cards are 14 feet long.

Occasionally, the sound broke up—sometimes in subtle ways and other times in gross ways. Bumping up the buffer size made the crackles and distortion go away but at the price of slower response.

Here are some things to keep in mind if you're going for those single-digit latencies. Using the ASIO drivers under Cubase and Logic Audio yielded somewhat better results than using the MME drivers under Sonar. Nevertheless, it's surprising that RME is able to get MME latencies in the same ballpark as ASIO. That's the job the newer WDM drivers are supposed to do.

Another thing to keep in mind is that plug-ins eat up buffers. If you want to monitor with reverb and EQ, you may have to settle for higher latency. On my machines, 1.5 ms was pretty much out of the question if any effects were running, but 3 ms was sustainable with a reverb. Of course, monitoring with plug-ins is the whole point of low-latency audio interfaces. If you're not going to sweeten the cue mix with some reverb, why not just use the direct hardware monitoring available on most interfaces?

Likewise, the more tracks I recorded, the higher I needed to set the buffer. Exactly how many tracks and how high the latency was varied from program to program and from desktop to laptop. Suffice it to say that even though I'm impressed with the Hammerfall DSP's performance, I'm unlikely to subject

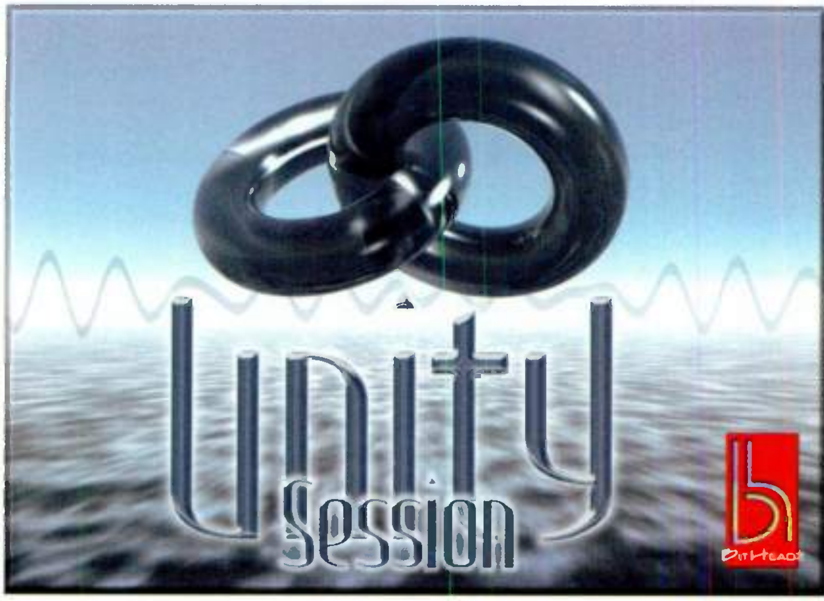
VINTAGE HAMMER

The original Project Hammerfall, aka the Digi 9652, is a great way to get most of the positive attributes of the Hammerfall DSP system for a few bucks less. For $665 you get a PCI card with all of the Digiface's audio I/O built in. The only connections you give up are the MIDI ports. To make room for all of the connectors, you need a second space on the back of your computer but not a second PCI slot. The second space houses the third ADAT port and the word-clock I/O. If you don't need those, you can save $90 by buying the Hammerfall Light, which is the PCI card without the extra connections.

What do you lose by pinching pennies? Aside from the DSP system's modular design and the convenience of placing your audio connections 14 feet from your CPU, you miss out on the TotalMix application, the aforementioned MIDI ports, and what RME calls *dynamic bus utilization*. That's designed to improve PCI efficiency in the Hammerfall DSP, and on my desktop, the original Hammerfall was slightly less happy with the lowest latency settings than the Hammerfall DSP was. It was not a huge difference, though—far from a disqualifier for the original Hammerfall.

Having worked out my installation bugaboos on the Hammerfall DSP, I was able to install the Project Hammerfall card in a snap. Its configuration utility is almost identical to its sibling's, and configuration within Cubase VST and Sonar is just as simple. For anyone with lots of Lightpipe gear to connect, the Project Hammerfall and the Hammerfall Light are well worth a look.

Obsession.



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BitHeadz

a client to the latency–versus–audio-quality trade-off any time soon. For the foreseeable future, I'll run at larger buffer sizes and rely on monitoring through a console or through direct hardware monitoring.

Direct monitoring with the Hammerfall DSP is made a lot nicer by the inclusion of an applet called TotalMix (see **Fig. 3**). It's a 1,456-channel (that's right, 1,456) software mixer that provides almost limitless flexibility in moving signals between hardware inputs, software outputs, and hardware outputs. You can create multiple headphone mixes, pass a live stereo reference mix to your DAT while you're multitracking, and do a whole lot more.

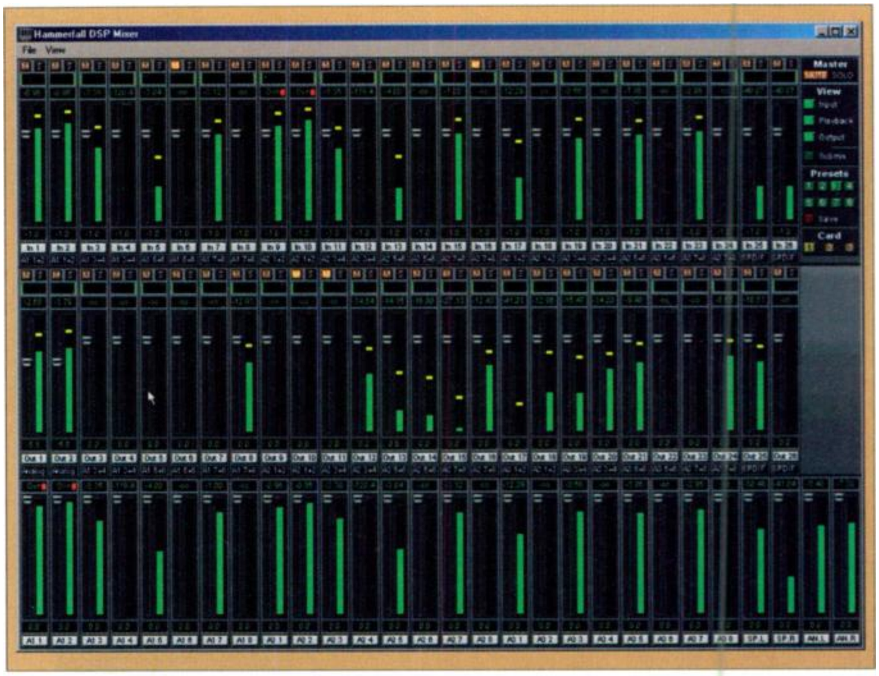

FIG. 3: TotalMix is a highly flexible software mixer that lets any Hammerfall DSP input or software channel be routed to any output.

CAN'T TOUCH THIS

Once the hardware was successfully installed, I had almost no problems with the Hammerfall DSP system, though my notebook didn't like seeing the CardBus card without the Digiface attached. It let me know by freezing and crashing for no apparent reason. Because CardBus cards are hot-swappable, I've developed the habit of stopping the device and removing it if I need the Digiface for my desktop.

RME deserves credit for its attention to detail. For example, the FireWire cables packed with the PCI and CardBus cards are 14 feet long. When was the last time you were able to sit back and ask, "Where is the most convenient place to put my breakout box?" Cables as long as 33 feet are available, and with three of those cables and two repeater boxes, you can place the Digiface or Multiface as much as 100 feet from your computer.

The Hammerfall DSP's driver CD is date stamped and filled with drivers and manuals (in English and German) for RME's entire line of products in all supported versions of Windows. It also holds a complete version of the company's Web site, including tips and tech info. That's particularly noteworthy, because it includes some great articles such as "Tuning Tips for Low Latency Operation" and five articles about using notebooks with the Hammerfall DSP system.

The Hammerfall DSP system, though not dirt cheap, offers an economical approach because of its modular design. Someone who operates desktop and laptop systems, can buy a PCI card and a CardBus card and share a Multiface or Digiface between them. Someone who owns a digital mixer can buy a pile of digital I/O without wasting money on redundant analog I/O. For the truly power hungry, multiple Hammerfall DSP cards can be installed in the same machine. (If you're on a tight budget, you can still buy the original Project Hammerfall system; see the sidebar "Vintage Hammer.")

Even though the much ballyhooed 1.5 ms latency proved impractical on my machines, I'd be happy to own a Hammerfall DSP system. Its modularity appeals to me, as does its efficient design and quality construction. Its combination of high resolution, numerous inputs and outputs, and expandability give it staying power in these gadget-of-the-month times. ❂

Hammerfall DSP Specifications

Audio I/O (Digiface)	(3) ADAT Optical (Lightpipe); (1) S/PDIF coaxial (AES/EBU compatible); (1) ¼" headphone output
Audio I/O (Multiface)	(8) ¼" TRS analog; (1) ADAT Optical (Lightpipe); (1) S/PDIF coaxial (AES/EBU-compatible); (1) ¼" headphone output
Synchronization (Multiface/Digiface)	(1) word-clock I/O (BNC); (1) ADAT 9-pin sync
MIDI Ports (Digiface)	(1) In/Out front panel; (1) In/Out back panel
MIDI Ports (Multiface)	(1) In/Out front panel
Audio Channels	(26) In, (26) Out @ 48 kHz; (12) In, (12) Out @ 96 kHz
Resolution	16-, 20-, 24-bit; 32, 44.1, 48, 88.2, 96 kHz sampling rate
Driver Support	ASIO 2.0 (Mac/Win); GSIF (Win 98/ME)

VIRSYN

VIRSYN 1.1 (WIN)

A powerful modular synthesizer with extensive sequencing capability.

By Len Sasso

Software-based VirSyn is a 12-part multitimbral modular synthesizer for Windows. Each Part consists of a 32-voice modular synthesizer with a step sequencer, an arpeggiator, and insert effects. VirSyn offers global reverb and chorus effects, a song manager, and a 12-channel output mixer. You can control all parameters with MIDI, and a unique, eight-dimensional automation screen (with four *x-y* controllers) lets you control as many as 64 simultaneous parameters.

VirSyn is available as a VST Instrument plug-in ($149) or a standalone application, which includes the VST plug-in and a built-in WAV file recorder. The extensive MIDI and sequencing capabilities of the standalone version make it well suited to live performance. Both versions come with more than 800 factory programs in a broad range of categories. You can download a limited-time, save-disabled demo of VirSyn as well as MP3 audio examples at the company's Web site.

For copy protection, VirSyn uses a key-disk scheme with separate key disks for standalone and VST installations. The authorization process is straightforward, though some dialogs are in German, which might cause a few anxious moments. You can deauthorize the program if you want to move it to another computer. Emergency authorization disks are supplied as insurance against lost authorizations.

As with all software synthesizers, performance varies depending on your computer's processor speed, available RAM, MIDI and audio drivers, and audio hardware. VirSyn supports ASIO audio output and DirectSound. In the VST version, performance and some features also depend on the chosen VST host. I tested VirSyn on a Pentium III/700 MHz Dell laptop computer running Windows 98SE with a Tascam US-428 audio/MIDI USB interface. Under those conditions, sound quality was good and latency was low but still noticeable.

VIRTUAL MODULAR SYNTHESIS

VirSyn uses a matrix modular format. Rather than cabling modules together in a graphical user interface, you connect the various components using context-sensitive menus for the inputs and outputs on each component's control panel. Because virtually any connection that makes sense is available, VirSyn lacks the limitations inherent in a system with a fixed array of modules. Although it doesn't offer the graphic feedback provided by a cabled system, VirSyn's format has a significant advantage in that the control panels are in the same positions regardless of the synthesis configuration.

It was difficult to trace VirSyn's signal and control paths initially, but once I learned where to look, I could quickly deconstruct and modify factory presets as well as build my own patches from scratch. One especially nice feature that makes programming VirSyn easier is that the controls do not move on any module that isn't being used.

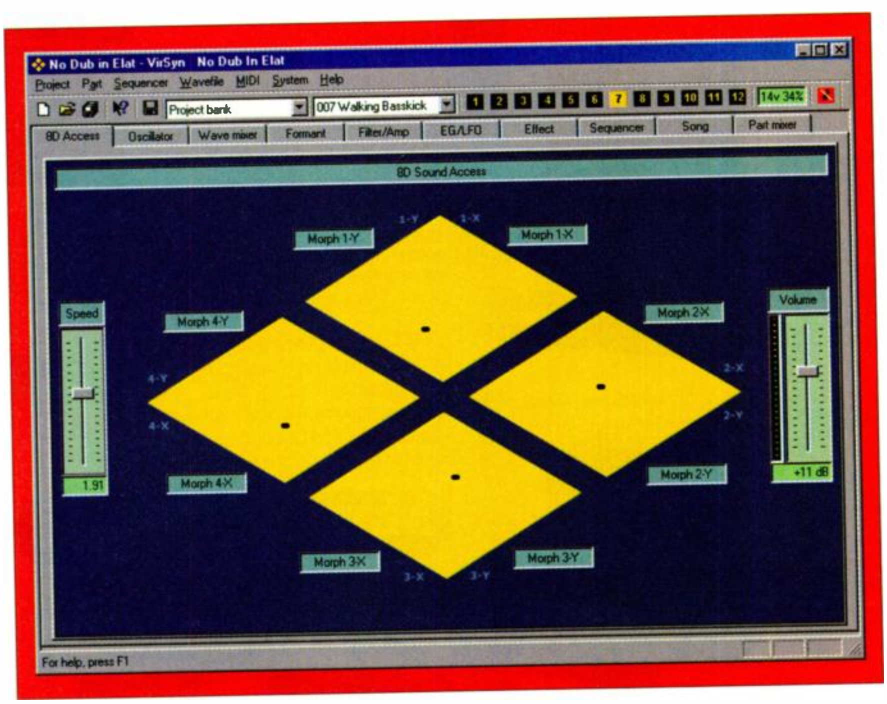

FIG. 1: VirSyn's 8D Sound Access system provides four *x-y* controllers, and each dimension controls as many as eight individual parameters. You can change settings with the mouse or with MIDI controller messages.

THE SOUND STARTS HERE

VirSyn's sound sources include four analog-modeled oscillators, a subharmonic pulse-wave generator, and white and pink noise sources. Three oscillators offer 64 waveforms with Wave Modulation, which mixes a variable-phase copy of the waveform with itself. Wave Modulation lets you produce a limitless variety of new waveforms, and you can modulate the phase differential with any control source (such as an LFO or envelope). You can set the oscillators to run free, or you can reset to the waveform's beginning with each new note, which is important for producing uniform drum sounds. You can also turn keyboard tracking on or off independently for each oscillator.

Oscillators 2 and 3 can be hard-synced to Oscillator 1, and any audio source can frequency-modulate them. The broadjam

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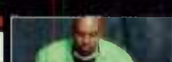
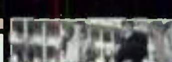
Artist: Impulse Ride **Song:** Funny
Genre: Alternative

Artist: Lenny Post **Song:** Spyder Web
Genre: Jazz



Artist: David **Song:** If I give my Heart to you
Genre: R & B

Artist: Steve McWilliams **Song:** Goodbye
Genre: Rock



Artist: Sabers & Roses **Song:** Sabers & Roses
Genre: Patriotic/Other

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fourth oscillator, Multi Osc, actually combines six detuned oscillators. With a little detuning modulation, you can quickly descend into the realm of truly clangorous timbres. The subharmonic pulse-wave generator produces variable-width pulse waves at Oscillator 1's first six subharmonics (the ratios of the subharmonics to Oscillator 1 are 1:1, 1:2, 1:3, 1:4, 1:5, and 1:6). Combine the subharmonic pulse-wave generator's signal with another oscillator tuned down three octaves for instant beef.

FILTERAMA

For signal processing, VirSyn provides four filters—two Multimode, one Formant, and one WaveDelay—together with ring-modulation and waveshaping modules. The Multimode filters come in 12, 18, and 24 dB–per-octave versions. They will resonate to the point of self-oscillation and produce a soft distortion when you push their input levels above 0 dB. Each Multimode filter is made up of several subfilters whose cutoff frequencies can be spread out using the Frequency-Shift (F-Shift) control, which slightly adjusts the slope or bandwidth of the filter, depending on its type. Controlling F-Shift with an envelope or LFO is similar to, but subtler than, controlling the filter's cutoff, and it quickly became one of my favorite VirSyn features.

The Formant filter is a three-filter parametric EQ that can be configured as three bandpass filters in parallel or three notch filters in series. The bands have independent resonance and relative level controls. In Bandpass mode, the Formant filter is useful for simulating vocal effects, especially when envelopes are assigned to modulate the band frequencies. In Notch mode, modulating the band frequencies with an LFO produces flanging and phasing effects.

WaveDelay, VirSyn's most unusual filter, is a feedback delay line with a pitch-synchronous delay time. You can vary its decay time from 50 ms to 200 seconds with the Feedback control in real time. WaveDelay has two inputs: one for the impulse that starts the WaveDelay feedback cycle and another

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Articulations:
Long Tones Detaché
Long Tones Expressivo
Bowed Tremolo Ordinaire & Sul Ponticello
Bowed Half Step & Whole Step Trills
Con Sordino (Mutes)
Staccato Martelé
Spiccato
Pizzicato Loose & Tight
Pizzicato Harmonics
Bowed Harmonics Natural & Artificial
Col Legno Tight & Loose
Sound Effects & Glissandi
Methods:
24bit-48k recordings. B&K 4011 front microphones with Schoeps CMC-6Ug and Neumann U87 spot microphones capture a rich balance of space and instrument detail. Up to 4 velocities per articulation with down & up bows. Release samples included to preserve natural instrument resonances and subtle ambient decay. Incredibly playable.
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FIG. 2: VirSyn's 64-step Pattern Sequencer holds 256 step sequences for each of VirSyn's 12 Parts. It also functions as a complex arpeggiator and chord generator.

for the signal to be fed back. Typically, the second input is assigned to WaveDelay's output, resulting in a standard feedback delay line. You can get interesting results by processing the output of the WaveDelay with, for example, one of the filters before feeding it back to the second input. You should protect your ears when you use WaveDelay; small changes in the settings can quickly produce uncontrolled feedback.

The remaining audio modules include an output amplifier, three insert effects, and a WaveMixer (a 5-channel submixer). The amplifier has built-in overdrive, downsampling, and bit reduction. The effects are overdrive (from hard clipping to soft distortion), delay/echo (a dual stereo delay with feedback), and modulation (chorus, flanger, and phaser). The WaveMixer is useful for submixing into the other modules as well as for the final mix (before the amplifier). Although WaveMixer provides three outputs (Mixer, Submix 1–3, and Submix 4–5), I would like to have had more submixing options.

TAKING CONTROL

VirSyn provides four envelope generators (EGs) and four LFOs. The EGs are DADSR (delay, attack, decay, sustain, and release) format, and all envelope parameters can be modulated. VirSyn provides three key-triggering modes (Normal, Reset, and Legato), and any LFO can trigger the envelopes. The LFOs offer sine, triangle, square, ramp-up, and ramp-down waveforms as well as sample-and-hold and random. In Sample-and-Hold mode, the LFO samples the output of one of the other LFOs, making it possible to build complex, repeating patterns quickly.

Twenty bipolar Modulation Matrices manage all EG and LFO routings, various MIDI parameters, and the Pattern Sequencer's control outputs. Output destinations include virtually all of the other module parameters. One nice touch is that the amounts for ten Modulation Matrices are provided as destinations for the other ten.

From a control standpoint, VirSyn's most unusual feature is its 8D Sound Access system (see **Fig. 1**). Each diamond is an *x-y* controller, and each dimension of each diamond can be mapped to as many as eight parameters, allowing you to control 64 synthesis parameters, each with its own range and direction. You can assign each of the eight dimensions to any MIDI controller or manage it onscreen with the mouse.

The Speed slider at the left of **Fig. 1** sets the transition rate from one position to another; its value affects MIDI as well as mouse control. The ability to specify a controller's response makes for smooth gestural control; it is especially handy when you're using the mouse, because a single click can initiate a gradual transition between positions. 8D Sound Access takes a bit of time to set up, but the results can be unique.

PARTS AND SEQUENCES

A VirSyn Project can have as many as 12 Parts. Each Part contains a complete polyphonic synthesizer (with 32-note maximum polyphony) and a Pattern Sequencer with 256 Patterns of 64 steps each. VirSyn also provides a 256-step Song Sequencer. Each Song step calls up a Pattern from the Pattern Sequencer, and you can specify the number of Pattern repetitions and which Parts are muted; that is, unmuted Parts play the chosen Pattern number from their individual Pattern Sequencers.

VirSyn's Part Mixer controls the overall mix, governing the level, pan, global reverb, and chorus send and return for each Part. The Part Mixer is also where you assign each Part's MIDI channel, MIDI note range, and number of voices. The MIDI channel assignments can overlap to create keyboard layers and splits among the Parts.

The Pattern Sequencer is quite extensive (see **Fig. 2**). Each Pattern includes a pitch sequence and two control sequences. The pitch sequence is hardwired to the synthesizer's overall pitch, but you can route the control sequences anywhere using the synthesizer's Modulation Matrices. Each step has its own

Minimum System Requirements

VirSyn

Pentium II/200; 32 MB RAM; Windows 95; DirectX 5.0; compatible sound card; compatible MIDI interface; VST 2.0–compatible host (for VST operation)

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For more about the MRS-4 and other cool Zoom products, unplug the toaster and check out www.samsontech.com gate time and note value (whole to 32nd note—standard, triplet, or dotted) as well as Skip and Mute buttons. VirSyn can automatically convert each step to a chord; it provides every inversion of almost any three- or four-note chord type.

One of the Pattern Sequencer's most interesting features is its Arpeggiator mode, in which holding as many as eight MIDI notes forms the basis for an arpeggio. You can assign each Pattern step to a specific arpeggio step (in pitch order) or to the next arpeggio step. The Pattern's pitch sliders set the transpose amount for the selected arpeggio step.

VERSATILE SYNTHESIS

VirSyn is a versatile package that offers much in the synthesis and sequencing departments. The emphasis is clearly on functionality; what the interface lacks in design aesthetics is more than made up for in ergonomics. For example, you can use the computer keyboard to adjust each control with three levels of precision. Pop-up indicators describe each control as the mouse rolls over it, but the pop-ups are delayed so that they don't appear when you're working quickly and presumably don't need them. In addition, detailed, context-sensitive help explains each module and control.

VirSyn's printed manual (yes, you actually get a manual) is complete, well written, and clearly translated, despite occasional lapses into German. VirSyn supplies plenty of factory patches to get you started. The six factory Projects (which match the demo MP3 files) are limited in scope, but they at least give you a running start.

VirSyn's price tag is relatively low for a modular software synthesizer with so many features. The VST version is a real bargain, despite the fact that it lacks a Pattern Sequencer and global effects—those are functions that are usually fulfilled by the VST host, anyway. Take a test drive and listen to the MP3 examples; it's definitely worth your time. ❀

PRODUCT SUMMARY

VirSyn
VirSyn 1.1 (Win)
software synthesizer
$249

FEATURES	4.5
EASE OF USE	3.5
AUDIO QUALITY	4.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Functional, well-thought-out user interface. Flexible and CPU-efficient synthesis engine.

CONS: No overall patch view. Somewhat limited audio signal path mixing. Key-disk authorization.

Manufacturer
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e-mail info@virsyn.com
Web www.virsyn.com

PEAVEY

KOSMOS

Bone-shaking lows and stereo-image enhancement in a low-cost box.

By Michael Cooper

Various *spectral enhancers*, which are devices designed to impart hyper-realism to full mixes but also to individual tracks, have been available for more than a decade. Typically, enhancers contain two or more processors—for example, one for lower frequencies and another for processing highs. One possible approach to low-end enhancement is to generate bass-frequency subharmonics (frequencies that are an octave lower than the source signal). Another approach, which affects the upper end of the audio spectrum more, is to expand the stereo image into a wider plane or even a three-dimensional perceptual field, typically by employing digital-signal processing (DSP) algorithms that manipulate signal phase.

The Peavey Kosmos, described as a "low-frequency energy and stereo image enhancement system," puts both of those types of processes into one inexpensive box. The dual-channel, 1U rackmountable Kosmos can be used in the studio to beef up individual tracks or an entire mix, or live to enhance the sound of a P.A. system.

DISTINGUISHED PANELS

The Kosmos's rear panel is not stingy with I/O. Each channel provides two balanced inputs and outputs, wired in parallel, one on XLR and the other on ¼-inch TRS jacks. (The TRS jacks can accept unbalanced lines.) In addition, the unit has a Sub Woofer output on a fifth balanced ¼-inch TRS jack. An IEC power receptacle and a detachable AC cord round out Kosmos's rear-panel connections.

The unit's front panel is divided into three sections: Input, Seismic Activity, and Sub Woofer. In the Input section, a continuously variable Level knob provides 10 dB of additional gain (plenty for processing –10 dBV signals) and "infinite" attenuation (off or muted output). The knob's unity-gain position, marked U, is detented. Two LEDs next to the Level knob indicate signal levels: a green LED, labeled 0, lights when the input level exceeds 0 dBu, and both the green and a red LED (labeled +10) light when the signal level exceeds +10 dBu. (Output levels should nominally be +4 dBu, though the unit's maximum output levels are specified as +22 dBu.) The Input section also provides a Global Bypass button and a corresponding red status LED (labeled By). The bypass disables all controls except those that adjust input for the unit and levels for the Sub Woofer output.

TECTONIC SHIFT

There are three continuously variable knobs in the Seismic Activity section: Quake, Thud, and Xpanse. Each knob is marked "min" in the fully counterclockwise position and "max" in the fully clockwise position, with nine hash marks between. To the left of the knobs are the Cut Sub Bass from Main and the Sub-Terranean Shift buttons. The Kosmos *Operating Guide* is cryptic as to what exactly most of the unit's controls do. That is in part to safeguard Peavey's trade secrets; fortunately, however, I did manage to wrestle some details from the company.

The Kosmos's Quake control takes bass-frequency content present in the input signal and shifts it down an octave to generate subharmonics. As you turn the Quake knob from its off position, you increase the volume of subharmonics in all of the Kosmos's output signals. A yellow LED lights when the subharmonic levels exceed –20 dBu. The frequencies that Quake adds to the output signal(s) are determined by the position of the Sub-Terranean Shift button. With that button disengaged (out), Quake processes a narrow band of frequencies in the 100 Hz area, shifting them down to the 50 Hz zone. When the button is engaged (in), Quake shifts frequencies in the 70 Hz region to about 35 Hz. Leaving the button out thus makes the Quake effect more audible on smaller speakers.

The Thud control knob boosts an unspecified band of bass frequencies that lies roughly an octave above the subharmonics the Quake function produces. This control is also off at the "min" position.

The Xpanse control produces two effects at once. As you turn its knob up from "min," the Kosmos simultaneously boosts high frequencies and expands the perceived stereo width of its left and right output signals. But Xpanse doesn't employ equalization per se; the high-frequency boost and stereo-width enhancement are instead produced by the Kosmos's phase manipulation of the input signal(s).

BASS IN YOUR FACE

The Sub Woofer section contains a lone, continuously variable Level knob. With its nine hash marks and its "max" label at the fully clockwise position, the Level knob is identical to the others, except that it is, curiously, marked "off" (as opposed to "min") in the fully counterclockwise position. The knob raises the level of the signal presented at the rear-panel Sub Woofer–output jack.

The Peavey Kosmos can be used live or in the studio to beef up your music's bottom end and widen its stereo image.

The Sub Woofer output's 90 Hz lowpass filter remains operational even with the unit bypassed. The Kosmos's Power button and accompanying blue status LED reside to the right of the Sub Woofer section.

The Kosmos's rear panel provides balanced I/O on both XLR and ¼-inch TRS jacks. Note the Sub Woofer output.

IT TAKES CONNECTIONS

When cutting tracks in the studio, you can route them to the Kosmos's left and right inputs and feed the unit's processed output signals to your MDM or digital audio workstation (DAW). You can feed your mixer's main stereo outputs to the Kosmos to process an entire mix (connecting the unit's outputs to your mixdown deck or DAW to record the processed mix).

For live-sound applications, you can route the Kosmos's left and right outputs to an amplifier that feeds the house monitors. The Kosmos's Sub Woofer output jack, which accommodates balanced and unbalanced connections, puts out a line-level signal that can be patched into a subwoofer. The subwoofer will receive the sum of three bass-range signals derived from the unit's separate processing blocks: the input signal filtered by an 18 dB–per-octave lowpass filter with 90 Hz corner frequency and the outputs of the unit's thunderous Quake and Thud circuitry. If, in a live-sound setup, the Kosmos's beefy bass output is clipping the house monitors, you can engage the Cut Sub Bass from Main button on the front panel to remove low frequencies from the unit's main outputs and then feed them to a subwoofer through the Sub Woofer output jack.

You wouldn't want to use that setup in the studio, because it would hype your subwoofer's bass response and thus skew your perception of bass content in the mix or on individual tracks. However, you could simply print the subwoofer output to a separate track and combine it with the original during mixdown. That strategy would allow you the flexibility to adjust the balance between the original signal and the Kosmos's low-frequency output.

QUAKE, RATTLE, AND ROLL

One difficulty in using the Kosmos stems from the somewhat deficient *Operating Guide*. The manual is too cute for its own good—it offers plenty of playful earthquake metaphors but falls short on critical application notes and other useful information. Although Peavey has since expanded the Kosmos *Operating Guide* to include three pages of helpful diagrams that illustrate various studio and live setups, if you are planning to buy the Kosmos, save this review for a better understanding of what the controls do and ways you can put the unit to use.

I began my tests by using the Kosmos to process a stereo mix. At 0 dBFS meter readings, my Yamaha 02R's +4 dB stereo analog outputs dish out +26 dBu levels, exceeding the Kosmos's maximum input spec of +22 dBu. Yet even with its Input Level knob in the unity position, the Kosmos handled all but my hottest (most compressed) mixes without distorting them.

On stereo mixes, I liked the Quake function the most of the three Seismic Activity processes the Kosmos offers. You should be aware that a little bit of the Quake function goes a long way—cranking the control too high could damage unprotected speakers. (I recommend putting fuses on passive speaker leads to protect the drivers.) Moderate Quake settings can add thunderous bottom to mixes, imparting a quality that cannot be achieved using EQ. You probably won't want to use Quake on every mix and musical style—although it worked great on some R&B and techno mixes, it just sounded weird when I tried it on country and folk stuff.

The Kosmos's Thud function can provide flattering midbass EQ boost. However, you can't adjust its bandwidth or center frequency, so its usefulness on stereo mixes proved a hit-or-miss proposition. The same held true when using Thud on individual tracks.

As for the Xpanse process, anything more than moderate settings changed the stereo material's spectral balance too much for my taste. The Xpanse function can open up the sound of a mix considerably, but the cost is often a thin or brittle sound. Unfortunately, there's no way to avoid the resulting EQ boost—and you also can't tweak the affected frequency range—as you turn up the Xpanse control to enhance stereo width, because both of those effects are irrevocably tied to the same control knob in a proportional

Kosmos Specifications

Inputs	(2) balanced XLR; (2) balanced ¼" TRS
Outputs	(2) balanced XLR; (2) balanced ¼" TRS; (1) balanced ¼" TRS (subwoofer)
Maximum Input	+22 dBu
Maximum Output	+22 dBu
Frequency Response (in Bypass mode)	<10 Hz–40 kHz (+0, –1 dB)
Total Harmonic Distortion + Noise	0.003% (Process mode); ≤0.002% (Bypass mode)
Signal-to-Noise Ratio	–101 dBu
Crosstalk	≤75 dB (@ 1 kHz)
Dimensions	1U × 9" (D)
Weight	7.1 lb.

PRODUCT SUMMARY

Peavey
Kosmos
low-frequency and
stereo-image enhancer
$299.99

FEATURES	3.0
EASE OF USE	4.0
AUDIO QUALITY	3.5
VALUE	3.5

RATING PRODUCTS FROM 1 TO 5

PROS: Affordable. Mono-compatible. Plug-and-play operation. Quake function can beef up the bottom end of tracks and mixes in a way that EQ can't. Balanced I/O can handle unbalanced signals.

CONS: Needs more parameter control. Xpanse function can make audio sound thin or brittle, even at moderate settings. Scant documentation.

Manufacturer
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Web www.peavey.com

relationship (that is, increasing one effect increases the other).

ONE-TRACK PONY

The Kosmos generated mixed results on individual tracks. The Quake and Thud processes gave me wonderfully beefy kick-drum tracks; indeed, that turned out to be my favorite application for the Kosmos. I could detect no processing delay, and the groove remained tight.

The Kosmos also delivered some extremely deep tones on electric-bass tracks. That worked well for some musical styles, but for others, it sounded like unwanted rumble. As with any processor, judicious and artistic use of the Kosmos's abilities must be prime considerations.

The Kosmos's Xpanse effect made hard-panned stereo acoustic-guitar tracks sound wider, but it also made them sound exceedingly bright. I obtained similar results using the Xpanse effect on drum-overhead tracks. Remember that simply boosting highs on hard-panned tracks will result in a wider-sounding mix. For example, I was able to duplicate the width enhancement and change in timbre produced by the Kosmos (with the Xpanse knob set to 11 o'clock) fairly closely by using simple EQ. I gave the hard-panned, dry tracks a +6 dB shelving boost at 5.04 kHz and a 5 dB shelving cut at 157 Hz. The tracks processed through the Kosmos still sounded slightly wider, but not dramatically so. In either case, the resulting brighter timbre was not desirable. Moreover, after I turned down the Xpanse knob to a point at which the timbre was acceptable, little stereo-width enhancement remained.

At Peavey's suggestion, I multed a mono vocal track to the Kosmos's left and right inputs, turned up the Xpanse process, and listened to the unit's stereo outputs, hard-panned in the stereo field. Contrary to what I had been led to expect, I heard virtually no stereo-width enhancement in the processed signals.

LIKE, KOSMIC

The Kosmos can deliver great sounds on stereo mixes and individual tracks; however, its fixed frequency bands and interaction of imaging and timbral effects often require compromised settings. Peavey is aware of the Kosmos's limitations and plans to offer an advanced model, tentatively dubbed the Kosmos II, that will grant greater control and flexibility.

For the time being, other devices on the market can deliver more dramatic 3-D effects than the Kosmos and do so with fewer timbral compromises. That said, the Kosmos is fully mono compatible, something many other 3-D devices cannot claim. Moreover, few other enhancers can compete with the Kosmos's low price. Finally, the Kosmos provides more than just stereo-width enhancement—indeed, its Quake function alone might be worth the price of admission. If urban and techno music are your specialties, the Kosmos could be your low-cost ticket to bone-shaking nirvana. ❁

CYCLING '74

MAX 4.0/MSP 2.0 (MAC)

A significant revision for this important multimedia development toolkit.

By Thomas Wells with Dennis Miller

Since its introduction in 1998, Max/MSP has quickly become one of the premier development tools for sound design and music composition. Max is a powerful, graphical-object-oriented, interactive programming language for MIDI. It was developed at IRCAM and has remained under continuous development by David Zicarelli and Cycling '74 since 1989. MSP expands on Max by adding a powerful suite of real-time audio Objects. Combined with the extensive support provided by third-party developers, the two programs give you one of the most complete programming environments for multimedia production available today.

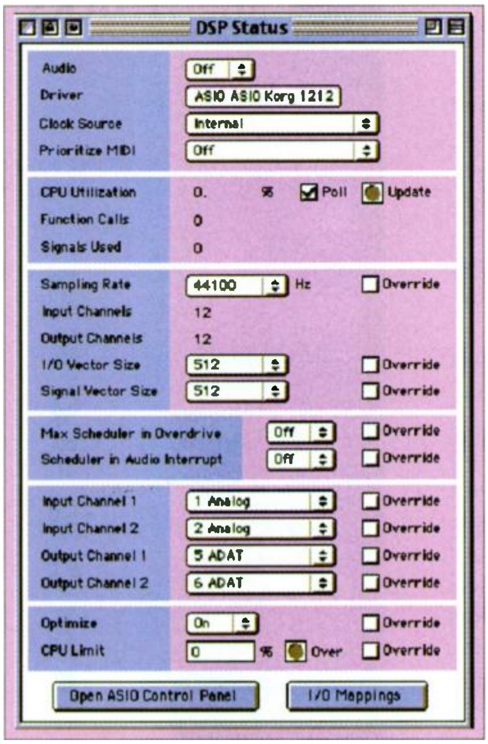

FIG. 1: The DSP Status window contains numerous settings for configuring the software to work with your computing and audio hardware.

Unlike sequencers and audio editors, Max/MSP does not have one primary purpose. You can use it to build your own application or to customize someone else's to fit your needs. In this review, we'll focus on the new features of Max 4.0 and MSP 2.0 and look at the software's ability to construct real-time synths and sound processors.

Although Max is available separately, Max and MSP are typically sold as a bundle. The bundle also includes a utility that lets you use the audio plug-ins you develop in Max/MSP within a VST, a MOTU Audio System (MAS), and, soon, a Real Time AudioSuite (RTAS) host. Max/MSP runs only on the Mac, but a Windows version is currently in development.

MSP has received two **EM** Editors' Choice awards, in 1999 for the first release of the program, and again in 2002. Max was also a 2002 winner. MSP 1.0 was reviewed in the October 1998 issue.

To write programs (which are called patches in Max/MSP), you can start from scratch or modify one of the plentiful examples. Programming takes place in the Patcher window, where you connect Max/MSP Objects (represented as boxes) together with patch cords. Max/MSP comes with several hundred Objects that perform a huge range of tasks, from adding two numbers together to waveform editing. You can write your own Objects in Max/MSP or in the C programming language. Hundreds of Objects written by users are available on the Internet.

KNOW THE CHANGES

Max 4.0/MSP 2.0 is more than just the first major revision in almost five years—it is an explosion of creative thinking and innovation. This release further advances a sound synthesis and MIDI environment that is terrifically deep, accessible to users of varying levels of expertise, and eminently well organized and thought out.

Minimum System Requirements

Max/MSP

PPC 604e/300; 64 MB RAM; OS 8.1

Users acquainted with previous versions should have few problems adapting to Max 4.0/MSP 2.0. Many changes involve enhancements to the editing interface of the Patcher window and the addition of tools that are used to create spiffy graphic interfaces. But with more than 40 new Max 4.0 Objects and 70 new MSP 2.0 Objects, the revision is far from just cosmetic. Important capabilities for the treatment of polyphonic Objects and for building frequency-domain processors are among the many enhancements.

SOUND INS AND OUTS

Whereas earlier MSP versions included specific drivers for various audio cards, MSP 2.0 relies on ASIO drivers (an MSP Sound Manager driver is still provided). Not surprisingly, users must provide ASIO drivers for their audio interfaces.

For this review, the software was tested on a G3/400 MHz using two audio cards: a Korg 1212 I/O and an Audiomedia III. Each card offered challenges involving Web scavenging and trial and error. For example, the ASIO driver for the Korg was hard to find but worked flawlessly when it was installed in the MSP ASIO Drivers folder. The driver for the Audiomedia III has yet to work with both analog and S/PDIF outputs functioning simultaneously.

The driver story doesn't stop with ASIO: four other audio protocols are supported in MSP 2.0. The ReWire driver lets you route MSP output to applications such as Steinberg Cubase and Mark of the Unicorn (MOTU) Digital Performer. The DirectConnect driver allows you to play MSP into Digidesign's ProTools (when used with version 5.0 or later of the DigiSystem Init).

The VST driver lets MSP talk to a special VST plug-in, which allows MSP to be used as an effects plug-in for a VST-compatible sequencer. Finally, the NonRealTime driver permits complex


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processing operations that exceed real-time playback to be recorded to disk—a nice touch.

Driver options are set in the first pane of the DSP Status window (see **Fig. 1**). The top pane includes Audio on/off, Driver selection, Clock Source, and Prioritize MIDI, a control that runs audio processing at a lower priority to permit more frequent MIDI interrupts.

MSP has 512 logical channels and as many physical-device channels as your sound card can handle—a decided improvement to the previous limit of 16. To assign the other 510 channels to your sound-card farm, click on the I/O Mappings button to open a subwindow that displays all channel mappings.

MENU STUFF

Next, let's look at the revision of items and controls in the Max/MSP menu bar. The Edit menu is much the same, though it now contains a clever and useful new command called Paste Replace. That command lets you replace a group of Objects with a single Object. The contents of other menus have been altered, and two new menus have been added.

The new View menu controls the appearance of the Patcher window during editing. There you'll find one of the program's most used commands, Edit, along with a number of other useful commands. Edit is used to toggle between locked and unlocked (Edit mode) states of the Patcher window. The View menu also contains commands to facilitate the manipulation of Objects in the Patcher window, such as Hide/Show Object Palette, Hide/Show Connections, and Restore Origin.

The new Extras menu is a handy and customizable repository for useful utilities such as Audiotester, which helps you diagnose audio-output problems; an input/output monitor; Quickrecord, for recording the output of a patch to an audio file; and a Tips section, which contains practical information about working with Max/MSP.

One of the most remarkable additions to the Max/MSP interface, Contextual menus, isn't found on the menu bar at all. A Contextual menu appears when you Control-click on an Object; the resulting pop-up menu lists the commands for that particular Object.

For example, Control-clicking on a patch cord produces a Contextual menu in which you can invoke the Align feature (to make neat, straight, or right-angle patch-cord paths), color the patch cord using the customizable ColorPalette submenu, or hide or show the patch cord when the Patcher window is locked.

Control-click on the blank area of a Patcher, and an all-purpose menu appears that lets you accomplish simple tasks, including Lock the Patcher, Select All, and Get Info on an Object. It also provides a pop-up, vertical version of the horizontal Object palette (usually seen running along the top of the Patcher window in Edit mode).

This all-purpose menu also includes a New Object option that shows a list of all the Max built-in Objects. Click on an Object in this menu, and it appears in the Patcher window. The New Object menu also duplicates certain items in the Options menu and includes a submenu to call up example Patchers.

There are other interface changes, as well. For example, almost all of the dialog boxes in Max/MSP have been replaced by Inspectors that open when you choose Get Info from the Object window.

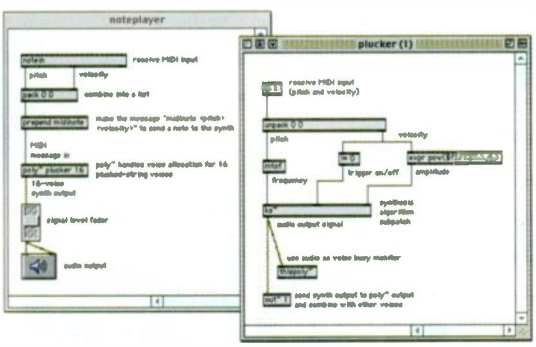

FIG. 2: MSP's poly~ Object, shown at left center, offers better management of polyphony than was available in previous versions. On the right is the structure of the Plucker subpatch that poly~ is managing.

PRODUCT SUMMARY

Cycling '74
Max 4.0/MSP 2.0 (Mac)
programming environment
Max 4.0/MSP 2.0 bundle $495
Max 4.0 $295
Max 4.0/MSP 2.0 upgrade $135

FEATURES	5.0
EASE OF USE	4.0
DOCUMENTATION	4.5
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent tool for real-time sound synthesis and multimedia development. Elegant user interface. Terrific documentation. Large number of third-party Objects and Patchers available.

CONS: Doesn't support Mac OS X.

Manufacturer
Cycling '74
tel. (415) 621-5743
e-mail info@cycling74.com
Web www.cycling74.com

Inspectors are special Patchers that let you change properties of individual Objects. You can inspect multiple Objects by using the new Floating Inspector, a window that remains on the desktop and shows the properties of each selected Object in turn.

A cool and timesaving feature lets you easily duplicate Objects. Select an Object from the Object palette, drag it over the Patcher window, and hit Control + Shift, and then everywhere you click, you will deposit a copy of that Object. All of the previously mentioned shortcuts, Inspectors, and pop-up submenus are real time-savers, and it's obvious that the Max/MSP designers put a great deal of thought into streamlining their software's human interface.

POLYFUN

A significant difference between version 2.0 of MSP and earlier versions relates to the management of polyphony. It was possible to achieve polyphony in previous versions, but doing so required a lot of copying and pasting and wasted CPU cycles. (In

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Load impedance: >1000 Ohm
Maximum SPL:131dB SPL
Noise: (Line): 27dB (A weighted)---17dB
S/N· 77 dB
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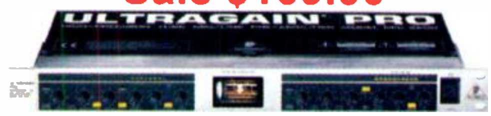

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earlier versions of the program, sound-processing Objects ran continuously, whether you heard their signals or not). MSP 2.0's new strategy for dealing with polyphony is the poly~ Object, which takes two arguments: the name of a subpatcher to be played polyphonically and a number that indicates maximum polyphony.

The patch on the left side of **Fig. 2** shows an example of poly~. This patch uses 16 voices of a patch called Plucker, which makes a plucked-string sound using the Karplus-Strong synthesis algorithm. By sending the poly~ Object a MIDI Note On message, notes are routed automatically to any Plucker patches that are not busy making sound. In hardware synthesizers, that is called *dynamic voice allocation*.

On the right of **Fig. 2**, you can see the structure of the Plucker patch. The actual synthesis part (written in Max/MSP) is encapsulated into its own Object, called ks~. The surrounding patch interfaces ks~ to the demands of the poly~ system by converting incoming MIDI note numbers to frequency and Velocity to amplitude. It also contains an Object called thispoly~, which reports when the patch is actually making sound. That is necessary for dynamic voice allocation to work.

That description gives some idea of how to work with the new MSP polyphonic management, but nothing can convey the fun of listening to the patch just described or the pleasure of working interactively in such a rich environment for sound production. It may be going too far to call it the nirvana or elysian fields of software-synthesis programming, but Max/MSP comes close.

PFFT!

The pfft~ Object greatly simplifies spectral-domain signal processing and lets you create subpatches that manipulate frequency-domain signal data independent of windowing overlap and fast Fourier transform (FFT) size. Pfft~, like poly~, works with a subpatcher as one of its arguments and is suitable for multiple applications.

Fig. 3 shows a pfft~ patch and subpatch (specdelay~) that use a delay line in the frequency domain, a spectral-processing algorithm that is similar to the one used by Native Instruments' Spektral Delay software. What does a frequency-domain delay line mean? Broadly speaking, the pfft~ Object processes FFT frequency bins sequentially from low to high frequency. When those bins are delayed, the energy in a low-frequency bin is shifted over to a higher-frequency bin, and eventually, the energy in the higher-frequency bins wraps around to lower frequencies. The effect is something like a filter, but not exactly. The specdelay~ subpatch also includes a feedback control that adds even more unusual effects.

The Examples section of the distribution CD contains many wonderful examples that illustrate the power of the pfft~ environment. Among the best of them are the pfft~ version of the Forbidden Planet Patcher and the Phase Vocoder Sampler. Forbidden Planet controls the output of the FFT bins with a graphic interface, which allows you to draw the shape of a filter on a graphic equalizer with as many as thousands of bandpass filters.

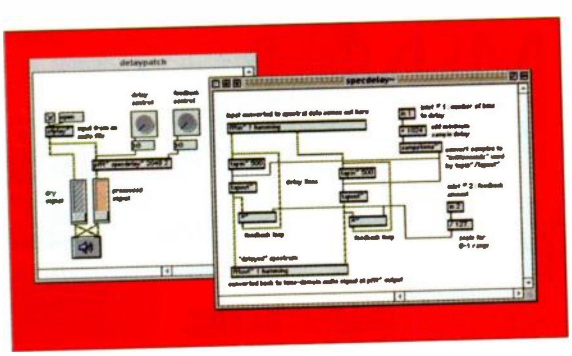


FIG. 3: Processing signals in the spectral domain is the job of the pfft~ Object patch, shown at left. This example illustrates a frequency-dependent delay. The structure of the subpatch, specdelay~, is shown at right.

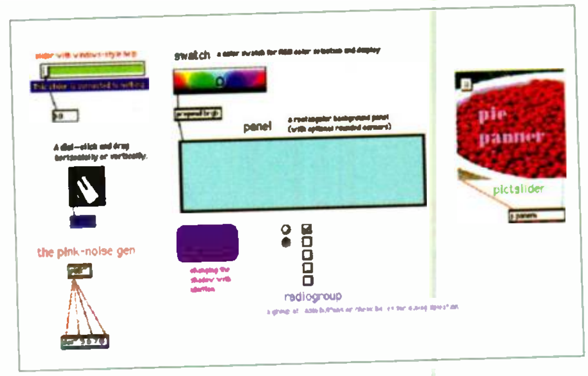


FIG. 4: Among the most important new features in Max/MSP are its greatly enhanced graphic capabilities.

GRAPHIC DESCRIPTION

Speaking of graphic interfaces, the new Max/MSP is a much more image-oriented and colorful place to work. A variety of new color options are available for elements such as patch cords and comments. You can change color attributes simply by opening the Contextual menu (Control-clicking) on the item and selecting a color. Colors are selected from a palette of 15 colors that you can customize using a hue and saturation control.

Fig. 4 shows some of those options at work. Clockwise from top left are a yellow slider with Windows-style pop-up help, a swatch Object that is connected to a panel to determine its color, a pictslider control with graphic image and slider, a radiogroup radio-button control group, a purple button with shading, Object boxes with colored patch cords, and a graphic dial (for a pink-noise generator) with a colored number box.

NEW OBJECTS

With all the new Objects in Max 4.0 and MSP 2.0, this review can only begin to communicate the wealth of processing capabilities they represent. Special attention was given to the all-important area of filters—not that the programs were lacking there before.

The centerpiece of the MSP 2.0 filter suite is the filtergraph~ Object. Filtergraph~ is not a signal processor by itself; instead, it generates filter coefficients that are then used by the MSP biquad~ filter Object. That permits you to build filters simply and directly and also allows you to input your own filter coefficients.

Other new filter Objects include fffb~, an efficient bank of bandpass filters; svf~, a state-variable filter that provides simultaneous output from different filter types; teeth~, a comb filter with independent feedback and feedforward delay times; onepole~, a single-pole lowpass filter; and buffir~, a finite impulse-response filter that uses a buffer to store data points convolved with the input signal.

There are other important new additions, as well. The oscbank~ and ioscbank~ Objects are noninterpolating and interpolating wavetable oscillator banks, respectively. Those Objects are capable of generating hundreds of sinewave oscillators simultaneously and are a welcome addition to the MSP Object repertoire. They are also very computationally efficient.

The sfplay~ and sfrecord~ Objects have been improved, with sfplay~ sporting new features such as variable-speed playback, sample-accurate cue looping and triggering, signal outputs for playback position, and correct playback of files with sampling rates other than the current MSP sampling rate. Those Objects support several new file formats—including NeXT/SUN(AU), WAV, and Raw Data—and new sample formats, such as 32-bit float, 64-bit double, and 8-bit µlaw. The sfrecord~ Object offers support for as many as 28 audio channels, and you can load multiple instances of sfrecord~ until you hit the 512 limit or your computer gives out.

There are also enhancements to the Patcher, specifically in the areas of scripting and automatic patch generation. The tutorials illustrate some impressive concepts, and the new feature shows terrific promise.

PATCH IT UP

Max/MSP's first major revision in almost five years is both a resounding and an unqualified success. New Objects abound, among them new methods for polyphony management and spectral-domain signal processing. The user interface has been improved, and new tools for designing custom graphic interfaces have been included.

The documentation for Max/MSP is thorough and complete, both in quality and in quantity, and weighs in at about 1,500 electronic (PDF) pages. A group of 88 tutorials included in the distribution represents a well-thought-out introduction to Max/MSP and is designed to assist those who do not have any previous programming experience. Manual pages are complete, with plenty of illustrations. A terrific Help facility with working examples and a library of example patches add another avenue to this fertile environment for multimedia development.

Thomas Wells *has been involved in computer music for more than 25 years. He teaches at Ohio State University.* **Dennis Miller** *is an associate editor of* **EM**. *Thanks to Professor Todd Winkler of Brown University for his assistance with this article.*

Quick Picks

CHICKEN SYSTEMS

Translator 2.5 (Win)

By David Rubin

Thanks to MIDI, samplers can easily talk to one another, but that doesn't mean they speak the same language. For myriad reasons, a universal sample format has never materialized, and manufacturers continue to cling tenaciously to their proprietary formats. As new hardware samplers enter the market and as powerful software samplers join their ranks, the number of sample formats has spiraled out of control, causing headaches and frustration for desktop musicians. Sample obsolescence runs rampant; selling your sampler and buying a different model usually means losing a perfectly good library of sounds. Moreover, many musicians are forced to use two or more brands of samplers in their setups just so that they can access sample collections in different formats. Fortunately, Chicken Systems has confronted this file-format Tower of Babel and has, to a great extent, broken the language barrier with Translator 2.5 ($149.95).

Readin' and Writin'

Translator is able to read from and write to a wide assortment of popular sample formats and can even tackle several formats that one might charitably refer to as obscure. Do you want to convert your Kurzweil K2000 library to Roland XV-5080 or Korg Triton format? No problem. How about Ensoniq EPS into E-mu EIIIx or Akai S3000? Again, no problem. Tascam GigaStudio users should be especially pleased to learn that they can now convert their old hardware-sampler libraries into Giga format. Translator supports dozens of formats, including AIFF, WAV, and SoundFont. Native-format support for Seer Systems Reality, Native Instruments Reaktor, Steinberg Halion, Emagic EXS24, and BitHeadz Unity DS-1 is also included along with support for Digidesign SampleCell and CreamWare Pulsar. (See the Chicken Systems Web site for a complete list of supported formats.)

Translator doesn't simply convert raw samples from format to format; it also converts keymaps, Velocity switches, modulation routings, envelopes, tuning, filter settings, and many other parameters. Of course, not all parameters convert directly from one format to another. As described in the documentation, Ensoniq supports bidirectional loops, Roland supports standard forward-and-release loops, and Akai supports multiple loops. If the destination sampler doesn't support the appropriate type of looping, Translator alters the samples to simulate the appropriate effect.

In a similar way, the program compensates (when possible) for differences in sampler architectures to yield a suitable result. For example, Translator automatically compensates for the Frequency Emphasis boost in Roland S-series samplers so incoming samples from other devices won't sound dull on a Roland instrument and so exported Roland samples won't sound abnormally bright in other formats.

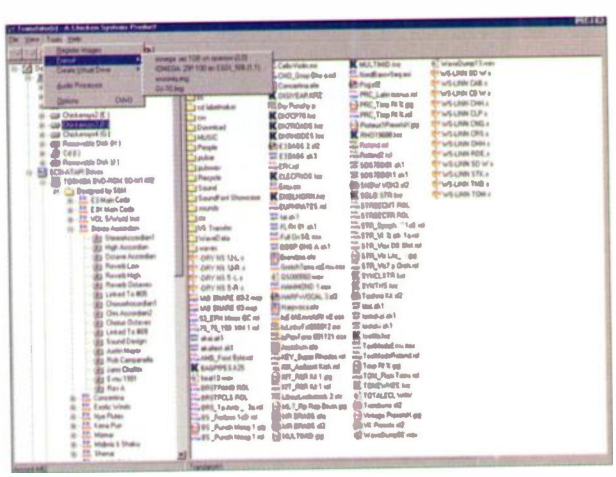
Language Master

Translator's user interface is simplicity itself. The main window presents a hierarchical file-tree display that resembles Windows Explorer. The computer's DOS-formatted hard drives appear at the beginning of the list, followed by any attached SCSI, ATAPI (IDE), or USB drives with proprietary formatting, such as Roland, E-mu, Akai, Kurzweil, and Yamaha. You can also create virtual drives (large files on your computer's hard drive) that are formatted in any supported proprietary format—a handy feature for burning your own sample CDs. To help locate specific files, you can audition samples directly from disk.

To translate a file, simply select and drag it from a sample CD or hard drive and drop it onto another drive (real or virtual) after specifying the new format. You can also right-click on your files to select and convert them in a similar manner. Translator lets you convert a single sample, preset, or program, and you can batch-convert banks, volumes, and complete directories. If you select an entire CD of samples and convert them all at once, you can sit back and watch as Translator whizzes through the files and writes them onto the destination drive.

Most of the time, the batch conversion goes without a hitch; occasionally, though, a patch has trouble, which triggers a dialog box explaining the problem. Chicken Systems is quite responsive to those glitches and encourages users to send in errant files for analysis.

Although Translator officially supports only CD-ROMs, hard drives, and magneto-optical (MO) drives, it does include four unsupported utilities that can read floppies from Ensoniq, Akai, Roland, and E-mu Emax sample libraries. The conversion process can be somewhat tricky, and the results are not always predictable, but the handy utilities may help you resurrect an old collection of patches that might otherwise be lost to the sands of time.



Chicken Systems' Translator 2.5 employs a Windows Explorer–style main window that lets you drag and drop patches, banks, and libraries to convert them from one format to another.

Sampler Helper

Translator's seemingly simple operation belies its complex and sophisticated inner workings. It's no simple task making samplers multilingual, and Chicken Systems accepts the challenge with zeal. As the company states, Translator is "the indispensable sampler utility that no sampling musician can live without." Unless you've never had to use more than one sample library and one sampler, I would have to agree.

Overall EM Rating (1 through 5): 4.5
Chicken Systems, Inc.; tel. (800) 877-6377 or (320) 235-9798; e-mail support@chickensys.com; Web www.chickensys.com/translator

MULTILOOPS

Naked Drums Rock, vol. 1, and Naked Drums Pop R&B (Mac/Win)

By Jeff Burger

Drum-loop libraries are proliferating at an astounding rate. Most contain stereo files, providing a no-muss, no-fuss approach but lacking the flexibility of multitrack recordings. Multiloops is one company bent on giving you greater control by delivering loops in 24-bit multitrack format. I reviewed two Multiloops products in one pass—*Naked Drums Rock*, vol. 1, and *Naked Drums Pop R&B* ($149 each)—because their concepts and delivery formats are identical. Each collection contains four CD-ROMs providing more than 2 GB of sounds.

The audio files are in Sound Designer II format on the Mac version and WAV format on the Windows version. (Multiloops sells the two platform offerings separately.) The products are optimized for Digidesign's Pro Tools; each folder contains a Pro Tools session file that makes it easy to audition all the offerings by clicking on memory locations.

The Mac version includes session documents in various forms to accommodate current TDM and LE systems, Pro Tools Free, and older 4.x systems. The PC version ships with one version facilitating Pro Tools Free and TDM or LE systems. Users of systems other than Pro Tools can import the results into any program that's compatible with the file format you purchased. The loops on the Windows version are also optimized for Sonic Foundry's Acid 3.0.

Drum Boogie

Each Multiloops library is organized by tempo, ranging from 60 to 165 bpm on *Rock*, vol. 1, and from 60 to 130 bpm on *Pop R&B*. (The latter includes one passage in a folder labeled 240 bpm, but the label and usability are questionable.) Each folder contains a variety of 4-bar themes (some closely related, some less so) as well as some fills at its designated tempo. Also provided in separate folders are individual hits, including multiple hi-hat hits featuring the ineffable variations that elevate live performances above drum machines.

Each loop, fill, or hit on *Rock*, vol. 1, is provided on seven tracks correlating to the mics for snare, kick, stereo overheads, and three toms. Each has a send engaged for its channel with Digidesign's D-Verb plug-in already in place. *Pop R&B* also has a separate track for hi-hat. (There is little overhead cymbal action in *Rock*, vol. 1, and the hi-hat seems to be isolated in the overhead channels, which is somewhat confusing.)

Multiloops' *Naked Drums* libraries are collections of 24-bit, multitrack drum loops for Pro Tools and Acid 3.0. However, you can import them into any software that reads Sound Designer II or WAV files.

Rather than naming the loops, Multiloops simply labels each loop numerically. To use a loop, you import all of the identically numbered tracks into your session document and line them up to the same beat. Although descriptive names can be difficult for a developer to divine and can taint the end user's creativity, the numbers-only system doesn't do much to help users pick loops from a veritable sea of choices.

Loop Editing

Both volumes were recorded and performed well, though I did notice some less-than-perfect timing on a few *Rock*, vol. 1, loops. Some tracks also exhibit what sound like phase problems. The sounds themselves are pretty standard, presumably what you want in a construction kit of this design. The biggest variation is in the use of a few different snares in some folders. One snare variation in the 120 bpm folder of *Rock*, vol. 1, rings so much that it's a little overpowering. A few tracks substitute a shaker for hi-hat; I would prefer having the shaker on an additional track supplementing the hi-hat.

The loops sound seamless on both products. Each loop is truncated precisely at the loop point, and Multiloops has already applied crossfades. There is a catch-22 to that approach, however: any instruments that would ring out at the end of a loop are clipped off. Leaving the tails intact would provide more realism when assembling and crossfading a series of loops into a song but would make it harder to audition loops using Looped Playback mode. You can always substitute single hits for any clipped sounds that you want to hear ring out.

Under Scrutiny

Deciding whether these products are for you entails scrutinizing the pluses and minuses of *Naked Drums*' multitrack format. Multiloops gives you plenty of puzzle pieces to edit together into your own compositions, both in terms of loop variations and access to each track and hit. However, flexibility translates into added work. You'll probably need to work with gates, compressors, and EQ to shape the raw material into something that sounds as polished as highly processed stereo-loop libraries. Most loops will also cleanly translate into Propellerhead ReCycle slices, but you'll have to translate each track individually unless you bounce them to disk first.

Pop R&B is more ambitious than *Rock*, vol. 1; however, that's probably more indicative of the difference in genres than a difference in the Multiloops performances. The more I worked with both products, the more they grew on me. (If you've already purchased the earlier two-disc *Rock*, vol. 1, set, the manufacturer will exchange it for the updated version for $20.)

The *Naked Drums* libraries are targeted primarily at home recordists who lack the resources to record live drums. The bottom line is that I found myself writing new song ideas as I simply auditioned various loops, and that makes them winners in my book.

Overall EM Rating (1 through 5): 3.5
Multiloops; tel. (615) 646-0150;
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SUMMIT AUDIO
TD-100
By Myles Boisen

EM ★★★★★ hot pick

Summit Audio has enjoyed a longstanding reputation for being a manufacturer of premium tube gear for studio recording. Its most recent offering—a tube direct-injection box that includes unbalanced high-impedance, balanced line-level, mic-level XLR, and headphone outputs—is equally at home onstage, at home, or in any professional or personal studio. The TD-100 tube DI and instrument preamp ($495) is a direct box with a difference—many important differences, actually.

Outside the Box

There are several useful features on the TD-100's exterior that are far from standard equipment. Most noteworthy is the continuously variable impedance control, which lets the user adjust how the TD-100 affects pickup loading and related tonal changes when it is used with guitars and basses. At the fully clockwise 2 MΩ setting, the device presents its highest possible input impedance, minimizing loading and typically allowing the clearest, brightest sound from most pickups. At the fully counterclockwise setting (10 kΩ), input impedance is relatively low and timbral changes caused by pickup loading, usually characterized as a dulling or warming of the sound, are easy to hear.

Intermediate impedance settings can produce subtle to drastic modifications of the instrument's tone, depending on the type of pickup and electronics, among other factors. For example, when I tested the TD-100 with a Fender guitar and bass, I heard little difference in tone between the 11 o'clock position (just below 1 MΩ) and the maximum 2 MΩ setting. Around the 10 kΩ extreme, however, both instruments' highs decreased dramatically, as they might when plugged straight in to a mixing board without the benefit of a direct box.

Other pluses include a Polarity switch for adjusting the pickup signal's phase; a Ground-Lift switch; and an unbalanced ¼-inch high-impedance direct output for connection to an amplifier or stompbox. The direct out (which is unaffected by the TD-100 circuitry) lets the unit interface with an onstage guitar or bass rig as a standard DI would while contributing its tube character to a house sound system through a standard mic-level XLR output.

A balanced ¼-inch line-level output is provided for studio applications, and a headphone out enables additional monitoring for practice sessions or limited studio setups. The levels of the rearmounted mic, line, and headphone outputs are controlled by the Output Gain knob, which ranges from 0 to +24 dB in stepped 4 dB increments. (There is no separate control for headphone volume.) Two indicator LEDs, located above the output gain pot, light up to show average signal presence and overload or near-clipping status.

On the inside, a 12AX7A/ECC83 vacuum tube imparts its glow to the signal path while the output-amplifier stage is discrete solid state. An internal power supply is a welcome feature that means a wall wart

Combining solid-state circuitry with a 12AX7A/ECC83 vacuum tube, Summit Audio's TD-100 DI and instrument preamp is designed for a variety of stage and studio applications.

is unnecessary. A standard detachable IEC cable provides the AC connection. All switches, indicator lights, and controls are mounted on the front panel, as are the high-impedance in and out. The TD-100 is one rack unit high and half a rackspace wide, making it eligible for mounting in a number of commercially available rack trays. Mounting screws are included.

Beyond Expectations

In a round of studio testing, the TD-100 received high marks. For direct guitar, the Summit was clean and airy, and on bass it compared favorably to a rack full of tube DIs and preamps with headroom to spare. In addition, the TD-100 imparted a convincing vintage punch to a sampled Rhodes electric-piano patch, making it a strong contender for processing a wide range of sample-based sounds. Overall, the build quality, sonics, and specifications of Summit Audio's first DI and preamplifier are impressive, especially considering its reasonable price and three-year manufacturer's warranty.

Overall EM Rating (1 through 5): 5
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BEST SERVICE

Ethno World Library (Akai, E-mu, Giga)

By Dan Phillips

The Best Service *Ethno World Library* sample CD-ROM ($299) opens up film composer Marcel Barsotti's private sample library to the public. The collection includes more than 1.4 GB of samples from 72 instruments and is divided into five categories: Stringed Instruments, Bell and Metal Type Instruments, Woodwinds, World Drums, and World Percussion. *Ethno World* is available in multi-CD sets for Akai, E-mu, and Giga formats; I reviewed the three-CD E-mu version.

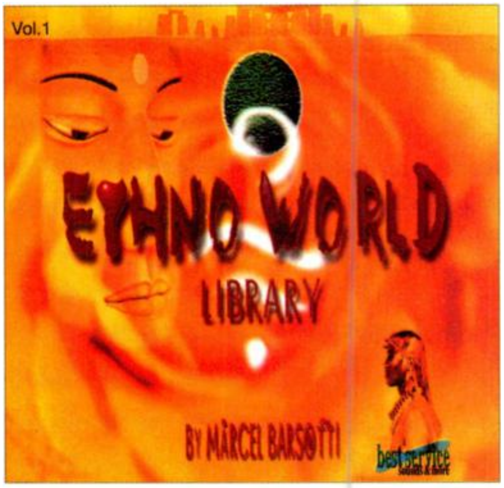
World of Drums

The World Drums banks are almost universally outstanding. Barsotti offers a wide variety of playing styles for each drum, including single and double hits, muted hits, flams, slaps, and rolls. In some cases, 50 or more samples are used in a single program to create expressive velocity splits, making them a joy to play.

The jangly tone of the tambourine drum is a standout. The samples cover a range from hits that play only the drum to others that emphasize the jingles, resulting in a very playable program. Other favorites of mine include the Moroccan derbuka; the deep, solid timbre of the African donn donn; and the medium-pitched, slightly resonant Indonesian ceremony drum. The unusually low-pitched tabla is less successful, marred by rattles and buzzing; the instrument might have benefited from a bit of repair before the sampling session.

I was pleasantly surprised by the elemental vibe of the German Big Hand Drum, with its deep, loose-skinned timbre. Two perfectly captured military snare drums include side sticks, flams, and press rolls. Although the hits are great, the phrases and beats would have fared better with a stellar percussionist at the helm. Many rolls have a slightly uneven swing feel instead of a steady left-hand-right-hand pulse.

In addition to the drums, the World Percussion section has 21 instruments. Its finger and hand cymbals, African rice shaker, rainmakers, gong besar, dream catcher wind chimes, and gamelan cowbell are all worth noting.



Best Service's *Ethno World Library* sample CD-ROM (Akai, E-mu, Giga) offers roughly 1.5 GB of instrument samples from around the world.

String Things

The Kantele category offers my favorite string samples. A Russian cousin to the dulcimer, the kantele's strings are plucked or struck with a stick. The resulting timbre sounds like a cross between bell and string, with an aggressive attack balanced by a sweet sustain. The sampled kantele phrases are sparkling and slightly mysterious, combining scraped string sounds with the other playing techniques. A Small Kantele bank focuses on special effects and loops well suited for horror-movie soundtrack material.

The twangy and rich saz multisamples are eminently playable with good dynamics. Chinese erhu, Irish truxa mandolin, German Framus banjo, and a couple of acoustic guitars are also included, but they seem less compelling after hearing the gorgeous kantele.

Bells and Metals

The Bell and Metal section is strong throughout, with great sampling, unusual timbres, and expressive programming. I instantly put the mellow metallophone to work in an ambient ballad. The shimmery and slightly chorused Bamboo Vibraphone bank is a favorite, especially the Bamboo Vibra 2 program, which features a distinctive double-hit at high velocities. The Shanghai Baby Piano is a rod-based toy piano with a bouncy, complex attack. It sounds as though the original instrument was in poor shape, but its unevenness lends the sound a charming, organic quality.

The Tibetan cymbals offer a good selection of velocity switched, sampled gestures, including multiple strikes and rubbed cymbals. The pitched versions stretch a single sample over the keyboard to create delicate fairy bells. The Tibetan singing bells are sweet and resonant.

Woodwinds Section

The Woodwinds section offers an unusual mixed bag of sounds, from Indian snake charmers to Irish whistles. Many of the woodwind programs offer a velocity switch between legato and staccato samples and are available with or without vibrato. I would have appreciated samples for different dynamics, as well. Uneven volume or tone mars some multisamples, and the programming should have compensated for those differences. Still, you'll find some good stuff, including the strong, complex tone of the Irish low whistle; the hollow, breathy alto recorder; and the susato tin whistle. A bank of flute phrases offers five octaves of swoops, jumps, flutters, and bird noises.

A World of Good

Ethno World offers an interesting mix of elements. The woodwinds and strings are uneven, but the drums, percussion, and bells offer a wealth of outstanding material. Those sounds alone are worth the investment.

Overall EM Rating (1 through 5): 4

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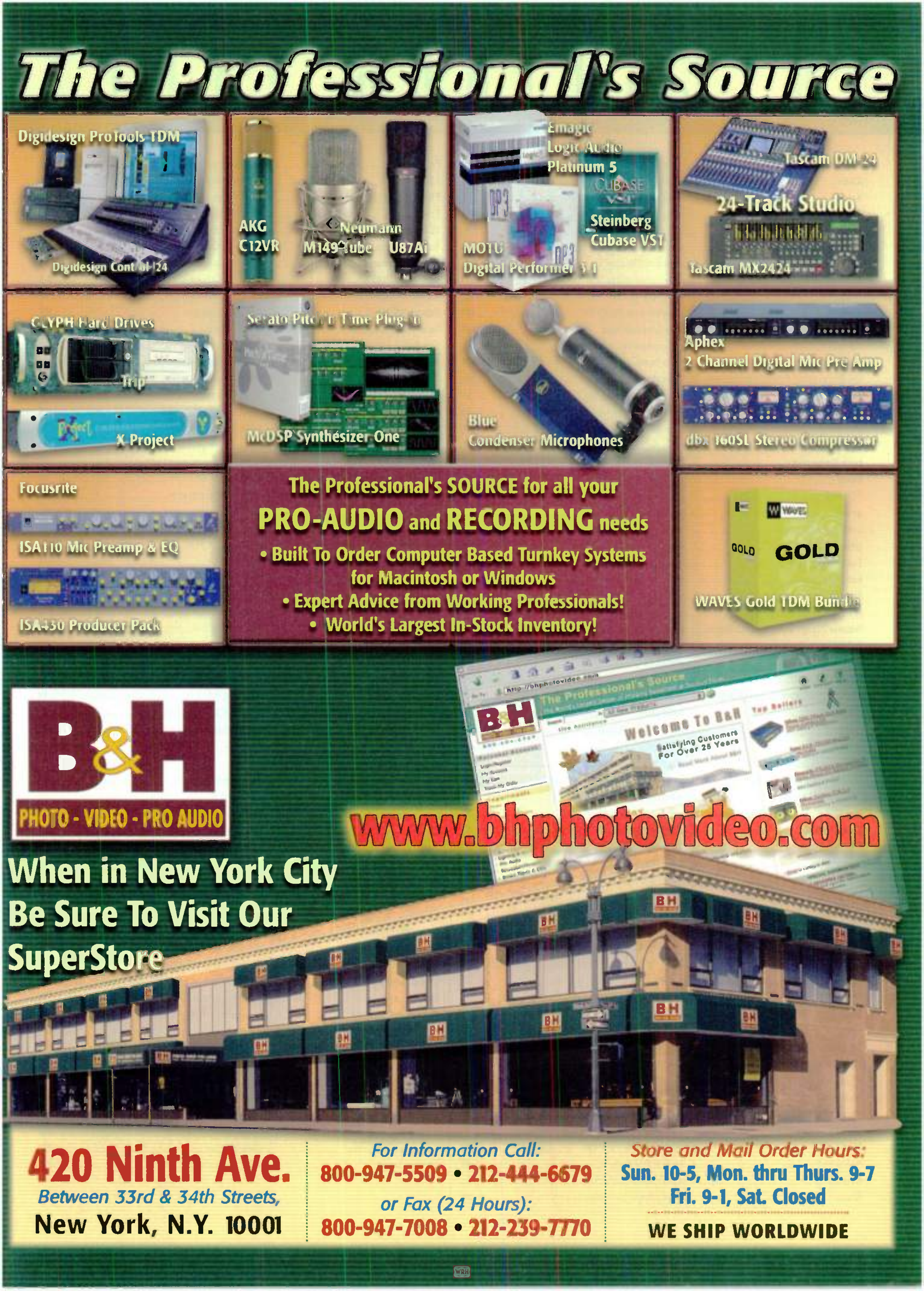
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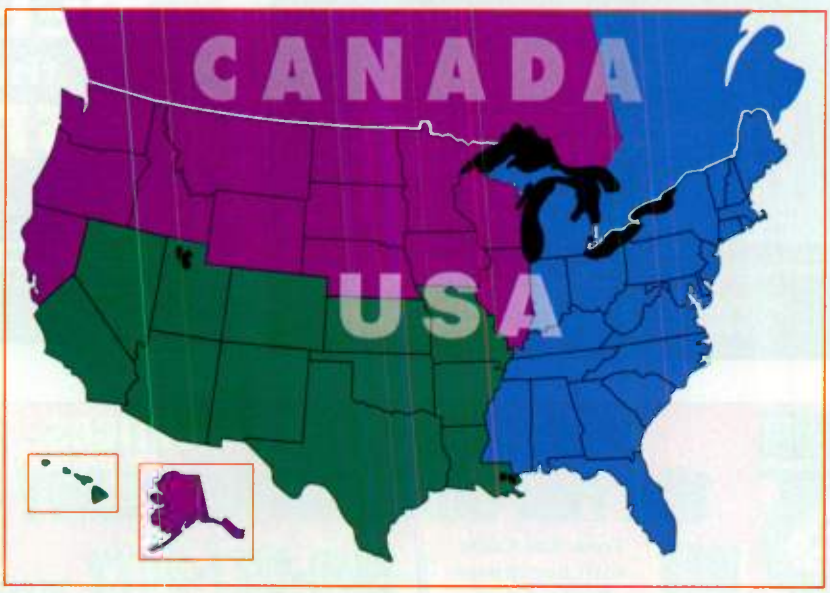

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PreSonus [audio electronics]

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  The most powerful and portable system ever
- **MOTU DP3 audio workstation software**
  Winner of 3 consecutive EM Magazine Editors' Choice Awards
- **MOTU 828 FireWire audio interface**
  Plug in the 828, plug in your guitar, mic, & synth and start recording!

SampleTank™ Plug-in sound module for Digital Performer

SampleTank has taken the Digital Performer virtual instrument scene by storm with its rock-solid compatibility with Digital Performer and a huge library of samples that it puts right at your fingertips. Not only does SampleTank sound incredible, it gives you lightning-fast access to hundreds of sounds, complete with 20 high-quality DSP effects with four assignable per instrument. The result? You spend more time creating and less time fiddling. Import your AKAI S1000/S3000 sample libraries, or choose from a growing number of SampleTank libraries from the top names in soundware development, such as IK Multimedia, Sonic Reality, Masterbits and the Advanced Media Group. New titles include Vintage Voltage ST™, Trance Grid™, Techno Grooves™, Loop Soup™, Terminal Head™, and many more. Use up to eight separate outputs per module for completely flexible mixing within Digital Performer's 32-bit virtual studio environment. Enjoy 128-note polyphony that is fully optimized for your G4 processor. Available in two versions: SampleTank L and the fully-loaded SampleTank XL. Call Sweetwater today.

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M Project Plus™ Audio-optimized FireWire hard drive storage with built-in FireWire hub

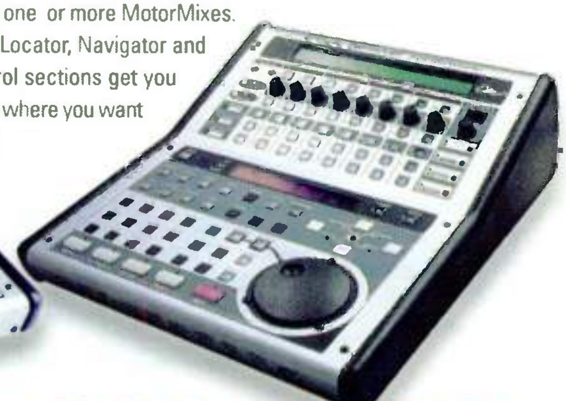
M Project Plus is a 1u rack-mounted FireWire hard drive offered in 40GB, 80GB or 120GB single-drive capacities. M Project Plus includes a 4-port FireWire hub, making it a perfect companion for MOTU's 828 FireWire interface. Up to three 828's can be connected to an M Project Plus drive, which is then connected to the computer, for a total of 24 analog ins/outs, 6 mic inputs (with preamps), 24 channels of 24-bit ADAT optical I/O and 6 channels of S/PDIF digital I/O - all on the computer's single built-in FireWire bus. Backed by Glyph's legendary service and support, M Project drives are the only MOTU-approved drives for MOTU FireWire and PCI audio interfaces, including the 828, 896, 2408mkII, 1296 and others. M Project drives are ideal for PCI systems because they free up a PCI slot. Call your Sweetwater sales engineer today.

GLYPH TECHNOLOGIES, INC.

MotorMix™ and DashBoard™ Automated mixing and editing worksurfaces for DP3

With custom software written specially for Digital Performer, MotorMix becomes a seamless, tactile extension of your MOTU software recording environment. Put your hands on eight 100mm motorized faders and rotary encoders to tweak your mixes in record time. Gain instant easy access to all MIDI and audio tracks with control banks. You'll never even think about mixing with a mouse again. Imagine having tactile control over most of Digital Performer's features with MotorMix's intuitive layout and easy operation. MotorMix gives you all the advantages of a professional mixing board, at an incredibly affordable price.

The perfect companion to MotorMix, Dashboard brings tactile speed and finesse to editing with DP3. Operates as a stand-alone worksurface, or it can be fitted to one or more MotorMixes. Dashboard's Locator, Navigator and Zoom control sections get you quickly to where you want to edit.



CM Labs

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A little Summit for everyone.

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Peak™ 3.0

Advanced waveform editing and mastering

BIAS Peak 3.0 for Mac OS 9 and X, is the ultimate editing, processing, and mastering companion for Digital Performer! Peak gives you lightning fast, nondestructive waveform editing with support for audio files up to 32 bits and 10 MHz, including 24-bit/96kHz files. Unlimited Undo/Redo gives you the freedom to work creatively. Select an audio region in DP, choose the "Use External Waveform Editor" command, and instantly switch into Peak! Peak's sophisticated options for on-the-fly marker, region and loop creation are simply unparalleled.

bias
sound creative

Advanced DSP and looping tools include Convolve, Repair clicks, Loop Tuner™, Loop Surfer™, Loop It™ and Guess Tempo™ and more. Process thousands of files, or just a few, using Peak's batch processor. Peak directly supports all MOTU audio interfaces and includes Roxio Toast™ Lite CD for burning your own redbook audio CDs directly from Peak's powerful playlists. Create web or multimedia content for export with multiple file formats, including Apple's QuickTime.

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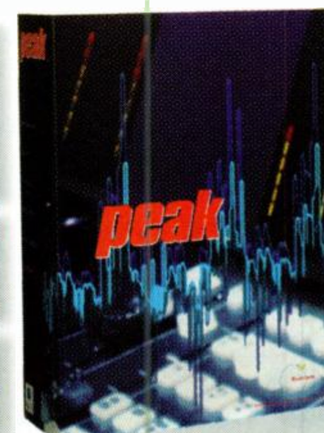

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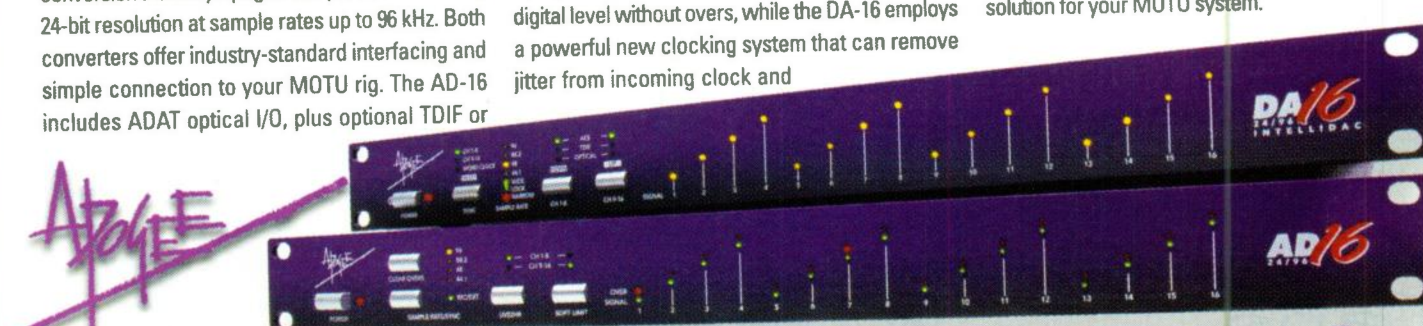
Apogee's AD-16 A/D and DA-16 D/A each offer 16 channels of the highest quality digital audio conversion that only Apogee can provide - including 24-bit resolution at sample rates up to 96 kHz. Both converters offer industry-standard interfacing and simple connection to your MOTU rig. The AD-16 includes ADAT optical I/O, plus optional TDIF or AES/EBU cards. The DA-16 includes ADAT optical, TDIF and AES/EBU built-in. The AD-16 includes Apogee's unique Soft Limit process for maximum digital level without overs, while the DA-16 employs a powerful new clocking system that can remove jitter from incoming clock and data for a pure, pristine, analog out for mixing or monitoring. Call Sweetwater today and ask about Apogee's AD-16 and DA-16, the ultimate high-end solution for your MOTU system.



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Supports 128 Digital Performer tracks with 8-band EQ and dynamics on every track

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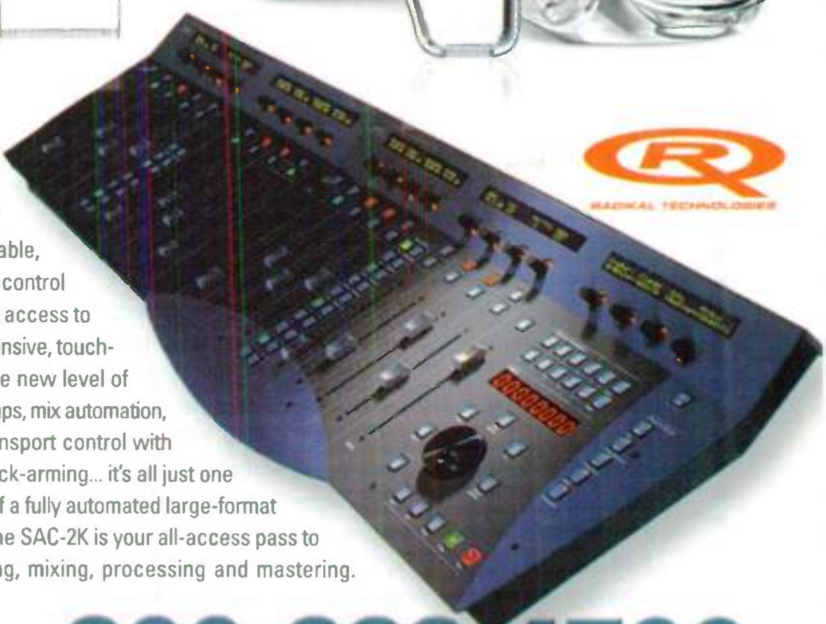

With the most complete support for Digital Performer currently available, the Radikal Technologies SAC-2K sets a new standard for hands-on control of Digital Performer with a custom plug-in for DP and easy, one-touch access to every element of the recording process in Digital Performer with responsive, touch-sensitive automated controls. Within minutes, you'll achieve a whole new level of interaction and creativity that you never thought possible with fader groups, mix automation, plug-in automation (up to 12 parameters at once), window sets, transport control with jog/shuttle, input/output routing, voice assignments, solos, mutes, track-arming... it's all just one touch away. The SAC-2K will make you feel like you're sitting in front of a fully automated large-format mixing console, but without the bloated size and massive price tag. The SAC-2K is your all-access pass to the world of Digital Performer-based recording, editing, arranging, mixing, processing and mastering.

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WRH

FINAL MIX

By Larry the O

The Scoop

STEVE CURL

I'm finally going to do it. For years I've turned to Scott Wilkinson's "Tech Page" column to glimpse out-there technologies that are likely to one day rock our little musicians' world. No matter what cool stuff I get wind of, Scott always turns out to have a year-old column about it.

Until now.

I just got off the phone with Scott and managed (through a sly ruse he won't even be aware of until he reads this) to confirm that I have the scoop on him, and he doesn't even know it.

Here's the deal: at an industry schmooze event in San Francisco, I conversed with Finnish recording engineer Norman Haardiheering. Haardiheering's a crack-up because he looks and talks like a Malibu surfer who should have taken an office job before it was too late, but he actually is a world-renowned expert on *spatial recording,* his early name for what is now called surround miking. Haardiheering has consulted for most of the world's best orchestras, as well as for theme parks (he developed spatial recording out of his theory of multidimensional acoustics), broadcast networks (primarily for sports coverage), and even, later in his career, the Grateful Dead.

Haardiheering's take on multidimensional acoustics was that the acoustics of any space could be modeled from the position of the original sound source as the sum of the contributions of multiple sound sources that are the boundaries of the space. The way he explained it to me sounded like a kind of inverse ray tracing.

Anyway, Haardiheering, being a classical singer since an early age, became interested in trying to make recordings employing his theory, and thus began a 32-year odyssey of experimentation with multiple-microphone ambient recordings. After a few experiments, he started getting work from classical-music labels, at first for chamber ensembles and then for full orchestras. Oddly, his breakthrough work was designing the 428-channel sound system for the Armstrong Linoleum exhibit at the 1988 Memphis World's Fair. Armstrong wanted fairgoers to have a "virtual" (a new word then) surround-sound experience of the acoustics in a bathroom.

Shortly after the fair, Haardiheering was approached at an Audio Engineering Society chapter meeting by a strange little man with a helium voice. The man was microphone guru Alexis T. Flondopowicz. Flondopowicz was a full professor of French history at Oxford but had long had a passion for making microphones. Eventually, he had been around long enough to be generally acknowledged as an expert.

Flondopowicz, fascinated by Haardiheering's recordings and theories, proposed a collaboration. Haardiheering, intrigued, got Flondopowicz's phone number and visited him days later.

In the ensuing five years, the two of them put their heads together, and sparks flew. When I ran into Haardiheering, he had just returned from England, having put the "finishing touches," as he put it, on their invention.

I was waiting for Haardiheering to get to the point when he did. "We have created," he intoned with obvious pride, "the world's first omnidirectional shotgun microphone!"

I thoughtfully replied, *"Say what?"*

Here's the weird part: once he explained it to me, it actually made sense. The microphone is essentially an array of capsules feeding a high-powered DSP matrix processor. With a bit of intensive, high-order phase manipulation, you can effectively control the polar pattern of each capsule individually, from omni to a rather tight shotgun.

"The omnidirectional shotgun allows you to zero in on fine detail, excluding surrounding noise, in a 360-degree circle," Haardiheering enthused. "If there is a bullfrog amongst the crickets, you can get just the bullfrog and hardly hear the crickets. If a dog barks in the trees on the other side of you, you can snag Rover but leave the leafy thicket behind. It really works; you have to hear it!"

I got Haardiheering's e-mail address and almost immediately dashed off a message asking when I could check out the mic, but I couldn't wait for a reply if I was going to beat Scott Wilkinson to the punch. I'll let you know how it sounds. ❂

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

"What sets the Dakota apart is the quality of its design and its ability to play well with others."
Electronic Musician Magazine

"After much critical listening, Tango24 was revealed to be a great-sounding interface. Its sound is smooth and detailed."
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- XLR main outs and phone jack with front-panel volume control.
- 8 channels of ADAT optical input/output with sample-accurate ADAT SYNC IN.
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- Word clock in and out; sample-accurate ADAT SYNC input.
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- Connect up to four 896s (72 channels at 48kHz) with no FireWire hub required. Add more units via 3rd-party FireWire bus expansion.
- Can be combined with the MOTU 828 FireWire interface.
- 10-segment level meters on all ins/outs, including ADAT optical.
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- Includes AudioDesk® sample-accurate workstation software for MacOS with editing, automated mixing & 32-bit effects processing.

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