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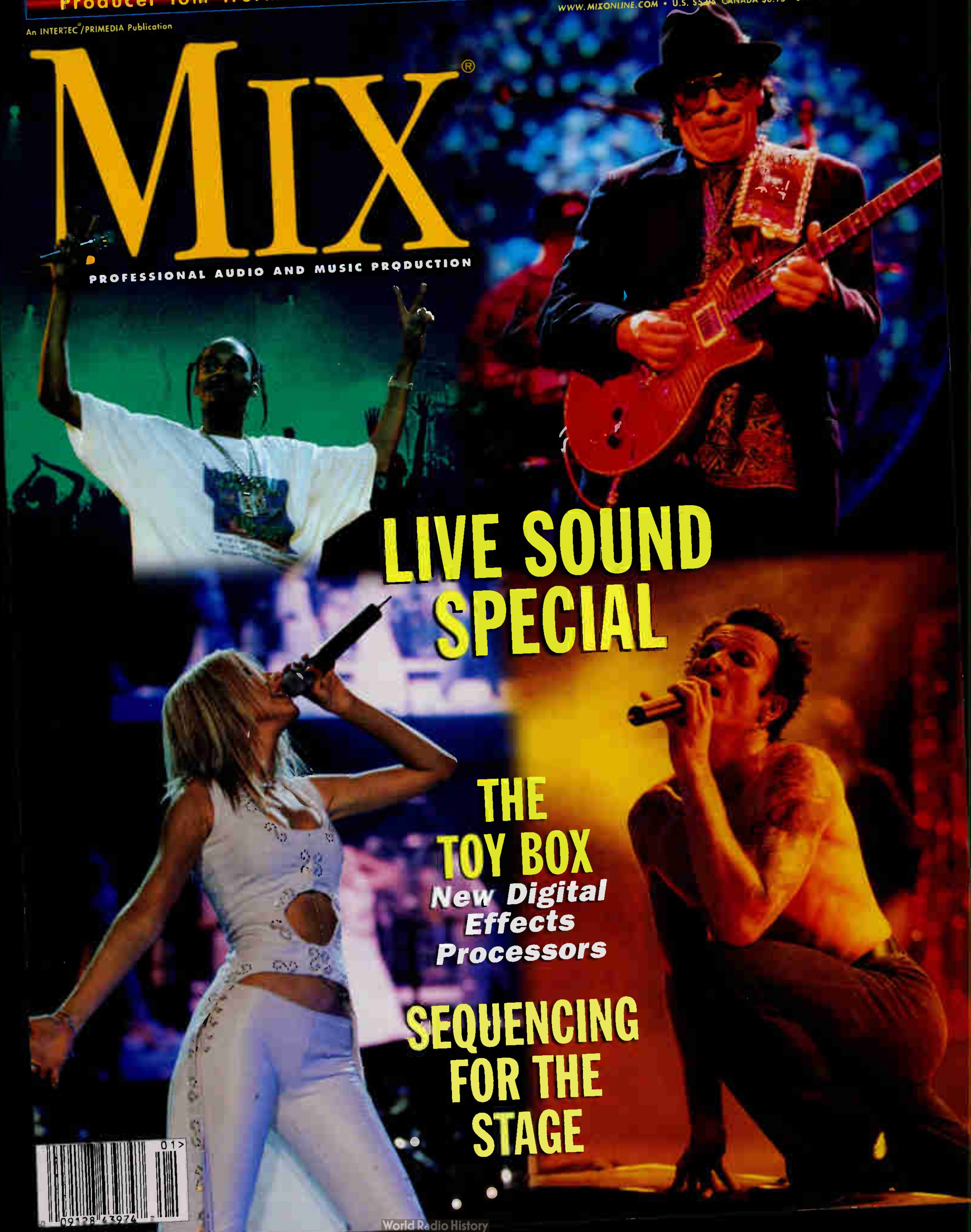

LIVE SOUND SPECIAL

THE TOY BOX

New Digital Effects Processors

SEQUENCING FOR THE STAGE

Producer Tom Werman · Ear Monitor Tips · Brian Setzer's Retro Cool

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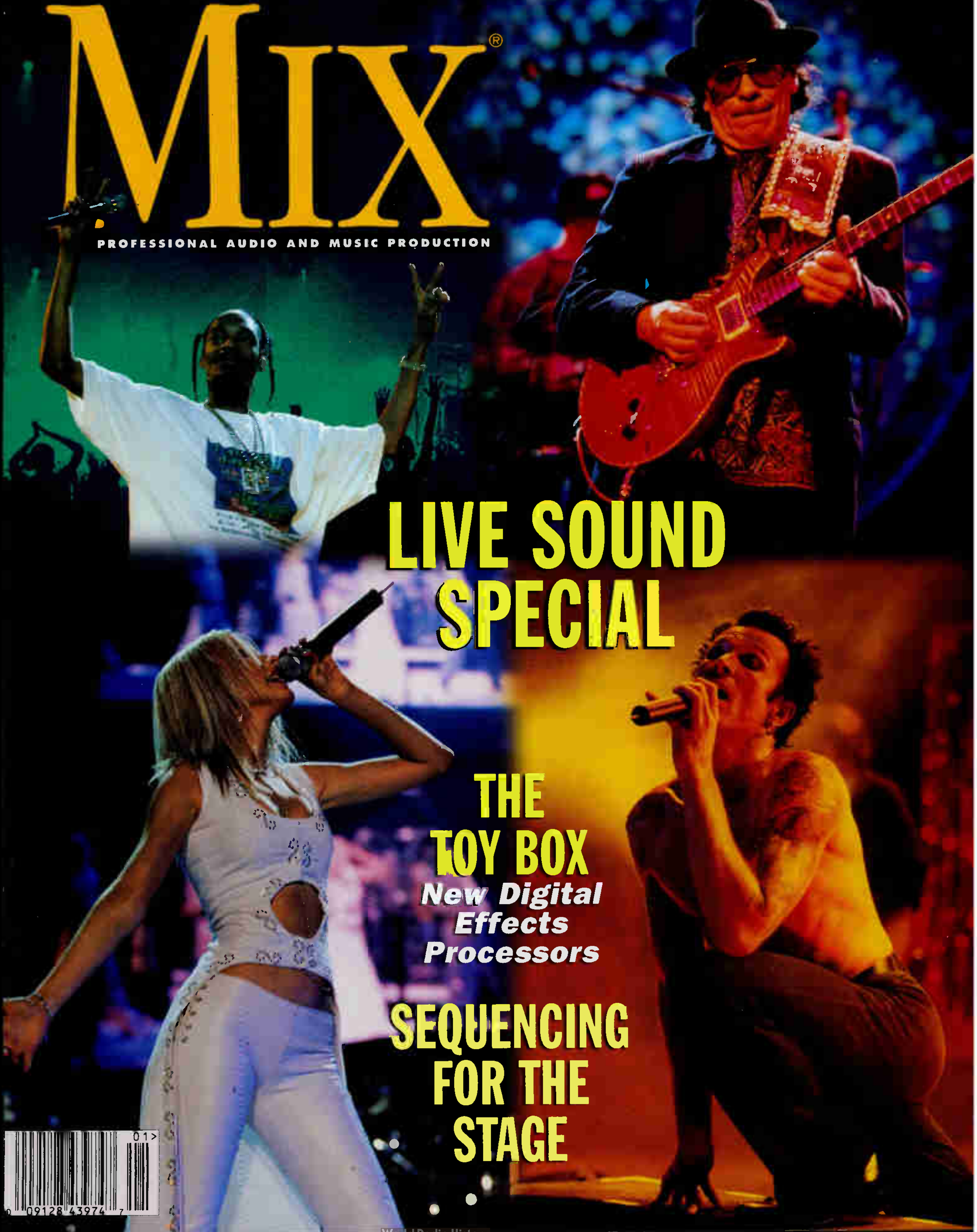

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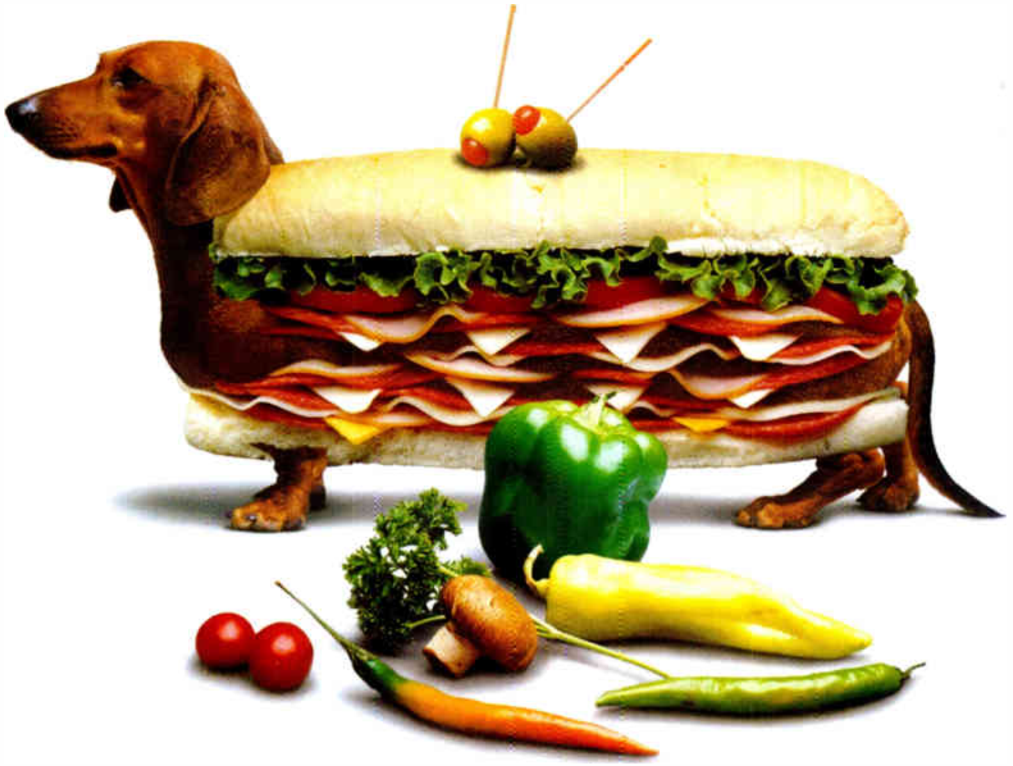

MIX®

PROFESSIONAL AUDIO AND MUSIC PRODUCTION

JANUARY 2001, VOLUME 25, NUMBER 1

AUDIO

PAGE 120


PHOTO: TRISHA LEEPER

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SOUND FOR PICTURE

LIVE MIX

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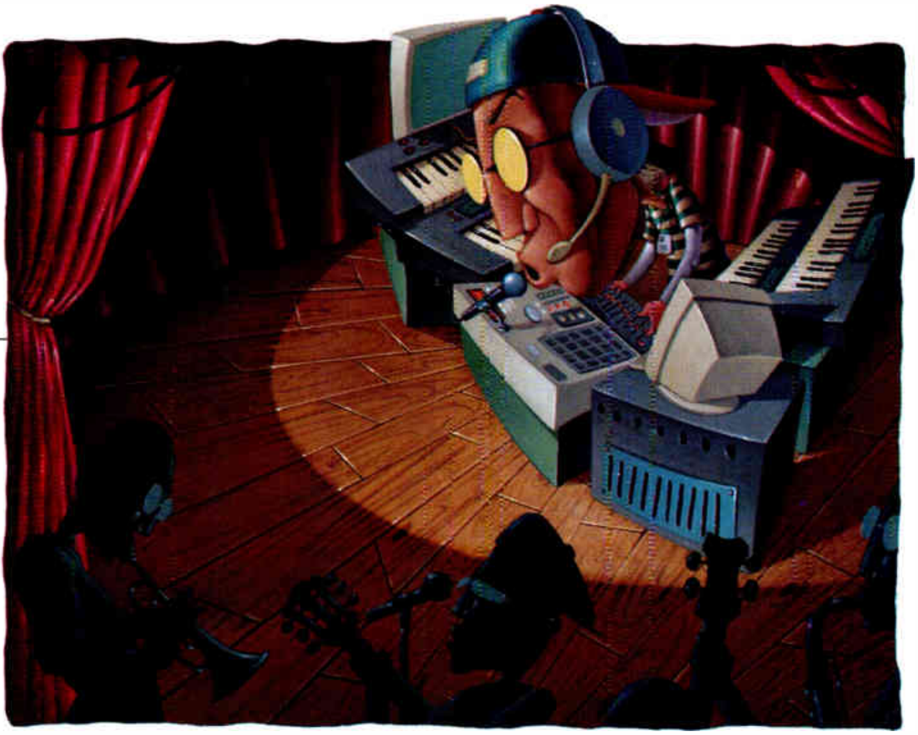

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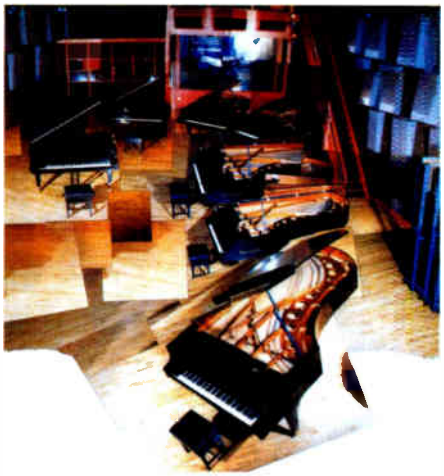
PRODUCTS



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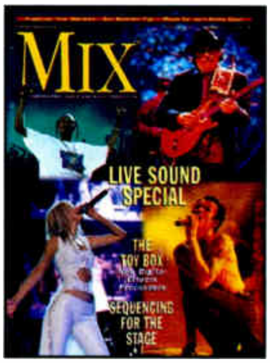
DEPARTMENTS

On the Cover: In the past year, artists have stormed across the continents in a whirlwind of what seemed to be endless tours. Pictured, clockwise from top left: Snoop Dogg, Carlos Santana, Scott Weiland of Stone Temple Pilots and Christina Aguilera. For more live sound coverage, see "Live Mix," beginning on page 59. **Photos:** Steve Jennings.



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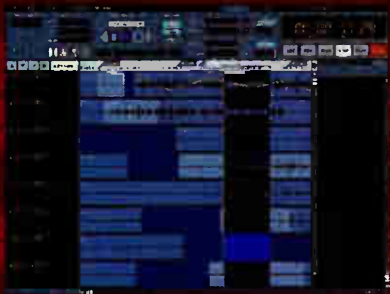

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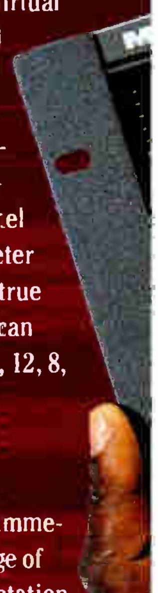

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FROM THE EDITOR

IT'S THE MUSIC, MAN!

On the classic Rolling Stones album *12x5*, Keith Richards and Mick Jagger wrote the simple phrase: "There've been good times/There've been bad times." The song deals with the ups and downs of relationships, but the same sentiment applies to the music business, and right now, the industry is definitely in the "good times" phase.

Much has been written about how the Internet is destroying the music business, providing a haven for illegal distribution of recordings, lyrics and software. No doubt, there are many unresolved issues to be settled—such as songwriter royalties on downloads—yet, Internet sites and online radio offer a powerful means of promoting new artists and exposing audiences to new musical genres and styles.

Given the ease of Internet shopping, traditional record stores need to step up to survive, perhaps by transforming drab retail sites into environments that enlighten, entertain and educate the consumer. Hey, this is show biz, remember? Stores could also provide high-speed kiosks, where customers could burn CDs with downloads of licensed, legal music and high-res graphics. Everybody wins: The store gets a sale, labels/artists get paid and, with no physical inventory required, a vast array of titles (including obscure out-of-print releases) could be available.

The situation is no different for MI retailers: Offer services and events that online sites *can't* provide—i.e., lessons, rentals, repairs, trade-ins, in-store clinics, etc.—and build that solid customer base. Meanwhile, 2001 bodes well for musical instrument sales, whether you're a player or retailer. It's ironic that today I can buy a Ludwig drum set or a Fender Strat for about the same price I paid in the '60s. On the technology side, today's MI, pro and semipro recording systems offer high quality at affordable prices. Figure in faster/cheaper computer systems, native processing and convenient USB interfacing, and a powerhouse desktop (or laptop!) recording system can be assembled at a rock-bottom cost.

Another positive sign for the music biz is Winter NAMM's arrival at the renovated Anaheim Convention Center this month after a three-year hiatus in Los Angeles. Judging from this show's record-paced advance registrations and sold-out exhibit space, NAMM's return to the Land of Disney is a welcome and much-needed change.

Speaking of change, we've got some of our own to announce. *Mix*'s managing editor Tom Kenny takes over the title of editor with this issue, while I move to a new editorial director spot. Tom will handle most of the day-to-day responsibilities at *Mix*, and I'll be focusing on the overall view of *Mix*, *Sound & Video Contractor* (now relocated here to Emeryville) and our related titles, such as *Internet Audio*, which begins quarterly publication this month, headed up by *Mix* technical editor Sarah Jones.

Meanwhile, mixonline.com has teamed with digitalmediaclick.com, a portal for all of Intertec's entertainment Web sites, including *Electronic Musician, Broadcast Engineering, BE/Radio, Millimeter, Entertainment Design, S&VC, Video Systems, Lighting Dimensions* and *World Broadcast Engineering*, with fast-track access to dozens of technology-based communities within our industry. Your mixonline.com bookmark still works; you just get a lot more for your click.

George Petersen

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FEEDBACK

THEATER SOUND DESIGN
MORE FROM THE PIT

I would like to respond to Bill Koggenhop's letter published in the October 2000 issue ("Or, We Could Just Forget the Whole Thing"). I am an arranger/orchestrator, and I have worked with sound designer Jonathan Deans on five musicals, including *Fosse* (for which I was awarded a Tony award and Jonathan should have been) and *Seussical*, which is about to begin previews on Broadway. In fact, I helped create the sound design concept for *Seussical* that Jonathan wrote about in his article and to which Mr. Koggenhop takes exception.

I believe Mr. Koggenhop, and those who share his point of view, are laboring under a false assumption, based on a romantic notion of what live theater is and is supposed to be. These folks believe that a live musical theater experience should, by definition, include "acoustic" rather than "canned" music, possibly because they assume the natural sound emanating from an orchestra pit is preferable to amplified sound. I hate to burst anyone's bubble, but, for the most part, the acoustic sound emanating from an orchestra pit is terrible; it gives composers and arrangers agita and designers nightmares. In fact, Broadway sound designers (for musicals, anyway) spend the majority of their time trying to overcome the sound coming from the orchestra pit and replace it with something better. Unfortunately for them, the audience for a musical prefers to listen at a more moderate volume level than say the audience for an Earth, Wind & Fire concert, making it difficult for the mixer to push the amplified sound to a level where it overtakes the acoustic sound. The result is that the acoustic sound that Mr. Koggenhop holds so dear often ends up getting in its own way, resulting in a less-than-wonderful aural experience for the listener.

On our last show, *The Music Man* (currently running on Broadway), Jonathan and I took an acoustic approach to the sound of the orchestra, in that we worked hard to simulate the sound that most theatergoers would identify as a "live" orchestra sound. Make no mistake, though: The sound of the orchestra in *The Music Man* is amplified. Why? Because despite the fact that I took great pains to orchestrate the show so that it would work acoustically, the live sound of the orchestra in the pit was simply not acceptable, given the high standards that today's audiences expect in live concert sound. To call amplified music "canned" is an insult to the craft of live sound design and the art of live sound mixing. The audience sophistication to which Jonathan refers to is, in fact, the result of audiences' exposure to high-quality sound design. Advances in technology and mixing skill now afford designers the opportunity to meet the general public's demand for the same type of definition and sonic brilliance in live sound as they are used to in recorded sound.

On *Seussical*, rather than forcing ourselves to overcome the live orchestra sound (as we had in the past), we decided to take the "radical" step of isolating the louder instruments (the brass and saxes) in order to better control their presence in the mix. (This is actually not a new idea; the notion of isolating instruments in the orchestra pit goes back to Phil Ramone's design for *Promises Promises* more than 30 years ago.) We decided to keep the strings in the pit so that we would have some live sound to work with (and so that parents bringing their children to the edge of the orchestra pit would have something identifiable at which to point). Also located in the "live" pit are the guitars, bass and keyboards (no amps—everyone is wearing headphones, listening to a custom mix), the drums (v-drums with "real" cymbals) and the percussion (behind gobos, but visible). The six winds and brass are in another room, watching the conductor on a video monitor (and being observed by him as well).

I include these details only to make the point that the audience's experience at *Seussical* is as "live" as a symphony orchestra concert: The orchestra, singers and audience interact in real time, the tempos vary from night to night, and the magic of live theater still occurs. The only difference is that we choose to sculpt the soundscape for our audience with the judicious use of technology, rather than leave it to the fates. I suppose Mr. Koggenhop would be justified in objecting to this practice if we did it tastelessly; however, our goal is to enhance the experience for the listener, not diminish it. I'd like to think we succeeded.

Doug Besterman
Via e-mail

THEATER SURROUND SOUND

I have been a sound engineer and designer for about 12 years. I have been reading *Mix* for about five years. I have designed for both bands and theater. I find it curious that in the world we live, in the "surround sound" world, your attention is not on theater. I have been designing shows that have included at least 16 channels of full-range and at least four LFE channels. You call home theater surround sound? Shame on you! We in the theater have been doing more with less technology to immerse our audiences. Plus, we do it live every night! Do we hear about this? Maybe once a year in your magazine. Do not get me wrong. I do like your magazine, and I take your articles and reviews seriously. However, there is another world that exists beyond bands, television and movies that I believe deserves recognition.

Christopher St. Hilaire
Via e-mail

SENIOR MOMENT

Oops! In my October article, "Adventures in Frequency-Conscious Compression," I wrote that a compressor's release time "is usually set for no faster than 1/10 ms to avoid distorting the waveform" when de-booming electric bass guitar. What I meant to say was 1/10 second, not millisecond. My brain's attack time was set to slow when I wrote that.

Michael Cooper

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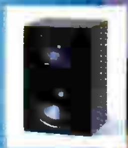
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CURRENT

CLAIR BROS. BUYS SHOWCO

Clair Brothers Audio Inc., of Lititz, Pa., has bought its most prominent rival in the sound reinforcement business, Dallas-based Showco Inc. According to Clair Bros. president and CEO Roy Clair, Showco's owners, Vari-Lite International Inc., decided to concentrate on that company's lighting business and approached Clair Bros. as a possible purchaser for the sound rental company. "We thought that the possibilities were very exciting," says Clair. "If you take two of the best companies and put them together, you should end up with a better company."

Clair Bros. will relocate Showco's equipment inventory to Lititz, where purchasing, workshop and maintenance operations for the two companies will be consolidated, though the proprietary equipment lines will be kept separate. "We are not interested in changing anything Showco is doing, because they are doing it well," Clair adds. "The phones will be answered as Showco, and their sales division will be taking care of their customers."

Showco will continue to operate a regional sales office in Dallas. "This is not an ugly situation at all," notes Showco VP of sales M.L. Procise. "The shop employees who are leaving were all offered positions with Vari-Lite or Clair, and we are keeping all of our road crews intact. We'll continue to service our 70-plus clients with Showco Prism® systems, and Clair will provide us with the financial resources for growth that Vari-Lite was no longer able to commit."

The merger is notable not only because Clair Bros. and Showco are two of the largest and longest established U.S. sound rental companies, but also because of the recent spate of takeovers and consolidations in the touring business. In recent years, the Production Resource Group (PRG) has consolidated the operations of ProMix, Burns Audio, A-1 Audio and Electrotec, and the SFX organization, now owned by radio conglomerate Clear Channel Communications, has consolidated most of the country's show and tour promotion companies.

—Chris Michie

EM EDITORS' CHOICE AWARDS

Mix's sister publication, *Electronic Musician,* announced its 10th Annual Editors' Choice Awards. Celebrating excellence in audio, musical instrument and computer products, the awards are based on products used or reviewed by the magazine's staff during the eligibility year ending October 1, 2000.

The 2001 award winners in 29 categories include: Alesis MasterLink ML-9600; Applied Acoustics Systems Tassman; Audio-Technica AT4047/SV mic; Big Briar CP-251; BLUE Dragonfly mic; Coda Finale 2000; Digigram VX-Pocket; FXpansion VST-DX Adapter 2.1; Hafler M-5 monitors; Korg D16 and MS2000R; Kurzweil K2600; Langevin Dual Vocal Combo; Lexicon MPX-500; Metasonix TS-21 Hellfire Modulator; MOTU Digital Performer 2.7; Near-field Multimedia/Buchla Marimba Lumina; NemeSys GigaStudio 160; Presonus Blue Tube; Roland ED U8 USB Digital Studio; Sounds Logical WaveWarp 1.2; Steinberg PPG Wave 2.V (VST) and WaveLab 3.0; Symbolic Sound Kyma 5.0; Tascam MX-2424 digital multitrack; Wave Mechanics Speed; Waves C4 Native and L2 Ultramaximizer; and Yamaha A5000 sampler. The Most Innovative Product Award was a tie, shared by Tactex MTC Express and Roland VP-9000 VariPhrase.

17TH ANNUAL TEC AWARDS CALL FOR ENTRIES

The Technical Excellence & Creativity Awards Nominating Panel is now accepting entries for product nominations for the 17th Annual TEC Awards, to be held September 22, 2001, in New York City. To be eligible, products must have been released and in commercial use during the period from March 1, 2000, to February 28, 2001.

Categories are Ancillary Equipment, Amplifier Technology, Mic Preamplifier Technology, Computer Software & Peripherals, Microphone Technology, Sound Reinforcement Loudspeaker Technology, Studio Monitor Technology, Musical Instrument Technology, Signal Processing Technology (Hardware), Signal Processing Technology (Software), Recording Devices/Storage Technology, Workstation Technology, Sound Reinforcement Console Technology, Small Format Console Technology and Large Format Console Technology.

Companies wishing to nominate products should send complete product name and qualifying category, date first commercially available (proof of shipment may be required; beta test sites *do not* qualify), and a contact name and telephone number.

Send all information to: TEC Awards, 1547 Palos Verdes Mall #294, Walnut Creek, CA 94596; fax 925/939-4022; or e-mail KarenTEC@aol.com. All entries must be postmarked by Thursday, February 1, 2001. For more information, call Karen Dunn at 925/939-6149.

NAPSTER, BMG ALLIANCE

The court battles against Napster may come to a screeching halt after Bertelsmann e-commerce Group and the online file-sharing music service announced a strategic alliance to change the original structure of the service by charging a fee.

The change is financed partly by Bertelsmann, which lent Napster an undisclosed amount to help implement a new business model and received an option to buy a stake in the company. Once the new service is in place, BMG will drop its lawsuit and begin plans to encourage other record companies to follow suit.

"This is a call for the industry to wake up," said chairman and chief executive of Bertelsmann Thomas Middelhoff. "It is not enough to fight file-sharing in the courtroom."

Under this agreement, Napster will compensate Bertelsmann and other record labels based on the number of their songs exchanged on the service by extracting part of the fee to pay for royalties.

It remains unclear when the new service will be implemented, how much the fee will be, how revenue will be shared and whether users will agree to pay for the service.

In other BeCG news, Matthew Katz's record label, San Francisco Sound, filed a lawsuit against the e-

—CONTINUED ON PAGE 16

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WORLD AUDIO NEWS

INDUSTRY NOTES

Soundtracs (Surrey, England) announced numerous distribution appointments: **HHB Communications** (Toronto) represents the company in Canada; **Sound & Vision** (Heerenveen, Netherlands) is the sole distributor in the Netherlands; **Danmon Norge A/S** (Kolbotn, Norway) covers Norway; **Audio International srl** (Milan, Italy) is responsible for Italy; and **TTS Professional Television AB** (Sundbyberg, Sweden) distributes in Sweden...**Lexicon** (Bedford, MA) named many appointments: **Larry Bennett** is the national sales manager for the professional division; **Steve Frankel** was promoted to technical sales manager and field applications engineer; the new executive administrator is **Robin Saragian**; new graphic designer **Jason Newton**; and **Jeremy Frost**, consumer product specialist...**Akai** (Fort Worth, TX) appointed **Sean Weitzmann** to the position of technical sales director at the Hollywood, CA, office...**Jeffrey Krull** was named **Shure's** (Evanston, IL) new VP of engineering...**Anne Bakoulis** leads the **Postworks** (New York City) sales team as the director of national and international sales...**Rhinoceros Visual Effects and Designs** (New York City) added **Harry Dorrington**, digital artist, and **David Barosin**, CG director and project leader...**Waves** (Knoxville, TN) promoted **Bob Reardon** to VP of marketing...**John Murray** returns to **TOA** (South San Francisco, CA) as product field specialist...**Saati Americas** (Somers, NY) appointed **Aida Boone** as the new bilingual customer service specialist...**Yamaha** (Buena Park, CA) announced many promotions: **Jim Wooten**, director of artist services; **Susan Davies Muhler**, artist services manager; **Craig Knudsen**, electronic keyboard marketing manager; and **Paul Calvin**, piano marketing manager...**Robert Hanson** joins **Leitch** (Chesapeake, VA) as senior VP of sales for the U.S....**ATTIK** (New York City) named **Ian J. Cohen** director of client services...The new senior product manager for **Rockford Corporation** (Tempe, AZ) is **Rex Whitehead**...**Raul Gonzalez** is the new product manager for tour sound, and **Simon Jones** is director of the European market for **JBL** (Northridge, CA)...**SADiE** (Nashville) brought aboard **Jayson Tomlin**, national sales manager...**Sam Spennacchio** was appointed to national sales director for **Crest Audio** (Paramus, NJ)...**Professional Audio Design** (Rockland, MA) announced new hires **Tony Belmont**, system consultant, and **Bruce Bartone**, director of marketing...**Full Compass** (Middleton, WI) promoted **Ron Rivers** to assistant national sales manager...**Panasonic** (Cypress, CA) announced that **Carl Marinoff** is the new area sales manager, and **Fred Jones** is now the national sales and marketing manager...**Ashley Audio** (Webster, NY) announced new sales representatives: **CM Sales Inc.** (Redford, MI) will cover Michigan; **Eakins/Bernstein** (Shawnee Mission, KS) will service Iowa, Kansas, Missouri and Nebraska; **New England Technical Associates** (Banford, CT) will represent Massachusetts, Connecticut, Rhode Island, Maine, Vermont and New Hampshire; **Signal Marketing Inc.** (Salt Lake City) will cover Colorado, Utah, Idaho, eastern Montana, Wyoming, Arizona, New Mexico and El Paso, TX; **B. Rich Company** (Penfield, NY) will service western, central, upstate and northern New York; **John B. Anthony Company** (Fairfield, NJ) will represent metropolitan New York and northern New Jersey; and **S.K. MacDonald Inc.'s** (Hunt Valley, MD) territory includes Delaware, Maryland, southern New Jersey, eastern Pennsylvania, the District of Columbia and Virginia...**DTS** (Agoura Hills, CA) announced that it is an associate member of Digital Video Broadcasting, a technical organization that creates standards for video and audio broadcasting...**D.Dino Virella** was promoted to VP of sales and marketing for **Universal Audio** (Santa Cruz, CA). ■

—FROM PAGE 12, CURRENT

commerce group, purporting that the alliance is aiding Napster to continue its file-sharing operations. Katz believes that this infringement on copyrights has adversely affected his business. Frank Sarfeld, senior VP and chief communications officer of BeCG, did not return phone calls.

HOT LINKS

Sabine announced that it has put an online book, *Digital Delay Advantage*, on its Web site, www.Sabine.com. It can be downloaded for free and explains delay theory, provides examples and offers how-to instructions, as well as several practical applications.

Fairlight launched its new Web site at www.fairlightesp.com. The site features support information, user manuals, user discussion forums, a distributor locator, news and events, and product information.

TC Electronic announced a new Web site dedicated to its System 6000 signal processor. The site, www.System 6000.com, contains software updates, customer support and user stories.

CORRECTIONS

Due to editing errors, the following schools had incorrect or missing information in the "Audio Education Directory" (November 2000).

University of Oregon, School of Music, 1225 University of Oregon, Eugene, OR 97403; Phone: 541/346-3761; E-mail: lgoren@oregon.uoregon.edu (undergraduate); gradmus@oregon.uoregon.edu (graduate); Web site: http://darkwing.uoregon.edu/~fmo. Degrees/Certificates Offered: B.S. Music Technology Option, Masters of Music in Intermedia Music Technology (final approval pending) and Intermedia Music Technology as a secondary area for doctoral students.

Full Sail Real World Education, 3300 University Boulevard, Winter Park, FL 32792; Phone: 800/226-7625; Fax: 407/678-0070; E-mail: admissions@fullsail.com; Web site: www.fullsail.com. Degrees/Certificates Offered: Associate of Science Degrees in Recording Arts, Show Production and Touring, Film and Video Production, Digital Media, Computer Animation and Game Design. ■

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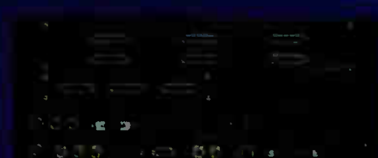

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INTO THE NEW MILLENNIUM WITH...MIDI?

KEEPING AN OLD FRIEND YOUNG

ILLUSTRATION RICHARD DOWNS

Well, this time it's for real. The new millennium. No more arguments about when it begins—it's here. The Y2K warnings and the Y2K jokes can be put away forever; the census has been taken, the presidential election circus (I hope!) is over and we're now in 2001, indisputably the 21st century.

Arthur Clarke and Stanley Kubrick were a bit over optimistic about what we were supposed to have by this year: There's no permanent space station above the Earth, and no Howard Johnson's restaurant in it (or anywhere else for that matter), and no one's building a nuclear-powered HAL/IBM-controlled starship to take humans to Jupiter's moons, or even our own. In his earlier writings, Clarke was right on the money about communications satellites covering the Earth, but he was dead wrong about the role of garbage at the turn of the 21st century: He saw it as a potential fuel source, not something to be delivered over microwave, twisted copper pair, coaxial cable and optical fiber to every home and office in every corner of the globe by the Petabyte. And he never foresaw the culture of the personal computer, the ubiquitousness of the Internet, the dynamic forces that are pulling the entertainment industry apart and putting it (maybe) back together in an entirely new way...or MIDI.

BY PAUL D. LEHRMAN

MIDI? "Why would anyone want to talk about MIDI in 2001?" I hear you cry. "I thought MIDI was like *so* last century!" Well, it was. But it's a little early to be digging its grave and dancing (no doubt using downloaded 124bpm loops) on it.

I'll admit it. I'm a MIDI-holic. Yes, my name is Paul, and I use MIDI. A lot. I compose with it, perform with it, mix with it, process with it, teach it and, yes, write about it. And except for the last one or two items, I'll bet most of you do many of the same things.

It's become so commonplace, so mundane that we don't even think about it anymore. And it's true that it's not very exciting, compared with the tools we now have for manipulating real audio. But even though messing around with MIDI data may not be as immediately gratifying as running old Rick James samples through Acid, it's still an important part of what our industry is all about (especially among those apparently dwindling numbers of us who value originality).

But MIDI is old stuff, right? And nothing's happening with it, right? Yes, it *is* old stuff (the MIDI Specification, after literally hundreds of changes in its 17-plus years, is still referred to as "1.0"), but to say that it's moribund is to ignore some very important work that's being done today to keep it useful in the age of digital video, T3 Internet connections and GigaHertz desktop computers.

A lot of the work, not surprisingly, has been on the consumer side of things, where marketers and manufacturers see potential numbers that exceed by orders of magnitude the size of the professional audio and music markets. MIDI is still viewed as an efficient and highly flexible way of handling music for games, Web sites and similar applications that require either low bandwidth or a high degree of interactivity. It's still a lot easier—and more convincing, if it's done right—to make a MIDI file instantaneously change the mood of a piece of background music in a game than it is with digital recordings, no matter how many tracks you might have to play with. And when a game designer has used up all available CPU speed and RAM on polygon generation and has forgotten to leave any room for audio, there's always enough space to slip in a MIDI file. As for the Internet, when you are dealing with typical dial-up connections (which most people still have), any audio file, even after you've crunched it through the compression algorithms of MP3 or Real Audio, goes down the pipeline *way* slower than a MIDI file.

While many consumers still associate MIDI with the cheesy FM sounds of early PC sound cards, even the cheapest "wavetable"-based chipsets of today sound a lot more respectable than that. (Wavetable is actually a misnomer for these devices, because they are, in fact, sample-based, and true wavetable synthesis is something completely different. But I won't get into that now.)

Much of the credit for the improvements can be taken by the MIDI industry's adoption—through its administrative body, the MIDI Manufacturers Association (MMA, www.midi.org)—of Downloadable Samples Level 1 (DLS-1). The significance of DLS-1, which is now almost four years old, is that instead of being stuck with the sounds a manufacturer puts into a synthesizer chip's ROM, or the 128 sounds in the original General MIDI specification, a composer or sound designer can create custom sounds in the form of samples. These can be downloaded as a block into dedicated RAM on the chip and then called up quickly and polyphonically from a MIDI file. In many ways, this makes for the best of both worlds: A 2MB sound set and a few hundred kilobytes of MIDI data can provide literally hours of high-quality, completely interactive music. (Another technology that follows the same general idea is Beatnik, Thomas Dolby Robertson's contribution to music on the Web.)

MIDI is still viewed as an efficient and highly flexible way of handling music for games, Web sites and similar applications.

But DLS-1 didn't solve everybody's problems. Even before it was developed, Creative Technology, the parent company of E-mu and Ensoniq, was working on its own version of this concept, calling it "Sound Fonts," which was similar to DLS-1 but with more advanced performance features.

DLS-1 and Sound Fonts threatened to cancel each other out, until Creative and the rest of the MIDI industry (as well as the MIT Media Lab and some other interested parties) came up with a higher functioning standard that was acceptable to everyone, and not proprietary to anyone (as Sound Fonts was). This is now known, not surprisingly, as DLS Level 2. The major improvements in DLS-2 are dynamic filters and matrix-based modulation, two features that are essential to any professional-level sampler or synthesizer. DLS-2 was formally adopted by the MMA in the summer of 1999 and has reached beyond the MIDI community to become part of the MPEG-4 standard, where it is called "Structured Audio Sample Bank Format."

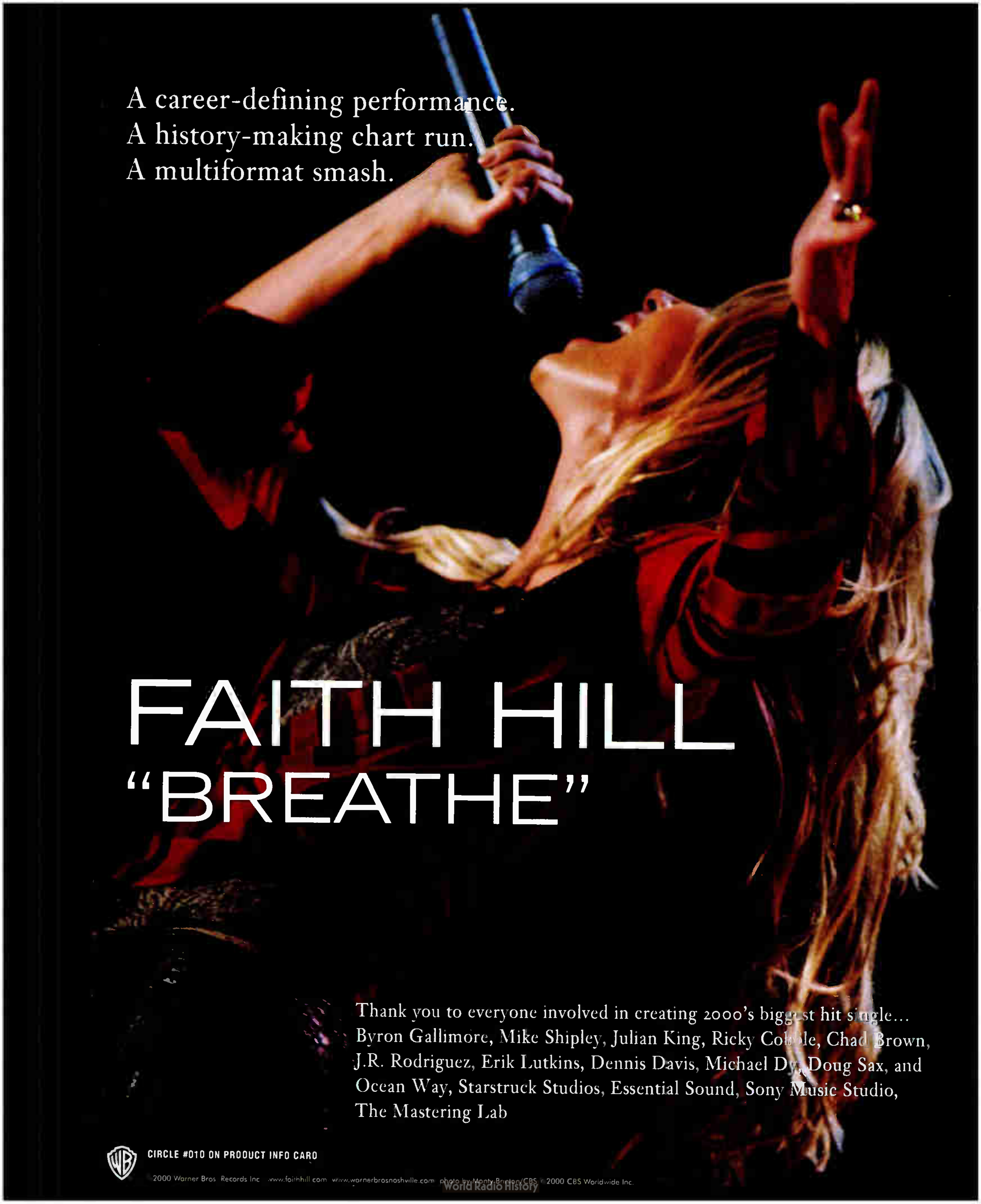
The first DLS-2 chips are about to hit the market, and one manufacturer claims that by the end of 2001, 40 to 60% of all computers being made will have DLS-2 sounds built right into the motherboard. On the game side, Microsoft is supporting the new standard in its upcoming X-Box platform.

Running parallel to DLS and DLS-2 has been the adoption of General MIDI Level 2. Before the ink was even dry on the original General MIDI Specification, which was supposed to ensure a high degree of file compatibility across many synthesis platforms (again, mainly in the consumer realm), Roland and Yamaha announced "extensions" to GM that were, of course, incompatible with each other. These extensions gave their devices more polyphony, effects like reverb and chorus, and an expanded sound palette. Other manufacturers of domestic keyboards and low-cost sound modules wanted to be able to improve the capabilities of their products, as well, but didn't want to have to license technology from Roland or Yamaha, or invent their own. So, they clamored for a nonproprietary expansion to GM.

GM Level 2 (formally adopted in November 1999) increases the minimum polyphony of an instrument from 16 to 32 voices, defines more controllers and more precisely than the original spec. For example, the new spec includes a formula for mapping MIDI volume controller values to amplitude in dB. (This is largely in response to a survey I designed in the early '90s on behalf of the MMA, in which it was found that controllers were being used very differently by different manufacturers.)

GM Level 2 also mandates and defines effects and significantly increases the number of available sounds, both instrumental and "rhythm," or percussive, using Bank Change commands to augment the 128 program changes. The advantage of a GM-2 instrument over a DLS-1 is simply that there is no sound set *at all* to download. So in applications where there isn't time or RAM for a downloadable sound set, music can play instantly; even at dial-up connection speeds, a MIDI file playing over the Internet is indistinguishable from one playing over a MIDI cable.

The first units to adopt GM Level 2 are from Roland, interestingly enough,



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and are the latest models in their Sound Canvas line, which started the whole General MIDI movement; and Korg—surprisingly, in its high-end Triton rack. More are expected to follow. Be sure to check out the product intros at winter NAMM this month.

But things are happening at the other end of MIDI, too—the professional end. The lowly MIDI cable, with its 31,250-bit/second speed, is ridiculously slow compared to today's networking and busing capabilities, and that fact has not been lost on the MIDI developer community. While MIDI over SCSI never was practical (SCSI is fast, but it works in spurts, which is okay for buffered digital audio, but not okay for the real-time control that MIDI requires), there have been strong efforts to incorporate MIDI with the newest networking protocols: USB and IEEE-1394, or FireWire.

USB MIDI interfaces have been around since early 1999. After Apple released the first USB Macintoshes, manufacturers like Emagic, Roland, Steinberg and Mark of the Unicorn scrambled to put out USB-compatible MIDI interfaces. Now there are a dozen or more on the market, from simple palm-sized 1-in, 1-out boxes to rackmount multicable interfaces with SMPTE and audio I/O. Happily, a standard method for putting MIDI on a USB cable is defined by the USB Implementers Forum (USB-IF, www.usb.org). Unhappily, the MIDI Manufacturers Association never endorsed the USB MIDI spec—and you'll see why in a moment.

USB has been very successful in replacing, or at least displacing, many of the disparate computer-networking formats like serial, parallel, PCI or SCSI ports. Printers, modems, scanners, removable media drives and gadgets we didn't even know we needed just a couple of years ago are now using USB cables. There are great advantages to USB, such as the ability to connect up to 127 devices of all kinds to a single computer (using bridges and hubs), automatic configuration (no more IRQ or SCSI ID nightmares), the ability to "hot-swap" devices, and higher potential throughput than any of the formats it replaces, with the exception of SCSI.

So what's the problem with MIDI? According to Jim Wright at IBM Research, a longtime member of the MMA Technical Standards Board and chairman of the organization's working group concerned with new transports, USB has timing problems that make it problematic for MIDI. He has conducted tests comparing "classic" (i.e., serial, parallel, PCI or PCMCIA) interfaces against USB interfaces, looking at their round-trip latency (the amount of time it takes for a MIDI event to get in and out of the interface) and their jitter (the variation in the latency). He found the latency in the USB interfaces to be between seven and eight milliseconds, about three times that of the classic interfaces. This is not in itself an insurmountable problem, because musicians adjust to small latencies in sound sources quite well—a bass player and a lead guitarist standing seven feet away from each other usually have no trouble staying together.

None of these solutions are compatible with each other, which negates the entire philosophy of MIDI and USB.

But the jitter in USB interfaces was also much higher than the older interfaces—about twice as high, meaning (to continue our analogy) that the two players could at any given moment be five feet away from each other, and the next moment be 10 feet away—and constantly moving. In another analogy, which Wright likes to use, imagine playing a slightly arpeggiated guitar chord: The jitter could make it sound as if one of your fingers jerked slightly while you were playing the chord. And for tight grooves and thick MIDI data streams with lots of aftertouch or controllers, this level of jitter is really unacceptable. Wright also found that when you add audio to the USB stream, the jitter goes up another 50%—so it's three times what MIDI musicians have had to deal with in the past.

Why is this the case? Well, the USB developers, according to Wright, came to the MIDI community very late in their development stage, and thus the MMA and its Japanese counterpart, AMEI, didn't have much of a chance to give their input about how MIDI on USB was going to be handled (although Roland, acting on its own, got involved much earlier). On a USB cable, MIDI uses asynchronous timing (that is, there's no underlying clock as there is with, say, AES/EBU digital audio), which means if there's a lot of traffic on the line, then the MIDI data will be delivered in fits and starts, and there's no guaranteed delivery time, even under the best of circumstances. (The same is true for a standard MIDI cable, but preventing this is what multiport interfaces are for!)

Audio on USB, on the other hand, uses isochronous timing, which means the delivery time *is* guaranteed. So the problem is further compounded by the fact that because they use different timing schemes, MIDI and audio data on the same USB cable can easily lose sync with each other. Getting MIDI and audio to work together in perfect sync is something software and hardware developers have labored hard for years to achieve, and now we're potentially seeing all those efforts being tossed away.

The interface manufacturers are not unaware of these problems—it's this very issue that's behind the huge advertising campaign that MOTU has been running promoting its "MTS," a proprietary system of time-stamping MIDI events as they enter the USB cable to overcome USB's timing problems. Time-stamping of MIDI events has never really been necessary before, because the latency and jitter of the synthesizers *themselves* have been greater than that of any delays in the MIDI network (or the resolution of MIDI itself, for that matter), but that's no longer true with USB. Emagic has followed MOTU's lead and is using its own version of time-stamping, and Steinberg is reportedly planning something similar.

But it's the same old song: None of these solutions are compatible with each other, which negates the entire philosophy of MIDI *and* USB. MOTU's MTS works only if you have the company's software and hardware and not with Emagic's hardware or Steinberg's software, and *vice versa, et cetera, ad infinitum.*

It's the computer manufacturers who are potentially in the best position to do something about this, and perhaps they will. Mac OS X might include time-stamping in its MIDI drivers, according to some sources. Doug Wyatt, the developer of the Opcode MIDI System, the best software driver for multiport MIDI on the Macintosh (and the primary casualty in the train wreck Gibson

—CONTINUED ON PAGE 244

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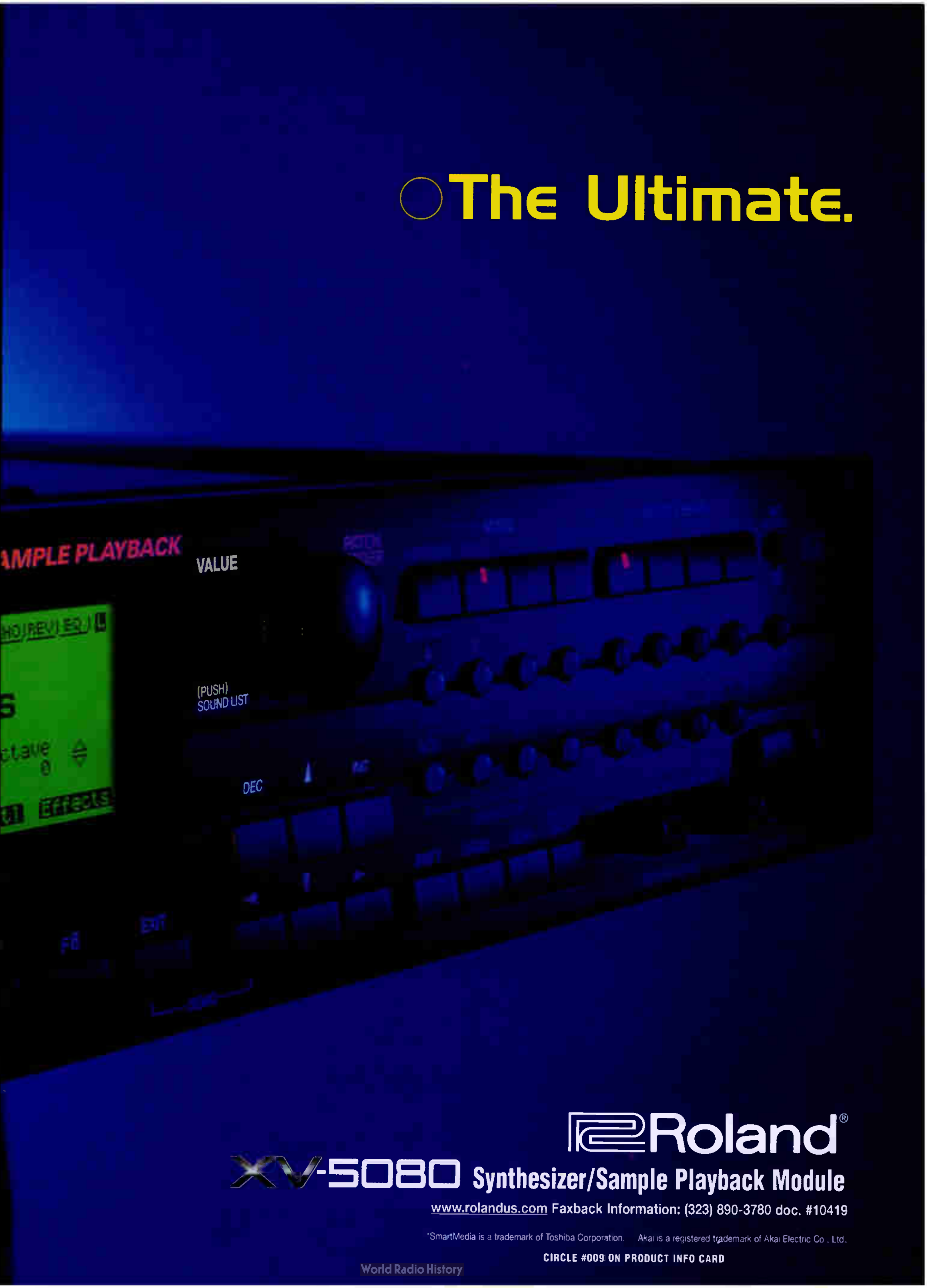
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MP THREATS

As I walk along,
I wonder what went wrong
with our love,
a love that was so strong...
— "Runaway," Del Shannon, 1961

Ah...one of my favorite songs. And now, you can just go get it online for free, along with every other song in the galaxy. Mmmm. Sweeet!

Not too long ago, I went out with a friend to look at new cars, and the BMW salesman presented me with a CD as I left—a compilation of songs that he thought I might like after he had spent an hour with me. All downloaded and burned onto a CD while we were talking. Nice!

At a small dinner party recently, I was introduced as an audio and music industry VIP to a friend of the hostess. The friend had no idea who I was but was anxious to make an impression because of *what* I was, so he launched into a detailed story of how he has downloaded more than *30,000* free MP3s from various sites on the Net, with Napster of course being his main source. He told me that he has a room full of CDs that he has burned that he shares with friends. He concluded his presentation with a request to be hooked up to buy blanks cheaper than 38 cents, as "he had only just begun." Excellent!

When I was younger, I knew of a guy who worked in one of those mall record stores. It seemed that if you were one of his friends, you could phone in a request list, and he would collect the 20, 30, 40 or more records that you wanted and drop them in the dumpster out back with the daily trash. All you had to do to get your tunes was drive by after 11 or so, stop by the dumpster, reach in and yank out your Christmas bag. And it was *free*! Awesome!

COUNTRY JOE AND THE FISH HAD IT RIGHT

One, two, three, what are we waiting for?
C'mon, we don't give a damn!
It's only the Record Man.

Well, times, they are a changin'. Everybody's Dylan up and downloadin' now.

The advent of MP3 downloading is forcing the music industry to restructure itself, to say the least. To say the most, it may well kill the music manufacturing and distribution industry as we know it. But to say something in between is probably more realistic.

But...aren't we all tired of the faceless record companies and their heartless commercialized approach to making records? I mean, we all know that a new artist doesn't have a chance any more. The labels will release that third *Best of The Jerryfish* compilation before they would ever dream of taking a chance with new talent. A fact of Modern Life. And now even a successful signed artist can't get his or her next album out if their last one didn't do as well as the one before, even if that last one returned good numbers.

It is clear that the time has come to change all this. The Man must be crushed so that Art and Creativity, and, yes, even experimentation can return and once again flourish in the new fertile soil of the free Internet! Eliminate the profit mongers, the megacorps with their formula marketing! Eliminate the middlemen! Hell, with Napster and the other pretenders, we can finally eliminate all the men. Music by the people, for the people. And I mean *all* the music, for *all* the people. Yes, we, as citizens of the new free virtual world, have a right to our music the way we want it...and we want it now, and we want it for free!

BY STEPHEN ST.CROIX

Bullshit! Every word of it.

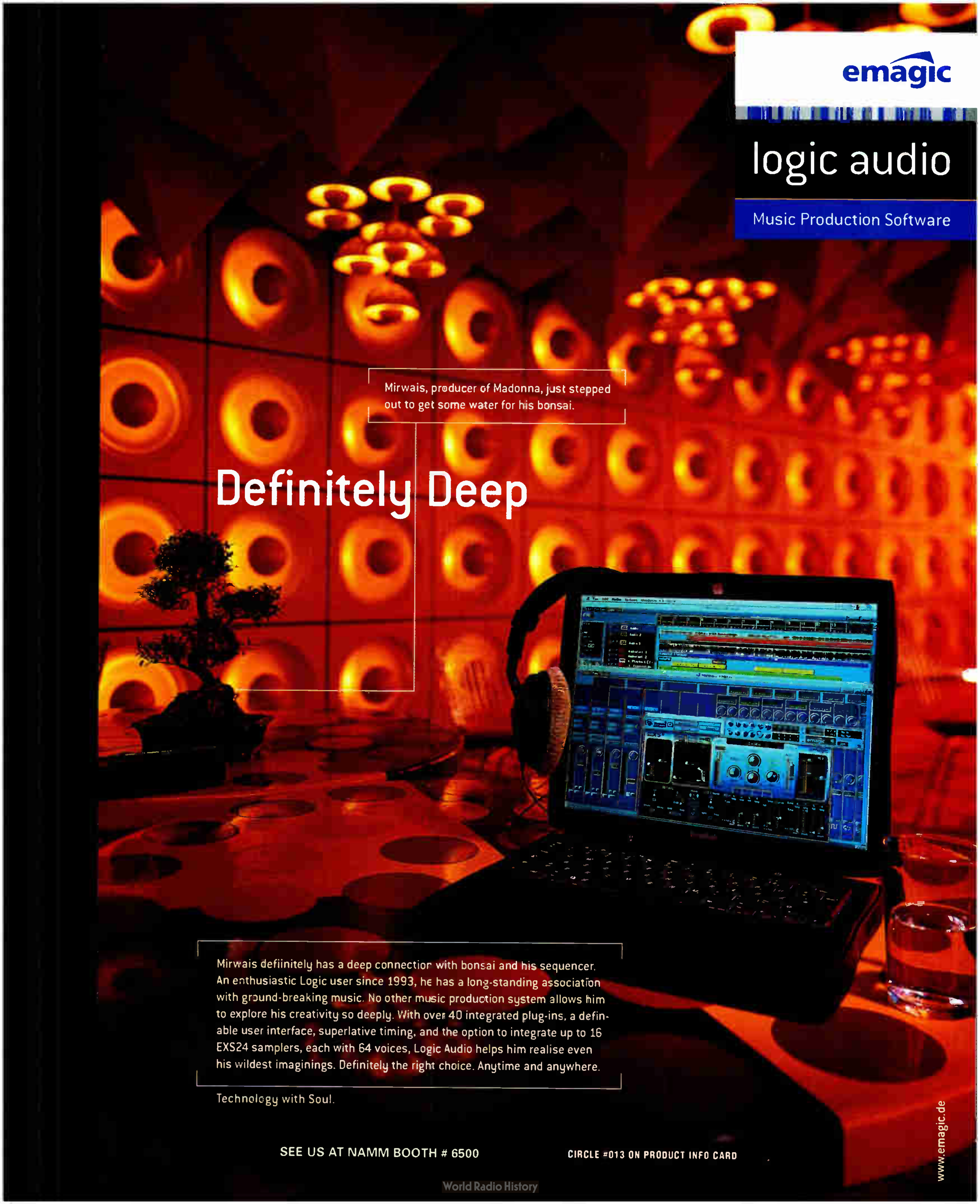
The MP3 arguments *on both sides* are so twisted, so evasive, so self-serving and so damned confusing that there is almost no real data there for a rational observer to use.

First of all, there aren't two sides. There are about 20 sides. And almost nobody has any friggin' idea what they are talking about. It has become a parody of itself. It's like a bunch of high-end audio reps working a promising customer: Each one tells how the things that their product *can* do are the most important things in the world, while the things their product *can't* do are impossible, meaningless hype, or just bad science. MP3 objectivity is just about the only thing that doesn't exist in all this rhetoric today.

MP3 objectivity is just about the only thing that doesn't exist in all this rhetoric today.

WARNING, WARNING, WARNING!

As you proceed, you must be made aware that for the first time in my adult life, I no longer have three feet of official sun-streaked, anti-establishment, hippie-surfer-musician-biker-type hair sticking out of my head. A little over a month ago I shaved it all off, against my will. And now that it is gone, I will never grow it back. I'm


emagic
logic audio
Music Production Software
Mirwais, producer of Madonna, just stepped out to get some water for his bonsai.
Definitely Deep
Mirwais defiinitely has a deep connection with bonsai and his sequencer. An enthusiastic Logic user since 1993, he has a long-standing association with ground-breaking music. No other music production system allows him to explore his creativity so deeply. With over 40 integrated plug-ins, a definable user interface, superlative timing, and the option to integrate up to 16 EXS24 samplers, each with 64 voices, Logic Audio helps him realise even his wildest imaginings. Definitely the right choice. Anytime and anywhere.
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now doing the shaved, side-spiked, bleached-top, snowboarder doo. So just keep my new appearance in mind as you read on.

Peace, love and flowers (and communal music) worked back in the old days when the world stretched as far as you could see. If Mary Jane copied an Airplane cassette for Sky Marshall, and he copied it again for Crystal when she moved into the commune, it did not force an emergency profit and loss meeting at The Label. The Label simply couldn't feel it.

Nevertheless, The Label's unbelievable greed and paranoia went a long way to keep Mary, Sky and Crystal against the establishment for decades. Federal lawsuits based on dire predictions of the collapse of known civilization if open-reel tape, and later cassettes and then DAT and CD-R, became openly available just agitated everybody.

But let's get back to the '60s, even if it is just so that we can leave them behind...It ain't the old days anymore. Now the world stretches as far, well, as far as it *is*. The Web truly makes the entire planet one place, one connected, immediate village. And everyone with a computer and a modem is a citizen. And it is just this, the incomprehensible *scope* of this new world, its immediacy and its accessibility, that makes it totally different from any previous model in history.

My new snowboarder self doesn't really care too much for the naive hippie view of what's right anymore. Like the anti-British minutemen hawks before them, the anti-establishment '60s doves are simply obsolete. The original reasons for either of their existences just don't apply anymore.

The British are not too likely to come storming over the fences of your farm these days hurling deadly boiled hamburgers, so it might not really be so important for every single person to carry a rifle to work every single day.

And the passive-peace commune culture is equally as obsolete. Those values no longer functionally apply in a world where everybody actually gets a public voice—and that voice is *instantly* available around the entire globe, with no censorship and no policing. Today, to remain passive is to disappear.

I speak of this from experience. Those supporting MP3 downloading claim to have similar values as the hippies before them, but as an ex-hippie in good standing, I can tell you that these are nowhere near our old values.

An act of subterfuge such as posting a song on the Web for free download has, by the very instrument of its availability, *global* implications, ridiculous arguments to the contrary notwithstanding.

THE CREATOR

I should say that I *still* have no love or sympathy for those mega-labels that refuse new artists for the lame reasons that they do. I too see the Internet and its naturally appealing immediacy as provocative enough to threaten the Old Ways. This is evolution, this is technological advancement, this is unavoidable growth. But...

If each song is stolen and posted for free worldwide download on the day of its release, then it won't take long at all until nobody can afford to make music anymore. These creators do this for a living.

In all the clouded skirmishes and ill-conceived legal posturing, the only point that matters is getting totally buried. Very buried. Is the MP3 issue about MP3? The compression format? Is it about who posts them? Maybe it's about how they sound, or how to bury copyright data in the stream to fingerprint them. Nah, that's the noise, that ain't what it's about. This is:

The illegal posting and free downloading of commercially released or illegally recorded music hurts the *creators*.

In all this banter, nobody seems to talk about the musicians and the engineers and studios that are part of the creative process. These people are the heart and soul of the end product. They had the visions, they literally *created* the concept, and they are the people who worked to translate the visions into finished songs specifically for the world to buy and listen to. *For the world to BUY and listen to.*

If each song is stolen and posted for free worldwide download on the day of its release, then it won't take long at all until nobody can afford to make music anymore. These creators do this for a living. And when they can't survive, they will simply disappear.

Now, I am totally aware of bands that do promotional and direct shareware-type marketing on the Web. This is great. It exposes the world to an endless wealth of new and experimental talent that would not otherwise be heard. But you know that I am talking about something else. Four thousand downloads of a single tune in a week without a penny in royalties going to the creators. This is insane.

And those lame-ass arguments that CD sales went up, not down, because of the free stolen downloads...I *love* statistical analysis. You can make any group of numbers say anything you want, ridiculous as they may be. "Oh, yeah, I stole your bread, but because you had to go out in the snow that night to earn more money to buy more bread for your family, you are now even stronger and more fit"...*This* is Napster's answer? What a spin!

Whatever the outcome, it is illegal to steal the musical bread, and those who choose to make a living at the expense of others, those who have chosen victimization as a career, should be prosecuted and neutralized. Isn't this type of protection one of the fundamental advantages offered by a healthy society?

As for me, every oldie that I want is out there, for free, right now. I have been looking for some of this stuff for a decade or more, but I choose to take the high road. I will not download, I will not steal from my friends and the creators that came before them.

But to be honest, it's not so hard to ride this high horse, as the audio quality of 90% of these things is so horrendous that the temptation disappears as soon as you hear them. But *that's* another column.

So, think it over and take your stand. As we have learned in Florida, every single vote *does* count. There are only two real choices here: Bow your head and accept piracy as part of your life from this day on; or stand by me and stop this attack on our industry and our lives. ■

SSC says, "damn!"

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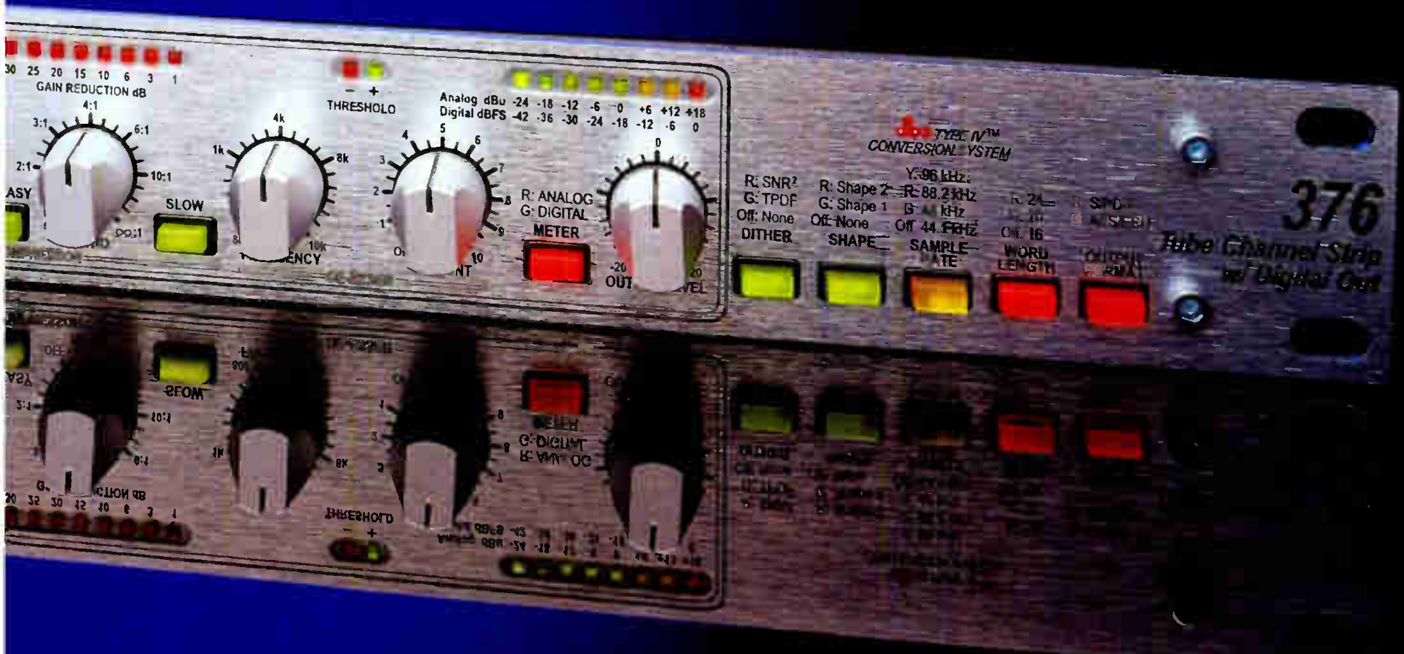

It's Time to Change the Channel...

In the digital world, you're only as good as your weakest link or channel. So, why settle for any less than the warmth and integrity offered by the new **dbx 376 Tube Preamp Channel Strip with 96k Digital Outs**? For more than 25 years dbx has been setting the standard, and the 376 is the culmination of those years of innovative success.

Tube Preamp Channel Strip w/ Digital Out

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With a feature list that includes a vacuum tube preamp section, three-band parametric EQ, compressor and the *real* kickers: built-in De-Essing and AES/EBU and S/PDIF digital outputs, make the 376 an all inclusive Channel Strip toolbox that's just as much at home in a conventional analog mixer application as it is in a state-of-the art digital workstation. With the 376 you can bypass the mixer all together, while producing warm and rich tube mic preamp tones in the digital domain by using dbx®'s proprietary Type IV™ A/D conversion system. Stop by your local authorized dbx dealer and tune into all that is available on the next channel.

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It's Alive!

It was another banner year for the touring industry, with a number of acts filling arenas, sheds and even stadiums across the country. This was the year that successful CDs by teen pop idols also translated into boffo box office. The most successful tour of the year was unquestionably 'N Sync's sold-out stadium and arena tour, a massive and magnificent production profiled in these pages in our November 2000 issue. Britney Spears and Christina Aguilera also crisscrossed the country to packed houses, not to mention playing on what seemed to be 100 different awards shows on television. The year began with the Backstreet Boys still touring behind their last CD; now they've put out another disc (*Black and Blue*), and their 2001 tour promises to be one of the most popular of the year.

ON THE COVER

The other big stadium tour was once again the George Strait Country Music Festival, which brought together such acts as Strait, Tim McGraw, Martina McBride, Dixie Chicks and Asleep at the Wheel. The Dixie Chicks and McGraw (with wife Faith Hill) also mounted very successful tours of their own.

If it was loud, aggressive rock you were looking for in 2000, then there was plenty to choose from, including tours by Metallica, Korn, Limp Bizkit, Papa Roach (see page 86) and veteran units such as AC/DC and Kiss (on their alleged farewell tour), among others.

Baby boomers shelled out sky-high prices for acts such as Crosby, Stills, Nash & Young, Steely Dan, Roger Waters, The Who, Joni Mitchell, Tina Turner and the Moody Blues. The unchecked rise of concert ticket prices remains a scandal that seems to only get worse with each tour by "name" artists. Only the hideously overpriced Diana Ross and the faux Supremes tour met with outright rejection by the public.


Christina Aguilera live

PHOTOS: STEVE JENNINGS

Other big tours of 2000 included jaunts by Phish, the Dave Matthews Band, Jimmy Buffett, TLC, Kid Rock, DMX, Mariah Carey and, continuing on from last year's tours, Ricky Martin and Bruce Springsteen (no, not together). And, of course, there were a zillion lower-profile tours that filled clubs and theaters—still the best places to see and hear music.

•••••

THE FOUR ACTS ON OUR COVER THIS MONTH:

Snoop Dogg was part of the popular Up In Smoke hip hop tour (see *Mix*, December 2000). The tour featured a flown V-DOSC system (from ProMix), subwoofer cabinets provided by Maryland Sound, EAW wedges and two Midas XL4 consoles (courtesy of Eighth Day Sound) for FOH and monitors, which were handled by Tim Colvard and Sean Sturge, respectively. Dr. Dre and Eminem were also featured on the tour.

Scott Weiland of **Stone Temple Pilots** did his thing through a Showco Prism system. The FOH engineer was Jimmy Huth at a Midas XL4 console; the Ultimate Ears Pro 5 in-ear monitors were handled by Maxie Williams on a Midas XL3. Ninety percent of the mics were by Shure.

Teen queen **Christina Aguilera**'s tour is profiled in detail in our "All Access" column on page 72.

Did anyone have a bigger year than **Carlos Santana**? His *Supernatural* album was bought by more people than voted for George W. Bush (okay, we're exaggerating a little), and he needed a Ryder truck to haul home his cache of Grammys, Latin Grammys, People's Choice, VH-1 and other awards. But mostly he did what he always does, which is tour incessantly. The world tour carried a Showco Prism system; FOH engineer Randy Piotroski helmed a Midas XL4 console, while monitor mixer Brian Montgomery worked on a Midas Heritage 3000. Mics were from Shure, AKG, E-V, Sennheiser and Beyer. ■


Stone Temple Pilots

THE TOY BOX

BY ROGER MAYCOCK

New Digital Effects Processors

Getting "your sound" involves much more than just what you print to tape, hard disk, MO or some other recording medium. The signal processing tools in your racks play a pivotal role in the overall sound of your finished product. During this past year, some slick, new digital signal processors have been introduced—something for just about everyone.

Alesis airFX™

The digital products included here are those that handle reverb, delay, assorted multi-effects, vocal effects, microphone modeling and intonation processing, rather than mic preamps, mastering tools or dynamics processors. Visit the manufacturers' Web sites for more information.

ALESIS AIRFX

Geared toward the DJ market but capable of integrating into any number of production environments, airFX™ from Alesis (www.alesis.com) is a thoroughly modern adaptation of the Theremin, whereby the device processes incoming audio or generates sound effects in reaction to the operator's hand movements. Employing the company's new Axyz (pronounced ax-is) technology, airFX incorporates an infrared beam that can be manipulated in three dimensions along X, Y and Z axes. Using triangulation, up to five different parameters can be controlled in real time in each of 50 preset programs. The unit's finish is a bright, anodized blue with a black dome, and a large, silver circular knob. Select your effect, wave your hand over and across the dome, and you're in business. MSRP: $249.99.

ANTARES AMM-1

Imagine having access to many of the best studio mics available—without having to rent or borrow them. That's the idea behind the Antares AMM-1 Mic Modeler (www.antares tech.com), a rackmount, hardware version of its popular Mic Modeler plug-ins. Using proprietary Spectral Shaping technology, Antares created a series of digital models from AKG, Audio-Technica, Beyer, Neumann and others. Depending on the selected mic, users can control options such as low-cut filter, proximity, windscreen, etc. A tube saturation parameter is also provided. MSRP: $995.

ART HPFX

The HPFX from ART (www.artroch.com) solves a common problem among computer-based recording systems—it enables performers to locally monitor themselves through digital effects, without having to experience latency delays caused by the computer's soundcard. The unit also lets performers control their monitor mix, including mic, effects and backing tracks, while leaving the recorded signal undisturbed. The HPFX features two monitor mic preamps, four independent and headphone outputs, and dual digital effects. Further, the unit can be racks or stand mounted, provides stereo or mono mix input, and can also be used as a conventional headphone amp/splitter. MSRP: $299.

CARVIN XP2

The XP2 from Carvin (www.carvin.com) is a dual-engine, 24-bit digital effects processor designed for professional studio and live sound reinforcement applications. The XP2 includes plate, hall and room reverbs, chorus, delay (with a selectable loop filter, precise delay time selection, and up to one second of delay per effect engine), flanger, phaser and echo. The unit also provides a rotary speaker effect and features the ability to change drum and/or rotor slew rate. Every function of the XP2 is controllable through the unit's extensive MIDI implementation. Additionally, the XP2 provides digital "serial linking" to eliminate noisy external patching. There are also provisions for footswitch access. MSRP: $395.

CRATE SM1-SP

The Crate (www.crateproaudio.com) SM1-SP is part of Crate's Studio Modules lineup, which consists of four additional mod-

ules—all of which are half-rack in size. The SM1-SP incorporates 32 programs, including reverb, delay, chorus, flange and rotary. One parameter for each effect can be edited in real time. Applications include sound reinforcement, recording, keyboard, guitar and bass performance, post-production and broadcast. MSRP: $179.99.

DYNATRON 255

Distributed by G Prime (www.gprime.com), the Dynatron 255 puts the classic reverb of the EMT 250 into an updated package. The original algorithms have been preserved, and the processor is now in an efficient two-rackspace unit ready for 5.1 Surround. Plus, new settings offer a greater range of creative options. The Dynatron 255 is the rebirth of the benchmark in electronic reverb, ready for the demands of today's studio. Initial delay settings are variable from 0 to 140 ms in 20ms steps. The unit's main reverb decay time setting is variable from 0.4 to 4.5 seconds. The unit's low-frequency decay ratio is adjustable from 0.5x to 2x that of main reverb time. The high-frequency decay ratio is adjustable from 0.25x to max (flat) of main reverb time. All inputs/outputs on the unit are standard AES3 digital audio format with XLR connectors. MSRP: $5,995.

DIGITECH RP100

The RP100 Modeling Guitar Processor from DigiTech (www.digitech.com) combines vintage and modern amp models with full-featured studio effects. There are 25 programmable effects to choose from, and up to 10 effects are available simultaneously. Each effect includes up to three adjustable parameters, and other features include a Rhythm Trainer sampled drum beat loop player and a chromatic tuner with four different tuning references. The RP100 has 40 factory presets and 40 user presets. The rear panel features an expression pedal input for real-time control of the RP100's parameters. MSRP: $129.95.

ELECTRIX REPEATER

While not a signal processor in the traditional sense, the rackmount Repeater loop-based sampler from Electrix (www.elec trixpro.com) offers fingertip control over up to four channels of audio loops with independent, real-time manipulation over tempo, key and track offset. Loops can be synched to any source using Repeater's MIDI clock or Audio Beat Detection, and Repeater uses CompactFlash media for easy memory expansion (up to 50 minutes of record time) and Mac/PC compatibility. MSRP: $699.99.

Eventide DSP7000 Ultra-Harmonizer

EVENTIDE DSP7000 ULTRA-HARMONIZER

With four times the processing power of the previous generation DSP4000, the DSP7000 from Eventide (www.eventide.com) is a stereo processor featuring 24-bit conversion at 96 kHz. The unit provides both analog and digital I/O with separate gain controls and metering for each domain. It offers the user extensive programmability and remote control capability. Over 150 user presets can be stored internally, and hundreds more can be stored on removable PCMCIA cards. The DSP7000 ships with over 500 factory presets.

The DSP7000 accommodates presets and preset library cards developed for the DSP4000 Series Ultra-Harmonizer units. Eventide's PC-based Vsigfile graphic editor can be used to create new effects for the DSP7000, and factory software upgrades can be downloaded to the unit from an Internet-connected PC. MSRP: $3,695.

JOHNSON J-STATION

The J-Station from Johnson Amplification (www.johnson-amp.com) puts the modeling technology from the company's "digital" guitar amps into a desktop unit for recording applications. Retailing at $449.95, the J-Station offers bass and acoustic guitar simulation, 30 factory and 30 user presets, compression, modulation, pitch shifting delay and reverb effects, seven gain and tone knobs for tweaking effect parameters, MIDI control (via a sequencer, external controller or included Windows-based editor/librarian software), and stereo analog and S/PDIF coaxial outputs.

KORG AX100G

The latest addition to Korg's (www.korg.com) ToneWorks multi-effects line, the AX100G is equipped with 80 multi-effects programs (40 preset programs and 40 user program locations) and 63 effects, plus a selection of features that include a built-in chromatic tuner, intelligent pitch shifter, phrase trainer, integral Expression Pedal that lets the player control 23 different types of effects in real time, and 50 PCM-based rhythm patterns.

The AX100G employs Korg's REMS technology (Resonant structure and Electronic circuit Modeling System) and allows up to seven effects to be combined per multi-effect program. Each multi-effects program features two changeable drive channels, allowing the player to switch between the clean and distorted mode without having to switch to another program. The Effect Select Mode can be used in conjunction with this feature for instant on/off switching of modulation and ambient effects. A cabinet resonator speaker simulator is provided for recording or P.A. use. MSRP: $250.

LEXICON 960L

The 960L is Lexicon's (www.lexicon.com) newest multichannel digital effects system designed for the high-end professional audio, broadcast, film, live sound and post-production markets. The unit features true multichannel surround sound processing with new algorithms and 3DPM Perceptual Modeling, featuring 24-bit/96kHz processing, expandable 8 in/8 out architecture, plus an all new LARC2 Controller with an intuitive user interface. There are 250 factory programs/1000 registers and a joystick for surround panning and parameters. Just released, Version 2 software for the 960L supports an additional DSP reverb card, provides mappable I/O, support for 16 channels of I/O, additional presets, dual LARC2 controller support and enhanced input metering.

Lexicon 960L

Lexicon has two models: the 960L and the 960L Digital Only, which eliminates the analog I/O capability of the 960L. The standard configuration for the 960L includes eight 24-bit/96kHz balanced XLR analog I/Os, four stereo pairs of 24-bit/96kHz AES/EBU digital I/Os, MIDI, wordclock, a 3.5-inch floppy and a CD-ROM drive for software upgrades. MSRP: $15,000 (960L); $11,995 (digital only model).

LINE 6 POD PRO

The POD Pro from Line 6 (www.line6.com) is a guitar effects processor that includes 32 amp models and 16 digital effects in a rackmount package. The unit includes 36 factory presets, 36 user programmable configurations and has a headphone output. Digital I/O includes 24-bit support for both AES/EBU

Line 6 POD Pro

media production system

product placement...

The most critical part of any audio production has always been the final mix and in today's world that means being able to work with surround sound. Until recently this type of mixing power was available only to a select few. But not any longer...

Nuendo's integrated 5.1 surround mixing means you don't have to buy expensive plug-ins or hardware to get the perfect mix. Nuendo lets you work in any surround format you choose with support for up to eight channels and includes presets for 5.1, 6.1, 7.1, LCRS and more.

Nuendo includes LCRS Encoder and Decoder plug-ins and works with surround encoded material right out of the box. A dedicated SurroundPan plug-in lets you freely position your audio within the surround field and features two modes: Position, which lets you work in classic cinematic situations with variable distances from the center, and Angle which defines an equal distance from the listen position. Nuendo's Master setup window offers total freedom to modify your studio's speaker placement for creating your own custom surround presets, plus any VST compatible stereo plug-in can be inserted into a surround configuration creating a whole new dimension in real life audio production.

Nuendo supports the VST 2.0 Surround protocol and using the optional Surround Edition collection of plug-ins from Steinberg Spectral Design you'll get up to eight channels of audio processing all in real-time.

Six plug-ins are included featuring a single band compressor, Loudness Maximizer, 7-Band Parametric Equalizer, a natural sounding reverb, plus an LFE Splitter and combiner for working with low frequency channels.

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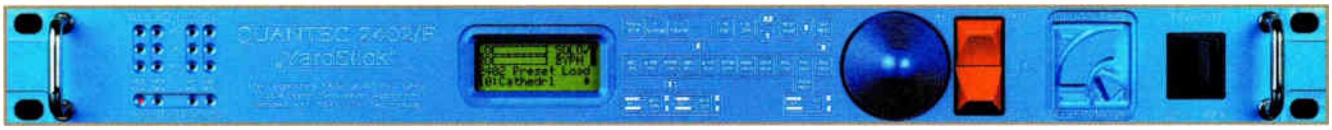
www.us.steinberg.net • U.S. (818) 678-5100 • Fax. (818) 678-5199 • Fax On Demand: (800) 888-7510 • Canada: (416) 789-7100 • Fax: (416) 789-1667 and S/PDIF. There is also a dual-mode XLR direct out. With the optional floorboard (model 99; $330), you can take advantage of remote channel switching, Wah and stomp box-style effects On/Off. You also gain a volume pedal, tap tempo function and tuner. MSRP: $799.

Quantec Yardstick 2402F



QUANTEC YARDSTICK 2402F

Now distributed by HHB (www.hhbusa.com), the Quantec Yardstick takes ambience and room simulation effects into the digital domain. The original QRS (Quantec Room Simulator) algorithm, along with the Room Size and Enhance parameters, are now DSP functions. There is also a constant density mode for special effects. Bar graph meters for input and output levels are available on both the front panel and remote control, and the unit provides full 24-bit stereo I/O for direct signal and delay lines. The QRS algorithm requires a minimum of 16-bits. Saving and loading of user presets is accomplished via RS-232 from a PC or via MIDI message. An optional variable-edge, brickwall, high-cut filter is available for simulating the timbre of the original QRS' 8kHz bandwidth. MSRP: $2,695.

ROLAND EF-303 GROOVE EFFECTS

Designed to bring pro quality effects processing to musicians, DJs and remixers, the Roland (www.rolandus.com) EF-303 Groove Effects provides real-time controls and groove-oriented effects, including a built-in DSP monosynth with Step Modulator, plus extensive MIDI control capabilities. The tabletop unit offers 16 synchronizable effects algorithms, including filters, flanger, phaser, delay, pitch shift, plus "groove" effects like Isolator, Slicer, lo-fi and Voice Transformer. The EF-303's design provides four control knobs, 16 sliders and a Grab switch. There is an automatic BPM counter and MIDI Start/Stop control, along with a Step Modulator for programming complex effect changes over time. There are 16 Patch memories for storing effects settings, and the unit accommodates a variety of input sources. MSRP: $595.

SONY DRE-S777 SAMPLING DIGITAL REVERB

While available for a year, the DRE-S777 Sampling Digital Reverb from Sony(www.sony.com/proaudio) never actually

—CONTINUED ON PAGE 218

AKG.EMOTION

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- D 550: New bass instrument microphone for kick drum, tuba, trombone, and bass amps.
- D 660 S: Price-leading, dynamic hyper-cardioid vocal microphone.
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- D 880: Super-cardioid vocal mic with great gain-before-feedback; new rugged steel mesh grille.
- D 880 S: Same as D 880 except with a silent on/off switch.

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Sequencing

Music has forever changed, and we've all seen it coming. Computer-manipulated sound has become a ubiquitous component in music production, and the sounds on many of today's records are so far removed from anything that could ever be duplicated in the "really-real world" that it's not even worth debating. And we're not talking about some obscure group of German nihilists banging away at MIDI triggers; pick up almost any album produced in the last two years that's cracked the Top 200 and ask yourself: Which instruments are real? What's been tweaked? And could *that* sound ever be reproduced live?

DAWS...THE NEW GUITAR?

Whether something was sampled, stolen or actually played by a human is a question that the record store employees of the world can answer. How to reproduce those sounds live, however, is a real-world problem that hundreds of musicians already face. Imagine that you've spent however many months in endless Pro Tools sessions crafting your *White Album*, and it's time to play out. An even more likely scenario for *Mix*-reading live sound engineers is that you're handed 70 minutes of material that has never seen the light of day outside of a hard drive and told, "We're going on the road in six weeks, make sure it sounds right!"

BY ROBERT HANSON

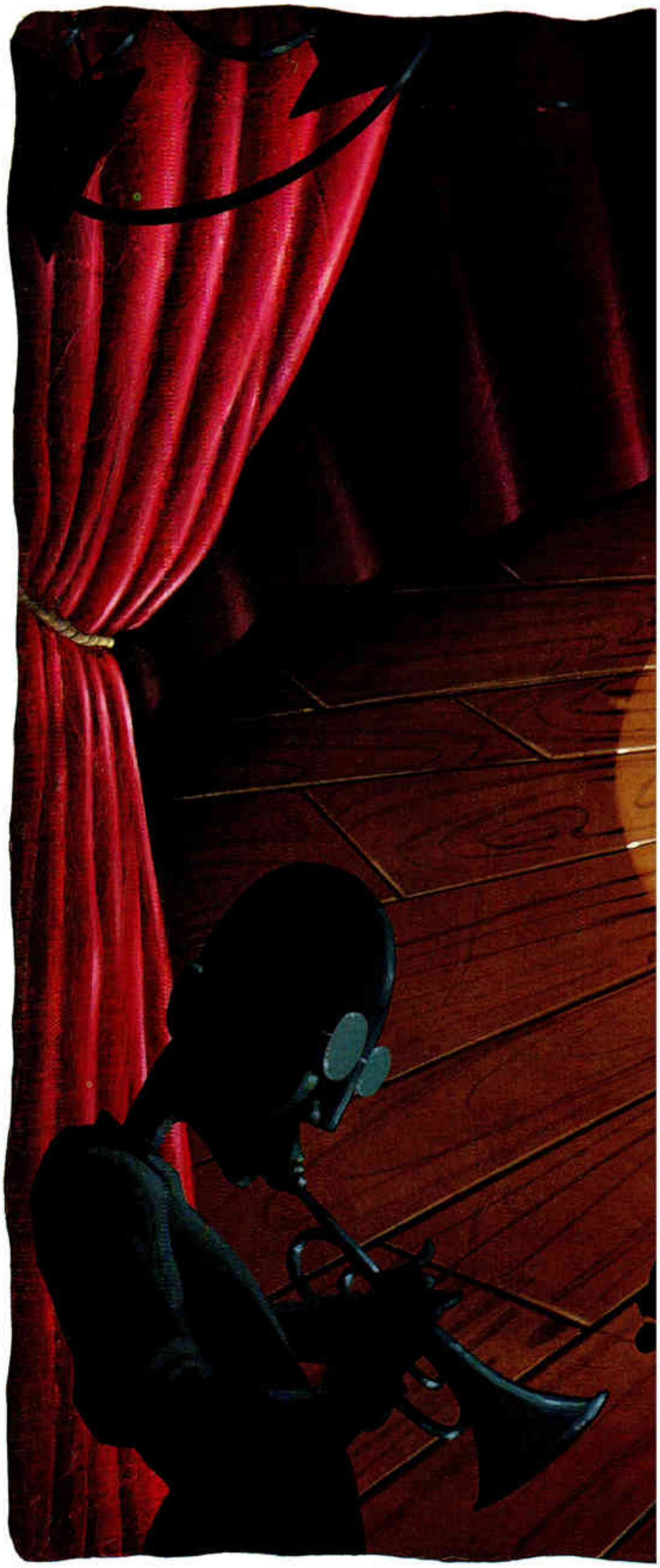

ILLUSTRATION BY NATHAN OTA

for the Stage

I can already hear what so many of you are going to say: "A *good* band doesn't have to worry about this." "A *good* band will never go out of style." "These are the worries of the Spice Girls and the Depeche Modes of the world." But this is not only the future, it's also the present. The crowds are moving with the music. Hip hop and electronic music are now the dominant forms of music, especially in many urban centers, and the trend shows no sign of slowing. With established acts like David Bowie, Melissa Etheridge and Tori Amos incorporating computer-generated elements into their music, gigs that require the combination of sequenced and live material are no longer going to be an exception, they're quickly becoming the rule.

So, how do you do it? Do you dump everything *sans* vocals and lead instruments to DAT, press Play and hide? Do you painstakingly create custom sample and synth banks and pray that the bird's nest of MIDI and SCSI doesn't hiccup? How to choose among ADAT, DA-88, Mini Disk, Cubase, Logic, Digital Performer and others? To shed some light on the issue, *Mix* picked the brains and racks of some of the artists and producers who do this day in and day out. Special thanks to producer/engineer Ray "SoL Survivor" Cham, engineer/producer/keyboardist Charlie Clouser of Nine Inch Nails and artist/producer Q of Uberzone.

Ray Cham did programming and sequencing for Christina Aguilera's recent tour.

SEQUENCING THE TEEN POP MACHINE

Ray Cham is just as comfortable manning the faders on an SSL, tweaking audio on a laptop with a pair of headphones or helping to bring teen diva Christina Aguilera to the arenas of the world. Cham, who co-produced Aguilera's hit single "Come On Over" and last year's Christmas album, did all of the live programming and sequencing for Aguilera's most recent tour and worked closely with musical director Alex "Adrenaline" Alessandroni and executive producer Ron Fair.

Cham began by transferring the original digital and analog masters into Cubase. He chose Cubase because of the program's integration of audio and MIDI and, the ability to work in both environments simultaneously. "While we were still editing and arranging the show, certain edits were happening on a daily basis," Cham explains. "We ran DA-88s as backup, but I was running Cubase for the live show. I was actually mixing and making mix changes live. At one time in the beginning, I'd be running 60 to 70 tracks per song during a show, just because we're still getting mixes together and making certain arrangement changes. Once we had everything locked in, we did end up dumping it down and condensing the tracks down to either 8-track format or 16-track format."

Cham's live Cubase rig consists of a rackmount G3 350, a Motor Mix control surface and a MOTU 2408 that interfaced with a DA-88 and sometimes an ADAT. "I would have a backup show running concurrently on tape," Cham continues. "But everything would be on input. So if anything ever happened with the computer for any reason, all I have to do is throw the whole thing out of input and the DA-88s will continue to run. Once the computer goes down, you're running off the prerecorded mix, but I have time to reboot the computer and prepare for the next song. I can reboot without missing a beat, because everything is backed up in DA-88: the click, the background vocals, everything."

For time compression and pitch shifting, Cham has recently switched from Digital Performer to the new ProSonic Time Factory, despite the program's admittedly lengthy processing times. Cham also maintains an "arsenal," as he calls it, of different sequencing software. "I don't believe there is one program out there that is doing all things for all people," he concludes. "I know cats that are using Digital Performer and cats that swear by Logic and cats who swear by Cubase. It's all about personal preference. In my situation, I've found that all of them have different algorithms that they use for their pitching and time compression, and at different times they all work great. It's a matter of not being locked in and saying, 'I only have one program, and I have to get it done as best I can with Cubase or Digital Performer.' At the end of the day nobody really cares how you get it done—just that you get it done."

THREE TRACKS THAT GO CLICK IN THE NIGHT

A long-time fixture in the NIN camp, as well as a renowned programmer/remixer, Charlie Clouser was instrumental in helping to bring the NIN opus *The Fragile* to the stage last year. In addition to a couple of co-writing credits on *The Fragile*, Clouser's recent credits include production and remix work with The Deftones, Rob Zombie and David Torn's SplatterCell, as well as contributing a track to the current NIN EP *Things Falling Apart*.

Clouser began by importing each of the multitrack files into Logic and divid-

PHOTO: STEVE JENNINGS

Producer/remixer Charlie Clouser

Who says you can't teach an old (road) dog new tricks?

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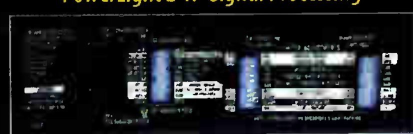

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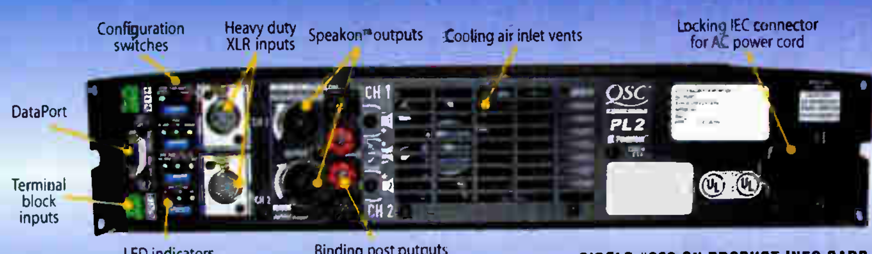

Rear view of PowerLight 2"A" amplifier

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ing the tracks into groups. One group consists of possible tracks that guitar player Robin Finck would handle; another group might be assigned to the bassist Danny Lohner, and so forth. "And [then] there's what's not playable live," Clouser explains, "i.e. 16th note 'typewriter' bass lines and 808 hi-hat parts that have crazy obviously machine-like bits to them." For the numerous keyboard parts, Clouser cut the multitracks into individual notes and exported them to an E4 sampler. The same process was used for a number of percussion elements that the drummer, Jerome Dillon, was intended to trigger live. The samples were dumped into an Alesis DM Pro drum module, which interfaced with triggers on the acoustic bass and snare, as well as an array of D-Drum pads.

For reliability reasons, the band uses DA-88s as a playback medium. "Back in 'the day,' it used to be a Tascam 234 4-track cassette deck," Clouser remembers. "It was a rackmount Portastudio with a mono click, a mono bass and a stereo pair of synths and ambiences. And we've expanded on that just a tiny bit; it's still basically three tracks: a stereo pair and a mono bass—those things that are going to have ridiculous low frequencies wind up in mono on their own track—and then a track of click. And this time out we were more daring and put timecode on because we had had some video elements we wanted to trigger at specific points in a song."

The process of assembling the sequenced tracks and assigning parts to be played live began with the band's initial round of rehearsals last summer in the Bahamas. "I had been making a lot of the live tapes for *The Fragile* stuff on a portable rig as we were in rehearsals. In the morning, I would usually confer with the guys and say, 'Here's three parts I've found for you to play, is that acceptable?' In many cases, there would be a bit of digital trickery involved because we find a lot live that you have to up the tempo by a few bpm. So then I would spend the morning doing my time compression and so forth and give the guys a really terrible version of the tape to play to. By the end of the day, after hacking through it a few times, we would have figured out the ultimate tempo."

The band then moved to full production rehearsals with the P.A. and a full crew, which included tape and trigger tech John Van Eaton. "We get the actual system down there," Clouser continues, "and we get an actual venue that's big enough to make it sound like it's actually going to sound." The band would then rehearse to either the DA-88 or Clouser's portable Logic rig, depending on how far along the band was with a certain song. Clouser's rig interfaced over separate ADAT bridges with a Yamaha 02R, which handled all the compression and EQ for the individual elements that would eventually end up on tape. The eight buses off the 02R would then be routed to the DA-88 for the final mix. "We can literally play along to the computer through the 02R. We can run though the song, and the FOH guy [John Lemon] can say, 'there is some tambourine loop that comes in there in the middle that is so freaking loud.' And I'll go, 'great' and walk over to the O2R and find that channel and pull it down and store that mix. And we would do that to very endless detail while we were in production rehearsals."

According to Clouser, another integral part of the process was creating a click track for the drummer to follow. "The drummer and I have worked out what his favorite sound is for the click and what sort of pattern he'd like it to play, because it's not just, 'one, two, three, four.' But it has eighth notes and 16th notes in there. And there are little extra sounds; lets say there is a long break where there is some weird ambiance playing, so that he'll know when to come back in, there is, unfortunately, my voice counting him back in. And there are many songs where he's counting the band in, because we all start playing together and none of us are lis-

PRERECORDED CUES FOR THEATER

AN ABBREVIATED HISTORY

by Jonathan Deans

If there are any historical accounts of the use of prerecorded sound to augment live performances, then they likely start in the dawn of live radio drama. Broadcast sound effects artists would typically supplement live effects with playback from vinyl records, adding realism or, in the case of the legendary anarchic BBC radio show "The Goons," surreal absurdity.

Outside the controlled environment of a broadcast studio, playback from vinyl was relatively impractical, so it was not until the advent of tape machines that prerecorded sound cues became common in live theater. During my early years as an ASM and later as a sound designer, Ferrograph 2-track tape decks were the first choice for playback, later replaced by the ubiquitous Revox. Two-track cartridge players soon became popular for providing spot sound effects, underlining the fact that theatrical sound playback devices must provide two distinctly different types of cues: long multitrack sequences and "instant" cues, which are typically short. And, on some projects, the sound design may call for a multitrack player that needs to be triggered instantly.

Though the laserdisc format has almost followed the 8-track and DCC into oblivion, laserdisc players were once a common choice for theme park playback systems. Laserdiscs can play back more than two tracks of discrete audio, and the format's robustness recommended it over multitrack tape for productions that typically included 10-plus shows a day, every day. However, the turnaround time to burn a new laserdisc was at best 10 days—and it was expensive. This usually had the effect of locking in the show early in pre-production (sometimes a good thing). Spot sound effects were typically stored on 360 Systems and Alcorn McBride devices, to name but two.

The appearance of hard disk-based recording systems offered sound designers a new set of tools. I first used such a system—New England Digital's Post-Pro—11 years ago for the *Seigfried and Roy* show in Las Vegas. The requirement was to play back all the prerecorded music on multiple channels, and, in fact, the system was capable of instant 8-track playback and stored the entire 90 minutes of show program in a towering rack of Patriot drives. This was considered quite a technical feat back then and came at an appropriately jaw-dropping cost—around $200,000. Backup was on MiniDisc. Ouch.

Other notable systems of the time included Doremi Labs' Macintosh-based DAWN system, an 8-track system with instant (buffered) playback. With Seagate hard drives and a lower

—CONTINUED ON PAGE 46

"These days there's a FIVE on every rider"
Ed Pratt, Pratt Sound, Salt Lake City, UT
CASCADE SOUND
"For us the FIVE's were an easy choice as we were seeing it appear on more and more riders. The board is so intuitive that visiting engineers can get on and mix really quickly. The FIVE was also great value for money for the features and performance you get."
Larry Carroll, Cascade Sound, Salem, OR
Midwest Concert Concepts
"Time and time again the FIVE kept appearing on the riders but before we went ahead we had demos from all the main console manufacturers. The FIVE had that great warm British sound and musicality along with the sonic purity that was superior to the rest. It's a very intuitive board, everything is where it should be and the layout is incredible."
Brian Houle, Midwest Concert Concept, Superior, WI
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Q of Uberzone

tering to a click except for the drummer. He follows the click we play to him. From our end, it's a very natural musical experience."

BEATS, BREAKS, DECKS AND SCSI

Recently picked up by Astralwerks, Uberzone is the brainchild of longtime artist/producer/remixer Q. Working out of his L.A.-based studio, Q has had a number of well-received, independently released UK and domestic dance singles. Q also remixed tracks by The Crystal Method, Sarah McLachlan and Herbie Hancock, in addition to picking up a co-production credit on the Afrika Bambaataa track, "2 Kool 4 Skool." Currently, Q is working on his debut effort for Astralwerks, slated for release this year.

"I approach [playing live] from the philosophy, first of all that I want to do something that is obviously recognizable and sounds like the original material but has a total fresh angle to it," Q states. "And then that emancipates me from having to figure out how to try and justify playing to backing tracks."

Q begins by making subgroup mixdowns of all the separate elements (drums, bass, synths, etc.) while he is still working in the studio. He then samples each of those elements and imports them into Acid for further tweaking. "You can take those subgroups," he explains, "put them in Acid and type in the bpm and take the sample, literally, and stretch it to whatever bpm you want to work at. Whatever bpm you're working at in Acid seems to really translate well to Cubase. So I'll take those samples that I've made in Acid that are perfect, perfect loops, and just export it to a new track and SCSI-dump it to the sampler. And if you write it in as a full bar loop, using Cubase, just drawing in a bar all the way across, it loops flawlessly."

Q uses the E-4 line of samplers and sequences live with Cubase running off of a laptop. "The laptop just becomes a sequencer," he continues. "I've got a simple MIDI interface on it. [The] E-4 acts as like ground zero for all the staple parts that are playing back lines and hooks that represent pieces of gear that I wasn't able to take with me. Like the Jupiter 6, all of its parts would be playing out of the E-4 now. A lot of the times you take a sound and sample it off a Jupiter 6, it kind of looses some of its impact. So I'll take the phrase and sample it. The E-4 is capable of like 128 Megs, so you really don't have any memory issues.

"Then I'm bringing a Jupiter 8000 and a Virus out as well to play parts that I'm able to tweak live. One of the things that I do, that makes the live performance more fun, is break it all out of the E-4 in stereo pairs of drums, synth parts, samples and that comes up in the mixer. I'll do it all in loops, and I'll do it so that I have sections of the song where I can move the locator points around in the sequence. I can extend a certain part of the song, and I can kind of do a live mix on-the-fly, using mutes, and I'll use the effect sends and send those out to time delays. So I can kind of play the mix. One of the other things I do a lot is use the trap-cap to trigger parts. So I'll have sounds in a second E-4 at the same bpm as the rest of the track, and I can trigger the loops off the trap-cap using the Lever function to start and stop breaks. Between that, being able to play the mix and playing keyboards parts gives me a lot of options and really involves you in what's going on onstage." ■

Robert Hanson, Mix's editorial assistant.

—FROM PAGE 44, PRERECORDED CUES

price than the NED system (the New England Digital company later dissolved), the DAWN system became the new standard, and I specified it for the *EFX* show in Las Vegas.

By using two complete systems, it was possible to program A/B rolls from cue to cue with eight discrete tracks of playback; all the short sound effects were loaded into a Roland S-770. This show has now been running for over five years with minimal failures—backup is on a bunch of spare hard drives. Another Las Vegas spectacular, the Rio Hotel's *Masquerade Parade*, uses Wild Tracks from Level Control Systems. This has the ability to play back 16 tracks of audio with cue-by-cue mapping of track requirements (asynchronous) and an online backup hard drive.

Two-track MiniDisc players are a popular choice for relatively simple shows, in part because of their relatively low cost and ease of use and labeling. It is also possible to perform basic editing on MiniDiscs with minimal equipment. However, the MiniDisc format has only two tracks, only offers compressed audio (sometimes an issue) and will only play one sound at a time. On Broadway shows, prerecorded sequences typically include a click track for the musicians or dancers or both, so a multitrack player becomes necessary, because the click will eat up one or possibly two tracks. Akai's DR8 and DR16 units are among my frequent choices for multitrack playback, and the DR16 offers the potential for automatically triggering mix automation cues in order to re-route sources and cue outboard effects via SMTPE/MTC. Another unit I currently favor is the Akai 6000, which can be used for short sound cues and will also play back straight from the hard drive for lengthier cues.

Rereading this partial list of equipment that I have specified and used over the years, I am struck by how many pieces of gear have come and gone. Hardware obsolescence is almost inevitable, but there is no reason why the cues cannot be migrated to new hardware. Archive source material in .WAV or .AIFF data formats, which are easily transferred among most modern playback units. And, above all, make sure that you can safely and efficiently back up your work after you have finished programming. ■

Jonathan Deans is an award-winning sound designer who can most often be found on Broadway or in Vegas.

David Thoener

MEMOIRS OF A FAMILY MAN

These days, you hear David Thoener's mixes everywhere. There's Santana's "Smooth," of course, the blockbuster single that spent 12 consecutive weeks at Number One on the *Billboard* Hot 100 and propelled sales of the *Supernatural* album to 13-times Platinum. Then there's Matchbox Twenty's "Bent," the Number One most played song in the U.S. for more than eight weeks; and the left-field "Teenage Dirtbag" by Wheatus, a Top 10 single on the Alternative chart.

You've heard a lot of Thoener's other mixes, too, from Aerosmith's 1998 smash, "I Don't Want to Miss a Thing," to Meat Loaf's "I'd Do Anything for Love (But I Won't Do That)," John Waite's classic "Missing You," John Mellencamp's "Pink Houses" and "Authority Song," and AC/DC's "For Those About to Rock." And don't forget those tracks he did way back as a young whippersnapper: the J. Geils Band's "Freeze Frame," "Centerfold" and "Love Stinks."

Wait, there's more: Thoener has mixed records for Billy Squier, Kiss, Sammy Hagar and Billy Joel, and he has recorded and mixed country artists, including Brooks & Dunn, Rodney Crowell, Travis Tritt and Ronnie Milsap. He's also known for his live recording and mixing with Aerosmith, Shawn Colvin, the Neville Brothers, Woodstock '94 and '99, Bob Dylan's 30th Annivarsary Concert Celebration and the double CD *Concert for the Rock & Roll Hall of Fame.*

And all of that is just a partial discography. It's obvious that Thoener has what many engineers aspire to, but few achieve: a career that has both diversity and longevity.

He rarely does interviews, but it quickly became obvious that he has plenty to say and a vast stockpile of great stories. It was also readily apparent that with Thoener, as with a great musician, it was best just to place the mic, roll tape and get out of the way. So that's what we did. Now, heeeere's Dave, taking it from the top.

I started at the [New York] Record Plant on April 4, 1974. I remember going for my interview in platform shoes, huge bell-bottom pants, and, of course, a psychedelic polyester shirt. Ed Germano [now owner of New York's Hit Factory] was the manager. When I left, I didn't think I had the job. But as soon as I got home, the phone rang; he wanted me to start the next day!

Just the atmosphere of Record Plant at that time was enough to excite you. The carpets were shag; the walls in Studio B had a weird map that looked like Bullwinkle. Records by Jimi Hendrix. Yes, Jackson Browne, Grand Funk Railroad, Don McLean, The Raspberries, John Lennon, etc., lined the walls.

The year before, I'd worked for a demo studio on Seventh Avenue where I'd learned disc mastering and engineering, but Ed G. started me out as a tape copier anyway. I remained in the copy room for about three months. The turning point came when I was asked to take a piece of music and make a cassette where it repeated over and over. I guess management expected me to record, rewind and record again, but I had a different plan.

I went into Studio C, took several mic stands, then created a huge loop that went from the copy room down the hall, past the restrooms and back. When (Record Plant owner) Roy Cicala went to the restroom, he had to pass my loop! The next day, Roy called me into his office. I thought I was getting fired, but instead I was made an assistant.

BY MAUREEN DRONEY

APOGEE DIGITAL.
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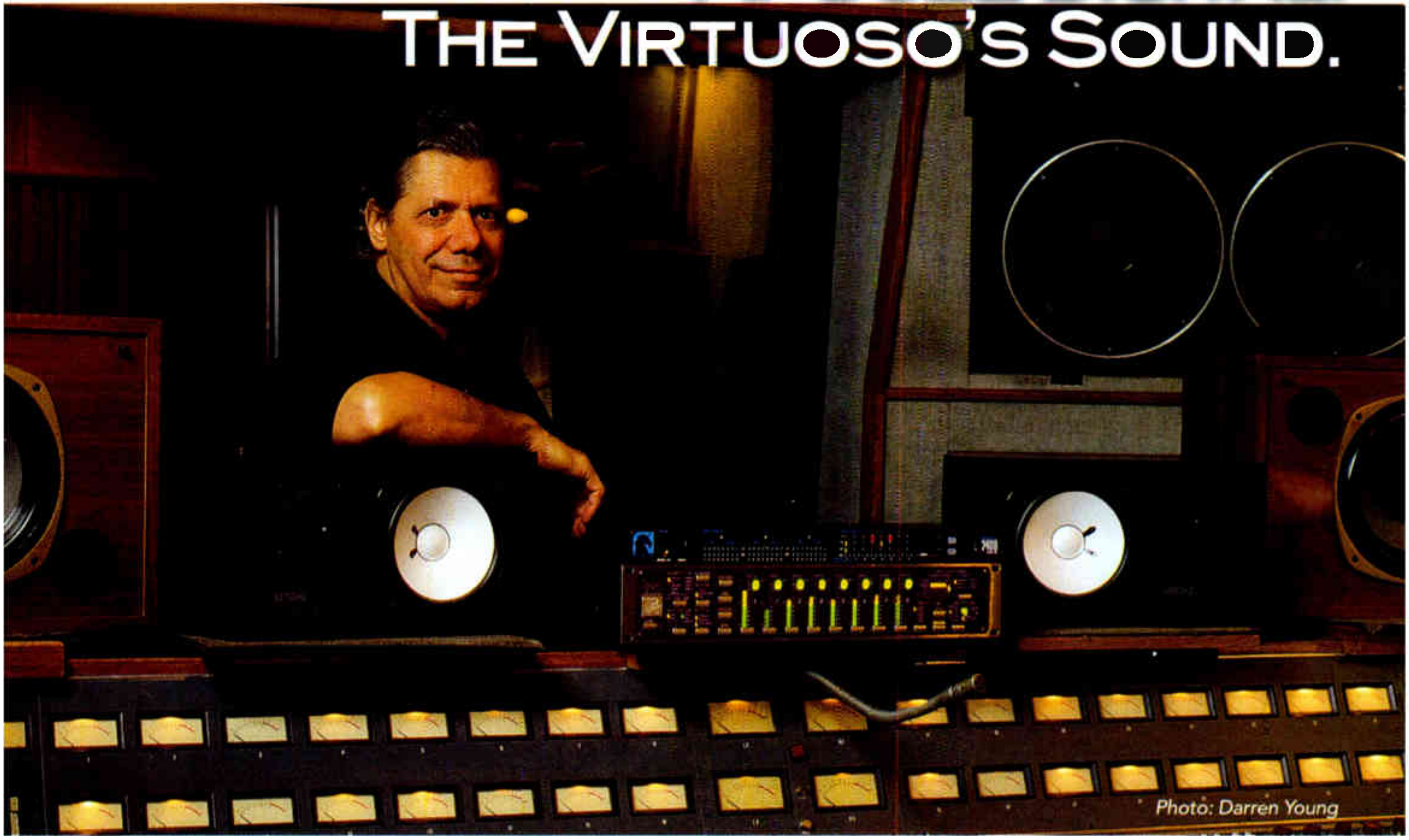

Chick Corea pictured in his Mad Hatter facility in Los Angeles with the AD-8000SE and MOTU 2408

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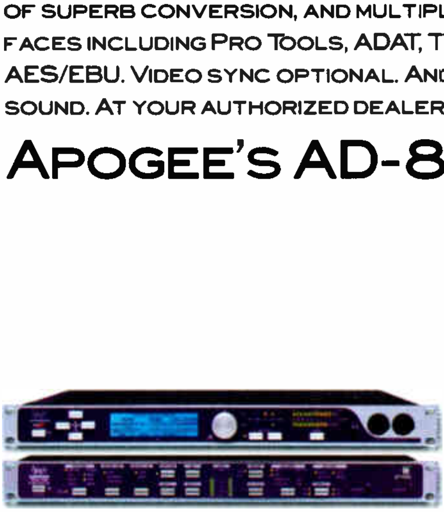

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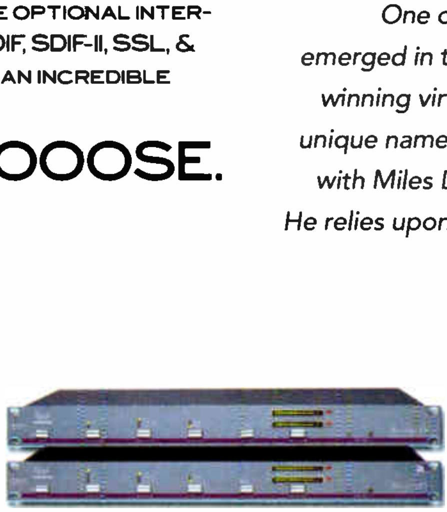

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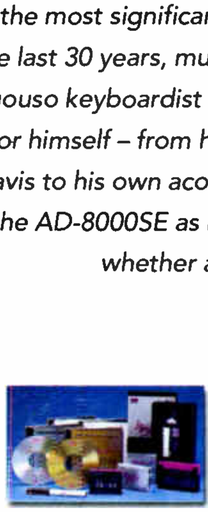

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Lucky for you, Roy appreciated initiative—and attitude.

Jimmy Iovine and I were assistants at the same time, and Roy made Jimmy his personal assistant. The chief engineer was Shelly Yakus, and since Roy had Jimmy, Shelly took me. That was great, because Shelly was one of the hottest engineers in New York. I got to work on some amazing projects. One of the first was The Raspberries with Jimmy Ienner producing. Then there was Johnny Winter, Edgar Winter, Rick Derringer, Chick Corea, Blue Oyster Cult and John Lennon, just to name a few. As time passed, Shelly would let me do some overdub engineering. He had me do the piano interludes with Chick Corea on "Where Have I Known You Before." I was only 21 at the time, so this was really great!

Then Jimmy got the chance to engineer Bruce Springsteen's *Born to Run* album. I remember him coming to me after a session, telling me the news and asking me to assist him. Of course, I said, "Yes!" All of the staff at Record Plant, especially the assistants, were very close, and we were happy when one of us got a break. One funny story that I hope Jimmy doesn't mind me telling was the night we were recording the lead vocal on "Jungleland." Jimmy had been working late for many days, and that night he fell asleep; he had his legs up on the console and he just dozed off. Jon Landau and Mike Appel were producing, and Bruce was ready to sing. They looked at me and said, "Dave, you can do this, right?" I said, "Sure," picked Jimmy's legs up, slid him over to the end of the console and recorded Bruce's vocal. It was an incredible performance; sitting in the control room watching Bruce deliver that vocal was a moment I'll never forget.

Soon Jimmy started giving me gigs—sometimes full recording and sometimes overdubs. I was very grateful to get anything. Not long after that, Jimmy started producing with Shelly engineering, and the rest is music biz history.

Now you were on your own.

I started getting booked with outside engineers because the management knew I could engineer, or at least explain how things worked to a visiting engineer. One was Martin Birch, and I did Richie Blackmore's *Rainbow Rising* album with him. Then I worked with Harry Maslin on David Bowie's *Young Americans*. But the engineer who introduced me to the band that would change my life was Bill Szymczyk. We worked on an album called *Hotline* by J. Geils in 1975. Then in '76, Bill was busy with The Eagles, so the Geils band called Roy Cicala to engineer, and he booked me as his assistant.

Thoener (right) with David Johannsen in 1977.

On the first day of recording I set up the band and set the console so Roy would just have to come in, sit down and hit the Record button. As the band started recording the first song, Roy looked at me and said, "I'll be in my office if you need me." Then he got up and walked out of the control room! I'm standing by the multitrack, like the assistant is supposed to, and I'm thinking he's joking, that he'll be coming back in to see if I'm cocky enough to sit down.

At the end of the first take, the band looks in the control room and asks if it was a good take. By this time I've worked on a lot of albums, so I give the typical producer response: "I think you should try it again!" They look at me, then at each other, and Stephen Jo Bladd, the drummer, counts off for another take! And I can't believe what's happening!

We recorded two more takes and the band walked in the control room to listen. They asked where Roy was, and I answered, "He's in his office, do you want me to get him?" They said, "No, you'll do just fine."

That's a "Glory Days" kind of story.

That album was *Monkey Island*, with a hit called "I Do." I thought I'd arrived. But that was my first album, and there weren't any bands lining up for me to do another. The management of Record Plant decided to put me on an assisting gig, but by this point my ego was too inflated, so when J. Geils asked me to go on tour with them, I quit the Plant and went on the road.

Well, that's a different perspective.

The road is a very good way to realize what's important. Example: At a show, the first priority is monitors. The band has to hear themselves and be happy with their monitors if they're to perform. Second is the FOH. Sometimes, as FOH guy, I was able to get a rehearsal and sometimes, if we were the opening act, I didn't. "How can you get great sounds without hearing the band?" you might ask. The answer: You wing it. From previous shows you have a good idea of what the survival EQ should be—the EQ that can get you started.

Okay, we know every show is different, but how about an example of what survival EQ might be.

For drums, maybe, roll-off some low bottom, roll out a few dB at approximately 500 on the kick and toms, and add some 4.5 to 5k on the snare. You might roll out a few dB of 200 on overheads, roll out 200 to 300 Hz on hi-hat, and on the vocal roll out between 300 and 800 Hz. Just a few dB, nothing radical. I'm more likely to take out than add. If something isn't bright enough, rather than adding top end, I'll find out what's making it cloudy and take that away.

Then you see where everything else sits. The most important fader is the lead vocal. If the drums are a little muddy, there isn't enough bass, or the guitar EQ isn't just right, they can wait as long as the lead vocal is heard. You can buy, like, three minutes, which is your first song. You better get it right by the end of the first song, where the enthusiasm of the band going onstage with all the screaming will help cover the fact that you're making the mix better. FOH guys have it down to a science. I have enor-

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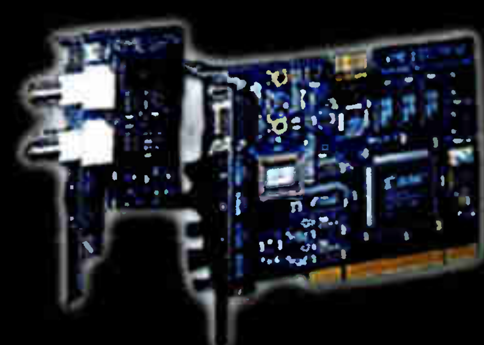

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www.Fostex.com • 15431 Blackburn Avenue, Norwalk, CA 90650 • Tel: 562-921-1112 • Fax: 562-802-1964 mous respect for them, because I know first-hand how hard it is. That's why I don't do it anymore! In fact, I quit after the first leg of the Geils show, because I thought they should use a more qualified person. I felt I needed to go back to the studio where I belonged.

But that experience paved the way for the live recording work you later got into.

The live recording engineer gets even less time than FOH. If the FOH guy needs time to get his mix together, his needs take priority. Sometimes, while he's getting his sounds, you can get yours. But if he needs to focus on a problem area, you may run out of time before you get your EQ together. How does an engineer get great live sounds if there's no soundcheck? Same answer as before, you wing it. Along with your knowledge of survival EQ, the most important focus is to make sure everything's going to tape!

It's always best to EQ on the monitor side of a live recording. Make sure everything is going to tape clean and at a proper level. It doesn't matter whether you're recording analog or digital; you need to maximize level to get the best signal on tape, and you need to make sure no distortion is coming from your side of the equation. As with FOH, it's best to get your act together in the first three minutes. That means EQ, compression, balance, effects. If you're doing a live broadcast like when I did Woodstock 2 and 3, the pressure is even greater. Your instant mix is going out to millions of people who have no idea what you're up against. Some of the Woodstock performances happened so quickly, many of us in the recording trucks only got a line check—you know, when a roadie goes out onstage and scratches the mics so you know they're live. Sometimes I've lost a mic just before the band goes onstage. Talk about heart failure! The only thing you can do is have your stage engineer try to locate the problem and switch out the mic, or the cable, and to do it between the time the lights go down and the band walks on! It has happened to me, and we've fixed it before the first note played. It's a stressful gig, to say the least, but it's also a tremendous thrill when it's all going down. You get an adrenaline rush—probably the best high I've ever experienced. And when it's over, the relief is just as strong.

Meanwhile, back at the studio...

Right. Well, after being on the road with J. Geils, I went back to Record Plant with my head hung low and asked for my job back. To my great surprise, they gave it to me. It was back to assisting, temporarily.

But soon, David Johannsen, who now goes by the name Buster Poindexter, asked me to do his album. It was called *Funky But Chic*. You might remember it? It got a bunch of airplay in New York, as David was in the New York Dolls, a kind of a cult N.Y. band at the time. After that, some more albums came along, and suddenly it was the next year and Geils was ready to record *Sanctuary*.

FOH guys have it down to a science. I have enormous respect for them, because I know first-hand how hard it is. That's why I don't do it anymore! I felt I needed to go back to the studio where I belonged.

This was the first of three albums we recorded at Longview Farms in Worcester, Mass. Longview presented a few challenges. It was a house, not the kind of acoustically designed place I was used to. We all lived on the premises, which was good because I didn't have to wait for the band to arrive, but bad because they also didn't have to leave. There were many long nights.

But that's a part of recording, and I love recording so much that it was more fun than pain. Also, the band treated me like a bandmember; I traveled with them, ate with them, partied with them. We were a family. I had great admiration for them, and they trusted and respected me and what I had to say about the work we did. I always tried to push the envelope sonically.

Like back at Record Plant when we were doing *Monkey Island*, we'd experimented with putting Stephen Jo Bladd out in the back of Studio A. It had a great sound because the walls were stone, and it was very live. The only problem for Stephen was that the garbage was stored there. We couldn't record there in the day because the sound would travel all over the building. So we started at night, when the garbage from all the offices was piled around him. Boy, did it smell! But it sounded great!

Sanctuary turned out well, but when *Love Stinks* started, I took it to the next level. *Sanctuary* was recorded in the house, which was the way the place was designed to work. When we started *Love Stinks*, we recorded in the barn. Now, I say barn, and it was, but it was split into two areas. The barn with the animals was on the top of a hill, and the place where the band stayed was next to it. The barn sounded great, but it had a very old Angus console and was meant to do demos and such. We loved the sound of the room, so I would bring the instruments up the mic pre's of the Angus and send them over line level to the MCI console in the house, which had the multitrack and the monitors. I couldn't see the band at all; I could only hear them. Every once in a while, I'd have to run over to the barn, which was three or four hundred feet away, and check a mic or explain something to the band that I couldn't explain without seeing them.

I remember putting a new mic that had just come out called a PZM [pressure zone mic] on the ceiling, experimenting with ambient drum sounds. I was to find out there was someone else experimenting at the same time. I remember driving up to Longview to finish overdubs, hearing what Hugh Padgham had done on Phil Collins' "Something in the Air," and thinking, "Damn, someone's beat me to it!"

Obviously, we weren't the first. Led Zeppelin had incredible ambient drum sounds in the late '60s. But the sounds had changed in the '70s; disco and dead drum sounds were in. Engineers put drums in drum booths—anything to suck up the ambience and create a dead snare, kick and tom. This was my attempt to bring back the live sound. I didn't have a castle to do it in, so it took some thought and experimentation.

Our guitars were also recorded in the barn, but in the part the animals stayed in. They would go out to pasture in the morning, so we would set up in the afternoon and record power chords full out. I had some of the mics in omni, and sometimes we'd hear a cow mooing over the guitar overdubs, especially at the end of a long, sustained note, and we'd have






letic also wanted me to be ready to go when he felt they were ready. We recorded about five takes and we had it.

What is that usual mic setup of yours?

Kick: D112 outside, 47 FET inside. Outside, build a tunnel with mic stands and blankets to keep cymbal leakage to a minimum. Snare: 57 top and bottom, bottom mic gated so I don't get the kick rattling the snares. Toms: 421s. Sometimes I might throw a mic on the bottom head of the lowest tom for low lows. Hi-hat: KM84. Cymbals: 421s. Room: C-12s. Percussion: 57s and 451s, sometimes an overhead 87. On bass: a Demeter DI, and on the bass amp an RE-20. Also on bass: dbx 160s with light compression, 1 dB or so at 2 to 1.

How did you record Carlos?

Carlos has his pedal that shifts sound from one amp to another with two different sounds on each, and he controls that, naturally. I put a 57, a 421 and an 87 about three feet back on both amps. Light compression with 1176s. I usually put the mics on different tracks, so that as the song takes shape I can alter the guitar sound slightly to work with the new overdubs.

What did you record to?

Pro Tools—Mark Dobson is our Pro Tools genius. We also used Apogee Special Edition 8000 converters. Matt Serletic owns his own Pro Tools system—actually the equivalent of three complete 64-voice rigs. He also brings

All of the sounds you record should be focused on trying to turn the artist's and producer's ideas into reality. Listen to the song.

all his own cables; it's Monster Cable on every microphone, and everything else going into the console. Every guitar cable is his own too, the kind by Monster Cable with arrows pointing the direction of signal.

Did you edit between takes?

Yes, we cut between performances, always taking the entire band performance of each section that we used.

Was Rob [Thomas] recording his vocal live with the tracks?

Yes.

So you recorded the basic track on the first day, and then...

The second day Rob came in and fixed a few vocal spots, and Carlos fine-tuned his solo. The third day, we did the horns and little extras. We flew to L.A. the fourth day and went to Record Plant. It was in the mixing stage that Matt decided to effect Rob's verse vocal with a setting from the plug-in called Amp Farm.

You mixed from Pro Tools to...

We mixed off Pro Tools locked to Sony 3348 HR through an SSL 9000, and we mixed through a 96k/24-bit dB Technologies encoder to a Genex MO.

There are a lot of parts on "Smooth." Did you have to use a lot of compression to keep everything in place?

I used compression, not a lot—just enough to keep things retaining their natural dynamic but able to work within the framework of the entire mix. I like the Neve 33609 as an overall com-

—CONTINUED ON PAGE 200

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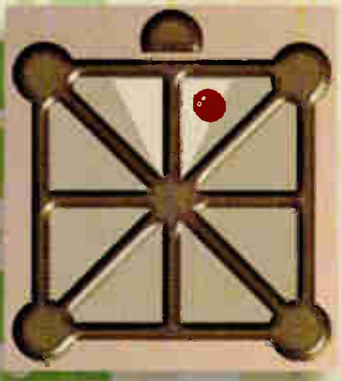

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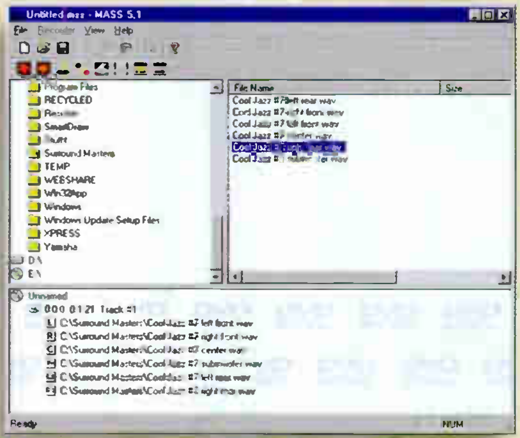

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LIVE MIX

Each month *Mix* covers live sound equipment and applications, usually from the FOH or monitor engineer's perspective. But for this issue, which is distributed at the NAMM show in Anaheim, we focus on the aspects of sound reinforcement technology that most affect musicians. To that end, we lead the section with Mark Frink's comprehensive article on in-ear monitors.

In-ear monitors, or IEMs, have brought about the first significant change in the performing environment since the development of quality wireless systems. Singers can now hear exactly the same mix, regardless of where they roam on- (and off) stage; if the whole band is on "ears," then FOH and monitor engineers need not struggle with amps leaking into open vocal and drum mics. And, because of the control, absence of feedback and potential for lower listening levels, IEMs can help preserve musicians' precious hearing.

But IEMs are not a complete panacea. As licensed audiologist and Sensaphonics president Michael Santucci notes, in-ear monitors are just as capable of damaging hearing as are traditional floor monitors. "There is an industry misconception that in-ear monitors, like earplugs, are hearing protectors by design," he says. "They can be used as a tool for hearing preservation—but only if used properly. Some in-ear monitors can reach levels of 125 dB to 140 dB! No one is going to protect their hearing at those levels, even for short exposure periods." As Santucci explains, one must refer to the tables developed by OSHA to determine safe exposure times and SPL levels. The longer the show or rehearsal, the lower the output of the in-ear monitor must be. Currently, the only method to measure these levels is with an in-ear probe microphone, typically only available to an audiologist.

Because IEMs, if improperly used, can exacerbate the risks to a musician's or sound engineer's hearing, we have rounded up a number of articles on hearing. Bob McCarthy writes humorously of his trip to the audiologist with his 85-year-old father-in-law. Audiologist Lisa Tannenbaum contributes a piece on protecting and monitoring your hearing health. And more information can be gleaned from the literature and Web sites of hearing education organizations H.E.A.R. and the House Ear Institute. Remember, hearing is priceless.

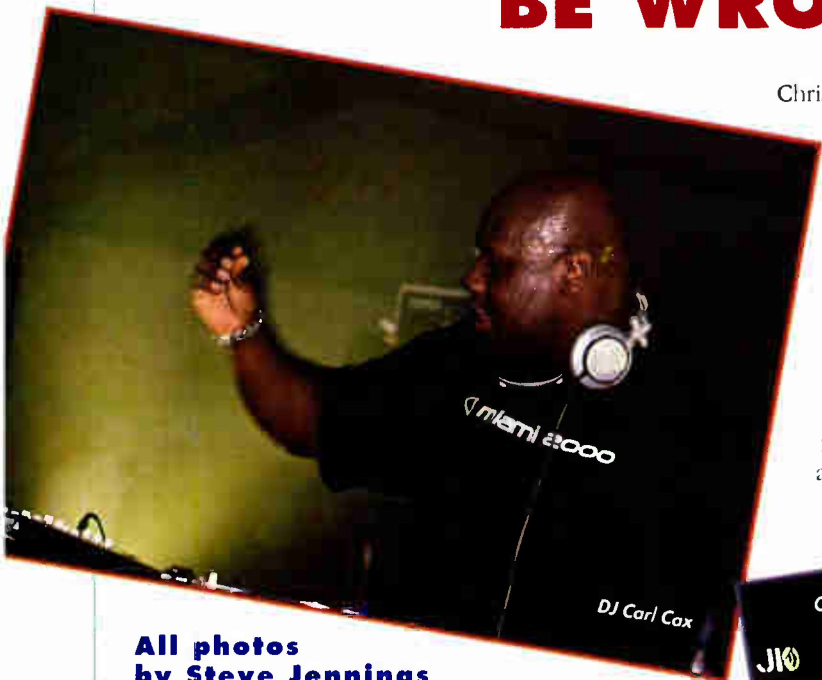
—The Editors of Mix

Live Mix Contents

TOUR PROFILE

MOONSHINE OVERAMERICA

COULD 30,000 KIDS BE WRONG?



DJ Carl Cox

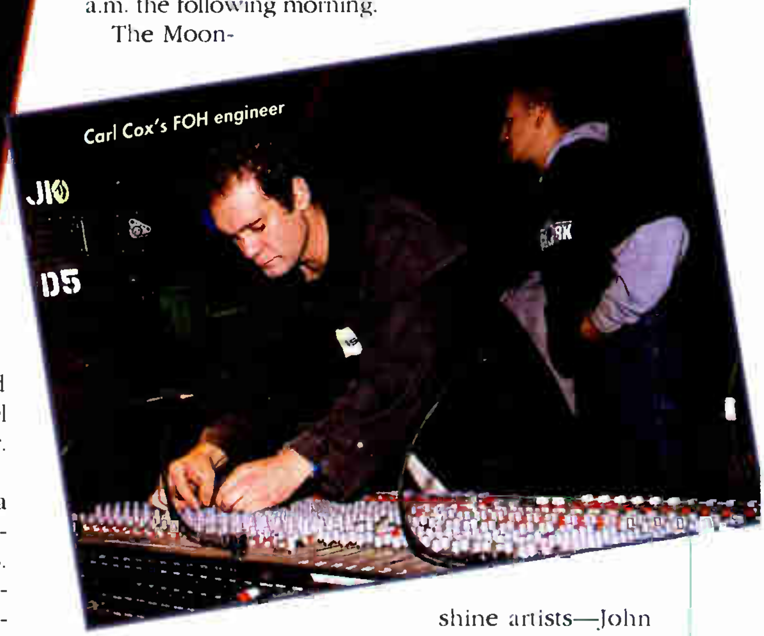
All photos by Steve Jennings

Back in 1997, Moonshine Music president Steve Levy noticed that the DJs and artists on his label were almost constantly traveling all over the world to perform. Recognizing that interest in electronic dance music was increasing substantially in the United States, Levy realized he could promote his acts and label simultaneously by putting his artists out on tour together. The result was the Moonshine Overamerica tour.

For the last four years, the Moonshine Overamerica tour has been an overwhelming success. This year's Listen.com-sponsored tour visited 22 locations in the U.S. and Canada in five weeks, and every show sold out. Unlike conventional rock tours, where the talent lineup generally remains the same from beginning to end, each Moonshine show featured a slightly different roster, consisting of three to eight acts. In total, 14 artists participated in the tour, with the band Cirrus and DJ Carl Cox logging the most miles. D:Fuse, Micro, Dave Audè, Dieselboy, John Kelley, Keoki and DJ Dara performed at least eight shows each, and Christopher Lawrence, AK1200, Charles Feelgood, Misstress Barbara and Frankie Bones also took part.

By far the biggest show of the tour was the concluding stop at the Los Angeles Sports Arena. Here, the Moonshine Overamerica show was just one component of the Monster Massive, a huge event featuring five separate areas and a wide range of DJs from all over the world. More than 30,000 die-hard ravers and dance music fans attended the event, enduring three- to five-hour waits in line to get in, extensive searches and the rather heavy-handed presence of the LAPD. Once inside, fans with enough stamina could dance until 6 a.m. the following morning.

The Moonshine artists—John Kelley, Cirrus, Carl Cox, Christopher Lawrence and Misstress Barbara—performed in a large "circus" tent set up in the Sports Arena's parking lot. The tent's sound system, which the Monster Massive promoters rented from an L.A. sound company, consisted of two stacks, each consisting of nine Cerwin Vega speaker cabinets—a combination



Carl Cox's FOH engineer

BY CHRIS GILL

of Prostax PS-215B and PS-15 SII enclosures—driven by an array of QSC power amps. Because Cirrus was on the bill along with the lineup of DJs, the sound company also provided a 32-channel Behringer Eurodesk MX3282A mixer for FOH.

"We don't tour with our own sound system, although I would like to," says Cirrus' FOH engineer

FOH engineer Ken DeSantis and Cirrus

Ken DeSantis. "Because this tour consists primarily of DJs, either the venue or the promoter is responsible for getting the equipment that we need and setting it up by the time that we get there. The clubs and theaters we've played at usually have their own in-house systems that are dialed in, but we've also played at several unusual locations, such as in big, open fields, abandoned warehouses with concrete walls and floors, or in a tent, like we did in L.A. As a result, I get to know the sound companies in every town pretty well."

Cirrus provides the promoters with a rider that spells out their requirements in detail. "I request a touring-quality board with at least 32 channels, such as a Yamaha PM4000 or a Crest, Midas or even a good-old Soundcraft mixer," says DeSantis. "We also require a full monitor

DJ Misstress Barbara

rig that includes a separate console, stage wedges and side-fill monitors. We require more than a normal DJ does. Since most event promoters aren't used to working with live bands, I have to do a lot of groundwork ahead of time to make sure that they understand what we need.

"The main problem is trying to educate promoters and DJs as to what the band's special requirements are. A lot of times my rider looks like Japanese to them. They don't understand what the different types of mics are for or why we need effects processors or EQs for each channel or even why we need a monitor

DJ Dara

board. They also don't understand that we need to do a soundcheck when we get there and that there's a lot of work involved before we're ready to go. It's not like setting up a couple turntables and a DJ mixer."

Cirrus does not travel with a monitor engineer, and they usually ask the sound company to provide someone to run the band's monitors. However, the sound company failed to provide a separate monitor rig for the L.A. show, so DeSantis ended up running Cirrus' monitor mix and FOH simultaneously. To make matters even more difficult, the mixing console was set up on the side of the stage, because the sound company wasn't allowed to run the snake cable across the floor area in the tent due to the hazard of dancers in the audience tripping over it.

"I had to do a lot of the FOH mixing that night through headphones," says DeSantis. "That's not what I prefer to do, but I used to do that a lot in the band's early days, so I'm used to it. Anyway, it ended up sounding pretty good. The whole show was pretty much a seat-of-your-pants ordeal. We got into L.A. late, so we couldn't change the setup, and because we were in a tent, they wouldn't let us turn the sound system up to full volume for very long while we were doing soundcheck."

Although the requirements for DJs on the tour were not as extensive as Cirrus', they had a different set of challenges to contend with. In addition to providing the house sound system, the promoters were also responsible for supplying a DJ rig of two to three turntables, a DJ mixer and side-fill monitors. While the DJs could count on finding industry-standard Technics SL-1200 MkII turntables in the rig, they often encountered a different mixer and monitor setup every night.

"The biggest challenge is getting accustomed to the monitors," says Christopher Lawrence, a DJ who specializes in trance and progressive house music. "DJs don't get soundchecks because generally we don't need them. All the sound company needs to do is hook up the turntables to the mixer, and as long as they've got a clean stereo signal coming from the mixer to the sound system, that's all we need. However, you only have about three minutes before the last guy's record ends to get accustomed to the monitors and how the sound on the floor will affect you. You don't have a lot of time to get your shit together."

Unlike a live band's performance, where the resulting sound is a collaborative effort between the performers, monitor engineer and FOH engineer, DJs act as a combination of all three. As a result, they have three sound sources to contend with simultaneously: the main house system, the side-fills onstage or in the booth and the headphone mix while they're cueing up records.

"I always turn the monitors up really loud," says Lawrence. "There's often a delay between the music coming out on the dance floor and what I hear in the monitors or over my headphones. You have to stay really focused while you're mixing—all it takes is one split second to lose your focus, and you could end up listening to the wrong set of hi-hats when you're trying to cue up the next record. All of a sudden you're cueing up off the sound from the dance floor, and all hell breaks loose!

"The optimal monitoring systems for DJs is having monitor speakers on the left and right side," he continues. "The mixer should have separate volume controls for the monitors, and what would be nice, but you rarely ever get, is having separate EQ controls for the monitors. A lot of venues that aren't used to working with DJs just give you a couple of crappy monitors. They don't realize that when you're mixing records, it's really important to be able to hear the bass, mids and highs. Different records have different-sounding kick drums, and unless you can hear the sound distinctly, you won't be able to make the records blend together. A lot of times promoters provide monitors that sound really shrill or distorted, and that makes it extremely difficult to mix, because you have no idea what the records really sound like."

Every DJ brings his or her own set of preferred headphones. Lawrence uses a pair of discontinued Sony MDR-V6 headphones, which he prefers for their excellent frequency response and ability to provide a clean, undistorted signal at loud volumes. "A lot of headphones designed for DJs have really heavy bass," he says. "When you turn up the volume, that bass just becomes a big farty sound, and you can't hear the records clearly. The bass response in the MDR-V6 isn't too heavy, and the midrange is really clear."

The biggest variable that DJs on the Moonshine Overamerica tour faced was the mixer. For the Los Angeles event, the sound company provided a Rane MP2016, a 6-channel mixer based on the legendary, discontinued UREI 1620. Lawrence says that most DJs require a mixer with a minimum of three channels, 3-band EQ for each channel (not just the master outputs), gain controls for each channel, a prefader level meter for matching the gain of a cued record to one that's playing, output meters, and separate volume controls for the monitor outputs and headphones.

"Sometimes I'll be mixing on a different mixer every night for a month," says Lawrence. "There are a lot of good models out there, like the Pioneer DJM-500, the Allen & Heath Xone mixers and some of the top-of-the-line Vestax mixers. I actually prefer Numark mixers over a lot of the other mixers out there, but you don't always have a choice. Some promoters think that DJs can get by with any old mixer with two input channels and a crossfader, but a lot of mixers have crappy components that distort the signal no matter what. They think we're just playing records, but there's a bit more to it than that." ■

Chris Gill is the Los Angeles-based editor of Remix *magazine. He used to have an active career as a DJ and remixer until taking over the publication, but he still manages to attend at least three raves a month.*

All Quiet on the Wedgeless Stage

Tips on Purchasing and Using Your First Set of In-Ear Monitors

BY MARK FRINK

In-ear monitors are now widely accepted as an alternative to wedge or side-fill monitors for live performance. First developed in the early '80s at the prompting of such innovative artists as Todd Rundgren and Stevie Wonder, the technology has now matured to the point that large equipment manufacturers have entered the market with lower-cost products, making "in-ears," or IEMs, affordable for a broad range of performing musicians. (It should be noted that the brand name Ear Monitors® is a registered trademark of Future Sonics.)

The major benefits of IEMs are several: When operated correctly, they can help conserve the hearing of both the artists and the monitor engineer; singers who risk straining their voices when booked for three or four gigs in a row now find that they can perform comfortably for five nights a week, a key factor in making high-overhead tours profitable; and entertainers with impaired hearing, who can find live shows especially challenging, can use IEMs to extend their performing careers by years, even decades.

For bands just starting out, IEMs can completely replace a rehearsal space sound system. Better yet, because an IEM system creates its own acoustic space between the ears, even the most claustrophobic practice space can be made quite tolerable. For the same price as decent amps, wedges and equalizers, an IEM system can serve a three- or even a four-piece band quite well. The same rig can be used in the studio and can also simplify club gigs and support slots, ensuring a level of consistency that is otherwise hard to achieve with a different wedge system at every show.

On the other hand, in-ear monitors are perhaps the most frustrating products to shop for. Unlike almost every other piece of equipment onstage or in the FOH and monitor sound systems, IEMs cannot be easily auditioned or A/B'd with comparable equipment, and when you're talking custom models, only one person can listen to the product at a time.

Setting up to use IEMs requires considerable preparation and forethought. Band members and management need to be convinced that the serious cash outlay will be well spent, and monitor engineers need to be sure that the budget will cover all of the specialized equipment necessary to provide each band member with a quality ear mix. And don't forget to include a small tackle box for keeping the ear-molds and belt-packs together, along with a supply of batteries and cleaning supplies for the molds. Misplacing an ear or pack can result not only in a serious financial setback, but can also be a real show stopper.

CHOOSING YOUR MOLDS

There are essentially two types of ear-molds: generic and custom. A variety of manufacturers now offer the generic, one-piece-fits-all models, and some can be upgraded to a custom fit with "tips" or "sleeves" from the makers of custom molds, which are made from an impression taken of the user's ears. One difficulty for those purchasing custom molds is that they cannot hear their molds before they buy them. The tendency is for first-time users to try someone else's generic piece. While that is an inexpensive way to check out the actual ear-piece driver, it may not be the best solution, but a much worse idea is to audition an IEM system through a $20 pair of Walkman ear-buds. Because properly fitted IEMs, whether generic or custom, will tend to block out ambient sounds on a performance stage, the system will allow for accurate monitoring at much lower volumes than wedge systems. Walkman-type ear-buds block out very little stage noise, so musicians typically turn them up really loud to overcome drums, guitar amps and P.A. backwash. At these levels, cheap ear-buds usually distort, and excessive levels, distorted or otherwise, will ultimately damage the user's hearing.

Yes, the difference in price between the least, expensive generic models (a couple of hundred bucks) and the best custom molds (about $700) is considerable. But many musicians have spent more than that on a poorly chosen guitar, digital effect or other toy of the moment. For a product that can both improve live performance and help preserve hearing—and will be more comfortable—the extra money is well spent.

Having said that, it is worth noting that good sound is a matter of taste. Everyone's ears are different, and variation in the size and shape of the ear canal will affect how any given transducer couples to the eardrum, with the result that the same product will sound a little different to each user. Also, each of us has had varying amounts of hearing recruitment (a less controversial

harder than dialing up a good headphone mix. The first step is to get used to them, and you can start using IEMs immediately, whether or not the band uses them. I stopped using a cue wedge at my console years ago, and I always monitor on "ears" during the show, even if the band is all using wedges. Apart from anything else, it's easier to shoo hangers-on away from the monitor mix position when there is no cue wedge to listen to. Plus, using a wireless cue system and the same kind of ear-bud as the performer gives me a high degree of confidence that I am hearing what the performers are hearing.

The ability to put together an ear mix quickly is a great asset, and monitor engineers new to mixing "ears" might consider getting some practice by creating dummy headphone mixes during idle moments while mixing traditional speaker-based monitors. After all, after the first three songs in a set there is often little for a "wedge" engineer to do. And because the difference between a precise and isolated in-ear mix and the typical ambient stage wash from multiple wedges is quite significant, it pays to be prepared.

When you do finally start a band on IEMs, present them with a mix that's already built, especially if it's their first time on IEMs. The experience of putting in ear-molds and finding nothing in the mix is disconcerting, even for experienced users; make a habit of preparing and checking each mix before it is handed out. Obviously, it is desirable to have a single style for the monitor engineer and the entire band so the engineer can reference each mix. Different group members will probably have varying needs and tastes, but a mix can be modified easily to an individual's taste simply by inserting an EQ across his or her output.

Mixing style can vary greatly between in-ear and traditional wedge-based monitors. Performers who rely on speakers are used to hearing both the stage acoustics and the sound of the P.A. coming back from the room. Players whose instruments are nearby are easily heard, but instruments on the other side of the stage will need to be in the wedge. Levels are all nominally about the same, but to hear an input in a wedge mix, it has to be at a higher level than the stage wash; most inputs in a wedge mix will be set at above 11 o'clock.

The in-ear mix, on the other hand, must contain everything the performer needs to hear, and small level changes are easily heard. Some inputs will be plenty loud at 9 o'clock, while others (usually the "me" inputs) may be wound nearly all the way open. Levels for in-ear mixes must be set much more carefully than for wedges, so the "velvet touch" will be appreciated. "Split the difference" is a frequently heard instruction for IEM engineers adjusting levels.

Depending on the band's arrangements and playing style, levels of some inputs may need to change from song to song, and for in-ear mixers this may require more work than is typical when wedge mixing. Inserting compressors in some channels may help, but it may be more effective to simply mix the show, as one would at FOH. The hardest decisions will be which sends are going to be pre-fader and which will be post. Anyone still on wedges will be better off pre-fader, because they'll require fewer inputs and level changes, and it's best to make each IEM performer's "me" inputs pre as well.

For bands just starting out, IEMs can completely replace a rehearsal space sound system. For the same price as decent amps, wedges and equalizers, an IEM system can serve a three- or even a four-piece band quite well.

A frequent problem with singers using IEMs is that they quickly learn that they can back off the mic and then ask for more level, a trick they can't get away with on wedges. The result is more stage wash into the vocal mic, which makes the FOH engineer's job harder and rarely improves the monitor mix. My solution is to open each singer's input up all the way and then offer to turn other inputs down in the mix. Plus, you can surreptitiously turn a singer's vocal down in his or her own mix, which usually gets him or her to hug the mic during quiet songs when he or she might otherwise back off.

Stereo mixes are vastly superior to mono for in-ear monitoring. It is much easier to identify the various components in a mix when they are panned across a stereo sound field, and stereo can help keep monitor levels lower, which will reduce ear fatigue and conserve hearing. Check this for yourself by switching a headphone mix back and forth from stereo to mono.

As mentioned above, it is easy to make an in-ear mix sound better by inserting a stereo graphic EQ. The response of the actual IEM driver may not be perfect, and most musicians will have some differences in the response of their ears, but minor EQ tweaks can quickly improve the perceived sound without having to change anything else. In fact, boosting highs to compensate for high-end roll-off in either an ear-mold or a wireless will provide a Dolby-like mechanism to cover up some of the noise in a wireless system. Wireless users can even walk over and tweak their own mix. After using mix EQ for a while with ear monitors, you may learn quite a bit about the combination of your hearing and the IEM products you've chosen.

SIGNAL CHAIN REVEALED

There are no shortcuts to quality. Though technology has improved over the past decade, the more expensive products tend to perform better. Once you put transducers right into your ears, the quality of all the components in a monitor system becomes more easily apparent. The mic preamps in most live consoles are satisfactory when heard through speakers, but the differences between desks are easily heard when listened to in the isolation offered by good in-ear systems. These differences are especially noticeable with vocals. The use of outboard studio-quality mic pre's for the money channels can provide a better listening experience.

Similarly, the qualities of digital reverbs are more easily heard on in-ear monitors. Most performers will enjoy hearing a little reverb on their voice or instrument, but sharing reverbs can be cumbersome and confusing, and simply inserting a reverb in the overall mix is

not sufficient. Dedicated quality reverbs allow the operator to give each performer exactly the mix he or she wants.

Though they eat up more inputs, reverbs definitely sound better in stereo. However, once you provide a stereo mix plus a reverb send for every musician using IEMs, the number of desk outputs required for in-ears nearly triples over that required for a wedge-based monitor system. A band with five musicians and three background singers might require as many as 24 output buses if they are all on in-ear mixes. Fortunately, there are strategies to reduce the number of mix buses. Reverbs dedicated to a single input can be fed an input from the channel direct output, and extra reverb send buses can be created by using an outboard line mixer to combine direct outputs from several channels.

Returning reverbs in stereo will either take up a stereo input (all too rare on monitor desks) or two mono inputs. Fortunately, many monitor consoles offer auxiliary inputs, typically used for line level signals like CD players, but equally useful for bringing in the outputs of a sidecar desk. Because these extra aux ins usually have dedicated volume controls and can be monitored on the cue bus, they can also be used for reverb returns. Multi-effects units often have enough onboard parameters to provide substantial EQ control, so a lack of board EQ on the aux ins may not matter.

Another way to reduce the number of monitor desk inputs needed, while also freeing the monitor engineer from operational overhead, is with additional small mixers onstage. Many keyboard players already use mixers in their rigs to give them independence from the monitor system and can happily program or noodle on headphones while the rest of the crew goes about getting ready for soundcheck. By sending band and vocal submixes to the keyboard mixer, you can probably give the mixer everything needed for an IEM mix; conversely, using the keyboard mixer's output instead of individual DIs on each instrument can free up quite a few input channels on the monitor desk. In most situations, background singers can be quite adequately served with a small mixer fed with splits of the vocal channels and an instrumental submix from the monitor board; a dedicated reverb may also be necessary.

COMMUNICATIONS

Clear communication between the operator and performers is a cornerstone of mixing monitors, and IEMs lend themselves to concise and private two-way discussions during the show. Clear-Com's TW-40 walkie-talkie interface allows radio traffic to be injected into the party-line intercom system. For my cue mix, I use a wireless in-ear system set to mono. One input is fed from the desk's headphone output, and the other has the intercom plugged into it. I can hear everything at once at a comfortable volume and in private, and the backline techs on the opposite side of the stage can communicate the requests of the musicians they work for without making a trip across the stage. Another simple way to facilitate communications is to place a spare microphone onstage so the musicians can talk to the monitor engineer between, or even during songs.

PROTECTION LIMITING

The use of "brickwall" limiters has been touted as a safety measure for hearing conservation, protecting the performer's ears from inadvertent transients. Set correctly, brickwall limiters will kick in before the IEM system's onboard limiters and, hopefully, before the onset of distortion. But the best safeguard is a vigilant, attentive operator who carefully provides a safe mix, and that is only one part of the equation. Because each performer has an individual master volume control on the IEM belt-pack, it is all too easy to turn up the mix to overcome stage volume or hearing deficiencies. However safe and stable the IEM engineer's mix, the higher sensitivity and lower impedances of today's ear-molds allows them to put out more than enough SPL to gradually cause hearing damage over time.

Perhaps the most important and overlooked feature of professional-quality ear-molds is the amount of isolation that they can provide. Even with brickwall limiters and good mixes, poor isolation will tempt performers to raise the volume. Long exposure to even moderate levels can damage hearing over time, so it is important that performers are aware of their volume control positions. A musician who learns to first try something other than simply "turning up" will go a long way toward preserving his or her hearing. ■

Mark Frink is Mix*'s Sound Reinforcement editor.*

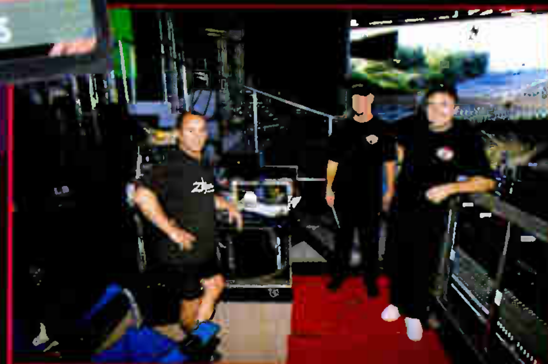
ALL ACCESS

PHOTOS AND TEXT BY STEVE JENNINGS

Nineteen-year-old Christina Aguilera, already a four-time chart-topper with singles from her debut album and a Best New Artist Grammy winner, has made an impressive start to what many predict will be a long and successful career. On tour last summer, Aguilera sold out arenas, sheds and state fairs across the country and drew consistently positive reviews for her well-paced show and extraordinary vocal talent. Opening the show with her hit single "Genie in a Bottle," Aguilera also delivered convincing readings of Etta James' "At Last" and Free's "All Right Now," as well as introducing songs from Mi Reflejo, a Spanish-language collection. Mix caught her trek through Northern California in mid-October 2000, when she played the Sacramento Valley Amphitheater and the Chronicle Pavilion in Concord.

Aguilera is sponsored by Shure and sings into a Shure UA Series wireless with a Beta 87 capsule. Background vocalists use Beta 87Cs, drums are miked with Beta 98s, and overheads and guitars get KSM32s.

(L to R) John Ciasulli (guitar tech), Chris Achzet (drum tech), Michael Bernard (tour programmer)



(L to R) Wade Crawford (system tech), John Porcelli (system tech), Stewart Bennett (FOH engineer). FOH engineer Stewart Bennett is using an unusual combination of FOH consoles—a Gamble DCX Event 40 and a Midas XL4. "The Gamble is an analog console with a digital control surface and is designed to be operated with a mouse," explains Bennett. "Being a little apprehensive at not having faders to mix with, we opted to interface the Gamble to the Midas XL4. Plus, I wanted to provide as conventional a layout as possible for the other engineers that would be using the system. Knowing what I know now, I probably would just use the Gamble DCX. It's got an amazing mic pre that is substantially better than any other console that I've ever used, and the EQ, compression and gating on each channel is equal to anything else out there. The combination ends up being just a wonderful small package that really takes a lot of the headache out of mixing. And Jim Gamble has given us great support."

On Aguilera's vocal, Bennett has been running four processors in series. "The way that Christina sounds on a microphone is incredibly bright, and the fact that she grabs the microphone around the capsule exacerbates the enhanced upper midrange," says Bennett. "And Christina's mic technique is such that she can have the microphone three feet away and then right on her lips the next second." First the signal goes to an Apogee parametric for a substantial amount of upper midrange cut. Next in line is a BSS DPR-901, then a Summit TLA-100 and finally a dbx 160 for hard compression. "It's embarrassing, to a certain extent, but it ends up being the right set of tools for the job," comments Bennett.

For FOH effects, Bennett has been using a Lexicon 224, an Eventide H3000, a Yamaha SPX900, a TC Electronic 2290 and a Lexicon PCM 81.

CHRISTINA AGUILERA

(L to R) Bill Chrysler (monitor engineer), Chad Prater (monitor system engineer). Monitor engineer Bill Chrysler, a 22-year-veteran of the road, was out with Limp Bizkit and Bob Dylan before the Aguilera tour. Chrysler is using a 52-input Midas Heritage 3000 to create in-ear mixes for the band, all of whom are using Ultimate Ears earpieces. Aguilera works from Showco SRM wedges and sidefills, which Chrysler has distributed around the multilevel stage. "I have four wedges down front and two sidefills," says Chrysler. "Up on the ramps I have one mix and another up on one tower. In the four speakers down front, Christina hears her vocals with a little bit of piano, drums and guitar from the center two wedges only. The outside speakers get a pretty regular mix, but with her vocal on top. The sidefills just have a regular FOH mix with her voice mixed right in there."

For processing, Chrysler uses a Yamaha REV7 on Aguilera's vocal in the wedges. He also uses a Lexicon PCM 70 on acoustic guitar and a Yamaha SPX990 on the drums, but only in the in-ear mixes. "And really that's it—just three reverb units. I also have some Brooks-Siren gates, some Behringer compressors and gates, and some Drawmer gates," says Chrysler.

The live musicians are supplemented with prerecorded tracks on a few songs. "On 'Genie in a Bottle,' you can't really duplicate that sound of her voice over a telephone wire," explains Chrysler. "The fans want to hear it just like on the record—it's a signature part of the song—so we use the hard drive for that."

Aguilera has been appearing in a wide range of venues, with capacities ranging from 12,000 up to 25,000. The Showco Prism system is configured in five columns a side for sheds and eight columns a side for arenas, supplemented by six subwoofer cabinets per side.

FOH engineer Stewart Bennett has been using Smaart Pro as a system analysis tool and to help match the Showco system to the various house delay and lawn systems. "We've had to use the house P.A.s at some of these state fairs," notes Bennett. "Smaart Pro makes it relatively easy to tweak the systems to match the mix on our console, as opposed to having to reset the console to adapt to a new sound system."

What the Audiologist Told Me

A SOUND ENGINEER EXPLORES THE NEW WORLD OF DSP-BASED HEARING AIDS

BY BOB MCCARTHY

Still wondering where to invest those unspent per diems? Looking for a technology sector industry in its infancy that will grow for the rest of your life? I have two words for you: hearing aids. The big all-star benefit for the new millennium will not be Live-Aid, Farm-Aid or Net-Aid. It will be Hearing-Aid.

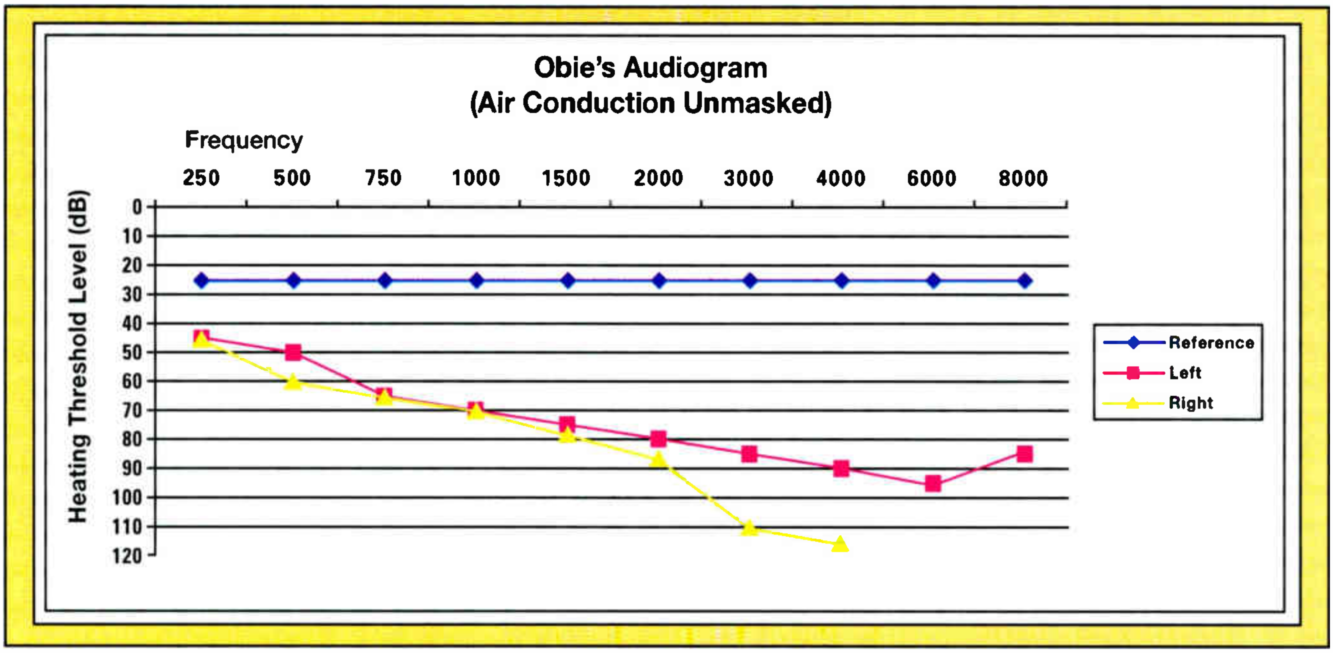
Baby boomers are getting old. They are going to outlast their ears. They are also accustomed to constant aural stimulation, and they are not going to suffer in silence. And the technology is ready. Hearing aids are rapidly becoming more socially acceptable and more affordable. But unless you are getting ready for retirement or know someone who benefits from a modern hearing aid, you are probably unaware of recent advances in hearing aid technology. I was too, until I helped my 85-year-old father-in-law get a hearing aid.

The audiogram below shows the results of Obie's hearing test. Each point on the graph is the loudness level at which Obie was able to first hear that frequency. Notice the downward trend in threshold at higher frequencies. This trend is typical of the age-related hearing loss known as "presbycusis." Notice also that the right ear is much worse than the left. This is believed to be due to Obie's exposure to the near-field cannon fire of the Sherman tank he commanded in World War II.

HEARING AIDS: THEN AND NOW

The human hearing mechanism has evolved to meet the needs of its owners. Until comparatively recently, most humans led lives that were nasty, brutish and short, and only a minority lived long enough to be bothered by the inevitable decline in their hearing. But thanks to modern health care, many of us will live long enough to be great-grandparents, and, thanks to a new generation of hearing aids, many of us will hear our great-grandchildren's first words. Technolog- ically advanced hearing aids offer the promise that we will be the first generation in human history whose perception of sounds does not necessarily have to degrade substantially as we grow older.

The key phrase here is "technologically advanced." As medical science has progressed, so have the solutions to our ills become more sophisticated. The first hearing aids were crude—



Euphonix TALKS WITH PHIL RAMONE

ABOUT HIS WORK ON ELTON JOHN'S 'ONE NIGHT ONLY'

The Elton John live event marked another milestone for Ramone: It was the world's first live, 96-track, 24-bit/96K recording and the first time the veteran recording console commander has deployed the Euphonix R-1 hard disk recorder

What made you decide to choose the R-1 on the final mix?

In a word: quality. *We were also helped by having 96 tracks in one place, so we didn't have to worry about synchronizing the machines, it was all on the R-1*

How does the R-1 compare to other recorders in the industry?

There is nothing on the market quite like it. Euphonix has created something that has the potential to enjoy a prominent place in the industry. It has a little cáche right now, and I think the quality of the Elton recording should quiet any detractors. I think every engineer in the industry should be exposed to this equipment and have a chance to work with it.

How did the R-1 compare to other recorders in the industry?

The layout of the machine was very comfortable to use and gave us a quick response. The R-1 saved us a lot of steps and undoubtedly some grief as well

PHIL RAMONE has his choice of ANY gear in the world. He chose the Euphonix R-1 96KHz/24bit digital recorder.

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cupped hands, an animal horn held up to the ear, then the illogically named speaking trumpet. Finally, early 20th century electrical technology ushered in the beginnings of modern hearing aids. Early systems used carbon mics, vacuum tubes, and large, expensive, nonrechargeable batteries that would last only a day. The advent of the transistor and solid-state technology meant that users were freed from unwieldy tube amplifiers, and designs became light enough to fit behind the ear or to be integrated into eyeglasses. Battery life was also greatly extended and sound quality improved as mics and speakers moved out of the carboniferous era.

In the case of externally worn devices, the appearance of hearing aids has not much changed, perhaps creating the illusion that the evolution of hearing aids has stalled. It hasn't. It's gone digital.

Inside a modern digital hearing aid is a dedicated DSP platform that rivals in scope the FFT analyzers that I have been using to align sound systems for the past 16 years. DSP offers digital filters, adaptive equalization, feedback suppression, noise suppression, multiband compression and more—all of the same hip stuff as in-the-ear monitors, except lower power and no radio transmitters.

DSP allows a hearing aid to be customized to compensate exactly for its owner's unique hearing problem. There are essentially two types of hearing mechanism problems. The first is mechanical damage in the middle ear, where the eardrum is coupled to the fluid-filled cochlea. These types of problems can, in some cases, be fixed surgically or with implants, but they are rarely solved with external hearing aids. The second type is inner ear damage, which is often a normal product of aging, or can be the result of a birth defect or excessive exposure to loud sounds. In fact, the most common cause of hearing loss is presbycusis, a type of age-related audiological decay characterized as the loss of inner-ear sensitivity. The effect is like slowly rolling in a hi-cut filter and lowering the fader, and it generally increases in severity after age 60. Another common cause of hearing loss is acoustical trauma, which, in our industry, is likely the result of mixing monitors for Black Sabbath. For my father-in-law, it was his role as a tank commander in World

HOUSE CALLS

SOME HEARING F.A.Q.S

Named for its founder, Dr. Howard House, the House Ear Institute of Los Angeles is a private nonprofit scientific center dedicated to improving the quality of life for people who have hearing and related disorders through research and education. Sound Partners is the Institute's hearing conservation program that focuses on hearing health issues in the music, pro audio and sound industries. To find out more, visit www.hei.org or call 213/483-4431.

How is hearing measured?
An audiogram is the graph measuring the lowest level that each ear hears a pure tone. Sound professionals should have their hearing checked annually in order to monitor any changes.

What frequencies are tested?
Most medical practices test each ear at 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 6 kHz and 8 kHz (the speech range), because hearing loss in these frequencies affects one's ability to communicate.

Why are the higher frequencies not tested?
Although hearing screenings at higher frequencies are available, *accurate* test results above 12 kHz are difficult to obtain, because no reference point for normal hearing thresholds is available. This is due in part to a lack of research to establish a standard method, equipment limitations and difficulty in reproducing testing conditions.

House Ear Institute
Research and Education... *so all may hear*

When does sound become hazardous?
The main factors contributing to a hazardous environment are high sound levels, the duration and frequency of exposure, and individual susceptibility.

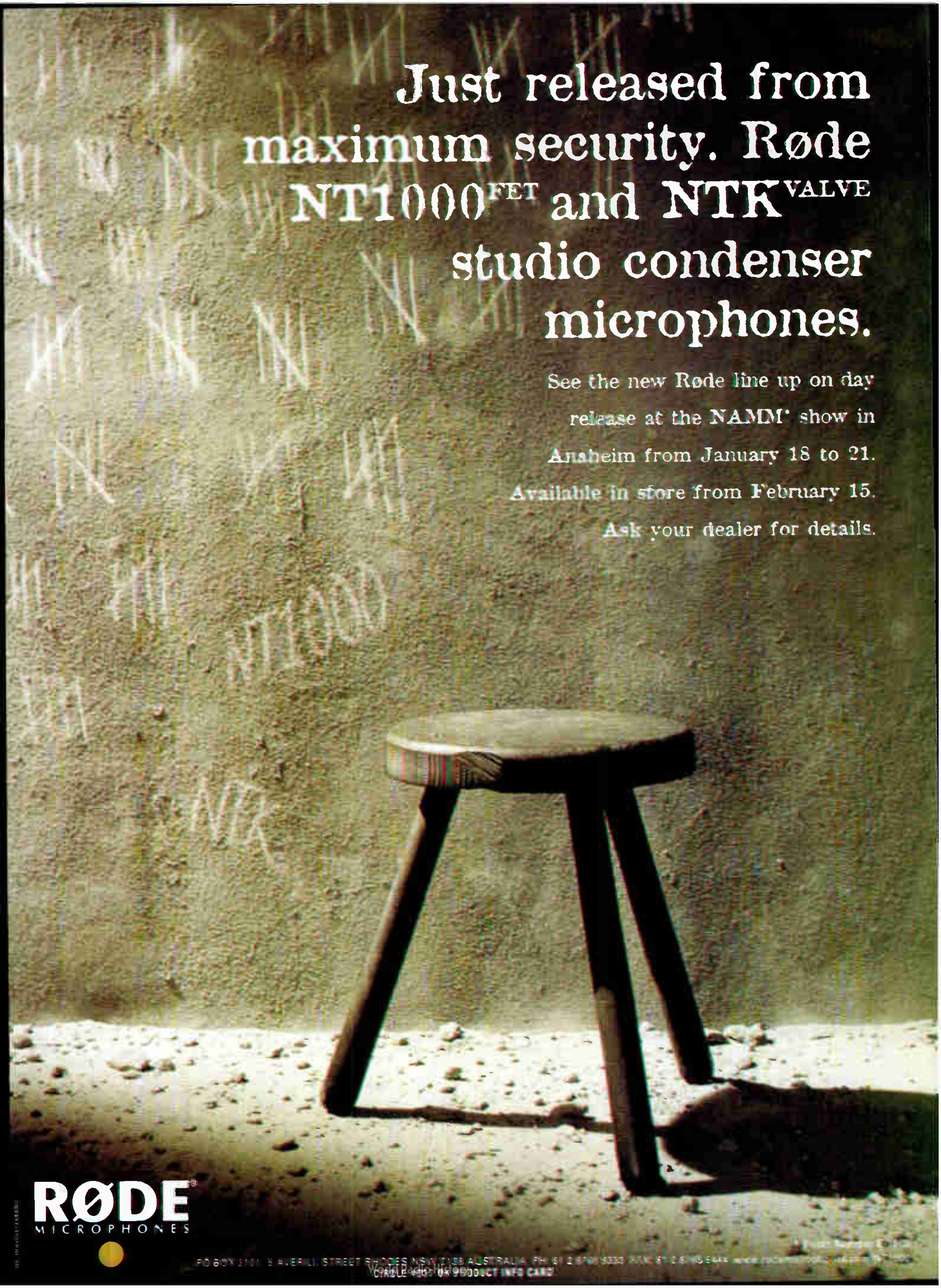
What is nerve damage?
Nerve damage, or sensorineural hearing loss, occurs in the cochlea or hearing organ. It is permanent and incurable.

What are the hair cells referred to in hearing?
The outer hair cells are the sensory cells in the cochlea and are so named because of the hair-like projection at their tip. When damaged, these nerve cells do not regenerate and permanent hearing loss occurs—although scientists at the Institute are working on this.

What is Tinnitus?
Tinnitus is a symptom, not a disease, and may be due to a number of ear- or hearing-related problems. It is a ringing, roaring or whooshing sound in the head that may be temporary or permanent in nature.

What causes Tinnitus, and is it treatable?
The simple answer is stress. Chemical stress can be caused by an excess of coffee, by aspirin or as a side effect of medication. Acoustic stress is caused by overexposure to loud sound. Physical stress is often the result of exertion or an illness. Pathological stress is due to a blockage in the auditory system (such as wax in the ear canal) and emotional stress. Tinnitus may be treated through a controlled diet, biofeedback and the use of electronic masking devices. It may be man-

—CONTINUED ON PAGE 78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War II that caused his hearing loss.

OBIE GETS HIS EARS

Ivan (Obie) Oberdan has been hard of hearing for 56 of his 86 years. In our family, we referred to him primarily as "hard of heading" due to his obdurate resistance to addressing the problem. He had bought a hearing aid back in 1966, but it amplified the background noise too much, so it was of little benefit and was a constant annoyance. We pointed out that a lot had happened since 1966, including the invention of the wheel and DSPs. We felt that the new technology should be given a chance. Obie finally agreed to a hearing test, and I happily went along to observe.

The audiologist explained to us, in language suitable for four year olds, how she would take the measured frequency response of Obie's ears and apply an inverse response into the internal DSP filters. I let her know that I was an audio professional familiar with the concept of equalization and that talking to me as if I was a three-year-old would be more appropriate.

She then measured Obie's hearing sensitivity with octave-interval sine wave tones. Obie's threshold of hear-

—FROM PAGE 76, HOUSE CALLS

aged, but it cannot be cured.

Can regular exposure to high sound levels affect your health?

Yes. Prolonged exposure to loud sound tends to make your heart beat faster, resulting in an increased flow of blood throughout the body. On a recurring basis this may have a cumulative effect and may cause headaches, irritability, insomnia, hypertension, fatigue, reduced efficiency and ability to concentrate, and even low morale. Temporary threshold shifts or a muffled sound sensation may be experienced for a few hours or even days after sound exposure. Continued unprotected exposure to sound may lead to permanent threshold shifts or nerve deafness. Tinnitus is also a common disorder associated with overexposure to sound with the emitted signal increasing steadily over time.

Does flying affect one's hearing ability?

Some people's hearing ability may be affected by flying, because their Eustachian tubes are unable to efficiently equalize the pressure in the middle ear. The ear pain or discomfort experienced when the airplane descends may be alleviated by yawning, chewing gum or sucking a piece of candy. Special equalizing ear filters are also available. It is advisable to avoid flying when one has a head cold, an allergy attack or a sinus infection, as blocked nasal passages may increase the discomfort and even lead to a burst eardrum. If flying is unavoidable, use a decongestant to clear the nasal passages 24 hours before departure and a nasal spray 45 minutes before the aircraft is due to land.

Are there any foods or beverages that affect one's hearing?

Caffeine, chocolate, alcohol and any form of stimulant have a diuretic effect that shrinks the vascular system. This may lower one's hearing threshold and increase the volume of a Tinnitus signal. An allergy to certain foods may also cause hearing loss.

Are there any dietary supplements that promote hearing health?

Niacin, a vitamin B-complex, increases blood flow and may be used to counteract poor circulation. Gingko biloba is often promoted as having beneficial properties, but there is no scientific proof that it enhances hearing.

Do medications have an affect on hearing ability?

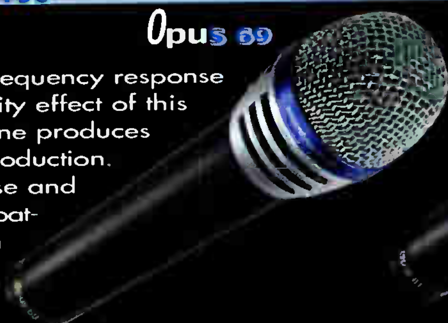
Yes. Aspirin and other salycilates act as stimulants and may increase hearing loss or Tinnitus. Anti-inflammatory agents, aminoglycosides such as neomycin, pain medications such as Vicodin, loop diuretics, steroids and quinine-based medications may also affect one's hearing. It is important to ask your physician if the prescribed medication is ototoxic or reacts adversely with another medication. ■

beyerdynamic introduces Opus, a selection of high performance touring and sound reinforcement microphones for the demanding musician and audio engineer.

The collective work of German engineering and ingenuity, the Opus range of microphones is a cohesive family of products optimized for music performance. Classic technology and 21st century materials combine to create a complementary range of microphones for the musician on stage or in the studio.

Vocal Microphones

Opus 69



A wide, uncolored frequency response and warm proximity effect of this dynamic microphone produces spectacular vocal reproduction. Its remarkable response and tightly controlled polar pattern enhances usable gain before feedback, producing a crisp, natural sound.

Opus 81

Advanced technology and extraordinary engineering has produced this state of the art condenser microphone. The Opus 81 can be used on the stage or in the studio. It produces a smooth, warm sound, accurately reproducing the rich textures of a vocal performance.

Instrument Microphones

Opus 51

A boundary microphone, the Opus 51 captures direct and reflected sounds from an instrument in perfect balance and at an equal level. This microphone is ideal for capturing the full sound within an instrument such as the piano or bass drum.

Opus 65

The Opus 65 is built with a 1.5 inch diameter diaphragm within a powerful neodymium magnet. The Opus 65 requires little or no equalization to produce the natural sounds of low frequency producing instruments, virtually eliminating the problems incurred by other kick drum microphones.

Opus 83

The Opus 83 is a condenser microphone ideal for drum and percussion overheads, cymbals, and acoustic instruments. Built from solid brass, the casing of the microphone protects the sensitive element from the rigors of the road.

Celebrating its 76th anniversary in professional audio, beyerdynamic adds to its legacy of audio innovation with Opus, the uncompromising choice for musicians who refuse to sacrifice their sound.

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ing in his right ear was beyond the edge of the chart, and his "good" ear was nearly there.

Throughout my career as an alignment specialist, I have lived by one constant principal: reproduce the original signal as closely as possible. A hearing aid's role seemed to me to be analogous to the filter set used to tune a sound system. However, just as in optimizing a sound system, although equalization is a crucial step toward creating a natural listening experience, taken alone it is not the intelligibility solution we seek. As we age, we also lose the ability of the ear to respond to a wide dynamic range. Not only do we lose sensitivity, but the elasticity of the basilar membrane also decreases, limiting its range of excursion.

This loss of dynamic range is, in some ways, more damaging than the loss of sensitivity at certain frequencies, because it limits our ability to discriminate between useful signals and noise. We live in a world of background noise. We are able to converse because our binaural hear-

HEARING HEALTH RESOURCES

Hearing Education and Awareness for Rockers (H.E.A.R.) is a volunteer-based, nonprofit organization. Since its inception in 1988, H.E.A.R. has worked to educate the public on the dangers of excessive noise levels in music. Many audiologists, otolaryngologists, musicians, DJs, sound engineers and equipment manufacture champion H.E.A.R.'s cause. These "H.E.A.R. affiliates" are listed on H.E.A.R.'s award-winning Web site, which also offers information on hearing protection, custom earplugs, in-ear monitors and hearing aids. For more information on H.E.A.R. and a comprehensive set of links to hearing- and music-related Web sites, visit www.hearnet.com. H.E.A.R. also maintains a 24-hour hotline at 415/773-9590.■

PROTECT AND PRESERVE YOUR IRREPLACABLE HEARING

BY LISA TANNENBAUM

Unlike sound equipment that can be repaired or replaced, you only get one set of ears. Once ears are damaged, you can't tweak, fix or replace them. Yet, despite the fact that ears are among the most valuable tools that musicians and sound engineers possess, they are often neglected.

Prolonged and frequent exposure to loud sound is one of the leading causes of permanent hearing loss. Sound engineers and musicians are obviously at higher than normal risk for noise-induced hearing loss. And, unlike those in other noisy occupations, their livelihoods and careers may depend on their hearing.

The good news is that noise-induced hearing loss is totally preventable. The bad news is that many sound engineers and musicians are not motivated to protect their hearing until they notice a problem. They may become concerned only after experiencing constant ringing in the ears or getting comments that their mixes are overly bright and brittle-sounding. Though it's never too late to commit to using hearing protection, the time to think about it—and act—is before problems arise.

MUSIC-INDUCED HEARING LOSS

Noise- and music-induced hearing loss is a function of exposure time, the average noise level and the peak level of very loud sounds. Damage from excessive noise occurs, for the most part, in the cochlea—the tiny, snail-shaped structure located in your inner ear. A healthy cochlea is lined with thousands of minuscule hair cells, which, when stimulated, brush against nerve endings that send sound messages to the brain. Hair cells located along various portions of the cochlea correspond with different frequencies. The hair cells that line the outermost portion of the spiral respond to higher frequencies, and those in the innermost portion respond to lower frequencies. The hair cells that respond to high frequencies are especially vulnerable to excessive noise and are usually the first to be affected. Once the hair cells are damaged, sound transmission becomes impaired, and once the hair cells are gone, they're gone—they don't grow back and cannot be replaced.

Noise-induced hearing loss occurs gradually and insidiously. After a loud gig or rehearsal, you may experience reduced hearing acuity; sounds and voices may appear to be muffled or distant for a day or so. The muffled sensation may be accompanied by an annoying ringing in your ears (Tinnitus). With appropriate ear rest, the ringing will probably subside and your hearing will recover. This is referred to as a "temporary threshold shift." But if this temporary threshold shift occurs night after night, week after week and year after year, the result may be "permanent threshold shift." If you notice ringing in your ears or even a slight change in your hearing, pay attention—these are red flags that should not be ignored.

Individual susceptibility to noise-induced hearing loss varies, and it's difficult to predict who will sustain permanent hearing loss and who won't. As professional sound engineers and musicians, you must respect the fact that the nature of your work puts you at a high risk for excessive noise exposure. The following questions may help you determine whether you are at risk.

• Do you need to shout to be heard by others during studio work, rehearsals or performances?

• Have you ever noticed a ringing noise in your ears for hours or even a day after sound/music exposure?

• Does music sound distorted toward the end of a long rehearsal or recording session?

—CONTINUED ON PAGE 84

ing system offers several complementary methods for screening out noise. Repetitive noises—such as HVAC systems and car noise or the sounds of whining children or nagging spouses—are unconsciously classified as information-free and are therefore ignored by our thought processes. Our binaural directionality allows us to localize an interesting sound source and focus on it, a trick that allows us to converse at a party while hundreds of voices nearby are transmitting in the same frequency range. This is known as the "cocktail party" phenomenon and actually requires a considerable amount of mental processing power. To get some idea of how mentally taxing this active listening can be, simply listen to a mono recording from a single mic placed in the middle of a room with several people conversing. For a person with compromised binaural directionality, the only way to sort out the important information is to concentrate on only the loudest signals.

With an undamaged human hearing system, which naturally has an extremely wide dynamic range (at least 120 dB), this discrimination is quite possible. But for someone with significant hearing loss—and a consequently diminished dynamic range—many of the conversations at a cocktail party will be heard at close to the same level, making for a poor signal-to-noise environment.

Fortunately, modern DSP technology provides the tools to deal with some of these problems. Equalization to compensate for lost sensitivity in the speech frequencies is fairly straightforward. Also, modern DSP-based hearing aids use as many as 14 frequency-conscious limiters, each controlling one band of audio so that overload in one range causes compression only in that range. This mimics the ear's natural limiting system and allows speech to remain intelligible, despite the presence of noise.

For the repetitive noise problem, a DSP chip can scan the incoming sound for repetitive signals and filter out those sounds identified as noise. Binaural localization can be immensely improved by providing hearing aids for each ear, each tailored to that ear's deficiencies. Another advance made possible with DSPs is the inclusion of speech recognition algorithms that can dynamically modify the vowels and consonants. For example, whereas we hear the word "stop," the hearing impaired person with limited dynamic range hears "stOp" with the vowel sound "o" way above the consonants. The speech-enhancement system can restore the balance of vowel and consonants by expanding the consonants and compressing the vowels.

At present, there are two principal types of hearing aids. The "in-the-ear" types are tiny units that fit discretely into the ear canal. However, these are suitable only for moderate levels of hearing loss. Because the microphone (input) and speaker (output) are so close together, in-the-ear systems can offer only low gain before feedback.

The "behind-the-ear" models also feature a miniature loudspeaker embedded within a custom mold that fits in the ear canal, but placing the microphone behind the ear yields a much higher gain before feedback. This was the only option for Obie.

Additional features of current models include a rear facing cancellation microphone, which can be engaged in noisy environments. DSP also allows for automatic feedback suppression.

Research continues, and aging baby boomers can be counted on to swell the

market for decades to come. As I look forward to my golden years, I take comfort in knowing that when I finally need a hearing aid, the state of the art will have advanced even further. If it all pans out as I hope, we will enjoy the power and fidelity of the audio industry's in-ear monitors, along with the intelligibility processing developed by hearing aid research and development teams.

As for Obie, I can tell you that this has made all of the difference in the world. He is part of our world and part of our conversations. As we left the audiologist's office on the first day of his new ears, we emerged into the parking lot as cars passed on the busy street. "What the heck is all that noise?" Obie asked. "That's called traffic. Welcome to the real world," I replied. Says Obie, "Now, if somebody can find a way to get rid of that, we would really have something." Ain't that the truth. ■

Bob McCarthy is a frequent contributor to Mix.

—FROM PAGE 81, PRESERVING YOUR HEARING

• Do voices sound muffled after you've been around music for a long period of time?

• Do your ears sometimes feel full or like they're "shutting down" following a long day of recording or rehearsing?

• Does your home stereo system sound defective when you return home after a tour?

If your answer to any of these questions is yes, you are at risk for noise/music-related inner ear damage. If you haven't done so already, have your hearing evaluated and embark on a hearing conservation program immediately.

HEARING TEST

The first step to protecting your ears is to find a good audiologist, preferably one who has experience with and knowledge of the unique needs of musicians and sound engineers. The audiologist will recommend that you have a baseline hearing evaluation. Although it's not unusual for music industry personnel to resist this step, the information it provides is critical. Without a baseline, the audiologist has no way of monitoring the efficacy of the recommended plan of action. The only way to objectively determine whether your hearing protection efforts are working is to retest periodically and compare the results to the baseline audiogram.

Even if you opt not to have your hearing tested, don't let it stop you from checking out your hearing protection options. Depending on the type of work you do, the amount of time you're around loud music, the levels you're exposed to and your budget, the audiologist can help you determine the most appropriate hearing protection device. Typically, the recommendation will be for custom musician earplugs (passive hearing protection), in-ear monitors (active hearing protection) or both.

CUSTOM EARPLUGS

By now, most seasoned musicians and sound engineers are aware of the custom earplugs designed by Etymotic Research (ER). For those of you who are new to the field, are unaware of the advantages of the ER plugs, or just haven't gotten around to getting fitted yet, the following explanation of how they work will, hopefully, motivate you to try a pair.

Conventional over-the-counter earplugs present three basic problems: First, most generic earplugs produce 10 to 20 dB more attenuation in the high frequencies than in the mids and lows, making music and voices sound muffled and distorted. Second, traditional earplugs tend to create a large occlusion effect, making sound transmitted by bone conduction (e.g., the wearer's voice) unnaturally loud and creating the perception of voice distortion and a plugged up feeling in the ears. Finally, conventional earplugs often provide more sound reduction than musicians and sound engineers can tolerate. Because of these problems, even the most ear-conscious musicians and engineers have abandoned efforts to wear them.

The Etymotic Research plugs are specifically designed for musicians; they allow you to protect your ears comfortably, without sacrificing fidelity. Fidelity is preserved by incorporating a small-diaphragm filter, similar to a passive speaker cone. The diaphragm, designed to follow the shape of the natural frequency response of the open ear (but at a reduced level), provides a nearly flat response, giving equal attenuation at all frequencies. The occlusion effect is reduced by the earplug's design, which requires a deep seal in the second bend of the ear canal. Finally, the filters are available in reduction levels of either 15 dB, 25 dB and, more recently, nine dB.

The ER-15s are intended for use in environments where the A-weighted sound pressure level is 105 dB or less. The ER-25s are better for situations that exceed 105 dB. The ER-9 plugs were developed more recently to address requests from musicians who wanted a high-fidelity earplug with a little less attenuation than the ER-15. The ER-9 provides noticeably less attenuation in the low and mid frequencies. Because the frequency response is not flat, the fidelity isn't quite that of the ER-15. The filters are all the same size and can be used interchangeably.

IN-EAR MONITORS

More and more performers and sound engineers are using in-ear monitor systems, which can (if used responsibly) reduce ear fatigue. With in-ear monitors, you can isolate your ears from ambient stage volume and virtually eliminate competitive monitoring. These high-fidelity, miniature speakers embedded in custom ear pieces enable you to hear the full dynamic range of your music without experiencing ear-shattering sound levels. If you use or are considering using in-ear monitors, ask your audiologist for guidance regarding safe use. Unlike the earplugs, which are a passive form of protection, in-ear monitors place an active sound source directly in your ears.

Thus, in-ear monitors, while they offer numerous advantages over conventional wedge monitors, should be viewed as hearing protection devices *only if used with guidance regarding safe use.* Safety can be best assured by obtaining an initial baseline hearing test, determining appropriate volume settings based on in-ear sound level measurements, using an earmold that provides a deep seal and good sound isolation, and rechecking your hearing periodically (at least once a year) to monitor any changes. Because in-ear monitoring is relatively new, many audiologists are unfamiliar with the technology and are ill-prepared to address questions on fitting and safety issues. If you need assistance in locating an audiologist who has experience in fitting in-ear monitors, contact Lisa Tannenbaum via e-mail at lisa@xcrash.com. ■

Lisa Tannenbaum, M.S., audiologist, operates Musician's Hearing Services, a music industry-oriented audiology practice based in San Francisco. Portions of this article have appeared previously in other publications.

BY BLAIR JACKSON

PAPA ROACH
The Infestation Continues

Surely one of the great success stories of the past year has been the steady and impressive rise of Papa Roach, the hard-rocking alt metal/rap band from Vacaville, Calif. (near Sacramento). Their *Infest* album on the Dreamworks label has sold nearly three million copies worldwide and has propelled the group from playing small clubs in the Central Valley of California to headlining in theaters and arenas around the world in just a half a year. As usual, this is one of those "new bands" that has actually been together for seven years. But a hit album changes everything—and *quickly*.

The last nine months have been a whirlwind for the group, who have toured nonstop in every type of venue and every kind of tour imaginable. They've headlined clubs, opened an arena tour for Korn, played second on the bill for a recent tour with Limp Bizkit and Eminem, done the late-night TV circuit, the MTV Video Music Awards, a gaggle of multi-act radio station arena and shed shows, a fistful of European dates and they even got invited to the most recent Rock In Rio festival.

So you can imagine that it's also been quite a year for Brendon Brown, who has been mixing the band live. Brown has known the guys in Papa Roach for many years; he was in bands in the Central Valley, and he also cut his teeth as a live engineer mixing sound at a Sacramento club called Harlow's. Later, he got an offer to go on the road as the FOH mixer for another popular Sacramento-area act, Simon Says, and he found "that I liked touring a lot; I liked traveling and I liked the challenge of mixing in different places," he says. "Right when I got back from Europe [with Simon Says], the Papa Roach album was getting ready to come out and I got a call from Brett, who's their manager, asking if I wanted to go out on the road with them. We headed out together and then the record really started to take off, and here we are."

It's not surprising to learn that Brown's approach to mixing this aggressive, elemental quartet—vocalist Coby Dick, guitarist Jerry Horton, drummer Dave Buckner and bassist

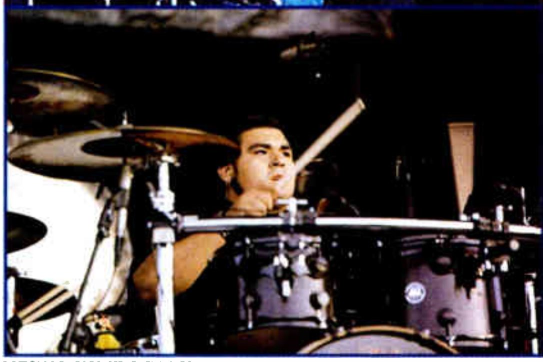
PHOTO DAVE FONDA

PHOTO DAVE FONDA

Left: FOH mixer Brendon Brown

Far Left: Lead singer, Coby Dick (holding an E-V N/D 767a) and drummer Dave Buckner.

Below: Lead guitarist Jerry Horton and bass guitarist Tobin Esperance



BOTTOM 3 PHOTOS STEVE JENNINGS

Tobin Esperance—"is to keep it simple. I like to get a raw mix going quickly—get the band up and going and work from there. I try hard not to over-mix the band. Of course it's heavy rock, but they also have a lot of hip hop feel to their music, in terms of the melodic bass lines and some of the drums, so I approach it like that with real heavy kick-snare-hat and the pounding bass underneath it. I have a couple of dbx 120 subharmonic synthesizers that I work on the kick drum and the bass to fill out the bottom end. It's important to try to keep warmth and definition there." In addition to the 120s, Brown's personal FOH rack contains a BSS 901 parametric compressor that he chains with a dbx 160 for vocals, and an SPX 990.

Brown says that on the current tour, "I'm using all E-V mics except for a Shure SM91 on the kick, in conjunction with an [E-V] 868, to get a heavier attack. It works really well on the bigger shows we've done, like on the Korn tour, and in arenas. Also, Coby likes using either an SM58 or an E-V N/D 767a on vocals. But otherwise it's all E-V, and I've been really happy with the way they've worked out. For the toms I'm using the new 468s, which are really nice because I go for a big, thick tom sound and those work well with the coated heads we use. On the snare top and bottom, I'm using the 478s, which I find are a little warmer than the regular SM57s so many people use. I also use the 478s on the guitar cabinets, left and right. For the bass mic, I use an 868 and then RE200s on all my condensers, which are hi-hat, ride and two overheads.

"I mix what they throw out at me onstage," he continues. "But I also try to mix it a bit so it sounds like the record. It's a conscious effort, because a lot of times you'll see a band and it's a great mix, but the vocals are too far out front. Coby doesn't like to be way out front; he wants to be right there with the guitars. You want to be able to hear and understand the lyrics, but not at the expense of the guitars and bass."

Generally, the band has not been carrying its own production, so Brown has worked on a lot of different FOH consoles the past year, and the group's big sound has been pumped through everything from smallish club P.A.s to a full-blown Showco Prism system. "It made for a very stressful tour," he comments, "but I've learned a lot. [Korn's FOH mixer] Bill Sheppell really took me under his wing. Talk about a guy who can really mix—he schooled me on the dos and don'ts, as far as arenas go, so I got my game down there. But it's tough. I don't even have a monitor engineer. After we get the wedges up, we get a house guy to run the faders. It's been a real challenge to work within the time restraints we've had on some of these shows—like on the radio shows [multi-act arena concerts sponsored by radio stations], you usually get about a 20-minute change-over—and to get everything up and slammin' for when the band hits the stage."

Brown's fortunes are likely to improve as the band plays more headlining shows and embarks on tours where they will carry their own production. For now, there's barely been time to come up for a breath. "I don't think the full impact of what the record's doing and how successful they really are has sunk in with the guys yet," Brown says. "But they don't seem like the kind of people who will be affected by that. They're really good guys, and they're really courteous to me and all the other crew guys. They take very good care of us. It's a family. I would think it will only get better." ■

Blair Jackson is Mix*'s executive editor.*

New Sound Reinforcement Products

ATM'S RIGGING GUIDE NOW FREE

The *Riggermeister Rigging Reference Guide,* published by the ATM Fly-Ware division of ATM Group, is now available for free at www.atmflyware.com. Downloadable in Acrobat format, the guide contains reference information on hardware, rigging application, rigging formulas and other topics. In addition, the Riggermeister Seminar slide show is available in PowerPoint format, and Riggermeister Software used for calculation of complex rigging formulas is available in Excel format. ATM Fly-Ware is making this information available as part of a revived effort to educate the industry about safe rigging practices.

Circle 314 on Product Info Card

MACKIE ACTIVE SUBWOOFER

Now shipping, Mackie Designs' (www.mackie.com) SRS1500 Active Sound Reinforcement Subwoofer is a compact, high-output unit featuring a Mackie/RCF (Reggio Emilia, Italy) 15-inch, long-throw, high-precision woofer. Powered by an integrated Mackie FR Series 600-watt amplifier, the SRS1500 has a frequency response of 45 to 120 Hz with 127dB peak SPLs and complements Mackie's SRM450 active loudspeaker, which can be pole-mounted directly above the subwoofer. Besides its compact 23.5x17.17x23.75-inch (HxWxD) size, the SRS1500 also features integrated active electronics with active equalization, time correction, phase alignment and speaker protection. Price: $899.

Circle 315 on Product Info Card

LA AUDIO DIGEQ RTA CARD

LA Audio (www.laaudio.com) offers the DPF-RTA, an optional plug-in DSP card for either the DPF3103 digEQ Master unit or the DPF3103R digEQ Remote Control. The DPF-RTA is a professional standard 31-band, ⅓-octave, filter-based Real Time Analyzer and, when fitted into the Remote Control unit, provides a highly portable, battery-powered system for collecting and storing data. digEQ systems equipped with the DPF-RTA and Digital Delay option can generate a click from the remote, useful for accurate time alignment. Analysis curves can be frozen and saved in any of 45 nameable memory locations, and four microphone offset memories allow non-reference mics to be used. The unit includes its own pink noise generator, and graphic EQ faders can be superimposed onto the analysis screen. Displayed range, reference level, filter averaging period and peak decay time can all be altered.

Circle 316 on Product Info Card

AKG WMS 40 UHF WIRELESS

AKG (www.akg-acoustics.com) has introduced the WMS 40, an affordable UHF transmitter/receiver package operating on a single fixed frequency between 710 and 865 MHz. The SR 40 nondiversity receiver features a 40 to 20k Hz frequency response, integrated SAW filters, squelch control and front-mount swivel antenna. Two LEDs indicate power and RF status; output connectors are balanced XLR (mic level) and unbalanced ¼-inch (line level). A rackmount kit is optional. The HT 40 transmitter includes a mic element similar to that in the Emotion Series D 880, and like the cigarette pack-sized PT 40 bodypack, the HT 40 operates on two AA batteries for up to 30 hours. Up to three WMS 40 systems may be used simultaneously. Price: $338 to $398, depending on configuration.

Circle 317 on Product Info Card

JBL EVO INTELLIGENT SR SYSTEM

New from JBL Professional (www.jblpro.com), the EVO™ Intelligent Sound Reinforcement System includes "intelligent"-powered speakers, mixer, system controller, wireless and wired mics, and all necessary cables and stands. Each EVOi.324 Intelligent Speaker contains two JBL 14-inch lo/mid speakers and an integrated HF horn, plus three digital power amps and a system electronics package. Up to four EVOi.324 speakers can be networked from a single EVOi.net Speaker System Controller, which optimizes the speakers for room EQ, delay, room and temperature conditions, and even includes feedback control. EVOi.324 speakers are available in white and black and may be purchased separately.

Circle 318 on Product Info Card

INNOVASON SENSORY COMPACT LIVE CONSOLE

Innovason (www.innovason.com) debuts the Sensory Compact Live digital console, with 32 mic and eight line inputs for live sound applications. Featuring the same input components and

control topography as Innovason's Sensory Live and Sensory Large Scale Series consoles, the Compact Live features rear panel XLR inputs, an alternative signal source to the A-to-D stagebox, which connects to the control surface via coaxial cable and may be expanded up to 72 physical inputs. All 32 input channels and 12 Group/Aux buses may be routed to the three master output buses (assignable as LR/mono or LCR). Mix buses may also be routed to 16 matrix output buses, each provided with comprehensive signal processing capabilities. Additional features include snapshot automation (transferable to other Innovason consoles) and a retractable built-in screen and a Hyper-Link™ system for cascading multiple Compact Live consoles.

Circle 319 on Product Info Card

E-V N/DYM UHF WIRELESS

Electro-Voice (www.electrovoice.com) is now shipping the N/DYM® SCU UHF wireless microphone receiver. A half-rackwidth unit provided with rack-mounting hardware, the N/DYM SCU features 10-channel frequency agility, detachable antennae, a 7-segment LED channel display, 4-segment RF signal strength and audio signal indicators, adjustable squelch control, diversity LED indicators and channel-change lock-out. Compatible with all E-V N/DYM transmitters, the unit also features ClearScan™ Auto Channel Select, which simplifies system setup by automatically scanning and selecting the clearest operating channel. Price: $544.

Circle 320 on Product Info Card

SIX NADY SPEAKERS

Nady Systems (www.nadywireless.com) has introduced six new professional loudspeakers for small venue applications. The Thunder Series™ models (THS-1512 and THS-1515) are full-range, two-way speakers with a conical 6x15-inch titanium horn. The THS-1515 features a 15-inch woofer and a 600-watt program power rating; the THS-1512 has a 12-inch woofer and 320W power rating. The ProPower Series™ (PS112, PS115, PTS515 and PFW12) offers similar performance to the Thunder Series but with fewer features and a lower price. All four models are full-range, two-way speakers with 12-inch or 15-inch woofers. The PTS515 and PFW12 have a 400W program power rating; the PS112 has a 300W power rating; and the PS115 has a 500W power rating.

Circle 321 on Product Info Card

RADIAN 2-INCH CD

Radian Audio Engineering (www.radianaudio.com) is now shipping the 950PB compression driver, featuring a 2-inch exit throat and up to 100-watts RMS handling. With a frequency response of 500 to 20k Hz, the 950PB features a 4-inch voice coil for high-power handling and a proprietary Mylar diaphragm suspension with aluminum dome that reduces harmonic distortion. The 950PB weighs 10.4 lbs., comes in 8-, or 16-ohm versions and is priced at $915.

Circle 322 on Product Info Card

NEW TEF SOFTWARE

Gold Line (www.gold-line.com) has released TDS Version 4.0.2 for Windows, a software upgrade for the TEF measurement system. The 32-bit program operates in all Windows-based programs, including Windows 95, 2000 and NT 4.0, but requires either Parallel or HI higher-speed ports; a TEF REV A is upgradeable, whereas a TEF with a GPIB connector is not. New features include the ability to tile or cascade files, allowing up to six measurements to be compared easily by switching tiles; and ETC and Frequency data can be displayed simultaneously. A Linked Cursor function moves the cursor on all tiled displays of the same type, and multiple measurements taken with different parameters may be overlaid on a single plot. A new TEST pulldown menu adds standard test functions (with provision for user-defined tests) and tone, and noise and generator icons add convenience. TDS V. 4.0.2 supports all modern color and B/W printers, and other new features include Pass/Fail Zoom and AutoCalibration functions, and integrated Help. Price of the upgrade for current TDS DOS version owners is $250.

Circle 323 on Product Info Card

TOM WERMAN

HEAVY METAL MASTER

Tom Werman has had a remarkable career by anyone's standards. As a record company A&R executive, he has signed and successfully launched at least half a dozen major acts. As a record producer, he has notched up sales of 40 million-plus, and his two-dozen Gold and Platinum records span the '70s, '80s and '90s.

Starting at Epic in the early '70s, Werman held executive positions at Elektra and EMI/Capitol, interspersed with successful stints as an independent producer. Though his credits range from power pop tunemeisters Cheap Trick to Southern rockers Molly Hatchet and Mother's Finest, from new wavers The Producers and Gary Myrick & The Figures to industry faves Jason & The Scorchers, Werman is best known for his work with an exten-

sive catalog of hard rock and metal groups. Starting with original Motor City madman Ted Nugent, Werman went on to produce key records by Mötley Crüe, Twisted Sister, Poison, L.A. Guns, Krokus, Stryper and Dokken. If not exactly the father of heavy metal, Werman was certainly not far away when it was conceived.

However, as heavy metal and its subgenres have splintered and evolved, Werman has found himself less in demand as a producer. But, if temporarily out of fashion, he is a long way from retirement. Most recently, Werman produced the music for the Warner Bros. Pictures project variously known as *Metal God* and *Rock Star.* Loosely based on a story that first appeared in *The New York Times* in about 1985, the script revolves around the lead singer (played by Mark Wahlberg) of a tribute band who is invited to replace the ousted singer of the band whose music he has been covering.

Tracks were recorded at Conway and the Record Plant in Los Angeles, with most of the overdubbing and mixing completed at Record Plant. Eddie DeLena engineered. "He's great. He's just great," says Werman. "He's focused; he makes it very easy in the studio. He's very sensitive to people's needs, wants and predispositions. And we do a lot of laughing."

Though the sessions may have been laughter-provoking, they were also fast and furious. "We did all the drum tracks, all the bass tracks and some of the rhythm guitar at Conway, all in a week. We steamed," recalls Werman. "All of this had to be very quick. We were still looking for songs and writing songs. So when we got to Record Plant, there were a couple of days when we actually got a song in, ran it down, recorded it and mixed it the next day. We did a couple of songs in 24 hours. We actually recorded 15 tracks. Some will be in the movie, but I think there are basically eight major songs. Parts of the others may be in the movie."

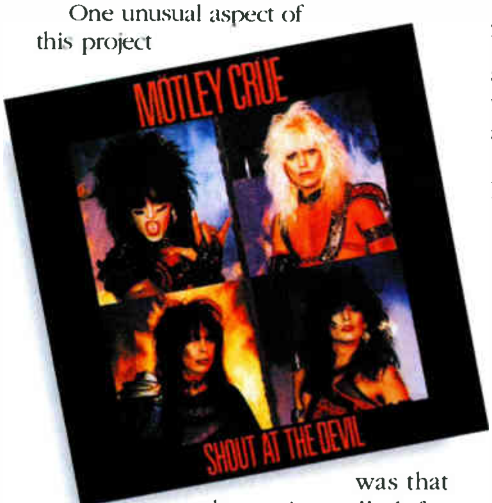
Tracks Werman has produced appear on several soundtracks, but this was the first time he had been involved in creating original music for a film. He recalls that it was

BY CHRIS MICHIE

Gary LeMel, president, worldwide music, Warner Bros. Pictures, who first suggested Werman to the film's music supervisor, Budd Carr.

"We initially spoke a year ago last spring," says Werman. "Since the film is set in 1985, it's not too surprising that they asked me to do it. So we talked about it, we exchanged ideas about the music, I read the script, and then they said, 'Okay, let's go.' I worked very closely with a guy named Bob Schaper, who does all the film mixing and all the magic. We met for breakfast, and he said, 'Just do what you normally do, and I'll do the rest.' And I was relieved to hear that, because I was a little bit concerned about the differences between movie music producing and record producing. Actually, there was no real difference, except that you are a service center, and you have to provide arrangements immediately on demand. So you can finish a song and deliver a song, and they can come back to you one day, in the afternoon, and say, 'We need this version of that song with a double intro, no vocal and an extended guitar solo—by tonight.' So there was no real opportunity to just work on our own schedule. And I think it's probably always that way."

One unusual aspect of this project



was that the script called for some of the same music to be played by two different bands. "That was the tough part," notes Werman. "We had two weeks to rehearse a band on eight songs. And this band was supposed to sound as though it had been together for 15 years. [Laughs.] And I think we did it. Then, on top of that, we had to record many of the same songs with another band, with only one musician in common. They were supposed to be very good, but not quite as good as the band to whom they were paying tribute."

For the "A" band, Carr and Werman recruited drummer Jason Bonham and guitarist Zakk Wylde. "Zakk was with Ozzy for quite a while," says Werman. "And he is unique. I would stake my life on the fact that he's among the five fastest guitar players in the world. His composition is excellent, and he's an astonishing player. His style is good for 1985, but it has a very current flavor to it.

"On bass, I picked Jeff Pilson, because I had worked with him previously in Dokken," he continues. "He's an excellent team player. He's a very focused musician and certainly familiar with the period. And when you get two guys like Zakk and Jason, who are fairly radical rockers, teamwork isn't always the first thing on their minds. So I wanted somebody like Jeff to be an ally and to kind of be the musical director in the band. And for the most part, everything came out as well as I would have liked, or better."

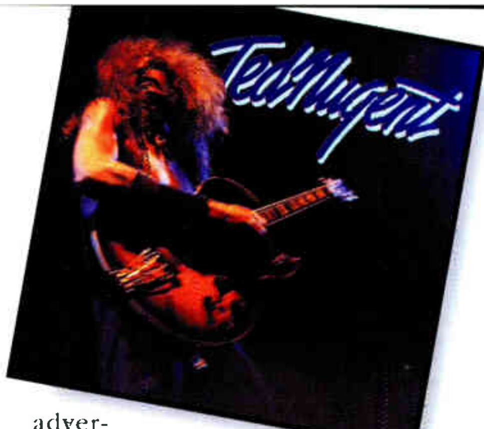
Arrangements were worked out by the musicians and Werman. "We did it as a team," he says. "They detuned drastically so that guitars were in sub-Metallica range. In other words, the bottom two strings, the E and the A strings, were, I think, tuned at least as low as Metallica, if not lower. About 10 years ago, guitar bands started detuning so that they could get a more aggressive, gnarly sound out of the guitar. Very little of that was going on in 1985, so the songs in the film are all in the year 2000 register, with 1985 lyrics and arrangements."

For the "B" band, Werman used Blass Elias, the drummer for Slaughter, and Brian Vander Ark of The Verve Pipe on bass. Nick Catanese, the rhythm guitar player in Zakk Wylde's band (Black Label Society), was the lead guitar player for the tribute band and had the unenviable task of trying to copy Wylde's licks. "And that is not easy," comments Werman. "In fact, it's almost impossible."

Despite the sometimes-frenetic schedule, Werman managed to meet his usual high professional standards. "There were very few compromises," he says. "I really like what we got. I think it's slammin' rock 'n' roll."

Let's go back to the beginning of your career. You played guitar in college, right?

Yes. I had a good collegiate music career. And in 1965, I jammed at Ondine's with a then unknown Jimi Hendrix—twice. I continued to play guitar in business school. And then I went into



advertising and hated it, and bailed after a year and went into rock 'n' roll at CBS Records.

How did that come about?

I simply wrote a letter to Clive Davis. I told him that I had a job, that I was in marketing, that I didn't want to be in marketing, that I knew a lot about rock 'n' roll, that I had an MBA, and that I thought CBS was the kind of sophisticated business organization, for a record company, that could use me. I interviewed with a few people over there, and they hired me to be the assistant to the director of A&R at Epic Records.

When I joined, Epic was billing 12-and-a-half million dollars' worth of records a year. When I left, it was billing, I think, 250 million. So there was explosive growth in that period for Epic. And it was just a fabulous place to work. A fabulous time, a time of incredible growth in rock 'n' roll's popularity.

I started in 1970, and I left in '82. It was just the most wonderful thing. There was no job description. It was a very, very informal and very seat-of-the-pants operation. My initial boss was a complete blues freak. He knew the blues and jazz up and down, back and forth. He knew very little about rock 'n' roll. So I was the rock 'n' roll guy. I was the all-purpose talent scout. I'd go out and I'd see bands and listen to demos, and I would have appointments with people who would come in and play their stuff for me. I immediately signed REO Speedwagon, and that was okay. I brought them three bands that they turned down, which went on to be gargantuan. And after six years, the head of promotion at Columbia was made head of A&R at Epic, and he asked, "What are you doing here? What have you been doing all this time?" I replied, "I've been editing all of the album tracks for single releases, and I've been trying to sign bands and being turned down. I guess I just didn't pound the table hard enough for these bands—I was turned down on Kiss and Lynyrd Skynyrd and Rush." And he said, "Is there anybody you like now?" And I said, "Yeah, I like

Ted Nugent," and he allowed me to sign Ted. Ted was on his own for the first time without the Amboy Dukes, and his manager was shopping him. I wasn't very excited until I saw him, and then I said, "Wow. We can make records with this guy." I kind of appointed myself as the associate producer—the manager had a production deal with Ted, so he called all the shots. But because I was with the label and could ask them for more money, I was able to remix the record after the manager mixed it. I remixed it with the engineer, and it went Platinum. So, bang! I was a producer.

That was the first record you had produced, right? Did you go in with confidence, or were you quaking in your boots?

No, I was rarin' to go. Because I had been listening to substandard records for six years and thinking, "Gee, I could do better than this." And I didn't. I had a lot to learn, and I did make some bad records to start with. But Ted's record was the first and that was closely followed by Cheap Trick. I signed them to Epic.

But the first album was produced by Jack Douglas.

Right. He was the one who called me and asked me to go see them. He had already started working with them, so he did the first album, which was great, but not terribly commercial. And then he was busy with Aerosmith, and they asked me to do the second. I did it and it was more commercial. My job at that time was to simply get bands on the radio. So that's what I did. In other words, anything in the song that was vaguely commercial, I had to bring out. And I did.

Apart from Ted, that was the first band that you had worked with seriously in the studio.

Cheap Trick? Yeah. It was all mine. I didn't have any co-producers. I was allowed to do whatever I wanted, which was the most fun I ever had, basically, in the studio. We recorded at the Third Street Record Plant in L.A.—the old one, the fabled, the famous one. The band told me they wanted to record in L.A., and I'd never been there to record. I came out, and we did the first part at Sound City and mixed at the Record Plant. And that was *In Color*.

Then, in 1978, I moved from New York to Los Angeles. I came out here, and it was completely different. In New York, the entertainment business is just one of many. Here, it's all there is. The entire focus is on the entertainment industry, and there are so many services that make it so easy to work here. So many rental companies, so many sound reinforcement companies, so many piano tuners, so many guitar stores, so many studios, so many engineers, so many musicians. It was just heaven. I came out here, and I moved into the Record Plant, and I did 16 records there. I hardly surfaced.

Tom Werman with Cheap Trick

So here you are on staff at Epic, and you've got a monster hit with Ted Nugent and you've got Cheap Trick going. Did this change your profile in the company?

Oh, yes. I started making a decent wage, although very little compared to what they make today. Then I signed Molly Hatchet and continued with them. So I had quite a few hits going on at the same time. Actually, three different acts in the Top 40 album chart at one time.

I had a couple of bands that I signed that, unfortunately, for one reason or another, were not successful. I still think they're great, and it's really too bad that they didn't get the treatment they deserved. And I also made some stiff records, no question about that. Not bad, just not good enough. Every producer can't bat a thousand. From a statistical point of view, I've probably batted about .350. More than one out of every three albums I did went Gold or Platinum. I did about 65 albums, and I have 25 Gold or Platinum records, including Best Ofs, at the moment.

I know that you have a band in the arrangements, and obviously the song selection and the radio edits and bringing out hooks and so on, but you're also listed all over the place as adding percussion.

Right. I do play all sorts of hand percussion. Everything but congas. It's just a little flavor, a little touch. And I always enjoyed it. I always made the same offer to every band: "I'll do it for free. You listen to it and, if you don't like it, we can hire somebody." And they always liked it. I did it in a hurry, I didn't spend any money, I didn't get paid. I stopped taking credit after a while. It's just another part of the arranging function, really.

And you'd typically bring in a keyboard player at some point?

Yeah, if there wasn't one. I used two main keyboard players, Jai Winding in the early days and then Paul Fox. I also enjoyed working with Michael Boddicker.

Tell me about your recording methods. Did you ever change your process?

I didn't change procedure that much, but what was recorded and how it was recorded was different according to each band. I didn't just say, "I always use a 47 for vocals and an 87 for...whatever." I don't do that. Each recording project is tailor-made for the band and the band's sound. We use different studios, but the procedure always remains basically the same, which is get the basic track done, get the drums and bass done, and put on the rhythm guitars and then put on the lead, the vocals, see what happens. Color it, arrange it.

—CONTINUED ON PAGE 170

Carbon 12 Studios

STEVE TUSHAR'S MIX OF MUSIC AND AUDIO POST

BY GARY ESKOW

Making it as a freelance sound designer/mixer in L.A. is no layup. The competition is fierce, and often post work will have to be supplemented with music projects—a fact not lost on Steve Tushar, whose remix of the Korn single "Somebody, Someone" has just been released in Europe. "I'm real happy with that mix," says Tushar. "Some bigger names had a crack at the song, but the label's going with my work, which will be released in the States later in the year."

Tushar works out of Carbon 12 Studios, his Hollywood space, splitting his time between record projects—he's now producing Kevin Moore, formerly of the band Dream Theater—and audio post. Last season he regularly logged up to 70 hours a week creating sound effects for the Pamela Anderson show *V.I.P.* "That was a tough schedule," he says. "I used Pro Tools exclusively on the show, running on a G4, but I've since switched over to Nuendo, Steinberg's native DAW."

Originally released in the mid-'90s on the SGI platform, Nuendo failed to generate significant interest among audio post professionals, who were flocking en masse to Pro Tools. The power of the current breed of computers has finally made native processing a viable alternative to chip-based DAWs, and Tushar has become a Nuendo enthusiast.

"Without a doubt, Digidesign set the standard as far as computer-based workstations go," Tushar says. "But Nuendo has some clear advantages. For one thing, although it relies on the computer for all its functionality, Nuendo—running on my 700MHz Pentium PC—is much faster than Pro Tools in every phase of operation, including redraws. I mixed the Korn track entirely within the system, and it sounds very warm."

Knowing that Nuendo would be retooled for a post-millennium re-release, Steinberg's programmers had the advantage of seeing that surround sound mixing was no fad, and the company is heavily touting Nuendo's capabilities in this area. "The surround mixing is superb," says Tushar. "You can enable the surround bus for any channel at any time, grab that channel, and use the onscreen joystick to place the sound anywhere in the matrix. I really like the fact that you can either pick a 5.1, 6.1 or 7.1 default speaker setup, or [you can] draw in the speakers as they're laid out in your room. If, for example, your room doesn't let you put the rears where they should be, you can draw their placement onscreen—the program actually has icons of speakers. I told Nuendo where my speakers were and found that the results were surprisingly accurate.

"Surround sound mixing wasn't an entrenched part of the process when Pro Tools was released," he continues, "and so it became an add on to the system. If you want to mix in surround, you have to enable plug-ins on each channel, which is a heavy price to pay. In Nuendo you simply switch to the surround bus. Also, let's say you're executing a 5.1 mix. When you're done—or think you're done—you can mix down to one 6-channel interleaved track, where you can view all six waveforms independently. You could also leave them as six separate tracks, which you'd do if you wanted to lay the mix, track by track, to a DA-88, for example."

Nuendo 1.5 offers VST instrument integration, as well as MIDI functionality. Tushar has been intrigued by many of the VST instruments he's used recently and may be shedding some old hardware as a result. "I wouldn't mind buying another computer, simply to run a bunch of VST instruments, especially since the software will allow me to sell some of my old devices. The Mercury-1, by TC Works, is a favorite at this time. It's got some great bass sounds, and the built-in distortion is excellent. It takes you into that Nord area. The Waldorf PG Wave 2.V is a fantastic replication of the original synth, and the Prophet 52 also sounds great."

The main console is a Spirit Digital 328 with a number of modifications: British EQ, a pair of TDIF I/Os, a pair of ADAT I/Os, 16 line inserts and built-in SMPTE. Tushar monitors through Yamaha MSP-10s with a matching subwoofer.

In his spare time, Tushar writes and records electronica music with his band, Glitch. "We're recording our album at this time, and it will be tracked and mixed entirely within Nuendo." ■

Gary Eskow is a contributing editor to Mix *magazine.*

Nathaniel Kunkel is one of today's hottest producer/engineers. It's his job to know good mics when he hears them. A few of Nathaniel's credits include: Robert Altman, Billy Joel, Little Feat, Lyle Lovett, Graham Nash, Aaron Neville, James Taylor, Anna Vissi, and dozens more.

Legendary studio drummer Russ Kunkel is Nathaniel's father. Russ's credits span four decades of illustrious session work for artists that range from A-to-Z – Herb Alpert to Warren Zevon. And as a successful producer of Jimmy Buffet, Graham Nash and many others, Russ also knows a good mic when he hears one.

"A-T's 40 Series are my 'go to' mics, without hesitation," says Nathaniel. "When I first heard the AT4050, I was very impressed. It was transparent, open and predictable. It is a stunning acoustic instrument microphone with the ability to handle extreme dynamic range very musically."

Take a professional tip from the Kunkels and hear what a 40 Series mic can do for your next session. Who knows? You might just be making musical history – like they have.

Special thanks to O'Henry Sound Studios, Burbank, CA

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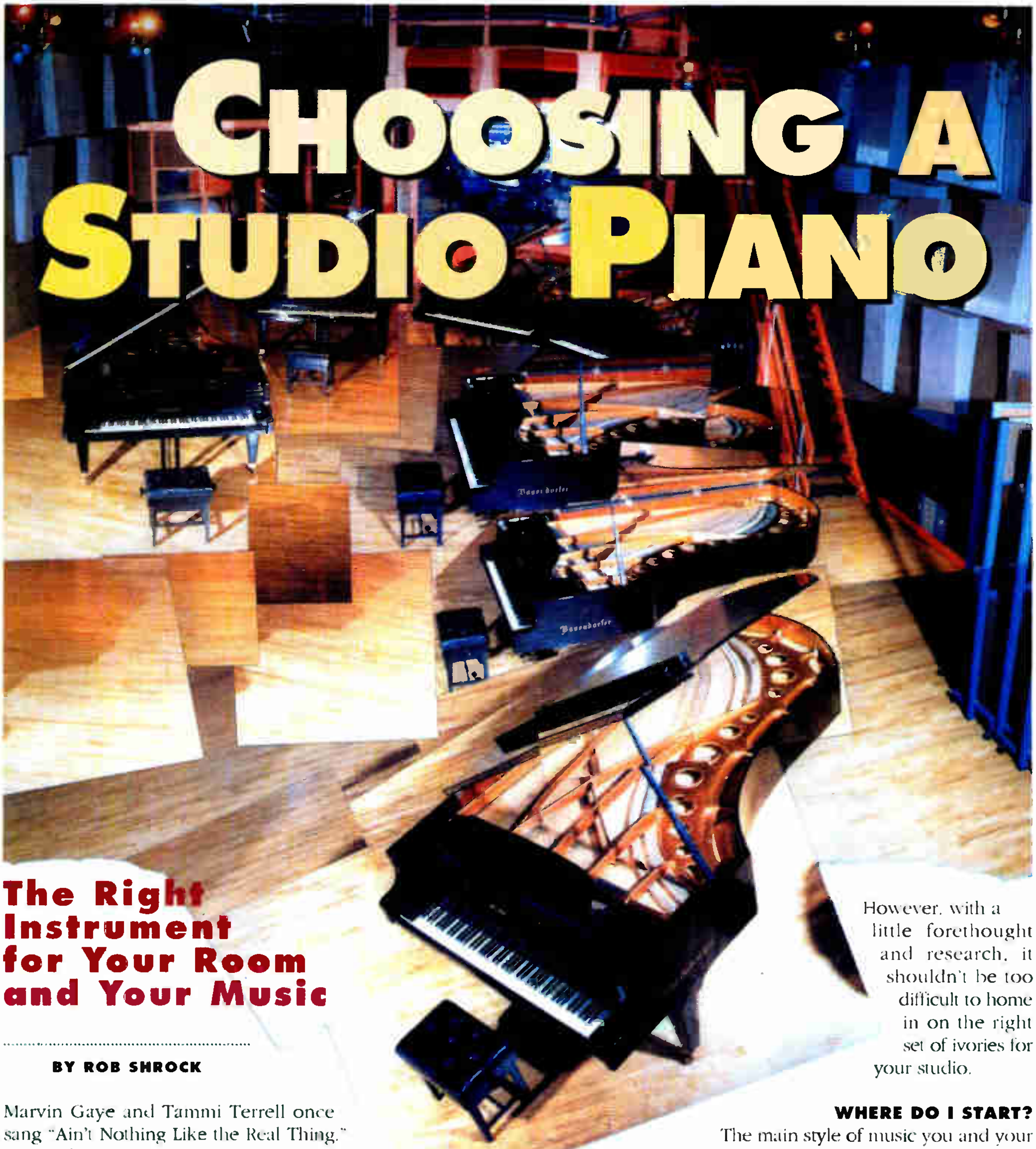
Choosing a Studio Piano



PHOTOMONTAGE COURTESY BILL PRICE/WESSEX STUDIOS

The Right Instrument for Your Room and Your Music

However, with a little forethought and research, it shouldn't be too difficult to home in on the right set of ivories for your studio.

BY ROB SHROCK

Marvin Gaye and Tammi Terrell once sang "Ain't Nothing Like the Real Thing." Granted, they weren't singing about pianos, but to many musicians, producers and engineers, that sentiment sums it up nicely. This article assumes you've already weighed the arguments for and against purchasing a bona fide, honest-to-God, real, acoustic piano vs. relying on a digital piano or library of piano samples. You've concluded that the expressiveness, uniqueness and aesthetics of a real piano outweigh the ongoing expenses of maintenance, tuning, and the additional investment in complementary microphones and preamps. Now it's time to find the instrument that's appropriate for your studio setting.

As with practically all acoustic instruments, every piano is unique. You'll have to accept the fact that it's highly unlikely that any one piano will be ideal for all musical situations.

WHERE DO I START?

The main style of music you and your clients record weighs in heavily on the type of piano best suited for your studio. Do you record mostly large orchestral ensembles and classical soloists? Are most of your sessions geared toward commercials, jingles or pop music? Do you deal mainly with rock music and/or creative artists who are more interested in recording interesting and unique piano sounds?

There are additional factors to consider, including the size of the physical space and the acoustic nature of the room surrounding the piano. Are you seeking a piano for upscale clients in a commercial facility with a large recording room? Will the piano be recorded in the main room or primarily in its own isolated piano lock? Are you in a project facility that can only accommodate an upright instrument in the corner of a small room? Is your recording space ambient or dead?

PHOTO: MUFFY KIBBEY

Unless you are an accomplished pianist, you are best advised to have one or two discerning pianists you trust help pick the actual instrument.

Clear answers to these questions will help you narrow down the multitude of options and choose the right instrument. The piano markets—both new and used—are nearly as overwhelming as the world of new and used guitars. Of course, your budget will further dictate your options; however, there are a lot of deals available for those open to financing or leasing an expensive instrument. We'll also look at the used piano market, where there are great bargains to be found with a little research and legwork.

UPTOWN

If your recording facility specializes in film scores, orchestral recording, and traditional or classical recording, then you're a prime candidate for a large grand piano. Assuming your recording space is large enough to accommodate large ensembles, you're looking at least at a 7-foot grand piano to provide the depth of tone required for orchestral and solo piano recordings, and an instrument as long as nine feet is not out of the question (although the largest pianos are more commonly found in performance halls). Pianos by Steinway, Baldwin and Bösendorfer easily come to mind, and odds are that you and your clientele won't be disappointed with the overall quality these brands offer. Pianos by Yamaha, Kawai, Fazioli and—my personal favorite—Feurich are also making more appearances in large U.S. recording facilities that specialize in traditional and classical styles, and they are gaining acceptance by both concert performers and classical engineers.

It must be noted that even two pianos that are identical in manufacturer and model can still sound very different from each other due to a range of factors, including setup and voicing, age, maintenance and even geographical location. I've found that Baldwin pianos, in particular, vary widely in timbre, action and actual volume output between exact models very close in age. Unless you are an accomplished pianist on the same level as the musicians who will be recording at your facility, you are best advised to have one or two discerning pianists you trust help pick the actual instrument before you make your investment.

A grand piano suited for traditional solo piano recording—which is usually acceptable for ensemble orchestral and some jazz recordings, too—should be rich in overtones, though not overly bright in timbre and crisp in action (as we have grown accustomed to hearing in pop, modern jazz and rock music). The voicing and action should be well-balanced from the low to high range of notes. You will also need to secure the services of a piano technician experienced in setting up pianos for classical performance to ensure the instrument responds appropriately to traditional repertoire.

More importantly, this piano should be free of any undesirable sonic anomalies that would be audible during very quiet solo passages. Don't underestimate the importance of this last point. I've come across several pianos that were otherwise fine solo instruments but emitted subtle thuds, creaks or odd mechanical resonances that even the best technicians could not eliminate. These unfortunate blemishes prevented the use of these instruments for any truly great solo piano recording. Even though traditional and classical recording techniques usually place the microphones several feet away from the piano, be sure that any mechanical imperfections (which practically all pianos possess) are not audible when standing six feet from the instrument.

Most likely, a top-of-the-line grand piano intended for classical and traditional recording will cost you the most (easily upwards of $40,000 to $100,000) but will not be the most versatile choice for other styles of music. Because classical pianos tend to sound "darker" than pianos used for pop and rock styles, it's hard to get a classical piano to "speak" enough for other genres. Even with EQ and compression, a piano designed for traditional music forms seldom creates satisfying tracks for rock and pop. In many upscale recording studios that need to host both types of sessions, the facility has two pianos. (One of the studios I worked in a lot during the early '90s had both a 9-foot Bösendorfer and a 7-foot, 4-inch Yamaha.)

One last point on choosing a piano for traditional recording: Be sure to choose a piano that has all three pedals. Some smaller grand pianos, like the Yamaha G Series, don't include the middle *sostenuto* pedal that is required for much Romantic and Impressionistic repertoire and is a necessity for many jazz pianists.

MIDTOWN

Most commercial facilities have to accommodate a variety of sessions from day to day and don't have a budget to invest tens of thousands of dollars in either a single piano that is not versatile or in the purchase and maintenance of multiple pianos. If your studio serves commercial, pop and rock sessions, then a piano with a brighter tone and

faster action is desirable. Both new and used piano markets are full of instruments well-suited to the task.

By far, the most popular piano that is versatile enough for pop, jazz and rock recording, yet is arguably expressive enough for successful traditional and classical recording, is the Yamaha C Series grand piano. The 7-foot, 4-inch model is my favorite pop piano, with a bright tone, strong overtones and a moderately strong bass. I come across a lot of these pianos on the road, too, and find them to be very consistent. The Yamaha C7 is seemingly less affected by age and geographic location than other instruments; a well-maintained Yamaha C7 almost always produces satisfying recording and performing results.

That's not to say that more traditional brands like Baldwin and Steinway don't make pianos capable of pop and rock styles; it just requires more of a search to find one that sounds right. (In fact, certain successful rock and pop artists either require or travel with their own Steinway or Baldwin as part of their sound.) Oftentimes, adjustments to the hardness of the hammers and other modifications are necessary to achieve the desired timbre. Another point to consider is that several Steinway models have a shorter than average depth of key throw, making the action feel shallow compared to other pianos; this feature could be uncomfortable to many players.

In addition to Yamaha, there are several other brands of pianos that would make an excellent choice for a midsized studio, especially for those with a modest budget. Kawai, Feurich and Young Chang all produce models in various sizes that record well. Again, it is necessary to ensure that there are no audible mechanical anomalies in your piano that will compromise your ability to make fine recordings.

For studios with a varied clientele, I suggest choosing a grand piano no smaller than six feet, four inches long in order to guarantee solid tone with enough sustain to sound good in a lot of styles. Although grand pianos smaller than six-feet-long are great for living rooms and parlors, they generally sound even smaller than they actually are when recorded. I own a 6-foot Yamaha G Series grand that sounds pleasing enough for composing and recreation, but it simply doesn't have enough tone and sustain to make great pop or rock recordings.

DOWNTOWN

Don't think you're stuck with playing piano samples if you don't have a studio or wallet large enough to own a big grand piano. There's no rule that says you have to make that type of investment, especially if you own a small commercial or project studio. Depending on your clientele or personal style of music, an affordable upright or smaller spinet piano may provide the right sound for your recordings, especially for a lot of rock and alternative styles.

John Lennon's "Imagine" and most of Jerry Lee Lewis' early hits were recorded on upright pianos. Artists such as Ben Folds Five, Fiona Apple and Rufus Wainwright often employ baby grand pianos and uprights to create more of a sonically stylized sound, often using drastic EQ, compression and other effects.

If your sessions are more about style and personality than purity, then you're really in luck. There are a ton of great small grand and upright pianos available that cost no more (sometimes less) than the average electronic keyboard. Most of the popular manufacturers of pristine, large grand pianos also make excellent uprights and baby grands. In addition to upping the cool factor of your studio, most upright pianos don't take up very much space, and you get a composing tool that doesn't need speakers, a computer or even electricity. If nothing else, just writing a song sitting at a real piano with your partner strumming an acoustic guitar is way more fun than auditioning loops on a CD or scrolling through presets on a synth module.

Don't assume that, just because you're "downgrading" to a smaller and less expensive piano, the resulting sound can't be great. Uprights are actually easier to record than grand pianos and usually sound better than small baby grands. However, quality varies greatly in this level of piano, and you are still obligated to perform routine maintenance chores like tuning, voicing and minimizing any audible mechanical noises.

BEFORE YOU BUY

There are a few additional points to consider before finally choosing a piano. If the piano will stay in its own isolated room, or piano lock, then care must be taken not to overload the room with a piano that is too large and/or too loud for the size of the space. Just as speakers and amps can be overloaded, a room can be overloaded by SPL to the point that a clear recording is impossible, especially if the isolation chamber is excessively live in nature.

The overtones of the piano will also interact with the ambience of the room size. This interaction will affect the overall timbre of the piano and is best left to be adjusted by a qualified technician in the room where the piano resides. However, if your recording room is fairly dead, then be especially sensitive to the overall tonality of the piano. A dark-sounding piano in an acoustically dead recording space is a recipe for lackluster, wimpy piano recordings. By contrast, an overly bright piano in an extremely live room will sound very smeared and unfocused.

Once you've secured the appropriate piano for your studio, it will most likely need to be caressed into sounding its best for your particular environment. First, the instrument will need several days—possibly a couple of weeks—to settle into its new environment after its move. Changes in aver-

—*CONTINUED ON PAGE 168*

RESOURCES

In addition to piano retailers in your local area, there are a lot of resources available to those interested in buying and selling new and used pianos. All of the major manufacturers have excellent and informative Web sites. There are also several Web sites dedicated to connecting potential buyers and sellers.

MANUFACTURERS

Fazioli: fazioli.com
Feurich: feurich.com
Kawai: kawaius.com
Steinway & Sons: steinway.com
Yamaha Pianos: yamaha.com/pianos.htm
Young Chang America: youngchang.com/yca/home.html

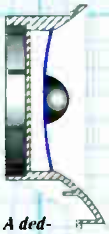
OTHER WEB RESOURCES

The Piano Page: ptg.org
Piano World: pianoworld.com
Piano Finders: pianofinders.com
PianoMart: pianomart.com
PianoNetwork: pianonetwork.com
PianoOutpost: pianooutpost.com

It *looks* like a regular system. It *sounds* like amplified reality.

The technology behind the SR1530 Active 3-Way speaker system and SR-VLZ Pro Series mixers.

Until the SR1530, there's never been a three-way active processed sound reinforcement speaker in this compact, fit-in-a-van size. In fact, as far as we know, there's never been another compact, active three-way speaker *period.*



A dedicated horn-loaded 6-inch transducer refines SR1530 midrange.

And until the SR24•4-VLZ® PRO and 32•4-VLZ® PRO, there have never been compact SR mixers with mic preamplifiers this good.

Put them together and you have a new level of sound system accuracy.

If you're the non-verbal, gotta-hear-it-to-believe-it type, audition the SR-VLZ-PRO/SR1530 combo at your nearest Mackie dealer. We think you'll agree that it definitely walks the walk.

If you're more of the tech type, read on while we talk up why this system re-defines how good a medium-sized PA can sound.

It's a three-way, not a two-way system.

Midrange is where fundamental vocal and instrumental frequencies are. A two-way system has to compromise by reproducing this range through a too-big low-frequency transducer and a too-small high frequency transducer.

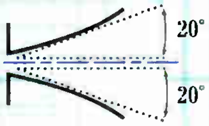
Our three-way system has a specialized transducer with separate horn that's sized perfectly for handling vocals and instruments.

When you listen to the SR1530, you hear the difference right away. Midrange sounds more accurate and natural especially at really high SPLs.

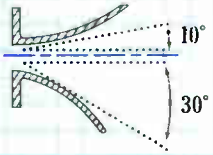
It's internally tri-amplified.

For clear definition, tight transient response and hard-hitting bass, nothing beats application-specific amplifiers for each transducer. Trouble is, until now that's required an outboard electronic crossover, three separate power amps and a lot of speaker cable.

The SR1530 has two 100-watt and one 300-watt internal amps, each optimized for an individual transducer. Moreover, we've built in electronic parametric equalization and time correction for accurate frequency response and phase alignment.



Symmetrical horn

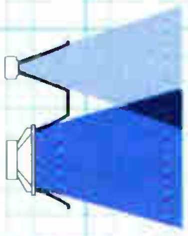


Asymmetrical Optimized Wavefront™ horn

Optimized Wavefront Horn.

Generating accurate high and midrange frequencies is only part of the story. They must be delivered to your audience in the right amounts and in the same place at the same time.

Instead of using separate horns, we created a one-piece 90°x 40° Optimized Wavefront system that combines high and midrange horns. Unlike conventional horns, which are symmetrical, the high frequency section of the SR1530's horn is shaped so that more of its output is directed down into the 6-inch midrange's dispersion pattern. This creates a focused, single wavefront with excellent phase and power response characteristics. Your whole audience gets even frequency balance, instead of lots of midrange up front and too much high end in the back.



Conventional horns fire straight out. The quality of sound depends on where you're sitting in the audience.

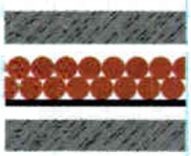
Optimized Wavefront™ system mixes high and midrange output for better detail and vocal intelligibility everywhere.

Inside/Outside voice coil.

Because we make our own transducers, we've been able to add technology that gives the SR1530's low frequency transducer (woofer) added thermal protection.

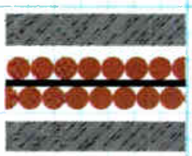
Conventional voice coils have two

layers of wire wound onto the outside of a voice coil former. When heat builds up, the wire and former expand at different rates, which can lead to catastrophic failure.



Conventional voice coil (Burnus Outus Potentialia)

The SR1530's LF transducer voice coil has one layer of wire on the outside and another on the inside. Both make full contact with the air, transferring heat to both surfaces of the surrounding magnet structure. Maximum cooling takes place…and the voice coil former is forced to expand at the same rate as the wire, relieving stress and preventing separation.



Inside/Outside voice coil (Coolus Cucumberous)

XDR

XDR™, the best mic preamps ever offered on a compact SR mixer.

With a speaker system as accurate as the SR1530, you can take maximum advantage of the premium, eXtended Dynamic Range mic preamps on our new SR24•4-VLZ PRO and SR32•4-VLZ PRO sound reinforcement mixers.

We spent over two years creating the first mixer mic preamp with specs that meet or beat those of mega-expensive esoteric outboard microphone preamplifiers. For example 0.0007% Total Harmonic Distortion, and frequency bandwidth from 5Hz to 100kHz ±1dB. At real-world gain settings (0 to +30dB), XDR preamps are capable of extracting more detail and better sound out of any microphone — including the dynamic type often used on stage.

And, as the name implies, XDR mic preamps have extended dynamic range to handle screaming vocalists or close-miked drum kits. Plus a full 60dB of gain to boost weak inputs without adding extra noise.

Even more live sound benefits.

Many conventional mic preamps deliver a different frequency response when presented with a 50-ohm load than with a 600-ohm load. Our XDR design is *impedance-independent*. In basic terms, this means that no matter what the combined impedance load of the mic and cable are, you'll get the same frequency response at the mixer. And, XDR mic preamps have the best RFI (radio frequency interference) rejection of any compact mixer brand. So you won't pick up stray signals or background noise when you use long cable runs.

Is your band worth it?

We'll be honest. The SR1530/SR-VLZ PRO speaker/mixer combo isn't the lowest-priced SR system you'll find in a music store (although it's far less money than a comparable tri-amplified passive system). We designed the SR1530 and SR-VLZ PRO Series mixers for musicians who want to sound their best at up to 126dB SPL. If you fit in this category, call toll-free or visit our web site for more info. Or better yet, visit a Mackie dealer and hear amplified reality.

32•4 VLZ-PRO NOW SHIPPING!

The SR1530 Active 3-Way:

■ Linear response from 45Hz to 18kHz ■ Total of 500 actual watts RMS delivered to internal transducers ■ Electronic EQ for flat frequency response ■ Time correction for accurate phase alignment ■ Optimized Wave-front™ high/mid horn system ■ RCF Precision™ transducers: 15-inch LF transducer with Inside/Outside heat-resistant voice coil and high-flux magnetic circuit ■ 6-inch horn-loaded midrange ■ 1-inch exit HF compression driver ■ Trapezoidal Baltic Birch plywood enclosure with rugged molded resin endcaps ■ Polymer-coated steel grille ■ Weight-balanced with side and top handles for easy carry and set-up ■

SR24•4-VLZ PRO & SR32•4-VLZ PRO:

■ 24•4: 20 mono channels, 2 stereo channels, 20 XDR mic preamps ■ 32•4: 28 mono channels, 2 stereo channels, 28 XDR mic preamps ■ Both: 4-bus design ■ New high-performance 2086 op-amps ■ 6 aux sends per channel ■ 3-band EQ w/swept mid & low cut filters on mono channels ■ 4-band fixed EQ on stereo channels ■ Mute, solo, signal present & OL LEDs on every channel ■ 4 stereo aux returns with EFX to Monitor feature ■ Separate Talk-back section with extra mic preamp ■ Double-bussed sub outs for easy studio multi-tracking ■

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NEW PRODUCTS

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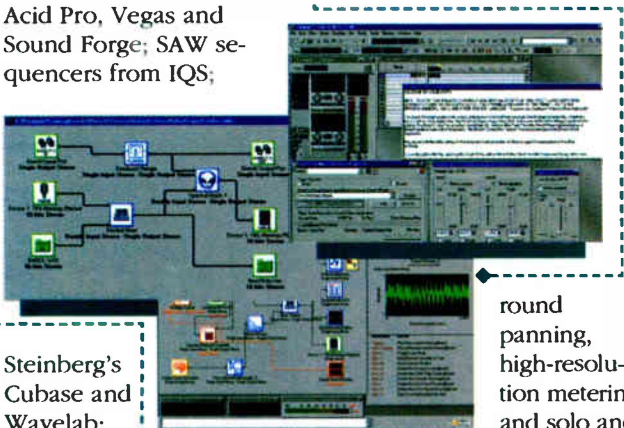
SOUNDS LOGICAL WAVEWARP 2.0

Sounds Logical (www.soundslogical.com) is now shipping WaveWarp 2.0 with full DirectX functionality. Now, in addition to using WaveWarp as a stand-alone effects processor, users can apply WaveWarp's modular architecture to create custom Direct X plug-ins for popular PC-based audio editors and sequencers, such as Cakewalk Pro Audio 9.0, Emagic Logic Audio, SEK'D Samplitude, Sonic Foundry Acid Pro, Vegas and Sound Forge; SAW sequencers from IQS;



Steinberg's Cubase and Wavelab; and Minnetonka MxTrax/Mx51. In addition to DirectX support, Version 2.0 includes two spectral morphing components, a digital filter component for zero-latency FIR filtering and an ADSR envelope generator with customizable envelope profiles. WaveWarp offers more than 260 modular audio effect components for $199; a scaled-back version is $99.

Circle 338 on Product Info Card

IQS SAWSTUDIO

Innovative Quality Software (www.iqsoft.com) introduces SAWStudio, the fifth generation of the SAW product line, featuring enhanced performance speed and a redesigned multitrack recording/editing interface that emulates a typical studio environment. The SAW Studio mixer has 72 stereo ins and 24 stereo outs, with phase reverse, swap L/R, 5-band stereo EQ, keying gate and compressor, surround panning, high-resolution metering, and solo and mute on each channel. Each stereo track is eight layers deep; the top layer is active in each track.

Effects can be plugged in and out of tracks in real time, and IQS' SoftEdge technology simplifies performing crossfades. A new SAWStudio API has been developed for effects plug-ins.

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NEW ADDITIONS TO THE SOUNDSCAPE R.ED FAMILY

Soundscape (www.soundscape-digital.com) announced two new R.Ed Recorder/Editor models specifically geared toward studio recording and radio, mastering and A/V markets at more affordable prices.

The two new models, R.Ed/16 and R.Ed/24, are derivatives of the full-blown R.Ed system (which offers 32 to 128 tracks) and comes with Soundscape's V. 3.0 Editor software. R.Ed/24 offers 24 tracks, 24-bit at 48 kHz (or 12 tracks at 24-bit, 96 kHz), with 24 digital I/Os via three TDIF ports. The unit has one removable hard drive bay, plus one position for an internal hard drive for up to 274 GB of audio storage and sync. An OMF Import/Export option allows project transfer between R.Ed and Pro Tools, SADiE, Fairlight and others. R.Ed/16 ($4,250) has 16 tracks, 24-bit at 48 kHz or 8 tracks at 24-bit at 96 kHz, with up to 12 inputs and 16 outputs via 2-in/4-out AES/EBU digital, plus eight digital in/out channels via one TDIF port. Options include the R.Ed analog interface board, offering 2-in/4-out balanced XLRs and 24-bit, 96kHz AD/DA converters. Like the full 32 R.Ed, R.Ed/24 and R.Ed/16 offer real-time DSP-based mixing and effects, full dynamic mix automation, compatibility with many hardware controllers and integration with video editing systems; many add-ons are available.

Circle 340 on Product Info Card

NEW BOMB FACTORY PLUG-INS

Bomb Factory (www.bombfactory.com) has introduced a host of new plug-ins. The

Fairchild 660 is a replica of the $35,000 compressor, the Pultec EQP-1A models analog EQ; the Joemeek SC2 Photo-Optical Compressor and VC5 Meequalizer re-create legendary producer Joe Meek's processors, and the Tel-Ray Variable Delay creates analog-sounding delay and echo. Three moogerfooger processors were introduced: the moogerfooger 12-Stage Phaser, a six- or 12-stage phaser; the moogerfooger Analog Delay; and the moogerfooger Bundle, which includes four moogerfoogers (including the previously released moogerfooger Lowpass Filter & Ring Modulator) at one low price. (Special pricing is available for registered owners of the moogerfooger Lowpass Filter & Ring Modulator bundle.) Also new from Bomb Factory: BF Essential is a suite of time-saving, trouble-solving studio essentials, all priced under $100, including a tuner, meter bridge and correlation meter. MAS, TDM, RTAS and AudioSuite are all supported.

Circle 341 on Product Info Card

AUDIO EASE NAUTILUS

Audio Ease (www.audioease.com) introduces its second pack of plug-ins exclusively for MOTU audio system. The Nautilus Bundle includes three modules: RiverRun (by the creator of the Thonk freeware application) uses granular synthesis and lets you scatter up to 1,000 streams of tiny samples across your stereo panorama to build big effects, PeriScope is a phase-correct equalizer and real-time spectroscope, and Deep Phase Nine is a true phaser that offers up to 24 notches per channel, beat locked sample and hold, and more. The bundle retails for $299 and is distributed in the U.S. by Mark of the Unicorn (www.motu.com).

Circle 342 on Product Info Card

UPGRADES AND UPDATES

The **PSP** (www.psp-audioware.com) **MixSaturator plug-in emulates analog saturation** with three algorithms: an analog saturation simulation algorithm, which enables the user to choose one of the seven curves of nonlinearity characteristic of valve devices, analog tape and digital clipping; the bass frequencies processing algorithm; and the treble frequencies processing algorithm...**Mark of the Unicorn** (www.motu.com) announced that **Mezzo Version 3.7**, the project backup system from **Grey Matter Response Inc.** (www.mezzogmr.com), **now supports Digital Performer 2.71...Syntrillium Software released "A Short Course in Digital Audio Processing,"** a free, animated tutorial authored in Director by Macromedia. The tutorial covers fundamental digital audio concepts, such as waveforms, sampling, bit depth and digital signal flow. A free download is at http://school.syntrillium.com/tutor/shortcourse.htm ...**Emagic** (www.emagic.de) released **Version 3.0 of its SoundDiver universal patch librarian**. New features include a redesigned interface created to be similar to Logic Audio 4.0, and a context-sensitive interactive help system...**Digidesign** has **licensed POW-r Consortium's POW-r** technology **for digital audio word-length reduction** for introduction into Pro Tools. In other Digidesign news, **a free version of Pro Tools**, Pro Tools FREE 5.01, is available. Visit www.digidesign.com for information...**Glyph's X-Project with FireWire**, with 15GB capacity, is now shipping. Glyph also introduced WildFire, a high-speed Fire Wire, tabletop CD-rewritable for audio and video files. Visit www.glyph.com... **Noren Products Inc.** (www.norenproducts.com) introduces the **NoiseLock, a fan-free isolation box** that cools using Noren's Heat-Pipe technology...**Studio Network Solutions** announced **an end-to-end storage area network**, A/VSAN, developed for audio, post-production and multimedia. Check out www.studionetworksolutions.com...**Radikal Technologies' SAC 2K software assigned controller is now shipping**. For more information, see www.radltec.de...**Swissonic's AD8 is an 8-channel, 24/96, A/D converter** with built-in mic pre's. Also new from Swissonic, the **WD8 word-clock generator/distributor.** Visit www.swissonic.com... **Medéa's AudioRaid Ultra2 SCSI disk arrays are designed for professional audio applications.** Check them out at www.medeacorp.com.■

EVEN DIGITAL RECORDERS GET THE BLUES

MDM AND DAT MAINTENANCE

Welcome to the new millennium! No Prince songs this year, but by now you must surely be tired of the 2001 theme, Richard Strauss' "Also Sprach Zarathustra."

In my first column for *Mix* last April, I predicted that no new DAT models would see the light of production. I was nearly correct, although Fostex proved me wrong by updating its PD-4 field recorder to Version 2, and, in an unprecedented move, recent Fostex ad copy includes the name of Rick Cannata, its primary caregiver—a positive note for location recordists. Rumors are circulating that Panasonic, HHB and Tascam are getting out of the DAT machine biz. That leaves Pioneer, Sony and Fostex. So, now more than ever, keeping your current DAT healthy is good advice.

Let's examine some basics of digital tape recorder maintenance, both for DATs and modular digital multitracks (MDMs).

A ROAR!

When they are in good condition, digital tape recorders have plenty of error-correcting headroom. Deterioration is gradual, so before your deck starts misbehaving in an obvious way, you should know that it may already be making tapes that are harder to play. The best preventive maintenance is to check the error rate often. (See the "Decoding Error Rate" sidebar for more details.) With MDMs, the error rate can be noted after formatting, and for DTRS models, I recommend formatting with all tracks in Record mode. You have to know what's normal to avoid surprises.

That occasional icky fuzz or mysterious error message could just be a head clog. Don't be afraid to use a cleaning tape when this happens or, better yet, learn how to clean the heads manually. There are many "head access" pictures on my Web site (www.tangibletechnology.com); early DAT transports are the least accessible, and all ADATs are the most accessible.

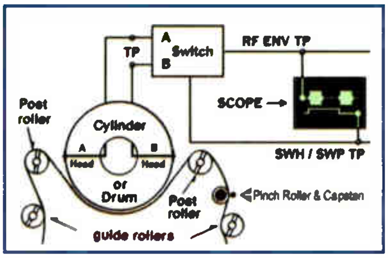

Figure 1: Digital transport basics are the same for DAT, ADAT and DTRS. Some symptoms are easily viewed with an oscilloscope via test points at the output of the head preamp.

CLEANING TAPES AND DTRS

Many users have an unwarranted fear of cleaning tapes. Considering the inaccessibility of many DAT transports, a cleaning tape may be your only option. As shown in Fig. 1, the "tape wrap" around a DAT head drum is only 90° or one-quarter of the head's circumference. (Tension is about six grams.) By contrast, both ADAT and DTRS formats have a 270° wrap—three times more contact—so, obviously, you want to be a little more careful. (Tension for those two formats is 25 grams and 10 grams, respectively.)

Figure 3: A racecar and a "flying head chip" have a similar profile. For head cleaning purposes, rotate the drum counter-clockwise only.

Note that cleaning tapes should only be used when there is either a sonically noticeable problem or when the error rate is high. Never rewind a cleaning tape. Simply play—10 seconds for DAT, five seconds for DTRS—and remove. Dispose of the tape when it reaches the end.

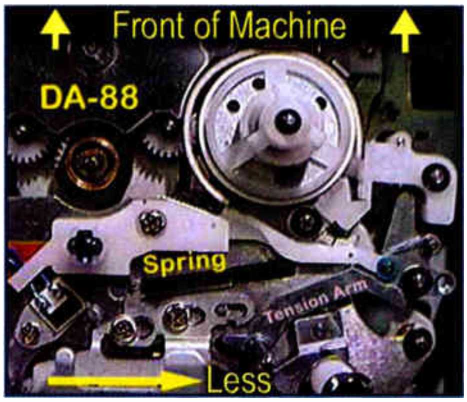

Figure 2: The horizontal arrow sits on top of a metal plate to which is attached a back-tension spring for the DA-88 and its kin. Two screws lock the adjustment, which should not be at maximum. When a new head is installed, the tension should be set to minimum to maximize head life.

Because ADAT head assemblies are so accessible, I never use a cleaning tape on an ADAT. DTRS heads, especially on the DA-88, require a little extra care. Unlike its newer siblings, the DA-88 has a linear, fan-cooled power supply. Unfortunately, the original design has the fan sucking air and dirt in through the "tape port." Tascam's solution was to add a clear plastic shield over the transport, but the shield is mostly in the way and not very effective. My solution is to reverse the fan, add an external filter and throw away the shield. Note that DA-38 and DA-98 heads are much more accessible.

BY EDDIE CILETTI

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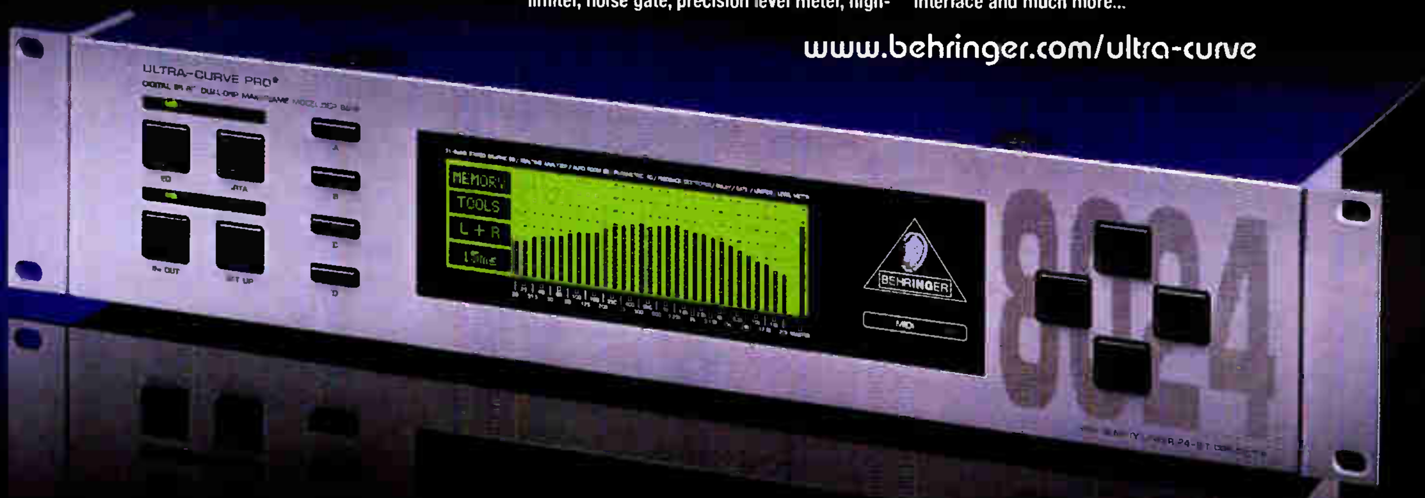

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Early DA-88s came from the factory with the tension set at maximum, which, when combined with an overly aggressive cleaning-tape policy, often resulted in accelerated headwear. Fig. 2 details the location of the back-tension (tape-to-head) adjustment. If it is at the maximum setting, the machine should be serviced. Ask the technician to set the tension between eight grams and 10 grams (maximum). Check the battery too, because leakage can cause all sorts of weird problems.

Remember that head cleaning in any form is not a panacea. The complex interaction of the head, capstan and reel-servo systems can be antagonized by accumulated dirt in clutches, belts that become stretched and slippery, cams and linkages that wear out or break, pieces from the cassette shell that break off, plus poorly applied labels. If a cassette gets stuck, all we technicians ask is that you don't freak out. Don't try to dig the tape out, and don't get blood on the heads. (Don't laugh—I've seen it all.)

Figure 4: Three steps to access error rate on an ADAT: 1) the normal display, 2) the brief display that pops up after pressing Set Locate and Record Enable for track 3, and 3) the actual error rate display and condensed tape counter.

Figure 5: Press Mode-Reset-Pause simultaneously on a Panasonic DAT, then press Mode until the "AB" appears. Copy protection is also indicated in this mode.

SWABBING THE DECK: COUNTER CLOCKWISE

Many technicians clean heads with a

DECODING ERROR RATE

Just so we're on the same page, Error *Rate* (and not an Error *Message)* is a numeric report of a machine's ability to recover data from tape. Errors are a normal fact of life, and most of the time they are "conceal-able," which means the data is correctly and transparently reassembled.

Most DAT machines have two heads, with the exception of models with "confidence monitoring." MDMs have four heads (one pair for playback, one pair for record) and still, errors are detected *on playback only,* because the play heads are in advance of the record heads. Not all machines provide error rate access, and those that do may not display the numbers in a meaningful way. Visit www.tangible-technology.com for more examples.

Both Alesis and Panasonic display error rate in an easily comprehensible fashion. Figs. 4 and 5, respectively, show the secret keystrokes that provide access to the error display. (The exception is the blackface ADAT, which only displays tape errors by blinking a decimal place.) Four zeros (0000) are good. ADATs show the combined errors of both heads. Panasonic lets you select either or both heads.

FALSE POSITIVE

For XT and LX Series ADATs, recording on reformatted tape may yield "false positives," that is, an intermittent "sun" icon indicating an overly high error rate. Check the error rate by the numbers, and you'll see fluctuations from nearly all zeros to three digits without any change in the audio. The problem is annoying but benign.

An oscilloscope makes it easy to see the cause of the many tape-related problems. Fig. 1 shows how to make the connections to view the RF envelope, the signal from tape. A dual-trace 'scope with 20MHz bandwidth is the minimum requirement. To view the cause of a false positive, a better' scope with a Delayed Sweep feature can zoom in. (See Fig. 6 a, b and c.) Oscilloscopes have vertical inputs (for amplitude measurements) and horizontal inputs (for sweep-related triggering and timing). There are many variations on a theme.

Locate and connect vertical input-A to the RF ENV test point and vertical input-B to the PG Delay/SWP test point. (Set probes to "x10" mode and trigger from input-B. The test points should be easy to find, near the head wire/preamp connections. When in doubt, see the service manual.) The square wave trace is the head-switching signal. The transition from high to low (and back) alternately selects either the A head or the B head.

If the orange asterisk (error correction) icon on your ADAT flickers cyclically on recycled tape—but not newly formatted tape—check the error rate to confirm that no asterisk means minimal (less than 0050) errors. High error rates for DAT machines (above 0400) and Tascam's DTRS/Hi-8 format should not be taken lightly.

Some machines will lose or gain unusual functionality while in Error Rate mode. The Panasonic SV-3800, for example, will not shuttle tape, and the Tascam DA-P1 will auto-rewind and play.

DTRS

Tascam's newer DTRS machines (DA-38 and DA-98) allow easy access to the Block Error Rate (BER, via menu), which is reported in the same four-digit fashion as the machines mentioned above. Access is much more difficult on the DA-88. (Press FF-Stop-Play on power up. Then press Stop. If "Test" is displayed, then press Remote. If not, keep trying. If a more serious problem exists, it will not be possible to enter Test mode.) When Play is pressed, the error rate will be displayed on meters A and B. The meters will jump to maximum at first but should settle down and "disappear" within three seconds. If any LEDs are lit, either the machine has problems (have it serviced) or the tape is marginal and should be cloned. Do not overdub on a marginal tape.

—Eddie Ciletti

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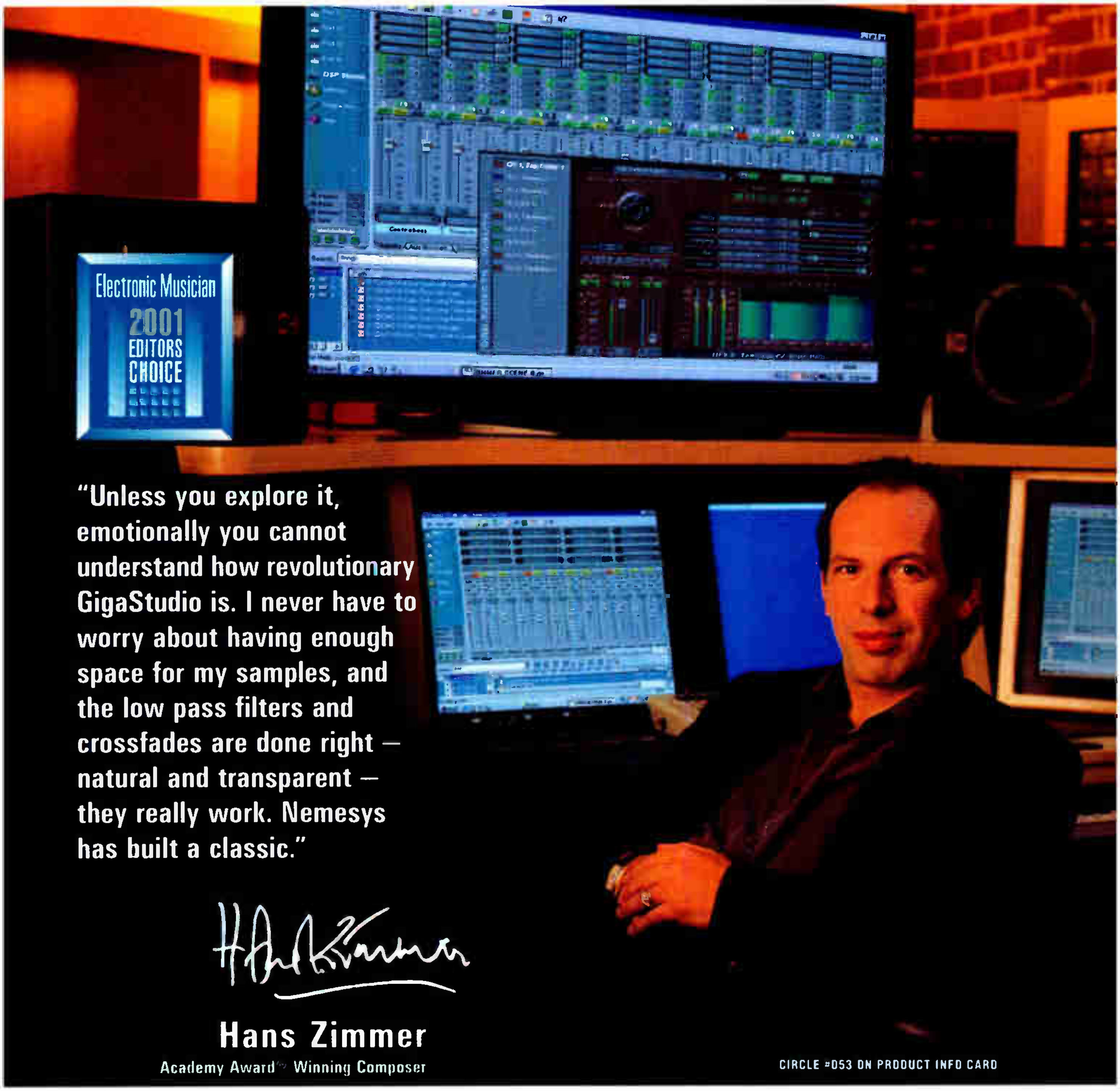

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chamois or chamois-on-a-stick (a Minnesota favorite). I use Twillwipes from Chemtronics (www.chemtronics.com). Moisten the cloth with anhydrous/99%

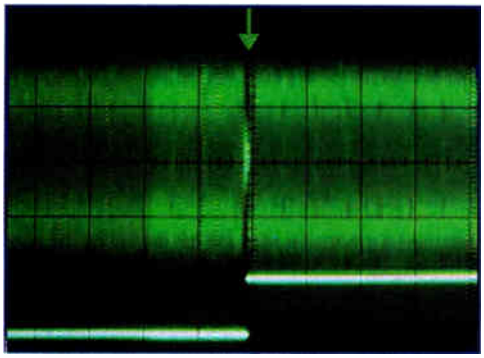

Figure 6a: The ADAT's RF envelope, including the output of both heads, highlighting the area to be scrutinized.

alcohol; denatured alcohol is an acceptable substitute. There may be other more aggressive/effective cleaners, but I work from home and choose human-friendly solvents. Apply the cloth to the head drum and turn the drum counter-clockwise only. Keep your hand absolutely steady—any "up and down" movement at this point can seriously damage the heads. I then apply a second dry cloth to wick away the remaining alcohol

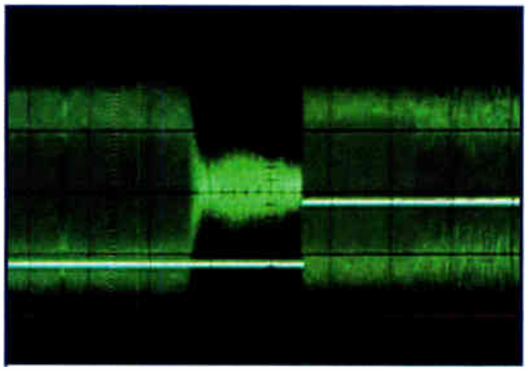

Figure 6b: Zoomed in to the head-switching transition, a nearly empty "gap." When recorded on new tape, the gap is a flat line.

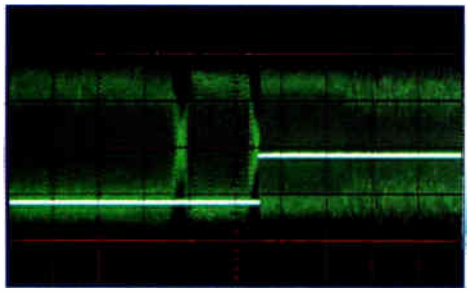

Figure 6c: Same zoom as in Figure 6b. Now the fragment of a previously recorded signal is obvious. Timing and sample rate differences are the reason this RF debris remains and, like a chorus/flanger, drifts in and out, disturbing only the error rate counter and the "sun" error alarm display, but not the audio.

and the oxide it dissolved. MCM electronics—www.i-mcm.com—is a good source for maintenance supplies, although their online search engine is lame. Request their paper catalog.

DTRS note: All DTRS head chips—this includes the DA-38, -88 and -98—have a "racing car" profile on the leading edge to slice through the air with the greatest of ease (as shown in Fig. 3, the 1984 Ferrari 288 GTO). If the drum is turned counter-clockwise, the sharp trailing edge will catch on the cleaning cloth, so don't do it.

DOCTOR, CLONE THYSELF

Paying attention to "data integrity" becomes especially important in a multitrack and multiple-machine environment. Often, I get complaints after repairs are made, not from a current machine problem but because the unserviced machines were making marginal tapes. This can affect everything from machine-to-machine lockup (and lack of same) to the ability to punch in.

I advise everyone to evaluate tapes by checking the error rate on a known good machine, then clone all the marginal tapes at the earliest convenience on new tape.

YOUR VOICES OF EXPERIENCE

Workstations may have taken some of the pressure away from tape, but that doesn't preclude the need to confirm both media and data integrity. At minimum, check error rate before and after head cleaning, whatever your method. Experience tells me that some of you put up with problems for weeks or even months before doing *anything*. Don't wait! The head, the tape and the transfer you save could be your own! ■

Eddie Ciletti hopes to be cross-country skiing by the time you read this. Visit him at www.tangible-technology.com.

PREVIEW

GENELEC 1093A ACTIVE SUBWOOFER ▼

The new 1093A Active Subwoofer from Genelec (www.genelec.com) offers a frequency response of 18-85 Hz (±3 dB) from a single magnetically shielded, 10-inch driver housed in a compact, 41-liter cabinet. It is designed to work with Genelec's 1029, 1030, 1031 and

1032 Active Monitoring Systems. Capable of a maximum SPL of 112 dB at 1 meter, the 1093A features an acoustic THD of less than 2% at the 2nd and 3rd harmonics at levels of 100 dB at 1 meter. A new 6.1 bass management system provides XLR I/Os for combined LCR Front and LCR Rear configurations, as well as a separate LFE input for compliance with Dolby EX and DTS ES 6 main-channel monitor formats. A built-in calibration tone generator allows users to check for correct phase using only a standard SPL meter. Additional features include comprehensive loop-through connections, LF response control, driver protection and a bypass function. The compact 21x12.5x23-inch (HxWxD) enclosure weighs 57 lbs. Price: $2,185.

Circle 327 on Product Info Card

SPL STEREO PARAMETRIC

The Stereo Q from SPL (distributed by Group One, www.g1ltd.com) is a stereo parametric equalizer featuring three fully parametric bands per channel with boost/cut ±15 dB and Q variable between 0.5 and 5. A subbass EQ (cut/boost ±14 dB, up to 200 Hz) and HF "Air" EQ provide additional tone-shaping functions, and a 3-D Enhancer circuit controls stereo width without affecting low-frequency components. All control functions for both channels can be adjusted with one control, and front panel controls feature detented pots. Lundahl input and output transformers are optional. Price is $899.

Circle 328 on Product Info Card

YAMAHA MLAN

Yamaha's (www.yamaha.com/proaudio) mLAN technology is a protocol and specification for connecting electronic musical instruments and audio devices via an IEEE 1394 high-speed digital interface. The mLAN system runs over a single line, providing "plug-and-play" cable connections capable of transmitting the equivalent of 100-plus channels of CD-quality digital audio, and music data equivalent to that carried by several hundred MIDI cables.

Three Yamaha products debuted at AES 2000: the mLAN-8P interface/breakout box, the mLAN-8E interface module and the CD8-mLAN interface card for connecting compatible Yamaha digital mixing consoles—the 02R and 03D—to an mLAN system. Each product operates at 200 Mbps (megabits per second) and offers sampling rates of 44.1 or 48 kHz, and 16- or 24-bit depth, depending on specifications. All products ship with mLAN patchbay connection management software, a mixer control application and mLAN Mac drivers for compatibility with the ASIO and OMS standards. Yamaha plans to develop mLAN product for both Mac and PC platforms via third-party developers. For details, visit www.yamaha.co.jp/english/news/00072002.html.

Circle 329 on Product Info Card

OTARI STATUS 2 DIGITALLY CONTROLLED ANALOG CONSOLE ▲

The Status 2 Digitally Controlled Analog Console from Otari (www.otari.com) is an affordable 5.1 surround-capable console based on Otari's existing Status console. Features of the Status 2 include 24 buses (up from 12), a five-way panner on each input module, a new automated joystick panner for surround panning and an M-Mon option. Standard features include master switching, dual-path equalized channel architecture, Image Recall, Eagle automation, VCA or motorized faders. It also includes frame sizes up to 96 automated inputs, and an optional dynamics package and patchbay.

Circle 330 on Product Info Card

ATC ACTIVE MONITOR

The T16 Active from ATC (distributed by Flat Earth Audio, www.flatearthaudio.com) is a two-way active monitor featuring a 6.5-inch soft-dome woofer with a 45mm voice coil and a 1-inch fabric dome tweeter with a neodymium magnet. Onboard amplification provides a total of 250 watts and maxi-

PREVIEW

mum continuous SPL is 108 dB; frequency response is 62-17k Hz (free-standing, -6dB down points). Additional features include a heavily damped aluminum casing (available in custom colors), front panel power-on indicator and user-configurable dynamics control for optimizing the system for multichannel use. The T16 is also available in surround format as part of the ATC Concept 4 system, which includes a matching subwoofer and center channel speaker. Price is $3,950 per pair.

Circle 331 on Product Info Card

TASCAM 24-BIT MULTITRACK RECORDER/PLAYER

The Tascam (www.tascam.com) DA-98HR DTRS-format MDM offers a range of 24-bit record/playback capabilities: eight tracks at 44.1 or 48kHz sample rates, four tracks at 88.2 or 96 kHz or two tracks at 192 kHz. The DA-98HR is record- and playback-compatible; all Hi-8mm DA-98, DA-88 and DA-38 tapes use the 16-bit, 44.1kHz or 48kHz standard. It is also fully compatible with the high-resolution DA-78HR. The DA-98HR includes a confidence monitoring feature, a comprehensive LCD, onboard 2-track mix function and an advanced internal electronic patchbay. Digital I/O is standard via two DB25 connectors available in both TDIF and AES/EBU formats; optional analog I/O boards are switchable among 44.1/48/88.2/96kHz formats. Sync functions include SMPTE In/Out with onboard TC chase synchronizer, RS-422 (Sony P2) interfacing, MIDI In/Thru/Out and MIDI Machine Control. Up to 16 DTRS units of any type can run in sample-accurate sync. Price is $6,999.

Circle 332 on Product Info Card

DBX TUBE CHANNEL STRIP WITH DIGITAL OUTS ▼

The 376 Silver Series Tube Channel Strip from dbx (www.dbxpro.com) features a tube preamp, 3-band parametric EQ, compressor, de-esser, and AES/EBU and S/PDIF digital outputs. The single-channel 376 also has +48V phantom power, phase-invert switch, 75Hz lowcut filter, -20dB pad, 8-segment LED meters, mic/line input switching, and a high-impedance, ¼-inch instrument input. The compressor section has rotary controls for Threshold and Ratio and a choice of hard knee or OverEasy compression characteristics. The frequency-tunable de-esser offers Frequency and Amount controls. Users can select from 44.1/48/88.2/96kHz sampling rates at 16/20/24-bits, with several noise-shaping algorithms and dither types available. The rear panel has mic and line inputs and wordclock I/Os. In addition to AES/EBU and S/PDIF digital outs, XLR and ¼-inch analog outs are supplied. Price: $599.95.

Circle 333 on Product Info Card

AUDIENT 5.1 MONITOR CONTROLLER

Audient (distributed by Audio Independence, www.audioind.com) offers the ASP510 surround sound monitoring controller, designed to add comprehensive 5.1 monitoring and mix capabilities to any stereo console. System integration is via a single-rackspace interface. The ASP510 supports three 5.1 and three stereo sources, accepts eight inputs from console bus sends, and provides eight outputs to recorders (5.1 plus stereo). Additional features include switchable Encoder/Decoder insertion connections, six speaker outputs, individual speaker cut/solo function, individual speaker level trims, guide track input and user-definable monitor reference level. State-of-the-art DCA gain control elements are used throughout, and an ergonomic remote control unit is standard. Price is $3,860.

Circle 334 on Product Info Card

DYNASTAR SPEAKER KNIFE II ▼

The Speaker Knife II from Dynastar (www.dynastarelectronics.com) is an adjustable, high-speed transient/overload speaker driver protector. Able to handle up to 2,800-watts peak power (1,400W RMS) at 8 ohms, the unit installs easily in-line between the amp and any stan-

PREVIEW

dard speaker, and it works with 2, 4, 8 and 16-ohm impedances. The SK II operates as fast as 65 nanoseconds, and, when a fault is longer than about 2 ms, the module will disconnect the driver from the power amplifier or circuit, automatically resetting if the fault is no longer seen by the module. Retail is $89.95.

Circle 335 on Product Info Card

SPIRIT DIGITAL 328 V. 2.0

Spirit by Soundcraft (www.spiritbysoundcraft.com) upgrades its Digital 328 console. New V. 2.0 software includes automation support for a wide range of sequencers and third-party hardware, and it now allows Digital 328 users to record, overdub and edit using any MIDI sequencing package. Audio from Control Room and Group Outputs may be routed via AES/EBU and S/PDIF digital outs. Version 2.0 offers support for Soundscape hardware, including the SSHDR, R.Ed and Mixtreme systems. The 328 now allows for control of Steinberg's Cubase VST 5.0 application, and the 328's Custom Tape Machine mode may be configured to use MMC, Note On, MTC and LTC in both Master and Slave modes. The Solo function has been enhanced, a wordclock feature added, and data dumps can be requested via SysEx to allow third-party applications to act as librarians. The new software is free to Digital 328 users.

Circle 336 on Product Info Card

API 2500 STEREO BUS COMPRESSOR ▲

API (www.apiaudio.com) offers the 2500 Stereo Bus Compressor, a dual-channel, rackmount unit featuring an all-discrete, fully balanced design. Offering threshold settings variable between -20 and +10 dB, and various compression ratios, the 2500 may be used for subtle compression or "brickwall" limiting. Attack time is variable from 0.03 to 30 ms, with release time from 50 ms to 3 seconds. ATI's patented Thrust circuit, located before the RMS detector, delivers a low-end punch. Compression is selectable between "Old" style (feedback-type of compression used by the API 525, Fairchild 600, etc.) and "New" style (VCA-modulated compression using an RMS detector, as used by dbx and SSL). Stereo coupling strength is variable between 100% and 0% in six steps. A high- or low-pass filter can alter the coupling signal and control the transfer of low end or transients from one side to the other. A (bypassable) automakeup gain control allows dynamic range to be manipulated without affecting output gain. Price: $2,695.

Circle 337 on Product Info Card

HOT OFF THE SHELF

SSL offers new software enhancements for the SL 9000 J Series SuperAnalogue console. Version 4.3 software enhances the Sony 9-pin interface and allows users to manage machine control from either the SL 9000 or from Pro Tools. Go to www.solid-state-logic.com or call 212/315-1111...**HHB has upgraded its 2.6GB and 5.2GB Magneto Optical (MO) disks** to deliver sustained and increased data transfer rates, offering significant advantages in high-resolution digital audio recording applications. HHB MO2.6GB and MO5.2GB media (identified by a 100% Certified stamp on the disk shutter and packaging) carries a lifetime warranty. Surf to www.hhbusa.com or call 310/319-1111...**Maxell's next-generation Betacam SX videocassettes** are backward-compatible with Betacam/Betacam SP equipment, and feature Ceramic Armor Metal magnetic particle and high-performance binders. The new tapes are available in both S (6, 12, 22, 32 and 60-minute) and L (64, 94, 124 and 184-minute) lengths. Visit www.maxell.com or call 201/794-5900 ...**Sony's CDR-W33 CD recorder/player** is a professional rackmount unit that includes 24-bit AD/DA converters, built-in DSP processing and CD-TEXT support. Supplied with a wireless or wired remote control, the unit features a range of onboard DSP functions, including parametric EQ, limiter and Super Bit Mapping. Call 201/930-1000 or visit http://bpgprod.sel.sony.com...**BGW introduces a new line of 19-inch racks.** The RN Series feature fully welded construction with 14-gauge steel tops and bottoms, 16-gauge steel side pieces, multiple cable lacing points, and adjustable or fixed rail-mounting positions. Call 800/468-2677 or try www.bgw.com...**Lexicon has upgraded the software for the 960L Digital Effects System.** Version 2 supports an additional DSP Reverb card, doubling the DSP horsepower of the system and providing for eight stereo or four surround reverbs at 48 kHz, or four stereo and two surrounds at 96 kHz. Version 2 also adds mappable I/O, support for 16 channels of independent I/O, a range of new factory presets, dual LARC2 support and several new cascaded configurations. For more info, call 781/280-0300 or check out www.lexicon.com. ■

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Mastering Engineer
"The Lodge" Studios, New York.

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TEC NOMINEE

KOBLO STUDIO9000

SOFTWARE SYNTH/SAMPLE PLAYER/DRUM MACHINE BUNDLE

Native software synthesizers were at one time dismissed for pro use due to their latency and poor sound quality. They were considered okay for occasional sound design but were shunned as real-time performance and composition tools. However, super computers—such as the Mac G4 and Pentium III class—have helped rewrite the reputation of software synths. Virtual instruments, as they are also called, now sport incredible synthesis engines and latency that is practically nonexistent with the right setup. Studio9000, Koblo's software synthesizer bundle, is an excellent example of the state of virtual instruments.

The Mac-based Studio9000 bundle has three modules—the Vibra9000 monophonic synth, the Stella9000 sample playback instrument and the Gamma9000 drum machine. Two scaled-down versions of the Vibra9000 are also included: the Vibra6000 and the Vibra1000. (The latter is also available as a free download at www.koblo.com.) All the modules share "Tokyo," a real-time engine that's put on your computer during the software install. An instrument's audio output can be routed to your digital audio sequencer's audio mixer via Steinberg VST2 or ReWire, MOTU MAS, or Digi's DirectConnect. To route the audio separate from your digital audio sequencer's audio mixer, Sound Manager and Direct I/O drivers are available. MIDI is handled via OMS or FreeMIDI.

Studio9000 has three modules: the Vibra9000 monophonic synth (pictured), the Stella9000 sample playback instrument and the Gamma9000 drum machine.

MAKING CONNECTIONS

Installing Studio9000 can be routine or complex, depending on your needs. Using the program as a stand-alone application is as simple as installing the program, booting up the Tokyo engine, opening an instrument, and assigning your audio outputs and MIDI inputs. Routing the instruments through your digital audio sequencer's audio mixer requires assigning Tokyo an interapplication communication bus (IAC). The manual clearly explains setting up an IAC bus for OMS—which I used—but doesn't mention doing this with FreeMIDI. Third-party audio drivers aren't automatically installed, but they are available on Studio9000's CD-ROM.

Koblo suggests a minimum system requirement of a Power Mac 604e, 120 MHz or better with 40 MB of available RAM, and OS 8 or higher. I'd suggest *a lot* more horsepower. I used a 400MHz G4 with 256 MB of RAM and OS 9.0.4. This setup was fine working with Tokyo by itself, but once I started routing Tokyo's audio through my digital audio sequencer and inserting effects plug-ins, memory and CPU power were tight. As with most virtual instrument applications, the more powerful your computer, the better the performance. I auditioned Studio9000 using Digidesign's Pro Tools Mix card with an 888|24 I/O converter.

BY ERIK HAWKINS

Authorizing Studio9000 is accomplished via a key disk or by challenge/response codes. Kudos to Koblo for including a key disk for folks that have floppy drives; it's nice to avoid the challenge/response routine if possible. Unfortunately, because the G4 and "blue-and-white" G3 Macs don't have floppy drives, the key disk isn't much good. Key disks don't often work with third-party USB floppy drives; consequently, most users will do the challenge/response anyway, registering a version and challenge code on Koblo's Web site. A response code is e-mailed back to you (mine came within 24 hours).

GLOBAL PATTERNS

The Studio9000 instruments aren't fashioned to look like any particu-

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WORKING FOR SCALE

If you've ever had occasion to pitch correct a melody line whose key was not exactly clear, or which had too many accidentals to fit comfortably into a conventional scale, you'll appreciate the ATR-1a's new Make Scale From MIDI function. Simply play any line into the ATR-1a from a MIDI keyboard or sequencer and let the ATR-1a automatically construct a custom scale containing only those notes that appear in the line. No muss, no fuss.

CHECK IT OUT

So if you're still looking for perfect pitch in a box, check out the ATR-1a at your Antares dealer now. Where the song remains the same.

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UPGRADE ALERT! *Got an original ATR-1? Except for the new cosmetics, current ATR-1s can be upgraded with all of the ATR-1a's new features by simply replacing an EPROM. (And if you purchased your ATR-1 after August 1, 2000, the upgrade is free.) Contact your local Antares dealer for all the details.*

ANTARES

WHERE THE FUTURE'S STILL WHAT IT USED TO BE

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lar type of real-world synth. Instead, they have their own look that is decidedly electronica—cool in my book! Though all the instruments have similar front panel layouts and identical graphical control elements, each module has its own color scheme, which makes it possible for the user to distinguish between the instruments at a glance. Stella is blue, Gamma is purple and Vibra is green.

Each instrument has a global parameter section with common controls, including master tune, pan and volume knobs; discrete mute and solo switches; a dedicated key for recording performances direct to hard disk; a MIDI panic button for clearing stuck notes; a Trigger for playing instrument sounds via a mouse click; and a Hold button that sustains the last note triggered. Because drum sounds are mostly transients, Gamma lacks the Hold button but has a global bend parameter. The Vibra1000 is the only instrument without the record-to-disk feature. (Remember, it's the freebie.)

Monitoring includes stereo master LED meters with peak hold and a MIDI activity light. Selecting any knob and a dedicated window displays that knob's function and Control Change number. (Every knob has a CC number for comprehensive automation.) The associated parameter value is shown as a large, red, alphanumeric LED. The current patch name has its own dedicated window, and the user's sound card's selected outputs are displayed there.

STELLAR VIBRATIONS

The filter, ADSR, LFO and modulation sections are similar on both the Stella and the Vibra9000. Each has eight multistate filters, three ADSR envelopes, three LFOs, arpeggiators and eight modulation sources/destinations.

The filters sound great and include the following types: 2/4/8-pole, Double and Quad (12 dB/octave multiple parallel), Notch (24 dB with split highpass and lowpass), and Saw and Square Comb with multiple resonant peaks. A filter output stage includes cutoff, resonance lowpass, highpass, bandpass, distortion and stereo spread parameters. Keyboard tracking and a Separation control can vary resonance and cutoff frequencies, depending on the filter type.

Each of the three ADSR envelopes can be inverted and is velocity sensitive. The waveforms for the LFOs are Ramp Up and Down, Triangle, Square, Sine and Random. Simple Attack/Decay envelopes are provided for each LFO. A parameter called Sharp applies a lowpass filter to the LFO's shape in order to smooth the waveform's edges. The LFOs can be synchronized to incoming MIDI Clock—a wonderfully useful effect. The eight modulation sources include all three envelopes and LFOs, velocity, aftertouch and the Mod Wheel, to name a few. Modulation destinations vary from Pan and Separation, to envelope times and even the LFOs themselves.

The arpeggiator is flexible and great fun to play with. Typical controls, such as tempo, octave range and sequence, (up, down, and up/down) are offered. In addition, there are several advanced parameters allowing sequences to be played back in a variety of ways. For example, rhythmic accents can be altered by changing note velocities, the sequence length can be set from one to 16 steps, portamento between notes is fully adjustable, and there are several settings that determine how octaves are played in an arpeggiated chord. The arpeggiator recognizes MIDI Clock for perfect synchronization with your digital audio sequencer.

The sound source for the Vibra9000 consists of two oscillators with five waveforms each: Saw, Square, Triangle, Sine and Noise. There is no oscillator sync feature, but amplitude and frequency modulation can be achieved by using Oscillator 1 to modulate Oscillator 2—a nice effect. Oscillator tuning spans an ample 15-octave range. Other parameters affecting the oscillators include portamento, detune, mix, bend and stereo doubling. A keyboard tracking control is available for Oscillator 2's frequency only.

Stella supports up to 32-bit, 44.1kHz samples in .AIFF and SDII formats and is 8-voice polyphonic (16 voices would have been nicer). It is a sample playback instrument only and does not record. However, I didn't find this to be a big problem, because there are plenty of sample sources on third-party CD-ROM, and, if you own a digital audio sequencer, you probably use that to record with anyway. A sample can be tuned to a 15-octave range and by semitones and cents. The pitch bend parameter goes from zero to 60, and a sample's start point can be adjusted. Reverse and loop are available, as well as a basic Attack/Decay envelope for the sample itself. Velocity can be set to modulate the sample's start point, pitch, volume and panning.

Samples are easily assigned for playback if the user clicks in Stella's sample window, where the name of the sample is displayed. This opens a typical Browser window where samples can be located. Unfortunately, there's no way to audition samples from this window, which is a bit inconvenient. Another inconvenience is that Stella only reads SampleCell keymaps. There is no way to map your own samples from within the instrument. This is a major drawback. If you don't have SampleCell to create keymaps, you're stuck working with just one sample when creating your own patches.

GAMMA GLOBULIN

The Gamma9000 sports seven sample slots (Koblo calls them tracks). The first six slots hold one sample each, and the seventh slot is for multisample keymaps. The samples in slots 1 to 6 are mapped from C0 to B0 (this is fixed), and C1 and above are reserved for the keymaps. Like Stella, Gamma can read SampleCell keymaps. But unlike Stella, if you don't have SampleCell to concoct keymaps, Gamma will automatically map a group of samples for you. All of the samples must reside in the same folder and be either .AIFF or SDII format (.WAV would be nice, too). Open the folder from within Gamma, and the samples are mapped alphabetically.

Each sample slot has several associated sound-shaping controls. These controls affect the samples in slots 1 to 6 discretely, but the keymapped samples in slot 7 are affected globally. Samples can be reversed, looped and have their start point offset. Modulation sources include a simple lowpass filter (called Tone), velocity and the instrument's master filter section. The modulation destinations are volume, pan, pitch, sample start offset and tone. There is a simple Attack/Release envelope, and each slot has its own volume, pan and pitch controls.

Less comprehensive than Stella and Vibra's filters, Gamma's master filter section still sounds cool and sports all the essential elements. There are three filter types—highpass, lowpass and bandpass—and dials to adjust cutoff, resonance and distortion. This filter affects all the samples routed to the instrument's master output. With a sound card with multiple outs, it's possible to bypass the filter by sending specific sample slots to different outputs.

—CONTINUED ON PAGE 216

ESSENTIAL READING

HHB CDR850/850PLUS CD Recorders

HHB CDR830 BurnIT CD Recorder

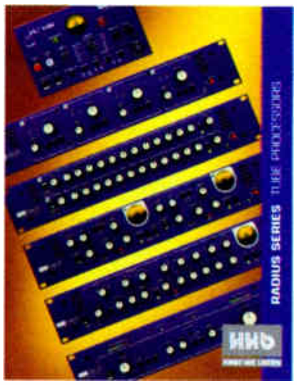
PORTADISC Professional MD Portable

HHB Circle Active/Passive Monitors

HHB Advanced Media Products

Genex GX8500 Multi Format Recorder

TLA VTC Vacuum Tube Console



HHB Radius Tube Processors

HHB Classic Tube Processors

Genex 24-bit/192kHz + DSD Converters

Quantec Yardstick Reverb

Rosendahl Nanosyncs Clock Generator

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FIRST WE LISTEN

CIRCLE #058 ON PRODUCT INFO CARD

SEE US AT NAMM BOOTH # 4268

AUDIX VX-10

VOCAL CONDENSER MICROPHONE

Over the past few years, Audix has emerged as a serious player in the field of live performance mics. The company's latest offering is the VX-10, a $599 cardioid model with a screw-on, field-replaceable, true condenser capsule in a body based on the same form factor as its familiar OM Series of dynamic mics.

I auditioned this model for use on Joni Mitchell's Both Sides Now tour. We were looking for the most natural-sounding vocal microphone on the market, because on the tour, as on the album, Mitchell is backed by a 60-piece orchestra. There have been many condenser vocal mics introduced in the past few years; we searched through dozens of models, so it took some time.

Most condenser vocal mics have an unnatural presence peak between eight and 11 kHz that is annoying to singer and listener alike. Many also have a proximity effect that clouds the response in the midrange. Some condensers use an electret design allowing operation on phantom power voltages below 48 volts, but many of these have an artificial sound that doesn't work as well for featured instruments or vocals. By the end of our auditions, only two other models were still in the running with the VX-10—a recent offering from Germany and a not-so-recent one from Japan—but the VX-10's open and transparent sound placed it ahead of those.

ON TOUR WITH JONI MITCHELL

The Mitchell tour proved the mic's strong points; it kept any leakage of the orchestra or floor monitors sounding as natural as the singer. The polar pattern seems tighter than what I would call cardioid, but, due to its smooth off-axis response, it is very forgiving. Mitchell's sultry, swaying vocal delivery is combined with four floor monitors and a 60-piece orchestra, so the mic had sound coming at it from all directions. Its smooth HF response eliminated the usual need for a de-esser. Low handling noise and a modest proximity effect make it a singer's dream come true. Our only objection was a slight excess of 800 Hz, which, given the mic's other strengths, was a simple shortcoming to fix with a tweak of the EQ.

AND WITH K.D. LANG

Moving on to k.d. lang's Invincible Summer tour immediately afterward, I wanted to use the VX-10 again. However, Ms. lang has a strong attachment for a legacy electret condenser that she's used her entire career, and she is very comfortable with its unique styling and steadily rising response. We did, however, put three VX-10s into play for background singers Amy Keys, Kate Markowitz and Windy Wagner, who especially enjoyed their sound because they were using in-ear monitors. The complicated three-part harmonies—a trademark of lang's production—are the second-loudest element in the mix. We were often asked what kind of effect was employed on them, when it was simply a stock Lexicon reverb.

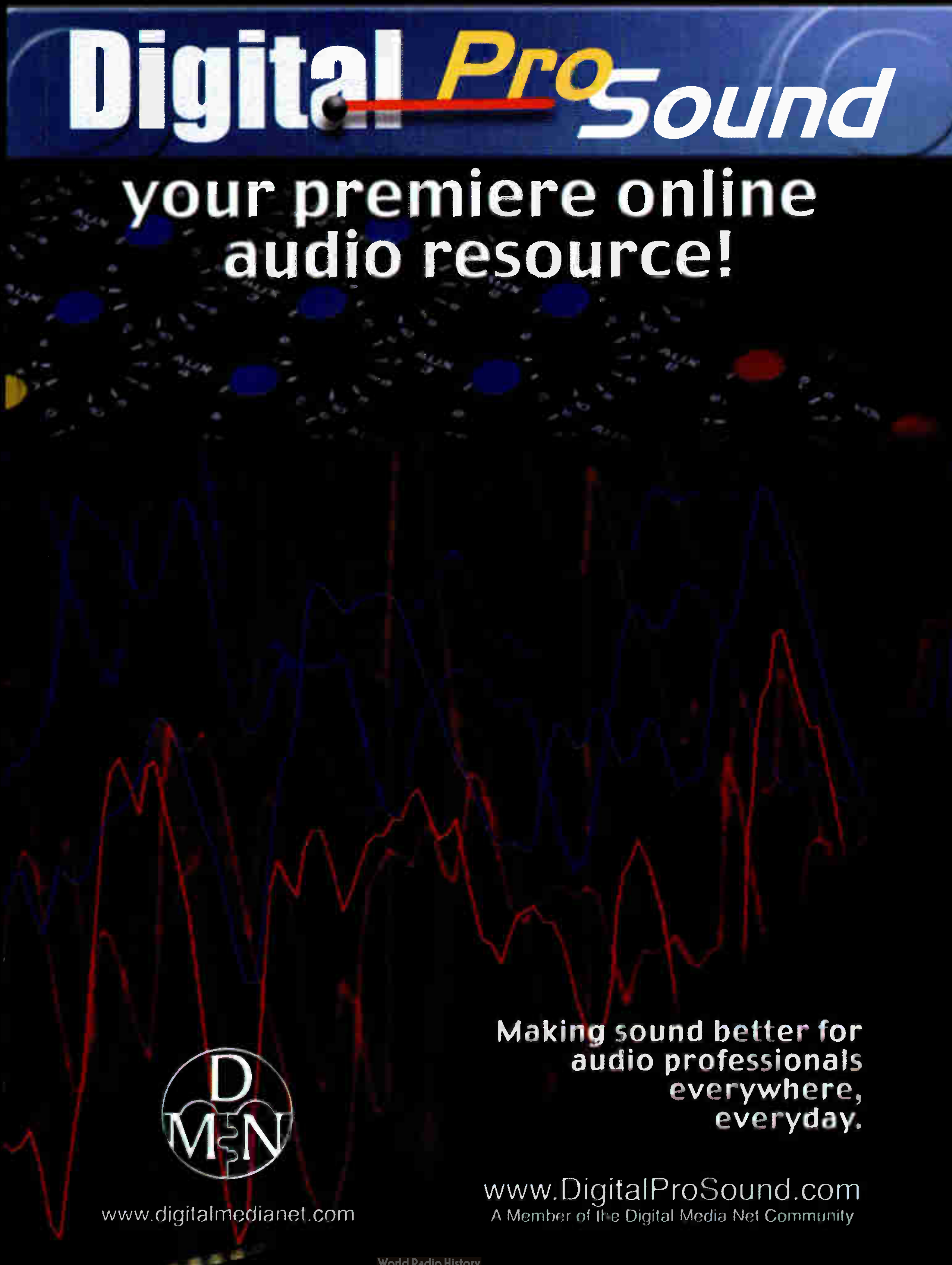
BY MARK FRINK

The VX-10 offers 10 to 20 dB more output than most other condensers. This requires less gain at the mic preamp for a cleaner sound. The VX-10 also has a great deal of headroom, so that vocals—from a whisper to a scream—are reproduced cleanly. The most striking feature of the VX-10 is its natural, transparent sound quality. With sound engineers having to fight so many elements to get vocals to sit cleanly in the mix, this mic offers an edge that will make the most jaded live engineer sit up and listen. Don't take my word for it; compare one to your current favorite vocal condenser, either live or studio, and hear for yourself.

BRINGING IT BACK HOME

Although the VX-10 is intended for live sound, broadcast and recording users will also appreciate its sonic honesty and robust output level. Engineers may also find it serves a variety of applications other than vocals. Musicians looking for an all-purpose condenser mic that can be used both in the studio and on the road need look no further than the VX-10. A nice addition to any inventory, this microphone would never sit on the shelf for long.

Audix Corporation, Box 4010, Wilsonville, OR 97070; 503/682-6933; fax 503/682-7114; www.audixusa.com. ■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PEAVEY SRM 2410 HC

SOUND REINFORCEMENT MONITOR CONSOLE

When it comes to headlines in the pro audio press, monitor consoles are rarely spotlighted; the "more important" house consoles generally get more attention. However, while FOH engineers typically need to focus on only one mix, monitor engineers may need to create four, five, six or more mixes simultaneously. Fortunately, monitor consoles have greatly improved over the years and often offer more complexity and features than FOH boards. Incorporating onboard mic splitting, low-noise circuitry, advanced feedback suppression capability and 10 mix output buses, the new Peavey SRM 2410 HC is an excellent example of that trend.

The SRM 2410 HC is a 24-input monitor board featuring 24 mic/line input channels (with phantom power switchable in three banks of eight channels). The 24 transformer-balanced input channels feature parallel male XLR "thru" splits on each channel (plus TRS send/return insert points), 4-band channel EQ with sweepable mid bands, and mute and PFL on every input. Each input includes an FLS (Feedback Locating System®) LED that indicates the channel with the highest level, and each of the eight mono submasters have an adjustable highpass filter, as well as two tunable notch filters and an FLS indicator to help identify the offending frequency.

In addition to the eight mono submasters, the Master section includes a stereo bus with pre/post switching and main L/R outputs addressable from any of the subgroups. A comprehensive talkback section allows the talkback to be routed to any of the output buses, and a ClearCom-compatible interface with standard 3-pin XLR intercom in/out connections, a 4-pin XLR headset jack and a large, bright "call" light eases integration with communications systems. Additional features include 12 peak reading LED output meters for the output buses and 12-volt XLR lamp sockets for board lights.

This 74-pound package is built into a 9.5x43.5x25.5-inch (HxWxD) Calzone ATA hard case (hence the "HC" in the SRM 2410 HC's model number). The board's internal universal power supply accepts any AC line source from 100 to 240V and ships with standard IEC AC cables with U.S.- and European-style plugs.

Setup and operations are straight forward. All of the main output bus signals appear on balanced XLR and unbalanced ¼-inch jacks, and the input transformers provided clean, transparent splitting. Phase reverse (polarity switching) is only offered on 18 inputs (1-6, 9-14 and 17-22). The board has no concentric controls so it's easy to navigate and make quick adjustments when necessary. The 75Hz lowpass filter and 4-band, ±15dB channel EQ is flexible, with plenty of overlap in the two sweepable mid bands (100 to 3k and 500 to 15k Hz), while the fixed 12.5kHz and 80Hz points on the HF/LF shelving bands are both useful and musical. Also handy is a set of "wedge" outputs that provides a separate stereo bus for monitors at the mix position, driven directly from any PFL or AFL button.

The feature that sets the SRM 2410 HC apart from other monitor boards is the inclusion of the FLS LED tuning indicators and 2-band, narrow-Q notch filters on each of the monitor output channels. The notch filters have a -15dB attenuation range and are sweepable from 500 to 10k Hz (high band) and 100 to 4k Hz (low band). The filters have a sizeable overlap, allowing stacked use for deeper notching. And, if desired, an external EQ can be patched into the insert points for each output.

Three LEDs (left/center/right) below each notch filter band indicate how to sweep the filter's frequency to eliminate feedback. If the left LED lights, then lower the frequency; if the right LED glows, then a higher frequency is required. Once the filter is set at the right frequency, all three LEDs light and the user merely adjusts the filter depth (attenuation) control until the feedback subsides. FLS is a simple, effective means of optimizing monitor levels, adding six to 10 dB of available gain, while system ringout can be handled in a matter of minutes.

My SRM 2410 HC want list is relatively short, with minor points such as differentiated channel EQ knobs (currently all six are the same color), including schematics and block diagrams in the manual and offering the board in a 32- or 40-input version for larger productions. *Note: Peavey also makes the RQ1606M, a 16-channel monitor mixer with built-in (nontransformer) splits, FLS channel LEDs, adjustable HPF on the subs and retail price of $1,166.*

But with its 10 monitor buses, clean audio performance, innovative feedback control and comprehensive set of pro features, the SRM 2410 HC offers a lot more than its $3,499 retail price would imply.

Peavey Electronics, 711 A Street, Meridian, MS 39301; 601/483-5365; fax 601/486-1278; www.peavey.com. ■

BY GEORGE PETERSEN

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TC ELECTRONIC TRIPLE C

DIGITAL MULTIBAND COMPRESSOR

Perhaps the Triple C should be called a Triple Threat. It is a digital full-range compressor, multiband compressor and envelope compressor capable of extreme alteration of attack and release. TC Electronic comes through again with another useful and modern recording processor with some unique twists on the compression process.

The Triple C comes in two different single-rackspace versions: a $699 single-channel unit (the one I tested ran V. 1.02 firmware) and a stereo version at $999. Two single-channel versions can be stereo-linked, but the stereo unit cannot be "unlinked" for two separate compressors. The Triple C accepts and outputs audio either as analog with the back panel balanced TRS jacks, or digitally with a pair of S/PDIF RCA jacks. Maximum input analog level is +24 dBu, and maximum output is +20 dBu balanced. A/D conversion is 24-bit with 128x oversampling, and the digital output can be dithered to whatever bit depth is required. The unit clocks to the incoming digital source or is selectable to a 44.1 or 48kHz internal clock when the analog inputs are being used. There is full MIDI implementation that can be used to offload and store the 50 factory and 100 user compressor presets called Styles, as well as control all the unit's parameters using Continuous Controller data.

KNOBS, NOT BUTTONS!

All adjustments are made with knobs—yes, knobs! Thankfully, there are no up/down value buttons here, just knobs that, for the most part, have a single function each. The Input Level control has a range of -6 dB to +18 dB, and input and output levels are displayed on two small vertical VU meters at the left side of the comprehensive digital display. Clip indicators show both analog clip at the input or output and digital clip (i.e., one or more samples at or exceeding 0 dBFS).

The compact display includes a 3-band, horizontal linear gain reduction/output meter that ranges from -18 dB to +18 dB; a graphical display used for the Envelope compressor; "Override" (a means of matching front panel knob settings to values held in the currently selected preset); and status indicators for analog or digital input, sample rate, quality of sync lock to an external clock, and stereo link to another Triple C.

Below the display, a well-lit, 23-character digital readout identifies and changes automatically to the knob and value being adjusted—a nice feature. This readout also shows menus for I/O settings and dither options, factory and user compressor presets, and certain internal fixed compressor parameters, such as the crossover frequencies for the multiband compressor. These parameters are accessible via the large System parameter wheel, a dual-concentric knob that is pushed in to select various actions. It took me a while to learn how to navigate around the Triple's system. However, after initial setup, I found little need to go there.

PROFESSIONAL COMPRESSOR

Like any other professional compressor, the Triple C has the required Threshold, Ratio, Attack and Release knobs. Threshold adjusts from -40 dB to 0 dB, and Ratio ranges from 1:1 to 1: infinity. Attack time is adjustable from 0.2 ms to 70 ms, and Release goes from 20 ms to 2 seconds. The Triple C has a Look Ahead feature in Multi Band mode, allowing for more precision with complex compressor tasks. However, this mode necessitates that the output be delayed by 3 ms. The nominal A/D processing delay through the Triple is 0.7 ms and should be considered when processing one or more sources within a multiple microphone recording where phase integrity is very important. This delay, although slight, could affect phase coherency in this special recording situation.

BY BARRY RUDOLPH

The Triple C is an RMS compressor and can be switched into a peak compressor in Multi Band mode. Softlim, available in all modes, is a peak limiter that works like the version in the TC Finalizer unit and is pretty easygoing. Finally, a master Makeup gain control boosts or decreases the entire processed signal by ±18 dB after gain reduction, and it has a clever display: The entire VU meter slides to the left with less makeup or toward the right when more makeup gain is applied. Meanwhile, the meter continues to show gain reduction, making this the best compressor metering scheme I have ever seen.

FULL-RANGE VS. MULTIBAND

The Triple C's most immediate mode is Full-Range, and all the controls work and act as you would expect, just like any good analog compressor. I usually started in this mode to get a ballpark setting and sound. Increasing the input level pushes more level and causes more compression. I liked the unit a lot in this mode, and I sometimes used the Digital Radiance Generator (another feature borrowed from the Finalizer) to introduce second harmonic distortion on rock guitar tracks.

The fun starts when you switch to Multi Band mode. For years, multiband compression has been used at radio stations and in live sound. Now, it is gaining popularity in the recording studio as a means of dealing with difficult dynamics. The user can compress the band most responsible for large level jumps without affecting the rest of the audio with unwanted compressor side effects.

In Multi Band mode, the VU meter splits into three parallel meters, each reading different gain reductions and levels for the High, Mid and Low bands. Without doing anything, you'll hear a noticeable difference in sound; if you were compressing a mix, the top end will open back up as if you were not compressing at all, and everything else will sound less squashed.

Using the Triple C on a solo piano recording, I first reset the crossover points of the low- and high-band compressors. The crossovers are both identical shelving filters that range from 19.95 to 20k Hz. I ran the high crossover out to 12 kHz, and, when I was moving the low crossover frequency, it was easy to tell where I wanted it. In real time, I heard and saw the frequency areas that were mostly triggering the compressor, and I could dial in crossover points.

The spectral balance is adjusted with the Lo-Freq and Hi-Freq controls. Increasing either of these controls increases the makeup gain for those respective bands, while maintaining the same gain reduction based on program content and the master threshold setting. Unlike some other multiband compressors, there is no way to set individual thresholds for each band, making the Triple C much easier to set on-the-fly. This action is indicated again on the sliding VU meters. Decreasing both controls together has the net effect of increasing the mid-band after you turn up the master makeup gain control.

ENVELOPE COMPRESSION

Envelope Compression refers to the process of increasing or decreasing the attack and/or release portions of the dynamic envelope of a sound. This is a process that works well on periodic and predictably consistent sound sources, such as kick and snare drums, individual percussion instruments, samples, loops or preprogrammed recurrent synth events.

When Envelope Compression is selected, the four main active controls become Attack Gain, Attack Time, Release Gain and Release Time. One note of caution: Turn your monitors way down when you are switching into this mode, because the sudden changes in level are very dramatic—especially if the two spectral knobs are turned clockwise. In Envelope Compression mode, both the Attack and Release controls set duration times for gain modification in the attack and release portions. Turning the Envelope Attack Gain clockwise increases the level during the attack portion of the envelope up to the maximum output of the Triple C, or +20 dB. Likewise, you can reduce attack level by 20 dB or more by turning the Envelope Attack Gain counter-clockwise.

Once the envelope is in a quiescent state—i.e., after the attack portion is over but before the onset of release—the signal passes unaffected. With the Envelope Release Gain, I could lift the level of the end of a sound for up to a full two seconds and up to the +20dB maximum output. Conversely, I could also reduce the sustain tremendously by turning the Envelope Release Gain control counter-clockwise. This was good for reducing reverb tails and unwanted sonic aftermath. It's a little like a downward expander, only much smoother and more musical.

The Triple C's Envelope Compressor is perfect for greatly increasing the attack or "hit" of a snare drum, or for bringing up the back end of a drum loop, or reducing excessive recorded reverb or room tone on any individual sound. Furthermore, any noise present will be greatly amplified along with the ring-out sustain. This feature worked more comprehensively than the SPL Transient Designer I reviewed in the January 1999 issue of *Mix*, and unlike the analog-based Transient Designer, the Triple C allows for control over both the length and level of the attack and release portions.

At $699 (or $999 for the stereo version), TC Electronic's Triple C is a new kind of digital processor that provides an easy-to-learn introduction to multiband compression and also offers an alternative dynamic control method with many creative possibilities. I like the new level of precision, adjustability and resettability made possible by the digitally based Triple C, previously only available within digital workstations and impossible with analog compressors.

TC Electronic Inc., 742-A Hampshire Road, Westlake Village, CA 91361; 805/373-1828; www.tcelectronic.com. ■

Barry Rudolph is an L.A.-based recording engineer. Visit him at www.barryrudolph.com.

FIELD TEST

Sonic Foundry Vegas Audio 2.0

DIGITAL RECORDING/EDITING SOFTWARE

Vegas Pro, Sonic Foundry's foray into multitrack digital audio recording and editing, established the company in the upper echelon of desktop audio production. Targeted for pro musicians, broadcast and audio engineers, as well as multimedia and Internet content developers, the application combined a highly configurable workspace with the ability to integrate into numerous production environments.

With Vegas Audio 2.0, Sonic Foundry introduces a number of refinements to the program's audio capabilities, while adding improved video support. It's also important to note that the program's name has changed to reflect the audio and video versions that are now available. Both Vegas Audio 2.0 and Vegas Video have video capability, but the higher-end Vegas Video is better suited for video professionals accustomed to working with applications such as Adobe Premiere, but who are looking for significantly greater audio capabilities. For the audio pro, Vegas Audio 2.0 is well suited to handle any number of audio production tasks.

Vegas Audio 2.0 supports 24-bit/96kHz digital audio, so the number of recorded tracks is limited only by your computer's CPU/RAM processing power. It also has the ability to import a considerable variety of file formats, including .WAV, .AIFF, MP3, AVI and QuickTime. Similarly, it will export to MP3, .WAV, .AIFF, and it supports file authoring for Internet streaming applications, such as Real Networks G2 (RM) and Windows Media Technologies (WMA).

Vegas is a PC-based application; system requirements are a 200MHz processor (400MHz recommended); 32 MB of RAM (128 MB recommended); Windows-compatible soundcard; VGA monitor (24-bit color display recommended); Microsoft Windows 98 SE, NT4.0 (Service Pack 4) or 2000; DirectX Media Runtime 6.0 or later (included on install CD-ROM); and Microsoft IE 4.0 or later for online help (Version 5.0 included on CD-ROM).

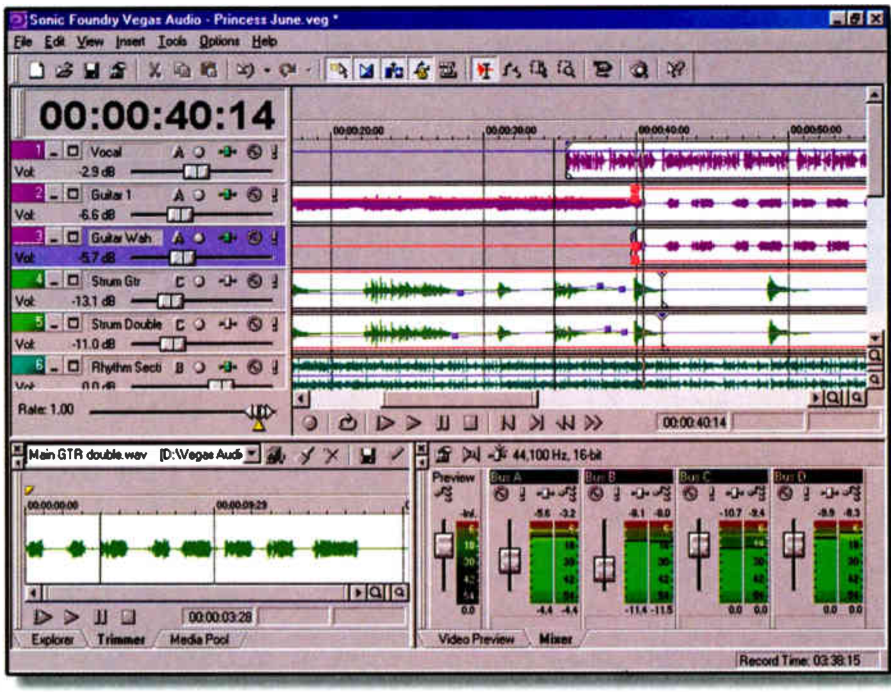

Figure 1: The blue and red horizontal lines running through events represent Volume and Pan for Vegas Audio 2.0's breakpoint automation. The lower left Trimmer window allows the user to isolate audio, and the Scrub control is just above the Trimmer.

For the multimedia or Internet content professional, Vegas Audio 2.0 has facilities for adding timeline metadata such as URL flips and captions. These facilities enable you to direct Web surfers directly to a specified URL and provide an excellent means of steering traffic to your Web site. The program not only processes multitrack audio, but it also enables one to perform dialog replacement for computer multimedia. The ability to open an AVI or QuickTime movie, replace the dialog, plus add music and sound effects, enables one to produce very high-quality media files with surprisingly little effort.

BY ROGER MAYCOCK

NEW FEATURES

Vegas Audio 2.0 provides a number of new features. For starters, the program now ships with Sonic Foundry's XFX 1, 2 and 3 DirectX plug-ins, providing 15 effects ranging from reverb, delay and chorus to time compression, noise gate and distortion. Of course—as DirectX plug-ins—they're accessible from within any other DirectX architecture audio applications. Also, in the new version, these DirectX plug-ins are assignable to both tracks and buses. Previously, these effects could only be assigned to the buses.

In Vegas Audio 2.0, an audio event is a segment of data on any track, while a track is nothing more than a container for the events. With V. 2.0, you can lock the volume and pan envelopes to these events. In Fig. 1, the red and blue lines running through the audio events on the upper right portion of the display are the Pan and Volume


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envelopes. Vegas Audio 2.0 uses breakpoint automation for volume and pan. By locking envelopes to events, you can move the audio data along the timeline, and any volume and pan assignments you made go right along with it.

Producers of jingles and on-air radio spots will appreciate the Rubber Audio feature, which can time-compress and stretch audio. Working with audio events, Vegas Audio 2.0 lets you change the length while preserving pitch, change the pitch while preserving length, or change both length and pitch.

If you need a click track, then Vegas Audio 2.0's new metronome is both a blessing and a curse. This feature lets you define time signature and tempo, but does not support time signature changes. To work around this, set your click to, perhaps, the eighth note value and then assign the same sound to both the normal and accented clicks. This will accommodate tempo changes between 5/4 and 4/4, for example, though you won't have any reference for the downbeat of the measure. If you find this distracting, you may wish to program a drum machine and print that track as a guide.

In the original Vegas Pro, all effects processing was accommodated by RAM and was, in essence, in real time. Vegas Audio 2.0 now offers destructive effects processing. This new function can be used to apply effects directly to the audio on a permanent basis. Doing so frees up computer resources for additional effects processing elsewhere.

In the lower left area of Fig. 2, there is an area for finding and accessing files directly from within the application. Clicking on the new Media Pool tab organizes all the various media files associated with your project in one convenient location. Once in the Media Pool, you can also change how these files are viewed, add and remove files, and view and alter file properties.

For the audio post professional, Vegas Audio 2.0 provides support for Apple's QuickTime 4.0 and OpenDML AVI files. Assuming you have the drive space, you can now open and work with digitized video files larger than 4 GB. This feature makes Vegas Audio 2.0 a viable tool for adding sound effects for TV and film projects.

IN SESSION

Vegas Audio 2.0 will record as many mono or stereo tracks of audio while

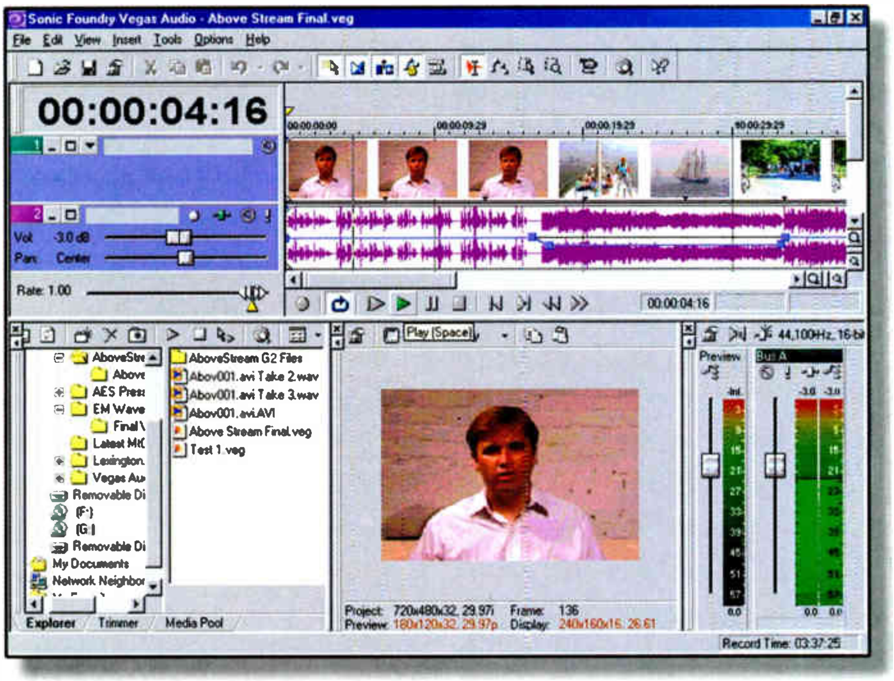

Figure 2: Window Docking in the lower portion of the screen helps the user organize workspace. Here, files can be accessed directly, within the application.

playing back existing audio tracks up to the performance capability of your computer and audio hardware. The factor that most directly affects simultaneous record capability is your audio card. If the card only supports four simultaneous inputs and outputs, then that's what you'll get.

Recording in Vegas Audio 2.0 is a nondestructive operation. Multiple takes can easily be recalled, auditioned and assigned to the track at a later time. You can think of these takes as audio layers. Punch-in recording is fast and easy to execute. After highlighting the desired record region (Fig. 3, track 1) with the Selection Bar (identified by two tiny yellow triangles), pressing "S" splits this event into smaller events. At this point, you simply drag the edges of the Selection Bar outward to create the Preroll and Postroll periods. Upon arming the track and clicking the Record button in the main transport area, playback commences from the Preroll point, and recording begins and ends at the edges of the highlighted section. The program automatically stops at the end of the Postroll.

Unfortunately, Vegas Audio 2.0 has no provisions for punch-on-the-fly recording—you must always tell the program in advance that you wish to record. Although this isn't likely to be an issue for everyone, it does, nonetheless, take some time to configure a punch. Perhaps Sonic Foundry could incorporate punch-on-the-fly into the next version.

Dedicated transport keys control playback, with a hot key to launch your sound editor application (the default is Sound Forge, if installed). The program has all the usual editing provisions you would expect to find—including Cut, Copy, Paste and Delete, along with unlimited Undo and Redo. A Trimmer window (see Fig. 1, lower left corner) can isolate material for these purposes. Data can be placed directly into this area from the Explorer or by right-clicking on an event and then sending that data to the Trimmer. This approach frees the user to work on a segment or full track without having to remember which tracks are muted, soloed, etc. With its hot-link to your sound editor application, this work area functions as if you had a dedicated editor operating as a plug-in.

A useful Window Docking feature keeps frequently used windows available, yet out of the way while you are working and can display up to three windows at a time. This lets you customize the look of the application, and is particularly useful when working with video files. While replacing music and dialog on an AVI file, for instance, Window Docking enables you to manage your workspace better. Without Window Docking, you would probably end up with a video preview window that interferes with your ability to see track data and bus faders clearly. By docking the video preview (see Fig. 2, lower section), you can view the video while maintaining access to the Explorer and bus faders.

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It was easy to ungroup the audio and video elements in this AVI file, clean up the audio, regroup the elements and render the enhanced file once again as an AVI file. Similarly, I could encode directly to RealVideo at a variety of bit rates. This application is wonderful for computer multimedia.

Vegas Audio 2.0 will scrub audio forward or backward at a constantly variable rate from—2x (reverse) through 2x (forward)—by grabbing the scrub control with the mouse and dragging right or left. It works smoothly and produces excellent results.

In addition to its myriad onscreen buttons and switches, Vegas Audio 2.0 also provides a comprehensive list of equivalent keyboard commands. While the point-and-click approach is quite comfortable during the initial learning phase, I was pleased that I could control many aspects of the program from the keyboard. Keyboard commands include basic transport functions, toggle between the Explorer and Trimmer windows, Cut, Copy, and Paste, resize track height, plus zoom in and out of event views. Sonic Foundry even provides a dedicated Keyboard Commands reference chart as part of the package.

MIXING

Vegas Audio 2.0 supports the grouping of multiple tracks, but they cannot be soloed or muted by selecting that group and then clicking either the Mute or Solo icons. This requires routing the tracks within the group—such as your drums and percussion section—to a dedicated output bus. Then, the entire bus can be muted or soloed. Although this is easy enough, it is an extra step that strikes me as odd.

I was impressed with the program's ability to mix internally. When you select Render to New Track from the Tools menu, Vegas Audio 2.0 processes the entire project to a new stereo track that it then places at the top of your track list. This sure beats mixing to DAT or some other external media, only to dump the material back into the computer for eventual mastering. When you select Preview in Player from the Tools menu, you have the opportunity to test various compression options when creating streaming media for the Internet.

SYNC AND EXTERNAL CONTROL

Vegas Audio 2.0 will slave to incoming MIDI Timecode (MTC), but it functions as a means of starting playback, or trigger sync. With this alone, Vegas Audio 2.0 and your external source (such as a stand-alone recorder) will ultimately drift. To maintain lock throughout the entire file, your audio card must be able to read incoming wordclock in addition to MTC. For slaving external devices to the program, Vegas Audio 2.0 outputs both MIDI Clock and MTC.

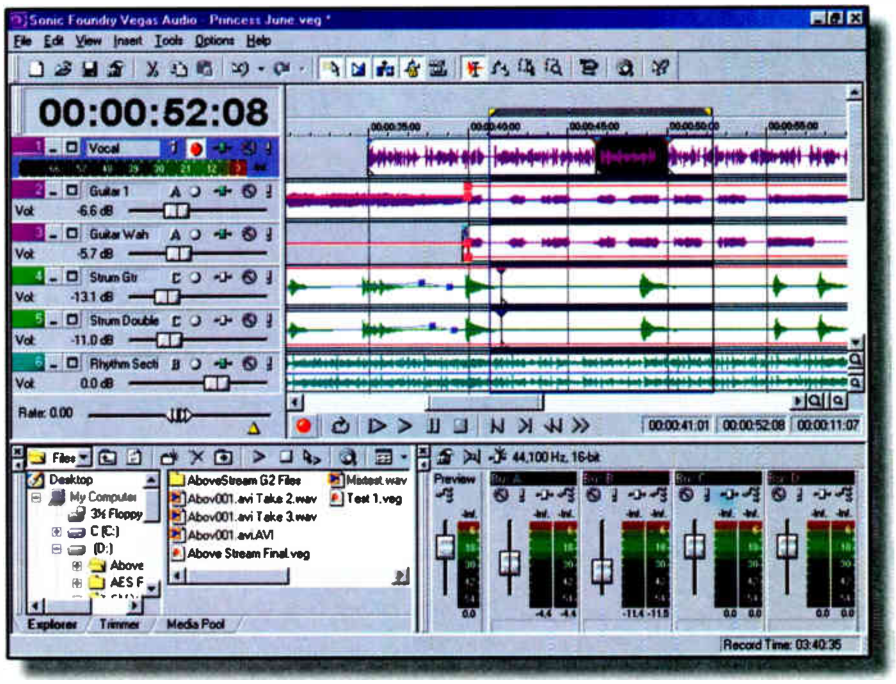

Figure 3: Recording in Vegas Audio 2.0 is a nondestructive operation, and punching in and out is fast and easy. Here, punch-in recording on track one is configured to include Preroll and Postroll. The selected area represents the audio data to be replaced.

With Vegas Audio 2.0, an external mixer no longer has such a vital role in the tracking and mixing process, but it is likely to be used for routing signals into the computer and as a monitoring device. Hence, it is disappointing that the application provides no support for MMC (MIDI Machine Control) transport control or MIDI Continuous Controllers. Compact mixers and control surfaces such as Roland's VM-3100Pro, Tascam's US-428 and CM Automation's MotorMix are ideal companions to this type of desktop environment. The ability to control the output buses via MIDI controller messages is something the company should implement at the very least.

VERDICT

Sonic Foundry's Vegas Audio 2.0 is a very capable program that is well suited to a variety of audio recording and editing tasks. I was impressed with its ability to handle music and dialog replacement on a video file. For the multimedia content creator, this program shines. Being able to preview various compression options for streaming media is extremely valuable, as is the internal mixdown feature.

I consider the inability of the application to respond to MMC transport and track arming commands or MIDI continuous controller messages for adjustment of the output buses to be its biggest shortcoming. Without this capability, no options exist for integrating the desktop production environment with more professional, dedicated hardware controllers. I'd also like to see punch-on-the-fly recording implemented.

Vegas Audio 2.0's ability to simultaneously accommodate multiple file formats and sampling rates can be a tremendous time-saver if you're grabbing music tracks and sound effects from multiple sources. Both the owner's manual and online documentation have improved considerably with this new version. Further, the program ships with a very useful tutorial CD. At a retail of $499, Vegas Audio 2.0 is a versatile, stable and intuitive recording tool that will serve a multitude of audio professionals very competently.

Sonic Foundry, 1617 Sherman Ave., Madison, WI 53704; 800/577-6642; www.sonicfoundry.com. ■

Roger Maycock is a former technical consultant for Mix.


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FIELD TEST

NEUTRIK MINILYZER ML1

HANDHELD AUDIO ANALYZER

Have you been avoiding the plunge into the world of geekdom? Neutrik has solved that problem by packing an arsenal of the most essential and powerful measurement tools into an affordable, easy-to-use package called the Minilyzer ML1. This is a pocket-sized audio analyzer designed for the many times you've needed to investigate an audio mystery but were afraid to call a technician.

Beyond the primary capabilities of this powerful little box—detailed in the seven major headings below—the ML1 includes a handful of little touches that make it as audio-friendly as a piece of test equipment can be. Test equipment is often unbalanced, a serious issue when trying to measure a balanced device under real-world conditions. The ML1 includes both balanced XLR and unbalanced RCA inputs. Three AA batteries supply power, automatic turn-off time is adjustable, and four user presets allow the ML1 to boot into your favorite mode. There is even a headphone jack.

Of the seven main functions, many are intermixed within the large, easy-to-read, backlit LCD screen. For example, the LCD is large enough to display "level" in large bold characters, while simultaneously including an analog-style linear meter at the bottom, a frequency counter in the upper left corner and an inspired "input balance" indicator in the upper right corner (see Function 5).

Note: Despite its piccolo footprint, the ML1 is quite capable of getting you into trouble. Translation: It is just as easy to take a bad measurement as a good one. Read the sidebar, "The Ultimate Test," to learn how to compile real data.

FUNCTION 1: LEVEL

The ML1 displays the RMS value of a sine wave test signal in millivolts (mV), as well as the relative level in dBu and dBV, referenced to 0.775 volts and 1 volt RMS, respectively. Additionally, the ML1 has a "relative" mode for precise comparisons of two or more signals—left and right, for example—with a signal-to-noise ratio as wide as 119 dB!

For years, technicians purchased the Fluke 8060A ($479 at www.imcm.com) for its ability to measure RMS volts and dBu. The Fluke doesn't do dBV; its noise floor bottoms out at -74 dBu (compared to -99 dBu for the ML1), and it's too old to have the DSP do all the cool things that the ML1 can do. Because the Fluke is a volt-ohm-mil-

THE ULTIMATE TEST

The carpenter's adage "measure twice, cut once" couldn't be more applicable to the art of acquiring good data. Measuring nominal level from a device is child's play, but measuring signals at or into the noise floor is quite another. This is not a fault of the ML1. Several attempts were made until the results were consistent. You might not think much about "impedance," but when taking measurements, both the source and the destination impedance must be addressed.

When measuring mic preamp performance at both minimum and maximum gain, the gain structure between it and the sound source (an oscillator) becomes quite critical. You can't simply turn down the level at the oscillator, because, in some cases, a buffer amplifier follows the level control. The oscillator's output amplifier has plenty of noise when looking down the high-gain barrel of a mic preamp. The top of Table 1 proves that reducing the level of the Minirator increases noise revealed in the THD+N measurement. Fortunately, the Third Octave analyzer helps to "see" the various noises.

I fashioned a quickie attenuator between the oscillator and the preamp using a 1-kilohm pot across pin 2 and pin 3 of the female XLR. Pin 3-f was connected to pin 3-m, while the wiper from the pot was connected to pin 2-m. With the Great River preamp set to max gain, I adjusted the trim pot so that preamp output fell just under +20 dBu (+19.9 dBu), keeping in mind the ML1's inability to tolerate anything higher. I then checked the attenuator output (-45.7 dBu) and calculated the gain to be 65.4 dB.

Once the gain was determined, I removed the generator and attenuator and connected a source impedance that represented the average mic. I chose 150 ohms, because it was used to create the preamp's published specs. To calculate Equivalent Input Noise, EIN = the noise floor + the gain setting. An older style signal-to-noise ratio (S/N) measurement was also made. S/N = the max output before clipping - the noise floor. ■

BY EDDIE CILETTI

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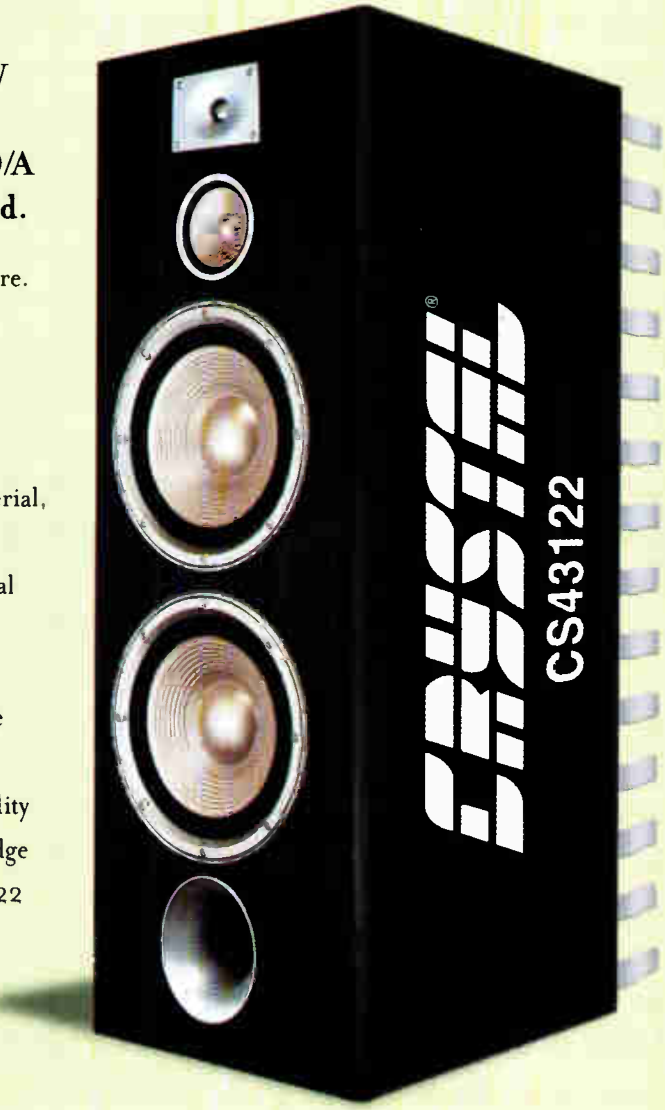

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CIRCLE #064 ON PRODUCT INFO CARD

liamp meter, it has no upper level limit—compared to the ML1's 7.75-volt/+20dBu ceiling. I would like to see this ceiling raised 10 dB.

FUNCTION 2: THD+N

When measuring Total Harmonic Distortion + Noise (THD+N), the "+N" implies that noise is included as part of the measurement process. How can it not be? Unless pushing a device purposefully into the red, you hope that both distortion and any noises—hiss and hum—are way down in the subbasement of random electron movement. The only way to separate distortion from noise easily is to use the built-in Third Octave analyzer, which will be discussed under the heading "Function 7."

Inside the ML1 is an A/D converter capable of 119 dB of dynamic resolution. The sample rate limits bandwidth (frequency response) to 20 kHz, as pointed out by Neutrik in the manual. This can potentially yield different (most likely lower) readings relative to more sophisticated gear, but only when in the Linear (flat response) mode. Analog test equipment is not bound to the sample rate. The Hewlett Packard Model 8903B Audio Analyzer, for example, has a 30kHz "window."

The ML1 also includes built-in filters so that the measurement window can be "weighted" (bandwidth restricted) to include what is relevant to the test. For example, the popular A-weighted filter (as per IEC 651) is often used, because it reflects the ear's sensitivity to noise. The other filter options are C-Message (IEC 468-4), HP22 (highpass at 22 Hz), HP60 (highpass at 60 Hz), HP400 (highpass at 400 Hz) and a Voice-band filter. These filters are available in the Level, THD+N and Third Octave modes.

Note: Searching the IEC on the Internet for specifications yielded nothing—their search and destroy mechanism is about as useful as Microsoft Help, *and* they want money for documentation I couldn't find. If you want an IEC clue, go to the heading "Function 6."

I used the ML1 to measure the THD of two oscillators—the GTC Tone Plug and the Neutrik MR1 Minirator—and two mic preamps—a Great River transformerless prototype and an Altec 1566 tube preamp. The Tone Plug is a handy "generator in an XLR plug" commendable for its size, not cleanliness. (Check out Table 1 for the results.) While the Minirator is respectable for its price range, to truly measure the Great River's performance, a better oscillator would be required. As you can see, there is almost no difference in the performance of the MR1 alone compared to its use with the Great River preamp.

FUNCTION 3: VU+PPM

The ML1 emulates three metering standards: mechanical VU meters (referenced to +4 dBu), Type I and "Nordic" Peak program meters (PPM, +6dBu ref), and Type IIA PPM (+8 dBu). The user can reconfigure all references. Both VU and PPM are simultaneously displayed. Each includes a numeric peak hold indicator, plus there are two integration time options: normal (Type I and

ML1: SAMPLE TEST MEASUREMENTS

Test	THD+N dBu / % @ "x" level	Signal-to-Noise	EIN = noise floor + gain	Notes
Minirator MR1: 1 kHz	-78/.013 @ +5.9 dBu -71.5/.026 @ -16.1 dBu			1
GTC Tone Plug @ 98 Hz	-26/4.663			
GTC Tone Plug @ 1 kHz	-32/2.389			
Minilyzer ML1		119 dB	-98.5 dBu (noise floor)	
GR @ 24dB gain (1 kHz)	-78/.013 @ +20 dBu	107.5 dB	91.3 + 24.2 = 115.5 dB	2
GR @ 65dB gain (1 kHz)	-72/.024 @ +20 dBu	109.1 dB	60.2 + 65.5 = 125.7 dB	
Altec @ 24dB gain (1 kHz)	(a) -60/.09 @ 0 dBu		79 + 24.2 = 103.2 dB	3
Altec @ 65dB gain (1 kHz)	(b) -51/.28 @ 0 dBu		53.5 + 63.4 = 116.9 dB	
Front-end overdrive (1 kHz)	(c) -30.6/2.96 @ 0 dBu		2 kHz@-31 dB, 3.15 kHz@-51 dB	
Panasonic SV-3700 DAT	-6 /.04	89 dB		

NOTES:

1. Using the Minirator's built-in level control, two tests were purposely made at different levels NOT to reveal distortion, but to show how output amp noise contributes to the "+N" of THD+N. Using an external attenuator can minimize oscillator amplifier noise.

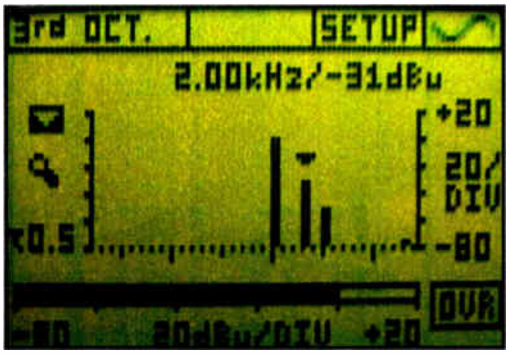
2. Maximum input to the Minilyzer is +20 dBu. Maximum output of the Great River preamp is +24 dBu. Signal-to-noise measurements were "assisted" by the Fluke 8060A.

3. The Altec 1566A preamp is "clean," up until 0 dBu. Preamp overload (test "c") yielded the most "pure" second harmonic distortion.

The chart compares measurements of preamps, oscillators and a DAT machine made with the ML1. The oscillators—Neutrik's Minirator MR1 and GTC's Tone Plug—were reviewed in the February 1999 and the June 2000 issues of Mix.

Nordic: 5 ms. Type IIa: 10 ms). In Fast mode, the integration time is 1 ms for all standards.

While observing the output of a Panasonic SV-3700 DAT, I immediately realized that the ML1 could use one additional metering standard capable of being calibrated to digital audio's 0dBfs maximum. For example, the SV-3700 has a -18dBfs nominal reference; the range should accommodate a "low" of -20 dBfs and a high of -10 dBfs. Neutrik could probably turn the VU+PPM function into a stand-alone stereo product with both analog and digital inputs (and a larger LCD screen). It would be a helpful mastering tool for viewing accurate peak information, while maintaining some consciousness of "volume" per the VU meter. The VU meter should not be solidly in the red while the PPM would be kissing 0 dBfs.



Even-order harmonics are generated by asymmetrical distortion as is common for some vacuum tube circuits. The cursor is over the second harmonic.

FUNCTION 4: POLARITY

The polarity test requires both the MR1 and the ML1. The MR1 generates a pulse that is easily detected even after traveling through the air. Selecting Polarity on the ML1 engages the input select option—a choice of either the XLR/RCA connectors or the built-in microphone. It works!

FUNCTION 5: BALANCE

Quite unexpectedly, the very first "if only" feature I thought of was an "input balance" indicator—not left and right, as this is a mono box—of the incoming signals on pin 2 and pin 3. This feature is a reality on the ML1, putting this gizmo and me on the right foot from the very beginning. A 6dB level problem in the analog world is not uncommon—active balanced outputs can become damaged or interrupted via a dirty patch cord or bad cable—and the ML1 will tell which pin isn't doing its share of the work.

Note: The ML1 "loses" what little headroom it has if the signal is not precisely balanced. I noticed this when testing the SV-3700, whose pin 2 and pin 3 outputs were particularly unmatched, reducing the max headroom in this case to +19.3 dBu.

FUNCTION 6: SWEEP

Sweep has two options, the traditional RMS level vs. frequency, or time vs. any of the following: level, THD+N or frequency. Getting this mode to function was most difficult, and the manual was not perfectly clear (perhaps due to translation). I got results simply by copying the example in the manual. Here, a picture was worth a thousand words. The best example would be to plot THD+N to show how distortion increases with increased levels.

FUNCTION 7: THIRD OCTAVE

The third octave analyzer can display the audible bandwidth from 20 to 20k Hz in 31 bands. As mentioned, THD+N does not separate distortion from noise. Harmonic distortion is a lack of sonic cleanliness relative to the input signal. *Some* vacuum tube gear is famous for its pleasing even-order (octave) harmonics. Input 1 kHz, for example, and push the device into its nonlinear region (not hard clipping) and watch the second harmonic (2 kHz) pop up. See Figs. 1 and 2.

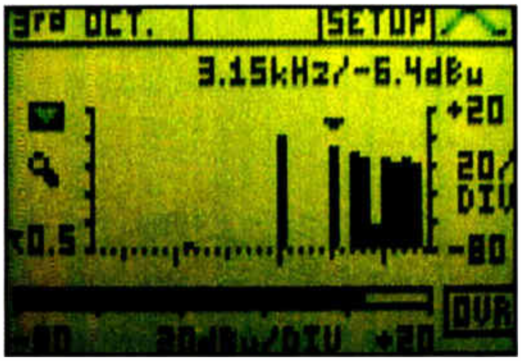
The cursor can be moved to each of the 31 bands to confirm both frequency and amplitude, the latter in both dB and %. An op amp circuit in hard clipping will produce odd-order harmonics (square waves are made from these), and the third harmonic of 1 kHz is 3 kHz. See the Altec preamp specs in the table on page 138.

FUNCTION 8: SCOPE

If you read my article on troubleshooting bad capacitors using square waves, the "oscilloscope-like" waveform display is the icing on the cake. Of course, the perfect companion to the ML1 is the $139.95 MR1 Minirator, featuring both sine and square waves. As with most LCD scopes, the ML1 does not have amazing resolution, further hampered by the 20kHz bandwidth, which softens square waves until 10 kHz looks like a sine wave. That's okay. About the cost of a cheap-but-real 'scope alone, the ML1 is much more likely to see active duty.

GOOD DADA

Reviewing the ML1 was a good brain exercise, emphasizing the relative ease of making a bad measurement compared to the work involved to acquire good data. That said, it would be helpful if Neutrik offered an external attenuator as an optional accessory to pad the output of the MR1 (or any oscillator) so that noise measurements would reflect the device under test and not the audio source. Otherwise, there is only one flaw. The maximum input of the ML1 is +20 dBu, which is not high enough and easily compromised if the signal balance is not perfect.



Odd-order harmonics are created when op amps hard clip. A 1kHz square wave produced this signature family of harmonics. The cursor is over the third harmonic.

The ML1 is small enough to be kept in a control room or clipped to a belt for those technicians on the move who are also looking to start a fashion trend. (PDAs might suddenly become less cool!) Consider how many times you've returned to the scene of an audio crime only to find no suspects and no problem? Now you can whip out this nifty little "tricorder" whenever a problem occurs. By creating the Minilyzer, Neutrik has given more people the power to troubleshoot. The more you use it, the more you'll understand that a little science never hurt anyone. If you weren't lucky enough to get one of these in your Christmas stocking last month, you can pick one up for about $399.

Neutrik USA, 195 Lehigh Ave., Lakewood, NJ 08701; 732/901-9488; fax 732/901-9608; www.neutrikusa.com. ■

Eddie Ciletti had too much fun reviewing the ML1. He's not giving it back! If he gets away with it, you'll see some cool pictures at www.tangible-technology.com. WOOF!

SABINE GRAPHI-Q

MULTIFUNCTION PROCESSOR

Sabine's Graphi-Q is a true multifunction digital processor, offering front panel controls for 31 bands of graphic equalization, plus sweepable high- and lowpass filters, 12 FBX Feedback Eliminator filters, up to one second of digital delay, and a compressor/limiter with ratio, threshold and gain. Graphi-Q comes in four models: single- and dual-channel GRQ-3101 and GRQ-3102 ($799 and $1,299 list), and both are available as Slave versions without front panel controls ($699 and $1,099 list). The single-channel models have a second output that provides all the processing *except* the delay.

After a decade of refining Sabine's FBX processors, the latest Feedback Exterminator algorithm is the best yet. Other imitators are poor substitutes—don't waste your money. The Graphi-Q is based on Analog Devices' 32-bit, floating-point SHARC DSP chip. Its 24-bit conversion and 110dB dynamic range compare favorably with a popular analog programmable parametric equalizer I was using in series when I first tried it. I since switched over to exclusively using Graphi-Qs for mix control.

Turbo mode is a setup procedure for automatically setting the FBX filters by "ringing out" a mix before it is used. It sets a limiter, increases gain and reduces output so the feedback that occurs while the filters are set is not as loud as if they were set manually. There are two kinds of FBX filters, Fixed and Dynamic. Once set for a frequency, the Fixed filters stay there and can be locked so they aren't driven deeper. Dynamic filters, once they're used, are reassigned to new frequencies as needed. The default setting (which I recommend for most monitor applications) is nine Fixed and three Dynamic filters and a width of 1/10 of an octave.

At first try, the Graphi-Q's 20mm sliders seem too short to do meaningful work, but as each adjustment is made, the cut or boost is shown in the delay's display, a feature called "Tweak-n-Peek." Other interesting features hidden in the manual include switching the sliders from 12 dB of cut and boost to 6 dB, with ½dB resolution. On the 2-channel model, pushing all the faders down all the way on Channel 1 makes the Channel 2 sliders their master.

Remote Control software provided with the Graphi-Q allows control of up to eight Graphi-Q units (up to 16 channels) from a single Com port using serial cables for RS-232 communication. Each additional unit is simply chained from the previous one with a DB9 male-to-female cable. Computer requirements are a Pentium 100 with Windows 95, so even five-year-old PCs will work.

The Remote software offers access to additional features. Width of the graphic filters can be squeezed down from the default setting of one octave wide to half an octave, which I prefer for monitor work, and cut/boost resolution increases to a half dB. Other software-only accessible features include the ability to change the FBX filters to standard parametrics on a filter-by-filter basis. Global parameters for the FBX filters can be adjusted for sensitivity, persistence and maximum filter depth. The high- and low-cut filters can be changed from the default 12 dB/octave to a steeper 24 dB. The software also provides additional attack, release and knee parameters for the compressor and independent adjustment of a limiter threshold. Internally, the Graphi-Q has three snapshots and 66 program memories. A rear panel, 8-pin Phoenix connector allows contact closures to choose any preset or the front panel controls. The software also allows powerful absolute or relative linking of parameters in multichannel systems.

Device Control via SIA's Smaart Live software is where the Graphi-Q really shines. Initially, I hesitated to use a computer to control the Graphi-Q, preferring to reach over and grab the sliders with my hand. However, to see a speaker's true response in Smaart's transfer mode, place a filter onto a peak by clicking the mouse and then watch the result. It is priceless. The Smaart device control module that SIA Software has crafted is outstanding. Almost anything you can do from Sabine's software is also available in Smaart. Filter resolution is increased further to make even finer amounts of cut or boost.

The old argument about graphic

MARK FRINK

vs. parametric filters is well-known to most of us. Graphic filters give us a familiar menu of tones we can easily identify, while parametrics are the only precise way to accurately smooth speaker and room anomalies. The Graphi-Q offers the best of both worlds and something for everyone. For monitor applications, I chose its graphic filters for basic equalizing and then ring out with the FBX filters. If there are large peaks centered on nonISO frequencies that need to be tamed, then I can switch one or more of the 12 FBX filters to become a simple parametric filter. For FOH applications where FBX filters aren't quite so important, more of the 12 filters can be used as parametrics to tune the speakers and the room, leaving the graphic filters for artistic tweaking or tone shaping that FOH engineers normally do.

Using the Graphi-Q without a computer offers challenging, but very workable, basic front panel controls for a graphic EQ, a delay and a compressor, plus fabulous FBX filters. Add a computer and get all the features of a powerful processor that you'll wonder how you lived without. I've long since gotten over EQ'ing with a mouse, which is outweighed *by far* by all the advantages a system of remote-controlled Graphi-Qs offers.

The combination of Smaart and Graphi-Q is a powerful tool that allows precise control of the speakers in my monitor system. The edge that FBX filters provides gives me extra gain-before-feedback and an invisible, lightning-fast hand to control the squeals of the occasional unforeseen accident. Presoundcheck time that I used to spend barking into a microphone is better invested getting an accurate picture of each speaker's response, tuning them to target curves and aligning the response onstage to work coherently as a system in conjunction with the main speakers. My front-of-house engineer often asks me, "Aren't you even going to talk into the vocal mic?" "Of course I am—I need to make sure it works!" I can't imagine pointing a speaker at a microphone without a Graphi-Q in between. You FOH guys might dig it, too, but get your own!

Sabine, 13301 Highway 441, Alachua, FL 32615; 904/418-2000; fax 904/418-2001; www.sabine.com. ■

Mark Frink is Mix's *sound reinforcement editor and is currently on the road with k.d. lang.*

SNAPSHOT PRODUCT REVIEWS

VIEWSONIC VA800 ▲
Flat Panel Display

Once available only to the rich and famous, large-format, flat panel computer displays are fast becoming more affordable. A recent entry in this field is the ViewSonic VA800, a sleek, lightweight unit that is compatible with Macs and PCs and offers a full 17.4-inch display with true 1280x1024 resolution.

As the studio environment becomes more and more populated with computer monitors, engineers are discovering the drawbacks of multiple displays in the control room. Besides the nasty effects of electromagnetic and radio-frequency interference when guitar or bass pickups are used too close to a typical CRT, monitors emit an annoying 15kHz whine and can generate a surprising amount of heat in a confined space. Placement is another issue—the weight of single or multiple 17- or 19-inch glass display can be hazardous to your meter bridge.

Weighing in at 18 pounds and only nine inches deep, the VA800's LCD is well-suited to the studio environment. Its 17.4-inch screen offers nearly as much area as a traditional 19-inch monitor and is free from the weight, footprint, heat and magnetic interference issues of standard glass designs. The display is sharp and bright with a useable horizontal or vertical viewing angle of 160° (±80° off-axis). Front panel buttons allow switching between two connected computers—a Mac and a PC in my studio. One thing I found somewhat odd was the small in-line external DC power supply—a fairly new concept for monitors. A wall-mount kit is an option that may be handy for some installations. The VA800 carries an affordable list price of $2,000 (retail is around $1,599), so there might be a flat panel display—or two, or more—in your future.

ViewSonic, 909/869-7976; www.viewsonic.com.

ART HPFX ▼
Headphone Monitor System

Sometimes technology creates as many problems as it solves. Computer-based recording systems offer excellent price/performance ratios, but the latency delays that occur when overdubbing on soundcard-based workstations can prove disastrous for operations as simple as running a touch of digital reverb to a performer's headphones.

The half-rack HPFX combines a 2-channel mic splitter and digital effects processor with a small mixer and four individual headphone outputs, which allows announcers or vocalists to set up an independent headphone mix while the dry mic signal is sent directly to the preamp or main mixer feeding the workstation. Two ¼-inch input jacks accept a stereo cue feed, so all the performer needs to do is set his or her own levels for mics, effects and overall volume while the studio engineer or workstation provides a backing track mix. No phantom power is available, but condenser mics can be powered from their preamps or the mixer—the split through the HPFX is passive and does not affect phantom powering.

In session, the HPFX worked perfectly, allowing artists to tweak their own headphone mixes—they love turning knobs, and it's simple enough for any singer to figure out. However, the HPFX also works just as well in "traditional" analog environments for recording or broadcast production, and if desired, the 24-bit effect processor can be used as a stand-alone unit to provide stereo reverbs or delays during the final mix (via a stereo TRS output jack). The headphone amp section can be used by itself in a variety of other situations. At $299 retail (with a three-year warranty), the ART HPFX is a useful problem solver that should find its way into all types of recording environments.

Applied Research & Technology, 716/436-2720; www.artproaudio.com.

SABRA-SOM SPK
Mic Mounting System

Described as a "Noise Protection System for Microphones," the SPK from Brazilian manufacturer Sabra-Som is a combination of products that form a versatile "Erector Set" to deal with mic placement, shock-mounting and pop filtering issues in the studio. The system includes the ST2 universal T-bar double support with ⅝-inch thread, an SSM-1 mechanical noise suppressor support and the SPF pop filter. Ad-

BY GEORGE PETERSEN

ditional modules or accessories can be added to create customized setups, or to add a second mic for M-S or conventional stereo arrays.

The SSM-1 shockmount consists of two rings with elastic bands that accept mics of up to two inches in diameter. The unit was effective in isolating mics from noise, rumble and vibrations and held most mics securely, though using larger studio mics will require that the lower shockmount ring is attached over the XLR input to prevent the mic from

sliding downward. Also, heavier, tapered mics, such as U87s, cannot be used *upside-down* with the SSM-1, because the mic can easily slide out. By mounting a second SSM-1 (optional) to the ST-2 T-bar, the system will accommodate even the longest shotgun mics. Additionally, the T-bar can be used alone as a stereo mic mount with a maximum 8-inch spread, ideal for X/Y arrays. All components attach to (non-rotating) hex rods for a solid lock.

The SPF (Sabra Pop Filter) is a dual-layer screen design that effectively removes breath pops and plosives, while protecting your classic mics from the onslaught of performer's saliva, etc. The SPF mounts on an articulated arm that swivels and locks precisely and securely into position, unlike other pop filters that use gooseneck-type mounts.

Priced at $129.95, the Sabra-Som SPK offers a flexible and elegant solution for all your mic hang-ups.

Sabra-Som, dist. by K-IV Enterprises, 201/828-9492; www.sabrasom.com.br.

KURZWEIL PC2R

MIDI Sound Module

Kurzweil is no stranger to creating great-sounding MIDI sound modules, and the PC2R—which puts the power of the company's latest PC2 keyboard in a single-rackspace chassis—definitely continues the lineage.

Features of the PC2R include 64-voice polyphony (expandable to 128), up to four split/overlapping zones under MIDI controls, stereo analog and 24-bit/48kHz digital output (AES or S/PDIF switchable), and real-time adjustable, dual onboard effects processors. A multiple bus architecture allows separate send levels per MIDI channel for effects and reverb levels. Effects include reverb, chorus, delays, distortion, rotary, phasing, flanging and dynamics.

Among the PC2R's 272 programs are useful presets such as triple-strike stereo grand pianos, newly recorded multistrike classic electric pianos, brass sections, drums, percussion, voices, Kurzweil's KB3 modeled tone wheel organ, mallet percussion, guitar, bass, Clavinet, harpsichord, synth sounds and more.

At a retail of $1,295, a few corners were cut, but fortunately not in sonic quality. The PC2R uses an external power supply, but at least it's an in-line transformer rather than a wall wart. But other than this minor hurdle, the PC2R was pure joy.

The triple-strike pianos are far better than anything Kurzweil has offered in the past—no easy feat—and are rich and full with natural decay and sustain that must be heard to be believed. The Rhodes and Wurlitzer electric pianos are also excellent, and the KB3 (B-3 tone wheel emulations) scream with nine drawbar settings, percussion control and realistic rotary speaker emulation. I was far less impressed by the PC2R's guitar samples—these are definitely *vin ordinaire*—but the strings, vocals (both choirs and Kurzweil's acclaimed Take 6 vocal samples), basses and percussion sounds, including traditional drum sets, Latin sounds and tuned percussion (vibes, marimba), are superb. The latter showed excellent punchy dynamics, long cymbal sustains and enough interesting new sounds to inspire creativity—just try "Virtuoso Percussion," "Woody Marimba" or "Aborigine Jam," and you'll see what I mean.

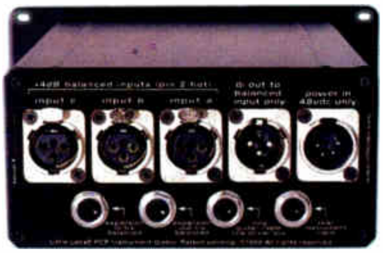
Anyone looking for lots of clean, useful studio sounds should give the PC2R a listen.

Kurzweil, 253/589-3200, www.kurzweilmusicsystems.com.

LITTLELABS PCP INSTRUMENT DISTRO

Guitar Splitter/Direct Box

There are too many "me too" products in the industry, so when something really different comes along, I'm inter-



ested. A good example of this is the Littlelabs PCP Instrument Distro, a unit that combines the functions of a direct box with a one in/three out guitar splitter providing transformer isolated, guitar level/high-impedance outputs, each with individual controls for routing, phase reverse, ground lift and level matching.

In its most basic incarnation, the PCP Instrument Distro can function as a high-quality direct box, but rather than providing a mic level output signal, the unit's XLR output is a +4dB balanced line for connecting directly to tape machines, consoles or pro out-

board gear. (By the way, the "PCP" part of the name stands for "Professional-to-Cheesy-Pedal.") The rear panel has three +4dB balanced XLR inputs, which can be summed together—with or without the instrument input—to feed three front panel, ¼-inch, high-impedance/guitar level outputs for routing any signal to guitar pedals/effects and or amplifiers for "re-amping." Additionally, a ¼-inch line driver output for driving long unbalanced cables allows the user to play guitar in the control room, while using amps in a distant iso booth or main studio. A pair of TRS expansion jacks on the back panel connect to a second PCP box, offering up to six instrument level outputs.

The PCP Instrument Distro is housed in a half-rack chassis (a rackmount kit is optional) with an external power supply. The front panel is laid out in a simple grid of pushbuttons that select which of the three XLR line inputs and/or instrument input is routed to each hi-Z instrument level output jack. Every switch has a corresponding two-color LED that shows all connections at a glance.

Whether tracking or mixing, the PCP Instrument Distro proved absolutely amazing, providing a degree of flexibility that's both creative and fun, especially to anyone who does a lot of guitar recording and has access to a variety of amps and pedal effects. On one session, routing a hat track through a Mu-Tron Bi-Phase into a little Danelectro 10-watt harmonica amp added just the right touch of weird to an otherwise straightforward rock mix.

Also, the possibilities of re-amping existing tracks is very cool. For example, after laying down a nice, clean DI-bass track, the PCP Instrument Distro let me route the audio back into the studio, connected to two miked amps—in this case, an Ampeg and a Fender Bassman. From there, I could choose from any combination of the three sources (DI/Ampeg/Bassman) during the mix.

Priced at $950 (including custom case), the PCP Instrument Distro offers a nearly endless variety of studio tricks. Besides introducing an entire Pandora's box of strange stomp boxes into your studio-mixing palette, with the PCP Instrument Distro, I suddenly have tons of outboard gear I never used before! I like this.

Littlelabs; 323/851-6860; www.littlelabs.com. ■