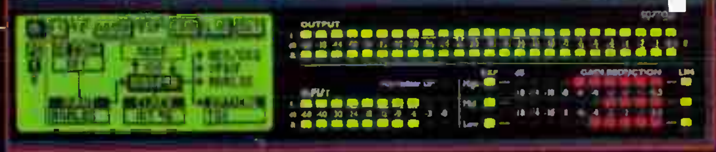
Hardware Mastering Processor **Roundup**



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JUNE • 2000

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Kind of Loud RealVerb

and more

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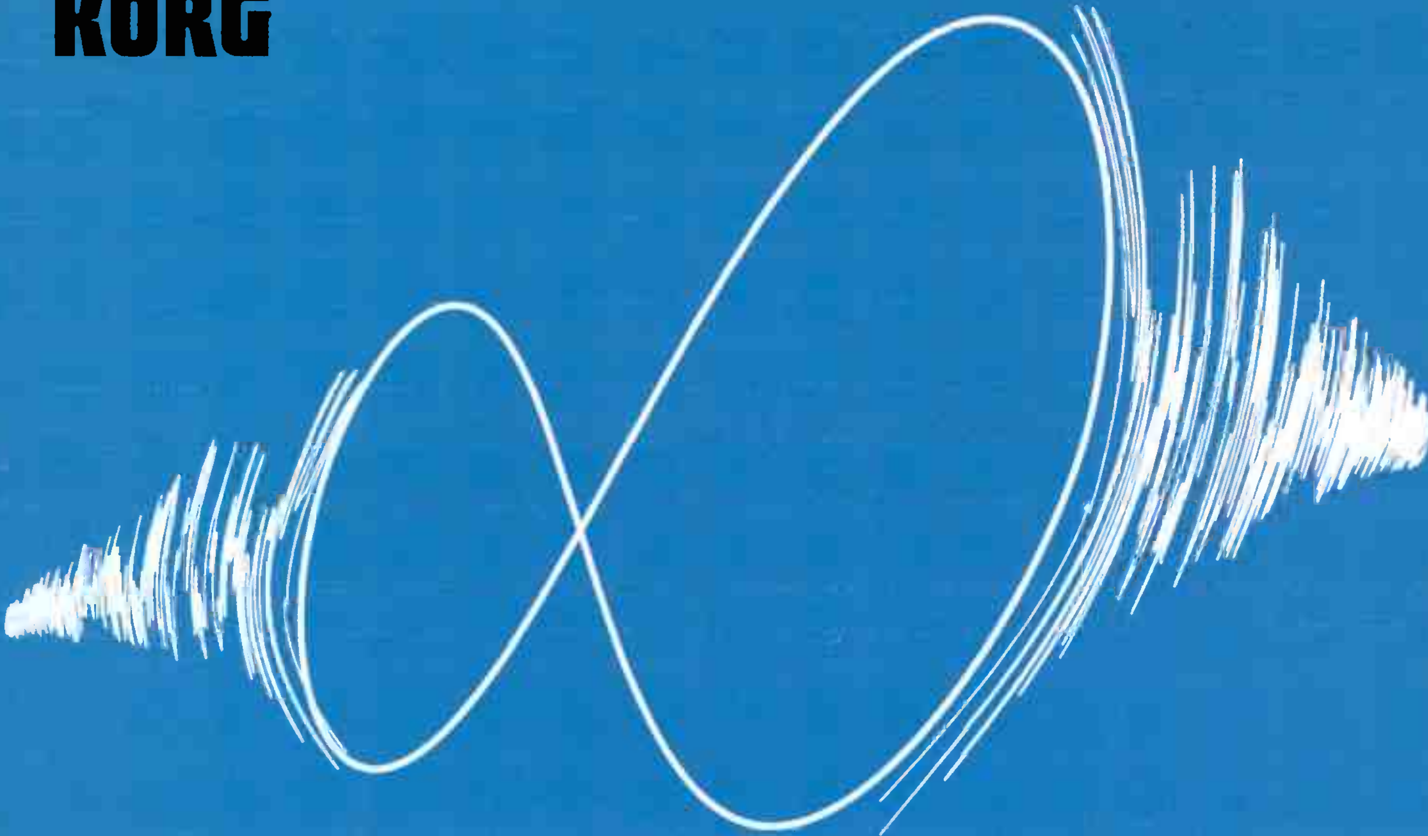

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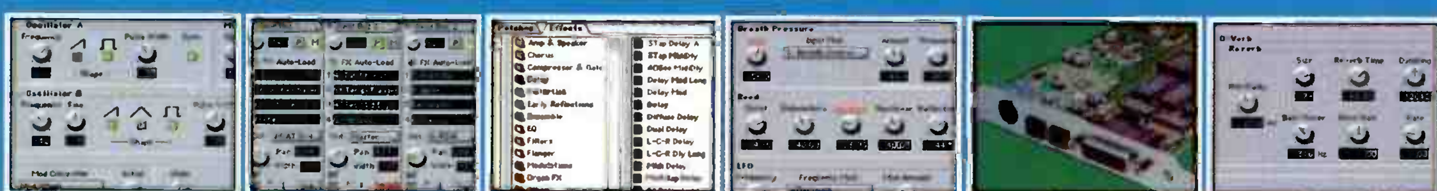

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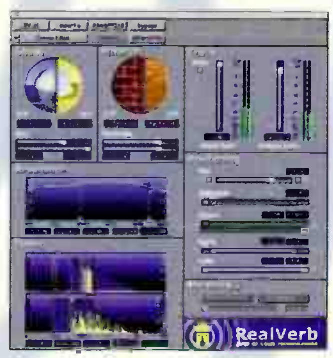

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Sound Designer/Mixer
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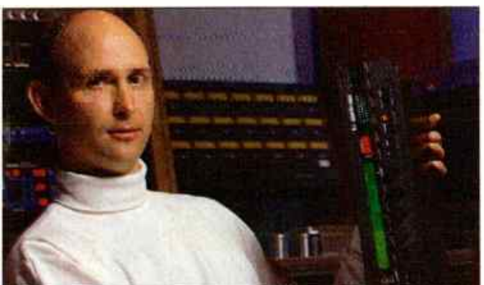
VOLUME 11, ISSUE 6
JUNE 2000



FEATURES

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COLUMNS/DEPARTMENTS

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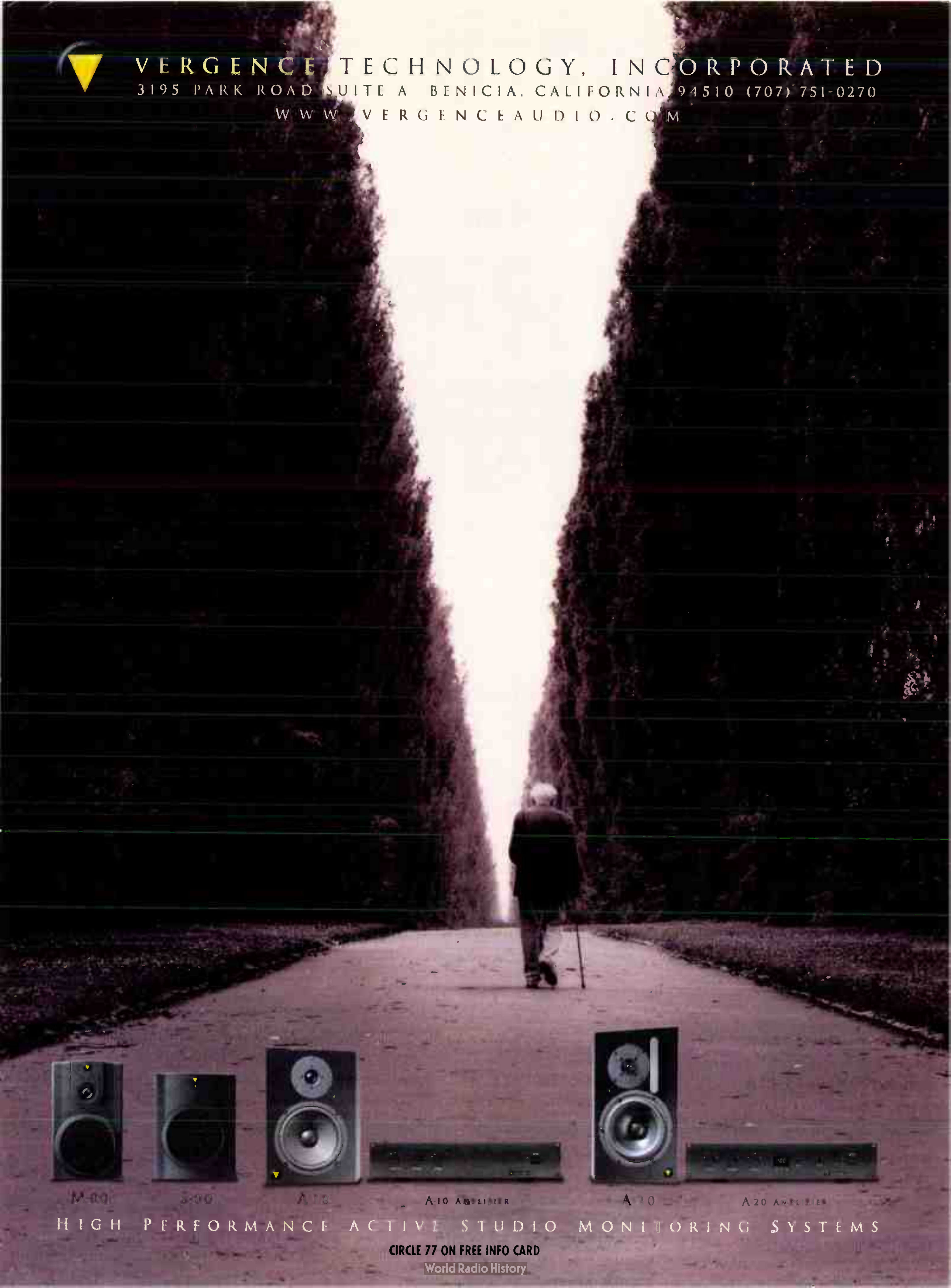
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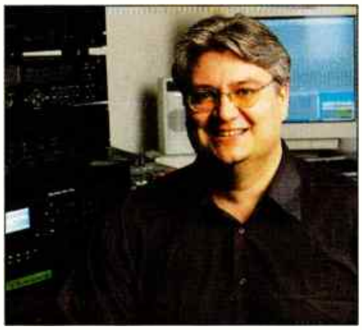
From the **Editor**



I'm not much of an outdoorsman. The last time I went camping as a Boy Scout, I stuck an ax in my left foot while chopping wood. This had a deleterious effect on my lust for outdoor adventure, to say nothing of severely compromising the integrity of my shoe, sock, and foot. Fortunately, the ax was left unscathed. As far as I'm concerned, the great outdoors were created primarily to keep buildings from banging into one another.

Still, one of my favorite humor authors, Patrick McManus, writes primarily about outdoor activities such as hunting, camping, and fishing. In his book *The Night the Bear Ate Goombaw*, Mr. McManus discusses the danger of *sequences*. A sequence, in McManus's parlance, is a chain of events necessary to accomplish a task that becomes so convoluted as to prevent the task from ever being finished — you're better off just going fishing. For example, let's say you want to trim your hedge. In order to do so, you'll have to borrow your neighbor's hedge trimmer. But if you borrow his hedge trimmer, he'll want you to return his lawn mower, which means you'll have to mow the lawn first. To mow the lawn, you'll need to get some gasoline to run the mower, requiring a trip to the gas station. On the way to the gas station, you really should...you get the idea.

Recently, I've been reviewing a bunch of software. Some of those reviews appear in this issue, others will show up over the next few months. In the course of doing those reviews, it seemed like a good idea to install a G4 accelerator in my G3 Mac. The newer Macs open easily, and it's a piece of cake to drop in a ZIF card accelerator. But in order to take best advantage of the accelerator, I found I should upgrade from OS 8.6 to OS 9. No problem, I had the required system CD-ROM at hand. Before proceeding, I made a backup of my current system — not that I'd need it. Soon the new system was installed, the Mac booted, and everything seemed okay. My mundane software (Word, Excel, Photoshop, etc.) all worked fine. My music/audio software wasn't so lucky — several pieces lost their authorizations, while other things wouldn't even start up. In the course of reinstalling them, it became clear that I should update everything to current versions to take advantage of the G4 Velocity Engine. Off to the Web to download the latest versions. As I installed them, many asked for new challenge/response authorizations; the old ones (which I'd kept in pack-rat fashion) were worthless. Off to the manufacturers for new responses. Meanwhile, each time the software was booted, it merrily stated, "Listen mister, I'm only going to run for a few more days. Authorize me, you moron." No problem, I'm used to being taunted by inanimate objects. But in one case, the engines for several products launched together, and attempted to taunt me simultaneously. This handily locked up the computer. One solution seemed to be to de-install, then reinstall separately with the new challenges. But, you guessed it, when I reinstalled, different challenge/response codes were required. Back to the manufacturers, *again*.

The sequence continued, *ad nauseum*. After three long days, much creative language usage, and a bit of soft sobbing, I gave up and reinstalled OS 8.6 from my backup. Yes, this required yet *another* trip to the manufacturers for challenge/response codes. As I write this four days later, my Mac is almost back to where it was before I succumbed to the sequence. I should have just gone fishing.

There's an old guitarist's cliché: "If I ever get this guitar in tune, I'm going to weld it." That can now be modified to: "If I ever get this computer working, I'm going to weld my hard drive shut." Words to live by.

—Mitch Gallagher
gallagher@psn.com

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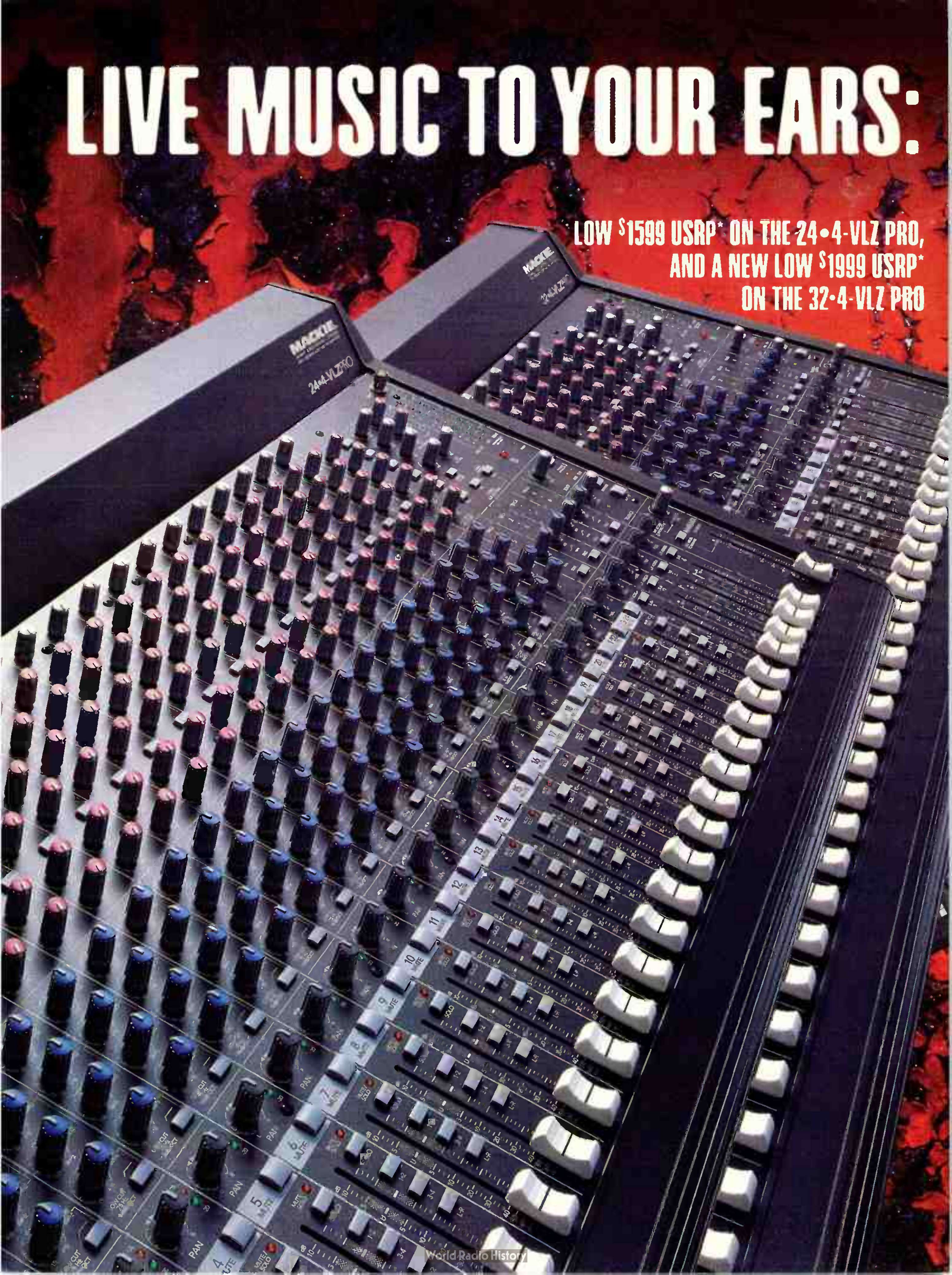
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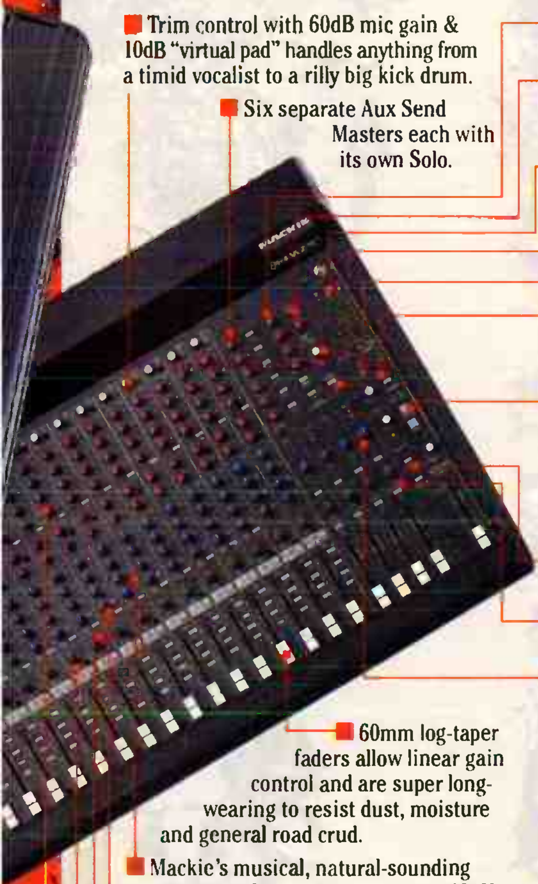

- Trim control with 60dB mic gain & 10dB "virtual pad" handles anything from a timid vocalist to a rilly big kick drum.
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- EFX to Monitor lets you send different effects or effects levels to stage monitors without screwing up your main PA mix.
- Easy level setting with In-Place Stereo Solo. Just solo a channel & adjust the Trim 'til the meter flickers at the Level Set arrow.
- Separate Tape Return level control.
- Global Aux Return Solo switch.
- Separate Solo section with level control, global AFL (post fader) or PFL (pre-fader) mode switch & Aux/Sub Solo LEDs.
- Separate Talkback section with level control, LED and switches for assigning talkback to Main Mix or Auxes 1 and 2. There's also a separate mic preamp input on the back of the mixer so you don't have to tie up a channel.
- Tape to Main Mix routes tape inputs to main outputs for music during breaks.
- Each Submaster bus has Solo switch, Pan control and Left/Right assign switch.
- Air EQ adds crispness and definition to high-end without boosting ear-fatiguing 8kHz-10kHz range.
- 60mm log-taper faders allow linear gain control and are super long-wearing to resist dust, moisture and general road crud.
- Mackie's musical, natural-sounding equalization. On mic/line channels: 12kHz Hi shelving, peak midrange sweepable from 100Hz to 8kHz (so it can also be used as a 2nd HF or LF control) and 80Hz Lo shelving. On stereo line channels: 12kHz Hi shelving, 3kHz Hi Mid peaking, 800Hz Lo Mid peaking and 80Hz Lo shelving.
- Sharp 75Hz 18dB/octave infrasonic filter on all mic channels cuts wind noise, stage rumble, mic clunks and P-pops.
- Super-twitchy –20dB signal present and overload LEDs on each channel.
- Constant loudness pan control.
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NEW 24•4 & 32•4-VLZ PRO MIXERS

- 4-bus design with 20 or 28 mono mic/line channels with XDR™ mic preamps and 2 stereo line channels
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- Muted channels can be soloed!
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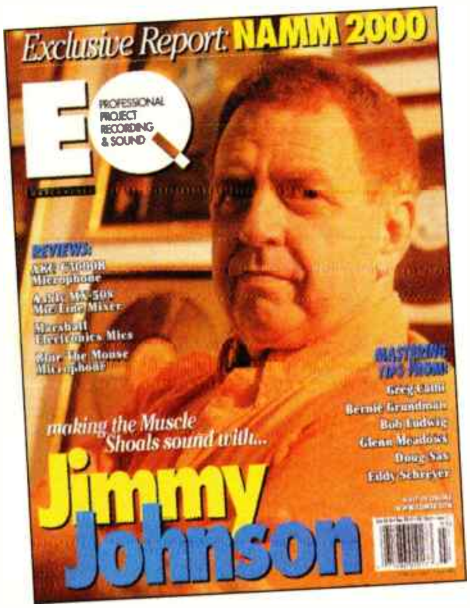
SWAMPER PRAISE

I would like to commend *EQ* for the excellent article on Jimmy Johnson in the March issue. I have had the pleasure of knowing Jimmy for the past 20-plus years and, though I seldom see or get to talk with him anymore, I still consider Jimmy a great friend.

I first had the pleasure of working with Jimmy in the '70s. He gave me a call after hearing some of the material by our group, "Carnival." We were from Mobile (the opposite end of the state for all you non-Alabamians), and he, Roger Hawkins, and David Hood came to hear us play in Birmingham. Shortly thereafter, we were fortunate enough to have them produce



us. I will always fondly remember the time we spent working with Jimmy, Roger, David, and Barry Beckett. Theirs is truly a unique story in the history of the record business.

As the years rolled by, I made many music-related trips from Mobile to Nashville. Though it wasn't the courteous thing to do, on occasion I would "detour" through Muscle Shoals unannounced. Jimmy and the guys made me feel at home. It always impressed me that, though he is a person of tremendous talent, Jimmy kept it all in perspective — something we could stand more of in the record business.

When you gaze upon the seemingly endless gold and platinum records adorning the walls at Muscle Shoals Sound, it's hard to not be overwhelmed by what Jimmy, Roger, David, and Barry have accomplished in that small town on the Tennessee River. I'm sure that they, individually and collectively, will continue to be successful in whatever they do.

Thanks again for a great article on Jimmy and for telling a small part of the Muscle Shoals story.

Jerry Powell
Southern Sound & Video Productions
Mobile, Alabama

WELL READ

I just had an enjoyable morning reading your May issue. As a studio owner/producer, I've been reading *EQ* now since the early '90s, and I must say I'm impressed with what seems to be a new content format. I like the inclusion of more "Room with a VU," "Techniques," and "From the Desk of" articles and more focus on musical/production elements from experienced pros. I find we learn more listening to the input of producers and engineers than we do with articles that concentrate purely on gear. So I appreciated your intro editorial and applaud your continued efforts. We are now reading *EQ* cover to cover, which I can't say about the other recording magazines we subscribe to.

Richard
via Internet

HOW MUCH IS ENOUGH?

Just wanted to say I thoroughly enjoyed Mitch Gallagher's May 2000 editorial. One reason we get so tweaky about gear is because we're always looking for ways to make things better. This is admirable, but often results in frustration: "How come my recordings don't sound like...?" It's easy to come to the conclusion that you're missing some magic ingredient. Why? The alternative is that you might be inadequate as an engineer. As long as there's someone telling you it's your equipment that's at fault, you don't have to take as much responsibility for how your recordings sound.

It also occurs to me that we are into this tweak obsession because it's *easy*, and, yes, it can be fun sometimes. You can sit around and research, quibble over, and obsess on minutia and feel like you're improving your music, skills, and studio, when in fact you're just procrastinating. It's *work* to make a better recording, get better on an instrument, or *really* improve your studio.

What I see is us spending time and energy debating and researching all this little crap because we don't know what else to do. There are no easy answers, so the next thing you know we're agonizing over polarized cables instead of basic production techniques.

It's tough because it's really hard to define what "enough" is. Having the judgment to know how far to go is the key. Too far and it gets too expensive (in time and money). Not far enough and the difference isn't easily detectable; a difficult quandary.

Have I rambled enough? Well, lets take the word count of this letter and divide by the time it took me to write it. Multiply by how much other stuff I could be doing, take the square root of my hourly earnings, and we'll probably find that the energy spent debating the validity of the debate is the biggest waste of all.

David Stewart
Sweetwater
Fort Wayne, IN

MORE MASSEY!

Howard Massey absolutely rocks. His recent John Leckie interview was so on point it made me laugh! I mean the stuff that came out in that short talk is so beautiful, so helpful, so perfect — I loved it. From small/large rooms to using shelving EQ only when tracking to 'verb sounding different in different rooms to vocal/snare balance...all I can say is keep it coming. It gave me lots of ideas. Looking forward to Howard's book. It's gonna be an industry best-seller for sure!

Ken Fordyce
Mirrorsound.com
Seattle, WA

PEOPLE MAKE MUSIC, NOT GEAR

Great editorial [May 2000]. The truest words I've read in a trade mag. in months. William Orbit makes magic happen with a 10-year-old Mac and a couple of 106's. Maybe you could devote an article to the act of "making" music, rather than how the gear "makes music." Keep up the great work!

Marc Battaglia
Creative Director
Dolphin Dance Music, NJ

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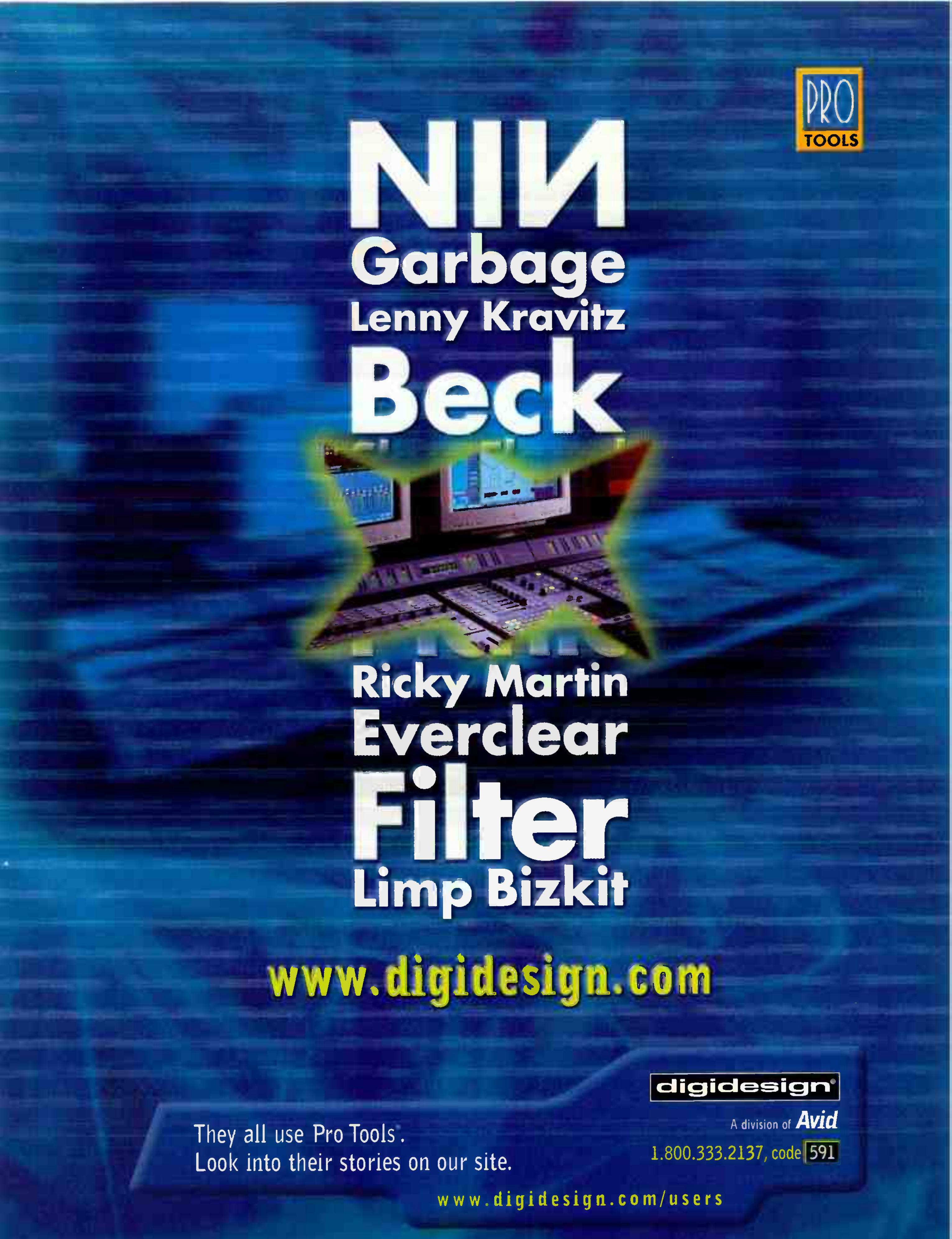
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IT'S ALL IN THE TIMING

Q *I'm running Cakewalk Pro 9 on an HP 450 Pentium III running Windows 98. I recently purchased an Akai MPC2000XL. I love it! I'm convinced my drum tracks feel better from the Akai than the same samples being triggered by Cakewalk (all quantized 100 percent). Could I be getting timing errors from the computer and Cakewalk? Should I consider a sequencing program from someone else, like Emagic or Cubase or…?*

I think that there is nothing running in the background in the computer, but I am not sure how to check. Windows has gotten so complex that I really don't know what's going on. Could you recommend a path for me to take to get to the bottom of this?

John Bergeron
via Internet

A Dedicated drum machines will beat a general purpose computer for timing accuracy because all they do is make beats, and the sounds are generally triggered directly rather than having to be triggered via MIDI, which creates its own delay issues. The most stable box in my entire studio is an Alesis HR-16, with a MIDI delay of 1.6 ms and less than 100 microseconds of variation. That's tight!

A problem with computers is that the machine is so busy interrupting itself to do various tasks (regardless of whether or not you think something is running in the background) that timing can become a problem, even with fast machines. A multitasking OS makes matters even worse. I remember talking to one of the members of the Amiga design team who said that it would never be able to do tight MIDI timing because of the multitasking kernal. The Mac has a slight timing edge for now because it doesn't do "real" multitasking, but we'll see what happens when OS X kicks in.

There are steps being taken to address this. Cakewalk has done an amazing job of optimizing the digital audio aspects of the program; hopefully they'll figure out some way to deal with MIDI next. Meanwhile, Emagic has introduced a system (AMT) that cuts timing inconsistencies to 2 ms or less, and MOTU has introduced a MIDI time-stamping protocol that claims to cut inconsistencies even further, into the microseconds range. At the Frankfurt Musik Messe show, Steinberg introduced its own timing solution. The catch is that all these solutions are dependent on having particular hardware; these are not universal panaceas. But it shows that manufacturers are aware of the problem, and have decided to do something about it. Surely Cakewalk, as well as other sequencer manufacturers, will come up with their own solutions to stay competitive, or figure out a way to work with existing hardware from other manufacturers.

Another solution is VST 2.0-compatible virtual instruments. These dispense with the MIDI connection, and interact directly with the computer rather than having to deal with MIDI-induced uncertainties. You don't have the range of products that exist with hardware boxes, but the world of virtual "plug-in" synthesis is heating up, with more and more options appearing almost daily.

Meanwhile, you now understand why so many beat fanatics love their Akai boxes — they really are tight. If you absolutely must use a computer, try to find an old Atari 1040. Don't laugh — it beats all the fancy G3 and PIII boxes for sheer tightness of MIDI timing, because MIDI was built right into the core operating system as a DMA process.

Craig Anderton
Technology Editor

DONGLE DILEMMA

Q *My question: Do you know if any software manufacturers such as Steinberg and Waves (who both use hardware copy protection) plan to issue keys/dongles in the form of a USB connection instead of parallel port?*

My problem: I use a PC. I currently use Cubase VST with the parallel port copy protection key. I also want to buy the Waves Gold Bundle, but they use a parallel port copy protection key as well. Can I use them both on the same parallel port? I am aware I could add a second parallel port via a card, but my system doesn't have the room.

A USB key seems like it would address this sort of problem since USB supports so many devices and seems more flexible. Any thoughts or info or possible solutions you can lend would be greatly appreciated.

Robert Hyman
via Internet

A I'm not aware of a conflict with using multiple dongles on one port (I have six dongles on my chain). A

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Does
your
tape based
system...
VSR-880 24bit DIGITAL STUDIO RECORDER
STATUS/LOCATOR/SCENE
Roland
SCRUB
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A.PUNCH
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CLEAR
INPUT (TRS)
PHONES
PREVIEW
TO
THRU
FROM
ZERO
REW
FF
STOP
PLAY
REC
SONG END
SHUT/EJECT

computer not being able to see a dongle is more common now with faster computers, so if this problem shows up, the first thing I would try is putting a regular printer extension cable on between the port and the dongles. What's also good about this is that you can bring the cable around and secure your dongle chain inside the computer, so that they don't stick out or get stolen.

If you have problems, you could also try changing the order of the keys on the chain, since our dongles have a throughput.

Costa Kotselas
Steinberg

A Waves is looking at many copy-protection options. Unfortunately, at this time there is no single USB solution that would work for both Mac and PC (which would reduce inventory for dealers and streamline manufacturing and development). There are rumors that such a truly universal USB dongle is being made. However, Waves is not limiting our investigations to only hardware protection systems. A decision will be reached later this year.

It is not our experience that multiple dongles won't work. Indeed, we have a machine for tech support that has both Cubase VST and Waves Gold bundle parallel keys on it. The only point to follow is to have the Waves key plugged into the port first, then the Cubase after it; that is, the WaveKey is "closest" to the machine.

Seva
Associate Founder/
Minister of Marketing
Waves

WHICH FORMATS?

Q *I very much enjoyed the review of the Alesis MasterLink disk recorder in the April issue of* EQ *magazine. My question is: Will it read CD-RW disks in order to copy songs to the internal hard drive? I save songs both in the CD-R and RW formats, so I'm looking for gear that will work with both formats.*

J. Larry Tyler
via Internet

A Glad you enjoyed the review. I didn't have a CD-RW burner to try this with, so I couldn't test it myself. According to Alesis, at this time the MasterLink does not officially support CD-RW discs — it may or may not work. My advice would be to take one of the CD-RW discs in question to an Alesis dealer and try it in the machine.

Mitch Gallagher
Editor

THREAD OF THE MONTH

As seen on the George Massenburg forum on the EQ Boards at www.eqmag.com

George, you build your own equipment, and you both record and mix records that sound very good. Why do you leave it to someone else to do the mastering? —*Swede*

Because by the time I'm finished with a mix (and I've usually also done the tracks, a lot of overdubs, and most of the vocals) I'm rather close to the project, to say the least.

I really like to have someone listen to my work and interact with me — I often learn a lot. In fact, I've made the point often that it's worth it to pay someone to listen even if they end up doing absolutely nothing (this has happened once or twice). —George Massenburg

So do you do any overall EQ and/or compression before you go to the mastering lab? —*Swede*

I very often use overall EQ. Sometimes I do a mix with overall compression. I almost *always* listen to my mixes with overall compression to better judge what is going to happen to them at a radio station. Often I'll mix with a compressor on for awhile, then turn it off and mix some more, the better to come up with something that works for either situation. —George Massenburg

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EQ NEWS

JUNE • 2000

Two Ton Shoe Voted Best Live Band on the Internet

Boston-based band Two Ton Shoe was the grand-prize winner of the 1999 "Best Live Band on the Net" contest. Sponsors included Fostex, Kaman, and other instrument and digital technology companies. Thousands of dollars worth of merchandise was awarded to the winner, as well as a recording contract with NewTechMusic.

"We've been in contests before, but nothing like this," says Jake Shapiro of Two Ton Shoe. "The equipment awarded as prizes is simply incredible. Winning the Fostex gear — a VR800 digital recorder, VM200 digital mixer, and CR300 CD-R burner — means we now have a complete studio, and one that's loaded with top-quality gear."

About 350 bands participated in the Web-based contest. Bands registered and uploaded their music from August through October of last year, with voting kicking off in November. Ten finalists emerged from this process, with the winner selected recently by a panel of artists that included Sheila E, Alex Acuna, Davey Johnstone, John Jorgenson, and Tom Dumont.

Two Ton Shoe was recently selected — for the third consecutive year — as winners of the prestigious Boston Music Award. Performing throughout the Northeast, the band has released two CDs on L-Shaped Records and is currently preparing the debut of their third release.

For more information, contact Fostex at 562-921-1112 or visit www.fostex.com.

The members of Two Ton Show show off one of their "Best Live Band" prizes.

Sonalysts Announces Recording Studio Auction Initiative

Sonalysts, Inc. announced that it is currently auctioning time in its world-class Power Station New England recording studio. Based on the concept of excess capacity sales, Sonalysts is pursuing an initiative to auction small portions of recording studio time at a deep discount.

"Similar to the hotel industry, for example, we often have small blocks of time between major recording contracts that are hard to fill using traditional selling methods," reports Tony Cowden, leader of the auction initiative. "We hit on the idea of the online auction format as a means to keep the facility full during these periodic down times." This initiative also provides an avenue for bands with smaller budgets to have access to a premier recording facility.

Designed by Tony Bongiovi, Power Station New England includes a replica of the famous "A" room at the original Power Station Studios in New York, where more than 400 gold or platinum albums were recorded. The control room is built around a Neve VR console, and tracking is done to a pair of Studer A800 machines. The collection of over 70 microphones includes many classic tube models from Neumann, AKG, and Telefunken. Outboard equipment ranges from vintage Pultec EQs, Urei limiters, and an EMT tube plate to the latest Lexicon and Eventide hardware.

"What we have done is create a product, a coupon worth 50 percent off our normal daily rate for the facility," says Andy Toriello, president of Sonalysts, Inc. "This is a low-cost way to move blocks of excess time, and we are very excited about the e-business potential of this initiative." Currently the coupons are for auction on eBay, and can be found by conducting a search for "POWERSTATION" at www.ebay.com. Future plans call for the auction to expand to other venues (Yahoo, Amazon, etc.), and the potential exists to auction other facility time as well, including Sonalysts's movie sound stages.

For more information, contact Sonalysts at 800-526-8091 or visit www.sonalysts.com.

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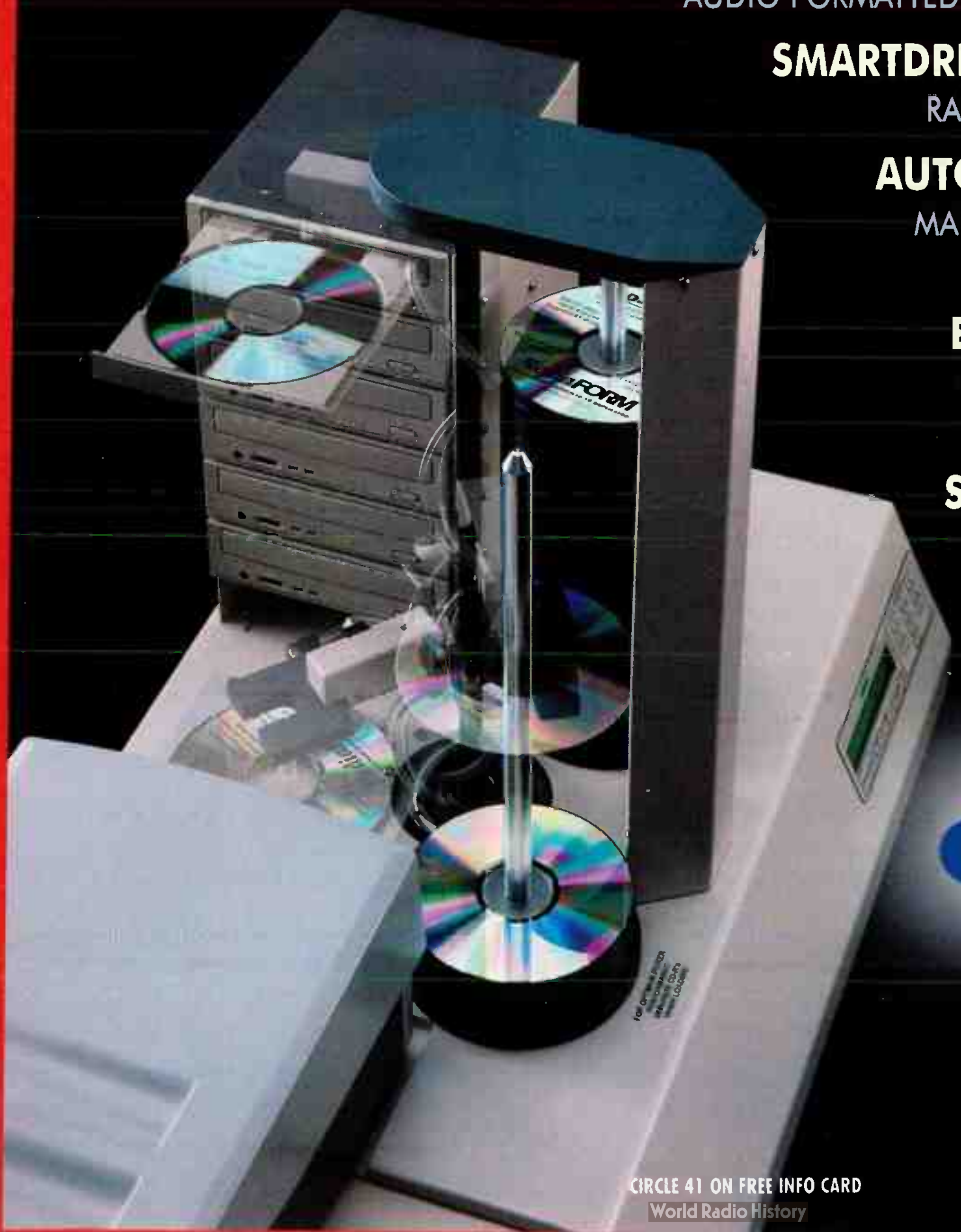

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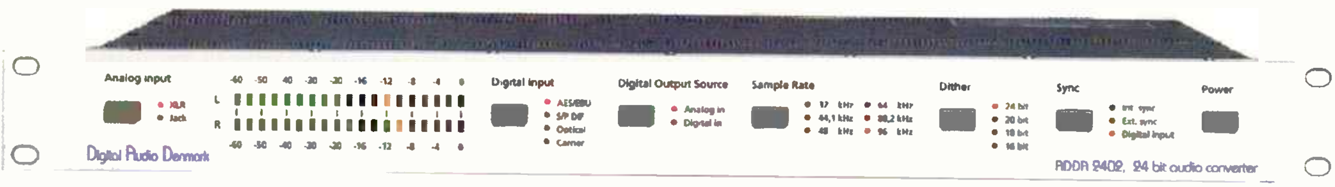
CIRCLE 41 ON FREE INFO CARD

EQ FRANKFURT REPORT

Each year, throngs of Europeans descend on Frankfurt, Germany, for one of the world's largest music industry trade shows, Musik Messe. Comprising the full spectrum of music-related products, Messe has exhibits covering everything from synths to software, guitars to percussion, mics to mixers, lighting to pyrotechnics, and effects to DJ gear. This year, intrepid *EQ* editors Mitch Gallagher and Craig Anderton made the trans-Atlantic sojourn to visit the show. As is the case every year at Messe, tons of new products were announced; here are some of the highlights.

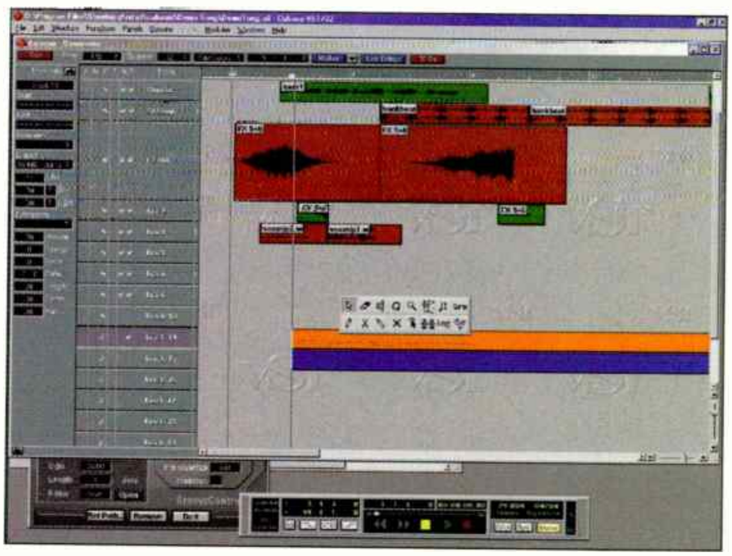


CONVERTING TO DANISH

The Digital Audio Denmark ADDA 2402 audio converter is tailored for high-quality audio recording on HD, DAT, and DAW systems. It supports sample rates from 32 kHz to 96 kHz in 24-bit resolution, with a dynamic range better than 117 dB (A-weighted). The A/D converter employs filtering for elimination of aliasing intermodulation distortion. The ADDA 2402 can be used as an upgrade to existing integrated converters, giving the benefit of its clean audio path. In addition, it has multi-format inputs and outputs and sample-rate conversion, as well as future-proof 24-bit, 96 kHz sampling capabilities. For more information, call Sascom (Digital Audio Denmark's North America distributor) at 905-469-8080 or visit www.digitalaudio.dk/www.sascom.com. Circle *EQ* free lit. #107.

SPARK YOUR CREATIVITY

The TC Works FXMachine allows a user to freely design a virtual modular synth with the included TC-O, TC-F, and TC-A modules — the user decides how many oscillators, filters, or amplifiers to use. The TC-O monophonic dual oscillator module supplies all the classic waveforms, offers pulse width modulation, an additional sub-oscillator, oscillator sync, ring modulation, and an integrated LFO with all the features needed for convincing synth leads, synth percussion, or strange "bleep" sounds. The TC-F filter module provides a 12, 18, or 24 dB low-pass filter with resonance, an additional high-pass filter, its own envelope, and key follow, plus single- and multi-trigger. The TC-A amplifier module offers the classic ADSR curve and more; "Drive" allows for analog-sounding gain in this module, providing a "phat" sound when desired. This drive stage employs TC Works' SoftSat analog emulation circuit. For more information, call TC Works at 805-373-1828 or visit www.tcworks.de. Circle *EQ* free lit. #108.



THE DO-IT-ALL AUDIO TOOL

The Musik Messe convention saw the launch of the Steinberg Cubase VST 5 Series for Windows and Mac. Cubase VST 5.0 is designed to be a comfortable tool for composing and arranging music; and for audio and MIDI recording, processing, and scoring. Cubase VST 5.0 offers a digital recording engine with scalable recording modes up to 32-bit floating-point resolution on 128 channels of audio in the top-of-the-line version Cubase VST/32. As the flagship in the Cubase VST 5 series, the VST/32 offers 128 channels of digital audio, and eight FX sends, each with four channel inserts and four master inserts. The handling of FX and plug-ins has been completely redesigned. The new FX rack hosts new control elements, and the new channel EQs can now be edited graphically or with virtual pots. For more information, call Steinberg North America at 818-678-5100 or visit www.us.steinberg.net. Circle *EQ* free lit. #109.

EQ FRANKFURT REPORT

CREAMY TECHNOLOGY

Musik Messe 2000 was the site for the introduction of the Creamware Pulsar II DSP system for Windows and Mac. The new Pulsar II PCI card features six Analog Devices SHARC DSPs. This permits the simultaneous use of more synths, more effects, and more mixer functions, or higher polyphony for the sound generators. Pulsar II now features a second independent S/TDM bus for seamless integration with other DSP cards from the SCOPE family. The inputs and outputs are situated on a daughterboard, which can be connected to the DSP card. In its basic configuration, the system features 20 I/Os (stereo analog, stereo S/PDIF and 2 x eight-channel ADAT). For more information, call Creamware at 800-899-1939 or visit www.creamware.de. Circle *EQ* free lit. #110.

CLAVIA'S CONTROL SOLUTION

The Clavia Nord Lead 3 is a 20-voice synthesizer offering a wide range of features. Some of these include Adjustable Unison without voice reduction, a redesigned oscillator to create two separately synced waveforms per voice, two separate multi-mode filters that can be routed in series or in parallel, a new modulation section, and a Morph function that lets the user continuously control defined ranges of up to 26 parameters using only a single control source. Clavia combined endless rotary knobs with circular LED graphs for solid control. The Nord Lead 3 also features four audio outputs. For more information, call Clavia's U.S. distributor, Armadillo Enterprises, at 727-519-9669 or visit www.clavia.se. Circle *EQ* free lit. #111.

RE-INVENTING THE TONEWHEEL

Native Instruments has launched the latest product for the virtual studio, the B4 VST-based tonewheel organ, which is said to capture the sound of the classic B3 organ and rotating speaker combo. Not stopping there, the B4 takes the tonewheel organ into new territory, with features such as 91 tonewheels, nine drawbars per manual, two manuals and pedal keyboard, scanner vibrato/chorus, percussion on any harmonic, adjustable keyclick, tube distortion sound, rotary speaker processing, and many parameters for fine-tuning the sound. System requirements for the B4 include Windows 95/98/NT/2000, min. Pentium 266 MHz, min. 32 MB of free RAM, VST 2.0 compatible host software, PC-equipped according to the specifications of the host software. On the Apple side, requirements are Power Macintosh or compatible system, min. 604e/250 MHz, and more. For more information, call Native Instruments USA at 800-665-0030 or visit www.native-instruments.com. Circle *EQ* free lit. #112.

EQ PRODUCT VIEWS

ISOLATED INSULATION

Middle Atlantic Products has introduced their new S24DG isolation rack, designed specifically to minimize the acoustic noise associated with technologies involved in the studio environment, such as computers and hard drives. The 24-space S24DG includes a gasketed Plexiglas front door, and gasketed solid rear door for optimum sound insulation. The rear door features a built-in quiet fan and filter, and a brush grommet at the door bottom to allow sealed cable exit. For more information, call Middle Atlantic Products at 973-839-1011 or visit www.middleatlantic.com. Circle *EQ* free lit. #113.

ANY ROOM, ANY SIZE

JBL Professional's LSR25P monitoring system is expressly designed to provide multimedia and digital audio workstation users with high-performance monitoring for smaller environments. Featuring many of the technologies found in the previous JBL LSR Series models, including an advanced high-frequency driver and JBL's low-frequency transducer technologies, the LSR25P is designed to establish a new standard for multimedia, audio-for-video, and edit room monitoring. It allows facility-wide implementation of LSR technology, standardizing playback environments throughout larger multi-room facilities regardless of room size. This provides for the accurate translation of a mix from the original dub stage through each phase of production and postproduction. Frequency performance is 60 Hz to 20 kHz ±2 dB with low-frequency extension to 48 Hz at –10 dB. For more information, call JBL Professional at 818-894-8850 or visit www.jblpro.com. Circle *EQ* free lit. #114.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EQ PRODUCT VIEWS

MONITORING PERFECTION

Genelec's latest addition to their line of active monitoring systems, the S30D, is designed to handle 24-bit/96 kHz digital audio reproduction. Each three-way enclosure features a 210 mm (8-inch) woofer loaded in a vented cabinet, allowing accurate frequency response down to 36 Hz; an 80 mm (four-inch) mid-frequency cone driver sealed in a cast alloy aluminum housing; and a proprietary ribbon tweeter with a moving mass of only 32 mg, with a frequency response extending up to 50 kHz. AES/EBU digital inputs, along with analog XLR inputs, are provided. For more information, call Genelec at 508-652-0900 or visit www.genelec.com. Circle *EQ* free lit. #115.

ON THE RACK

Two new balanced power rack systems from Equi=Tech have been designed to meet the power needs of larger sound applications. The new models ET 7.5R and ET 10R are large-capacity systems designed to furnish continuous power for systems that require between 60 and 85 amps. Rack system features include a magnetically shielded toroid isolation transformer, surge protection, overcurrent circuit protection, GFCI protected power distribution, and appropriate UL listing. Optional features include a front-panel-mounted LED voltmeter, an input voltage selector switch, and twist-lock outlets for portable power setups. Models ET 7.5R and ET 10R are priced at $3,689 and $4,389, respectively. For more information, call Equi=Tech at 541-597-4448 or visit www.equitech.com. Circle *EQ* free lit. #116.

GATEWAY TO THE FUTURE

The most recent addition to Crest Audio's Network Distributed Processor family of products is the NDP-ENG, an Ethernet to NexSys Gateway. The gateway satisfies inter-network requirements between Ethernet PCs and NexSys EIA485 amplifiers. The gateway also allows NexSys+MediaMatrix systems to be integrated in the MediaMatrix Windows NT mainframe computer. The gateway is compatible with 10BaseT or 100BaseT networks and may be used in a dedicated "NexSys control" network, or it can co-exist with other equipment in standard Ethernet LANs or WANs. Each gateway can control up to 32 NexSys amplifiers, with no limit to the number of gateways in a system. The NexSys NDP-ENG has a MSRP of $1,500. For more information, call Crest Audio at 201-909-8700 or visit www.crestaudio.com. Circle *EQ* free lit. #118.

RADAR REVISION

Otari announced the release of a significant new software upgrade for their RADAR II hard disk recording system, version 2.09. The revision includes several feature enhancements, including waveform display, the implementation of DVD-RAM, Mammoth and AIT backup drives, Seagate and IBM 9 GB and 36 GB qualified SCSI hard drives, selectable digital inputs, macro keys, auto offset calculate, and improved backup and restore implementation. A hardware upgrade is required in order to run waveform display, which includes a faster Pentium processor, 64 MB of RAM, and new key caps for the RE-8II remote. For more information, call Otari at 818-594-5908 or visit www.otari.com. Circle *EQ* free lit. #117.

Millennia Media **Origin STT-1**

Get tube warmth or solid-state cleanliness with Millennia's hybrid circuitry tracking channel

BY STEVE LA CERRA

Over the past few years, Millennia Media has introduced several unique audio processors based around what the company calls Twin Topology circuitry. Recognizing the trend for using tube-based gear as a front-end for digital recorders — while still acknowledging need for the "honesty" of a clean solid-state design — Millennia Media's engineers produced the NSEQ-2 (parametric equalizer) and the TCL-2 (compressor/limiter) processors. Both of these devices have an all-tube, Class-A signal path, plus a discrete Class-A, solid-state signal path *simultaneously* in the same chassis, with an associated front-panel switch to select which path is used for a given application. So it's only natural that Millennia Media has introduced the Origin STT-1, a Twin Topology tracking "channel" that offers a multitude of signal processing options for recording straight-to-tape (or disk, if you prefer).

Millennia's engineers accomplish Twin Topology through the use of a dual-amplifier circuit design. Rather than building two completely identical circuit paths, Twin Topology features two functionally equal amps with identical input and output requirements; one of these is solid-state, and the other is based around vacuum tubes (obviously, these each have their own sonic characteristics). All other circuit components such as resistors, capacitors, potentiometers, point-to-point wiring, and PC boards are shared. Since Millennia Media's minimalist design utilizes only one amplifier in each audio "module" (a compressor or EQ, for example), the change from solid-state to vacuum-tube circuitry is accomplished with a single switch.

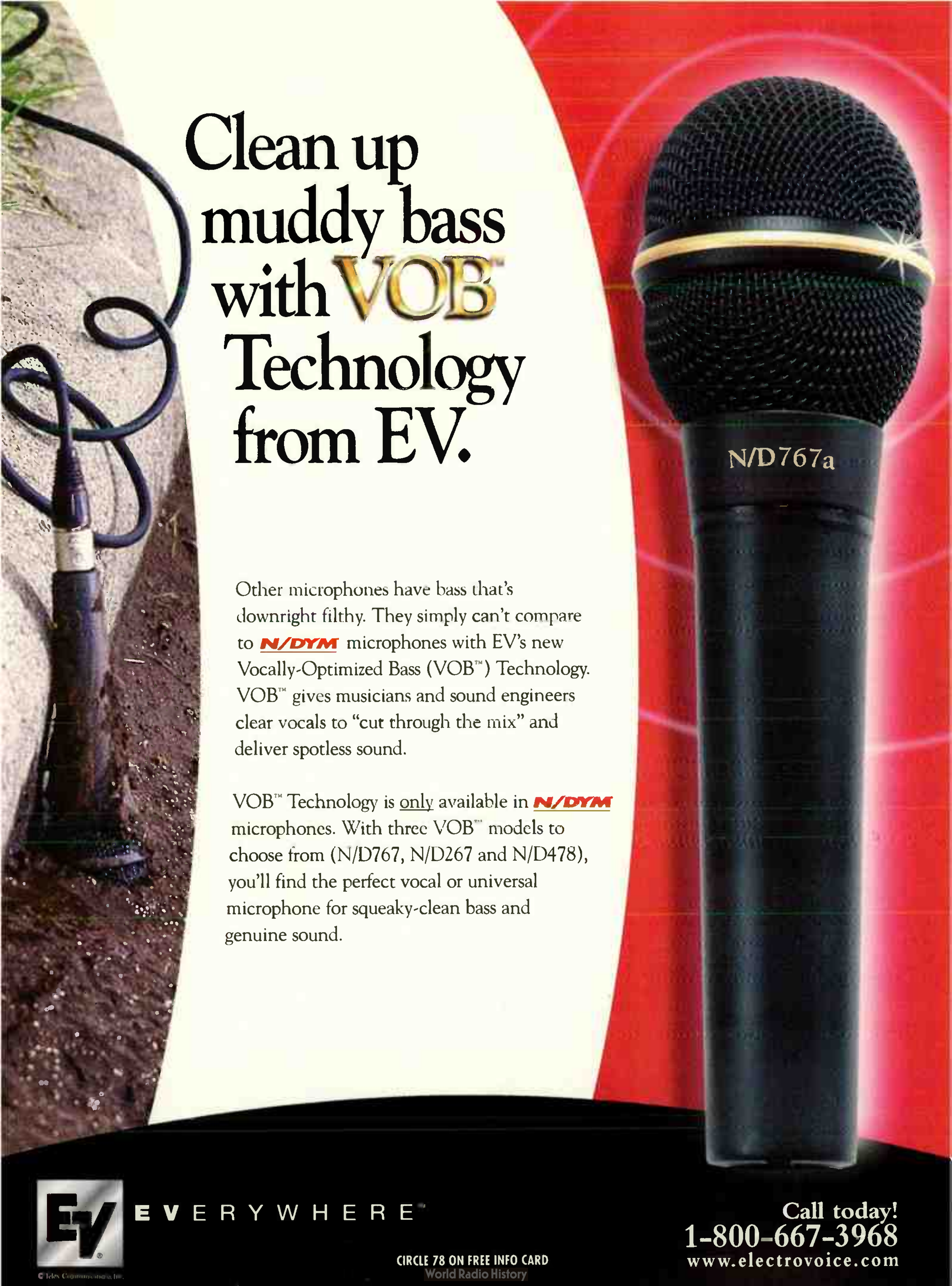
The STT-1 lives in a two-rackspace chassis, and includes a mic preamp, equalizer, compressor/limiter, and de-esser. All of the tube circuitry in the STT-1 employs 350-volt, hand-selected triodes, while all of the solid-state circuitry employs discrete, Class A J-FET amplifier modules. Starting with the mic preamp, the STT-1 contains Millennia's highly-regarded HV-3 solid-state mic pre, as well as their M-2b vacuum tube mic preamp. The HV-3 is optimized for minimal coloration and maximum headroom; its padless input stage is capable of handling signal levels upward of 20 dBµ. The M-2b tube mic pre offers a frequency response of 4 Hz to 130 kHz and a noise level of –116 dB (EIN) with gain set at 50 dB. Either of these mic pres can accept mic-, line-, or instrument-level input (mic and line via rear-panel connectors, instrument via front-panel 1/4-inch jack). A front-panel switch routes the input signal to either the solid-state or the vacuum tube pre. Additional switches are provided for +48-volt phantom power on or off and input signal polarity reverse.

Increasing the number of available circuit configurations, the STT-1's input stage also has an audio transformer, which may be switched in or out of the path. This transformer has been specifically designed by Millennia Media to emulate the coloration characteristics of some of the more popular "vintage" recording consoles. Purists can bypass the transformer for increased transparency.

The parametric equalizer employed in the STT-1 is Millennia's NSEQ. Comprising four bands — low-frequency, low-mid-frequency, high-mid-frequency, and high-frequency — the NSEQ has only one active stage in its signal path. The two mid-bands offer boost/cut, bandwidth, and frequency selection, while the low and high bands are fixed-bandwidth peaking type with a switch for shelving operation. Each of the four bands may be individually switched in or out of circuit, or the entire EQ may be bypassed with a single switch.

Front-panel controls for the TCL compressor include threshold, attack time, release time, and compression ratio. Bypass is available for the comp/limiter, as is a control labeled "Flip Dynamics." This function changes the position of the EQ in the circuit path relative to the compressor and the de-esser, allowing the user to "patch" the EQ pre- or post-dynamics. A "Meter GR" switch enables the STT-1's VU meter to show gain reduction or output signal level. A master output control is included for setting levels to the recorder.

Millennia Media's STT-1 is now shipping at a suggested retail price of $2,895. For more information contact Millennia Media at (tel) 530-647-0750, (fax) 530-647-9921, or visit their Web site at www.mil-media.com. Circle EQ free lit. #101.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Yamaha mLAN Digital Network

An emerging standard simplifies studio interconnection

BY STEVE LA CERRA

At the recent tradeshows such as the Musik Messe in Frankfurt, Germany, and NSCA in Las Vegas, Nevada, Yamaha Corporation has been showing a development that could prove to be as important to the professional audio industry as the introduction of MIDI almost 20 years ago: mLAN (pronounced "em-lan"). Designed specifically for music and audio applications, mLAN is a local area network through which electronic musical instruments, audio recorders and mixers, personal computers, and even consumer audio/video equipment may be interconnected for exchange of data. Physical connection of mLAN-compatible devices is made via IEEE 1394 serial bus, more widely known as Apple Computer's FireWire or Sony's iLINK.

Anyone who has ever wired a studio is going to quickly appreciate mLAN for the ease with which various devices are patched together. mLAN may be wired daisy-chain-style (as with SCSI devices), or in a branched configuration; the only topology not supported is the loop. mLAN gear utilizes a single port for input and output, negating the need for separate connections and cables for each — you simply plug one mLAN cable into the device. But more striking is the fact that a single mLAN cable carries MIDI, audio, serial data, and video information simultaneously between devices. Equipment supporting mLAN may be "hot plugged," meaning it can be connected or disconnected when its power is turned on, without any detrimental effects.

THE SPECS

mLAN's numbers are quite impressive because IEEE 1394 already supports data transmission rates of 100, 200, and 400 Mbps (Megabits per second). For those of you who like to number-crunch, that rate is 8 to 32 times faster than the Universal Serial Bus (USB). With a data rate of 200 Mbps, IEEE 1394 is powerful enough to simultaneously transmit as many as 50 to 100 channels of audio data (depending upon bus conditions) and up to 256 ports of MIDI data (that's more than 4,000 MIDI channels!) shared by all mLAN devices in the bus. As you might expect, audio is routed digitally via mLAN for maximum transmission quality and minimal loss over the length of cable. Using standard copper wire cable, maximum distance between one "node" (device) and another is 4.5 meters; fiber-optic long-distance adapters are available to facilitate 500-meter-lengths between nodes.

Although not required for use of an mLAN network, a computer may be used with software developed by Yamaha for control over routing and patching of mLAN devices in a system. Reconfiguration of a studio setup is possible without the need to physically disconnect or reconnect any of the cables. When AC power to an mLAN system is turned off, the configuration will be remembered for recall when the system is powered on again.

THE PRODUCTS

Yamaha already has several new product releases planned for the launch of mLAN, all of which transmit data at 200 Mbps and include mLAN Patchbay and mLAN Mixer software. One of these units is the mLAN8P audio/MIDI processor, a device designed to allow existing MIDI and audio hardware to communicate via mLAN protocol. The mLAN8P is capable of routing eight channels of digital audio in and out, as well as two ports of MIDI input and one port of MIDI output (16 channels each port) to IEEE 1394. A single cable connects the mLAN8P to an IEEE 1394-compatible computer. In addition to the three mLAN ports present on the mLAN8P, a serial interface is provided for interfacing existing computers that don't include the IEEE 1394 interface.

Yamaha's mLAN8E expansion board is designed to fit into the expansion slot of an mLAN-compatible MIDI instrument (some products may require a firmware update), and adapt the instrument's MIDI and audio capabilities to an mLAN system. When installed in the MIDI instrument, the mLAN8E provides a 16-channel MIDI port, eight channels of digital audio I/O (this is subject to the limitations of the MIDI instrument the mLAN8E is attached to), and a 16-channel mixer, enabling the expansion board to serve as a submixer for multiple-output keyboards and samplers.

Also planned for Yamaha's initial mLAN introduction is the CD8-mLAN interface card, which will allow existing Yamaha digital mixing consoles to access an mLAN system. A serial connector is also provided on the CD8-mLAN for linking the card to personal computers; dedicated application software will be packaged with the CD8-mLAN.

Yamaha mLAN system products will be compatible with Mac OS as well as Windows-based systems, and with OMS and ASIO drivers under development. Shipping dates and retail pricing were unavailable at press time. For more information contact Yamaha at (tel.) 714-522-9011, (e-mail) info@yamaha.com, or visit their Web site at www.yamaha.com. Circle EQ free lit. #119.

BOTH SIDES NOW: Two views of the mLAN demonstration at Musik Messe.

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CIRCLE 61 ON FREE INFO CARD

Tape Talk

by Jean Tardibuono
Vice President of Sales and Marketing
Studio and Broadcast Products
EMTEC Pro Media, Inc.

One of life's greatest resources is time. Although plentiful in its never-ending supply, it seems we never have enough of this hot commodity when we need it. I'm sure you agree that recording sessions are certainly no exception. I wish we could report to you this month that we have invented a way to have more than 24 hours in a day, or even more than a half an hour lead time before an artist makes a surprise appearance at the studio, but alas we can't. What we can report, however, is that we have taken one step toward making your life a little bit easier by giving you some extra time through the use of our new formatted DTRS Master for Digital Multitrack recorders. Since its release in January we have been inundated with requests for the Formatted DTRS and we feel fortunate that we have tapped into a need that many, if not all engineers share.... the need for more time.

Because the BASF Formatted DTRS eliminates the need for real-time formatting before recording it allows engineers and artists to go straight to tape. The hours of production time are not the only thing you will find you are saving by using the DTRS. We are getting rave reviews with the release of this tape due to the fact that "drum-on" hours of real-time formatting are being eliminated, therefore decreasing the wear on tape heads.

The BASF Formatted DTRS Master is currently available in 113-minute length with a sampling rate of 44.1 and we expect to offer the complete line by the end second quarter of 2000 as well as address the 24-bit issue in the near future.

So, now that you've got some extra time, go out and enjoy yourselves! ■

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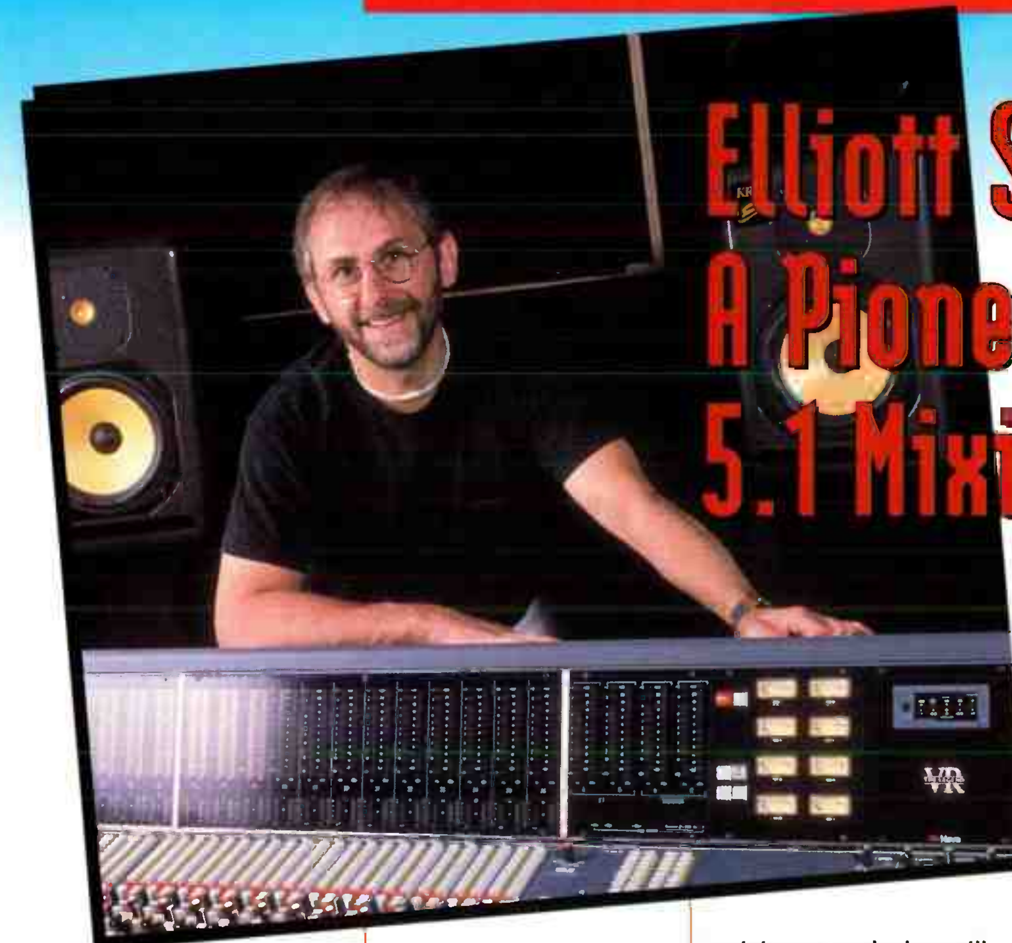

Elliott Scheiner A Pioneer in 5.1 Mixing

There are pioneers in every field of business imaginable, and of course pro audio sound is no different. When thinking of the people that have defined the way we hear often names like George Massenberg, Ray Dolby, and Rupert Neve come to mind. It has become increasingly clear that 5.1 sound will usher in the 21st century and eventually dominate as the medium we listen through. A name that has been closely associated with 5.1 mixing from its inception is that of Elliot Scheiner. Scheiner, a producer /engineer at the heart of this revolution has emerged as one of the early pioneers of this burgeoning technology and is currently mixing records that will more than likely lead us into the future of sound and redefine how we listen to music. One such record is already garnering praise from the music industry's highest court, NARAS, and its affiliated Grammy® Awards. For only the second time in history, this year's Grammy® Awards included a category for "Best Engineered Album." Perched atop the nominee list was Scheiner, for his work on Toto's current project, *Mindfields*. This Connecticut-based engineer, who is a long-time BASF tape loyalist credits his tape of choice as being a major factor in the success of the Toto project. "All I use is BASF," Scheiner relates, "Whether I'm going analog or digital. The Toto record I did entirely on BASF. We ended up mixing to analog ½ inch. The project was done on analog using BASF SM 900. We then transferred everything to digital and did all the overdubs. When it was

Continued on next page...

FROM THE TOP

CD-R MEDIA

Is there a difference between a CD-R that costs $.99 and one that costs $1.50?.... We think so!

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Joe Ryan

President • EMTEC Pro Media, Inc.

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time to mix, we basically went back to analog for the original tracks. We mixed the basic tracks from analog and had the overdubs coming from the digital machine. And we mixed to ½ inch," he concludes. "It's the first project I've done entirely on BASF tape and it's nominated for a Grammy®, which is pretty cool."

Unfortunately, Scheiner wasn't able to party too hard at the famed awards show, because his trip to the Grammys® was two-fold. "I was actually working the Grammys®," he confides. "They asked me to mix 3 or 4 of the performances in 5.1. So although I am honored with the nomination, it was a night of work." It is not surprising that Scheiner leads an "all work and no play" type lifestyle. His popularity as a traditional mixing engineer as well as a 5.1 guru has increased dramatically in the past few years as he continues to be looked upon as one of the leaders in the field.

1999 was filled with a string of steady projects for Scheiner, including mixing Sting's current release *Brand New Day*, in 5.1. "One of the more memorable projects of this year was the record I did for Sting in 5.1," Scheiner recalls. "That came out really good -- I'm proud of it. I was able to do it in Connecticut at a studio called Presence, which is about 15 minutes from my house, and I love that -- I mixed it on their Neve VR."

Presence is not the only studio on Scheiner's preferred list. He counts legendary Los Angeles studios such as Conway and Capitol among some of his favorites. However, it's east coast facilities like Hit Factory and Sony that should be stocking up on their BASF tape right now, because Scheiner's post Grammy® plate is full and he's ready to begin studio hopping all over New York mixing a variety of projects.

His upcoming project calendar reads like a who's who list of the finest in every genre of music. In addition to mixing New York Voices a new jazz vocal quartet and Matchbox 20's upcoming release in 5.1 on the Hit Factory's SL9000J, he has been tapped to do some unusual projects that will no doubt benefit from his expertise. "I'm doing an IMAX movie where they are trying to recreate a concert," he explains. "They're going to have Santana and Rob Thomas, Sheryl Crow, Sting, and Mary J. Blige. They're going to simulate the show and just do it in IMAX."

Later in the year he will once again lend his talents to a Steely Dan project, as he mixes a live concert on PBS for them at Sony Studios' big stage. It was only 3 short years ago that Steely Dan's record, *Gaucho*, was a guinea pig of sorts for Scheiner, who mixed the acclaimed record in 5.1 as one of his first offerings to the young medium.

One project that Scheiner is anxious to sink his teeth into is a 5.1 project for Queen. "I'm going to be mixing Queen's *A Night At The Opera*. Which has 'Bohemian Rhapsody' on it. That's going to be really amazing for to me." He will be returning to Presence to mix this project on their Neve. "I can't wait for this one," he concludes.

Although Scheiner may "studio hop," he confides that he isn't hard to please in the studio. However, he is adamant that they supply him with his tape of choice as he believes it is a vital part of his recording process. "I have learned the hard way that tape is not just tape... there is a difference." ■

In session With... Jim Scott

All of this Grammy® fever got us thinking about BASF customer, Jim Scott who is certainly no stranger to the NARAS awards show. Scott, an LA based engineer has been nominated for several Grammy's® in his illustrious career including Sting's 1985 hit Dream of the Blue Turtles and won his first Grammy in 1995 for Best Engineered Album, Tom Petty's Wildflowers. Scott has been a part of countless album projects that have topped nominee lists, not to mention the charts like Natalie Merchant's Ophelia, The Red Hot Chili Peppers' Californication and Santana's comeback smash, Supernatural, for which Scott took home his second Grammy® for Album of the Year. Although he shows great diversity in the projects he chooses, his recording process rarely varies. His approach is relatively simplistic and of course catered to the project at hand. We found that he has unearthed the secret to creating a successful recording project and that is to "find out how an artist hears themselves and get it on tape that way." Easier said than done you say? Read on for a few "how to's" from this brilliant producer/engineer who we caught up with in between mixes at Cello Studios in LA.

I was looking over your discography and you've done some amazing work. I'm curious, with Natalie Merchant, her vocal sound is so undeniably distinctive. How did you capture her signature vocals?

"I've done three projects with Natalie. The first one was the mix of Ophelia. I think a big part of the sound that she has is... the way she hears her own voice. Her speaking voice is similar to her singing voice, there is not a world of difference. It's a wonderful voice. Secondly, she has a home recording studio where she has 16-track heads on her 2 inch machine. So she's going even a step further more retro than the rest of us by using big fat heads on big fat 2 inch tape. That adds to the dimension of her voice, in addition to the Neumann tube microphone she uses.

And that Neumann Tube mic is what you used when you went into the studio with her?

Yes.

How different is it for you when you go from an artist like Natalie to the Chili Peppers?

The only thing that's different is giving people what they want. I don't feel like I have to do anything in particular. I don't have a system. I just take it as it comes and I just want it to sound good at the end of the day. The secret is getting down to 'What do they think they sound like' and capturing that... With the Chili Peppers it's fast, hard and funky. With Natalie it's lush, beautiful and focused on the piano and her voice. She really doesn't care about the drums and bass as much. If it was up to her, if she wasn't making pop records she probably wouldn't use them! She's not going to sit there very long and fret over the drum sound. The Chili Peppers were more concerned with 'Is this exciting, is this funky, does this have impact'. They weren't worried about how big or little the record sounded, they just wanted it to sound like what they hear when they're sitting around in Flea's garage rehearsing.

When approaching a project, do you usually go analog or digital? Do you find yourself switching formats?

No, I don't switch. I generally go into all my projects analog. That's certainly my first choice.

And your tape of choice is?

At the moment I've been using BASF SM 900 for everything. The reason why I switched originally was their 1/2" tape. I was using 456 and 499 for years and thought it was great. I use those tapes for different types of sound, different types of bands and different types of impact. The tape is as musical as anything available and you have to sort of pick your spots. Like with any tape, sometimes you get a hot reel and sometimes you get a cold reel. I was feeling like there was a little too much inconsistency. Enough so that I risked a change.

And you found more consistency with the BASF SM 900?

Yes, with the SM 900 I have for the last couple of years.

Are there any studios that you call home?

Yes. My home away from home is Cello Recording Studio, Studio III. Also a big favorite is Cello Studio II. Another favorite is Sound City and I always love going to The Village, Studio A. The similarity is that they all have vintage Neve recording consoles in the 80 series...8038 up to 8078. These boys have got a lot of courage, a lot of guts, and a beautiful sound and to me, they're the cornerstone of rock and roll.

Can you give us insight into your micing techniques?

My micing technique is to put the best mic you have in front of the best sounding thing you have and record it. And get on with it! No really, I generally use stock studio typical microphones. When I was an apprentice at the Record Plant from 1980-1984 I learned from Lee DeCarlo and Andy Johns, guys that had made classic rock and roll records. They were making a lot of records, and I was at a place where I was assisting them and I just discovered if you got a really great sound, put up a good mic and record it. I think you can really stop the flow if you have to stop and change things. I think you should rely on your experience to get things in position quickly, and if you can't get it into position the first or second time, quit going out there and hire someone to do it for you.

I don't use anything special. Whatever the best mics are in the house. Cello has an incredible mic collection. A lot of vintage tube mics which are up to standard and sound great on down to the kind of crummy everyday stuff that everyone uses. Sound City has a lot of classic rock mics like U87 and C12s, 421s...that's my technique. Bring me your box of microphones and let's put 'em up! ■

God, the FBI, and Janis Ian

Folk legend Janis Ian records her latest album, *god and the fbi,* in her L.A. project studio

STUDIO NAME: The House at Hazen
LOCATION: Los Angeles, CA
MAIN PEOPLE: Janis Ian, singer/songwriter; Marc Moreau, engineer
CREDITS: Since the mid-1960s, Janis Ian has released more than 20 albums, including *Between The Lines, Breaking Silence, Hunger*, and, most recently, *god and the fbi*. In addition to *god and the fbi*, Marc Moreau has worked on records by Roger Waters and Patrick Leonard.
MIXING CONSOLE: Mackie: 32•8, 24•8, and 1604VLZ (the VLZ was used for headphone monitoring)
MONITORS: Mackie HR-824 self-powered monitors
RECORDERS: Yamaha MD8 MiniDisc recorder
OUTBOARD GEAR: Digidesign 888 I/O
EFFECTS: TC Electronic 2290 delay; Yamaha SPX-990; Eventide Harmonizer; Lexicon Jam Man [3]; Dribble and Drool studio pedal board with The Amazing Hotcake Pedal; Dunlop Cry-Baby wah wah pedal
MICROPHONES: Audio-Technica AT4060; Shure: KSM-32 and SM57
MIC PREAMPS: Avalon VT737SP
SAMPLERS/KEYBOARDS/MIDI MODULES: Yamaha EX-5 and FP-8 keyboards; Emu 6400 sampler; Roland SP808 sampler
COMPUTERS: Apple PowerMac G3 with Glyph 18 GB hard drives for storage [2]
SOFTWARE: Emagic Logic Audio
INSTRUMENTS: Kevin Ryan acoustic guitar; Martin acoustic guitar; "loads of other cheap guitars (electric and acoustic)"; Buddy Rich's old drum kit, courtesy of Artie Butler; Takamine B-10 upright bass
MISCELLANEOUS: Sabine tuners; D'Addario strings; Monster Cable; Lloyd Baggs direct boxes and pickups; Davitt and Hanser microphone stands
PRODUCTION NOTES: According to engineer Marc Moreau, "The recording sessions for the Janis Ian album revolved around a G3 Mac running a Pro Tools Mix|24 system, with [Emagic] Logic Audio Platinum for MIDI and audio. The entire project was recorded digitally using 24-bit converters and was stored on two Glyph 18-GB hard drives so that there was the flexibility of working on any song at any time. Additional digital samples were from an Emu 6400 and Roland SP808, as well as those created by the group.

"Janis's vocal setup consisted of an Audio-Technica AT4060 microphone through Monster Cable to an Avalon 737SP mic preamp, and then into Pro Tools. There was no limiting during the recording process. The rest of the instruments we recorded with Shure, Audio-Technica, and old Neumann microphones run through the same audio chain using the Avalon.

"We originally intended to use the living room for a control room, and record vocals in the bathroom, guitars in the bedroom. However, by the end of the

continued on page 132

Picture This

The composers of Sherman Oaks, CA's Forte Muzika use their project studio to compose for today's big releases

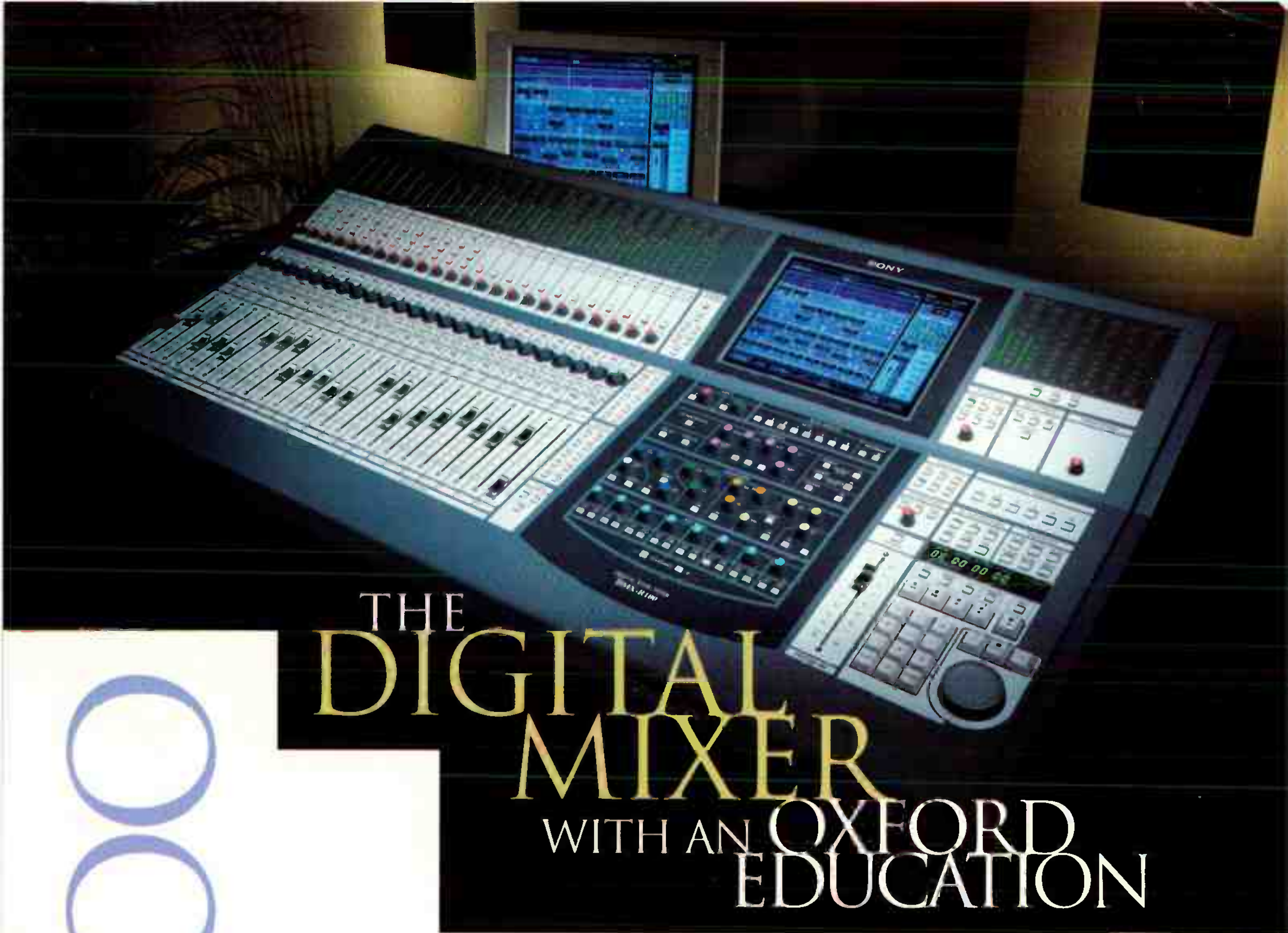
STUDIO NAME: Forte Muzika
LOCATION: Sherman Oaks, CA
KEY CREW: Elia Cmiral, composer; John Whynot, engineer/mixer/MIDI guru; Mike Flicker, music editor
CREDITS: Elia Cmiral has composed music for *Ronin, Stigmata, The Wishing Tree, Six Pack,* and *Battlefield Earth*; John Whynot has worked on scores for *Battlefield Earth, Austin Powers, Austin Powers: The Spy Who Shagged Me, The Astronaut's Wife, Price Of Glory, Ready To Rumble,* and *Wild Things,* as well as CD releases from Bruce Cockburn, Blue Rodeo, and Tamara Silvera; Mike Flicker's credits include *Stigmata, Lady In Question, Six Pack, Austin Power: The Spy Who Shagged Me, Judging Amy, Battlefield Earth, Lansky,* and *Beggars And Choosers.*
MIXING CONSOLE: Yamaha 02R [2, with a third one on the way]
MONITORS: Yamaha NS-10M; Tannoy System 12
AMPLIFIERS: Hafler Pro 5000
RECORDERS: TASCAM: DA-88 [3] and 122 Mk III cassette; Panasonic SV3700 DAT
OUTBOARD GEAR: Digidesign 888 I/O; BBE 662 Sonic Maximizer; dbx 1066 compressor/gate
EFFECTS: Lexicon: PCM90, PCM70, and Alex; Eventide H3000D/SE; Korg DRV3000
MICROPHONES: Neumann TLM 103, U 87, and KM 84
MIC PREAMPS: Avalon 737 [2]
SAMPLERS/KEYBOARDS/MIDI MODULES: Yamaha: AN1X and FS1R; Akai: S5000 [7] and S1000 [1]; Roland: JV-1080 and S760 [4]; Waldorf Microwave XT
COMPUTERS: Apple: PowerMac 9600 (with G3 accelerator card), PowerMac G3
SOFTWARE: Mark Of The Unicorn Digital Performer; Digidesign Pro Tools 5
VIDEO EQUIPMENT: Sony SVP9000; JVC BR-T1000
POWER CONDITIONING/BACKUP: APC
PRODUCTION NOTES: Elia Cmiral says, "Aside from the obvious requirement of great music, I have found that the key to a successful score that fully supports the director's vision is music preparation and organization. First step is spotting the film and deciding when and what to play, and when not to play. This is where the foundation for the whole concept of the score is laid.

"Second, I develop a palette of sounds from my extensive library and decide on the instrumentation of the orchestra and other live performances. The choices I make here will determine the voice of the score.

"The next step is extremely important to me. A successful film score is not a couple of well-written, emotive themes strung together. The score must have a form that supports the dramatic structure of the story and film. This is a great challenge and one which — if done well — makes the actual composing process pure joy.

"To aid me in designing the structure, I group the cues into thematic families. The cues that best represent each thematic group are attacked first, establishing early on the unique voices of not only each group, but also that of the full score. This allows me to evaluate the entire score as well as to effectively communicate my music to the producer and director right from the beginning."

PHOTO BY JOEY SKIBEL/FILMTRAXX



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RCA MI-6206-E

A spotlight on one of RCA's Aeropressure Microphones

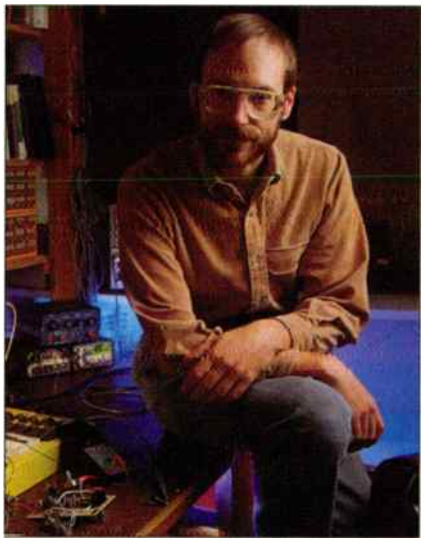
MICROPHONE NAME: RCA MI-6206-E
FROM THE COLLECTION OF: Bill Meredith, Cinesound Company, NYC
YEAR OF MANUFACTURE: Circa late 1930s
TYPE OF MIC: Moving coil dynamic
POLAR PATTERN: Non-directional
FREQUENCY RANGE: 60 Hz to 10,000 Hz
EFFECTIVE OUTPUT LEVEL: –57 dBm, (0 dB=0.001 watt) for a sound pressure of 10 dynes/square centimeter
OUTPUT IMPEDANCE: 250 ohms
HUM: -109 dBm referenced to a hum field of 1 x 10^{-3} Gauss
DIMENSIONS: 6 inches (length) x 2.0625 inches (diameter); baffle diameter is 4 inches
WEIGHT: 2.25 pounds
MIC NOTES: Primarily intended for public address and broadcast applications, this MI-6206-E microphone is one of several RCA microphones known as "Aeropressure Microphones," and features RCA's "Dark Umber Gray Metalustre" exterior finish. Other mics with similar response and directional characteristics include the 6206 (the same mic with a black finish), the 6207 (high-impedance output with black finish), and the 6207-E (high-impedance output with Dark Umber Gray Metalustre finish). The MI-6206-G and 6207-G models are similar to the "E" designations, except that the supplied cable had a length of 6 inches. The MI-6206 was built to withstand mechanical shock and wind disturbance resulting from outdoor use.
USER TIPS: The RCA MI-6206 is essentially non-directional at frequencies below 2,000 Hz, and was supplied with a removable baffle that served to vary the directional pattern of the mic. The baffle could be mounted on the microphone in one of two manners: When placed with the concave surface forward as shown, the mic is at its most directional. When the baffle is reversed and the convex surface is facing the sound source, the mic exhibits less-directional characteristics.

Technical data furnished by Clarence Kane of ENAK Mic Repair, Pittman, NJ, and Jim Webb.

PHOTO BY EDWARD COLVER

Frankfurt Trends

Find out what's going on the European music scene — and how its affects the States



BY CRAIG ANDERTON

The Frankfurt Musik Messe, held each winter in Germany, is not just the world's largest music trade fair, it's also a window into global trends. Although NAMM is becoming more internationalized, Frankfurt is still the place to tap the planet's musical pulse.

Check out this month's Product Views for more on new gear at Musik Messe, and there will also be saturation coverage at www.MusicPlayer.com. But let's take a look at some of the main trends at the show.

INVASION OF THE PLUG-INS

Although there is an increasing number of specialty hardware signal processors — tube-driven devices, optical-based analog compressors, high-end digital reverbs — it seems the heyday of the "Swiss army knife" multieffects is over. People are willing to spend a bit more for top-quality, single-function hardware devices; but, for everything else, plug-ins rule.

Three main factors have come together to make plug-ins the dominant lifeform in signal processing, and, increasingly, signal generation. First, Steinberg's VST 2.0 specification has enjoyed near-universal acceptance on both the Mac and Windows platforms, and DirectX — while not as powerful for musical applications as VST 2.0 — is a second *de facto* standard for Windows simply because of Microsoft's dominance. Therefore, the standardized formats are in place that lets the industry move forward.

Second, prices are getting very affordable. Unlike the early days of TDM plug-ins, which could easily cost $500 or more, VST and DirectX plug-ins enjoy the kind of economies of scale that allow for reasonable pricing. And third, computers have become powerful enough that using plug-ins doesn't have to be a frustrating experience involving stuttering audio and flaky fidelity. Even entry-level computers (by today's standards) allow running multiple plug-ins fairly smoothly.

SYNTHS GO SOFT

A similar split is happening in synthesizers. The high-end hardware synth is making a comeback (*e.g.*, Alesis Andromeda, Waldorf's top of the line synths), and although some solid budget synths are appearing — who doesn't want a Nord Micro Modular — software synths are bridging the price/performance gap. Some of these are stand-alone applications, such as the VAZ Modular synth (check out my review in the July 1999 issue of *Keyboard* magazine) but, increasingly, the trend is toward plug-in synths that work under VST 2.0.

The arguable star of the show was Native Instruments' B4, an uncannily realistic emulation of the ever-popular B3, but samplers, synths, and much more are showing up as virtual devices. TC Works even showed plug-in synth modules (oscillators, filters, etc.) that plug into the FX Machine matrix in Spark, their Mac digital audio editor.

And VST isn't the only game in town. CreamWare prefers the hardware approach, as it places less demands on the host computer; now-legendary synth designer John Bower (Prophet-5, Prophet VS) has a company dedicated to designing ultra-cool modules for CreamWare's Pulsar system.

Between virtual processors and virtual synths, the trend toward virtualizing the studio is gaining what appears to be unstoppable momentum. With one-giga-Hertz processors about to become commonplace, this trend will continue.

THE MP3 DJ

Although vinyl-oriented turntablists tend to scoff at MP3-based DJ setups, bringing computer technology to the DJ is also starting to happen in a big way. The premise is simple: Get a laptop with a sound card and a secondary 10 GB drive, fill it with 4,000 or so tracks in MP3 format, load in some DJ software that lets you create playlists of the tunes and do beat-matching, and you're ready to gig.

Sounds good, but much of the DJ experience is about hands-on control — and the big news is that these MP3 setups now typically include controllers that resemble standard DJ mixers. Not only do they let you do crossfading and mixing, they also have "turntable controllers" that vary the clock speed so you can do pitch-oriented changes like pseudo-scratching.

Overall, the concept of the DJ is starting to mutate. Some are more like conductors, essentially putting together music in a mix, while others are more like players — they do the turntable manipulations, play with effects, and so on. These are the DJs who increasingly incorporate groove boxes and processors in with their turntables, CD players, and/or MP3-based setups.

IS IT REAL OR IS IT VIRTUAL?: Native Instruments' B4

Pitch'n Time

New in version 1.1
Phase coherent processing of up to 48 tracks.

TIME STRETCHING THE NEXT GENERATION

After years of research, Serato have developed a completely new, patent pending algorithm that time stretches and pitch shifts audio with totally unprecedented quality. Now, for the first time, you can say goodbye to the multitude of distortions time stretching has produced in the past. This powerful technology is being made available for the first time, as "Pitch'n Time™", an easy to use Audiosuite plug-in for the Pro Tools platform.

HISTORY

Up until now if you tried to time stretch a sample, you didn't have much choice about the quality you'd get. If you were lucky you might pull it off, but more often than not the processed sound would suffer from a whole range of distortions. From the more obvious warbling, harmonic distortion and echoes, to more subtle rhythm and timing fluctuations, very rarely would you get a professionally usable result. That's because traditionally, time stretching algorithms have worked by repeating or dropping blocks of samples (micro editing) in an attempt to extend or contract a sample's time domain waveform.

UNPRECEDENTED RESULTS

Pitch'n Time™ produces impressive results regardless of the source material, whether it's music, speech, ambience or sound effects. From a single note to complex stereo mixes, Pitch'n Time can be relied upon for professional results every time. Pitch'n Time™ can pitch shift up or down 12 semitones and simultaneously time stretch from 50% to 200%, all without having to adjust any confusing or non-intuitive parameters.

FEATURES

- Modify tempo from 50% to 200% of original and simultaneously shift pitch by ±12 semitones.
- Unprecedented processing quality.
- Unique Patent-Pending Time Compression/Expansion and Pitch Shifting algorithm.
- No loss of timing accuracy.
- Process monophonic and polyphonic material.
- Process stereo tracks without phasing.
- Process Dolby™ matrix encoded tracks without losing surround information.
- Select time stretch by % tempo change, % length change, target length, or target BPM.
- Select pitch shift by % frequency change or semitone shift.
- Preview changes in real time.
- Full Timecode support.

SYSTEM REQUIREMENTS:

- PowerPC® Macintosh
- Pro Tools 4.0 or higher
- No minimum speed for offline processing.
- 200MHz PowerPC® required for real time preview of mono tracks.
- 350MHz PowerPC® required for real time preview of stereo tracks.

PERFECT PRO TOOLS INTEGRATION

Pitch'n Time™ integrates into your Pro Tools session so you can preview and make changes in real time. The novel "capture length" function makes it easy to match sample lengths with just a few clicks. To pitch shift as well, just enter the shift in semitones and cents, or adjust the pitch directly with the slider or jog wheel. You can use the built-in tone generator to help tune by ear, all in real time.

PRO TOOLS | Mac

download the demo at
www.pitchntime.com/demo/

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CIRCLE 51 ON FREE INFO CARD

DOT-COM MANIA

The "UploadYourMusicHereAndGetDiscovered.com" concept is catching on all over Europe, although, as in the States, the whole genre is in its infancy; it remains unclear just how effective this new medium will be in breaking artists. But the Frankfurt show administrators decided the dot-coms were important enough to merit a specific exhibition space, which, although only a tiny fraction of the size of a standard exhibit hall, is nonetheless a first.

THE IMAC STRIKES BACK

The renaissance of the Mac is happening over in Europe, but the emphasis is definitely on the iMac. Europeans are suckers for good industrial design, and, if the iMac is anything, it's the epitome of superb industrial design. This bodes well for continuing support of the Mac from premier European software developers such as Steinberg, Emagic, etc.

CULTURE TRENDS

The dance music phenomenon is so entrenched in Europe that generic dance music has become sort of the continent's Muzak. Whether you're on hold with background music, at an airport, shopping at a department store, or turning on the pop music TV shows, the four-on-the-floor kick drum is everywhere. As a result, the genre is once again mutating in order to remain fresh. Hardcore techno is getting harder, DJs are adding an even more eclectic combination of vinyl into their sets, and the hip-hop influence is being felt more and more in mainstream music.

But one of the largest, and perhaps most disturbing trends, is the growth of CD piracy. The recording industry estimates that about 30 percent of all music that people play has been pirated, not purchased, and this doesn't just seem to be the voice of paranoia. Some independent labels I talked to confirmed that it's getting harder and harder to survive when CDs are being duplicated on such a massive scale.

Although initially somewhat skeptical of these claims — after all, it's easy to blame a scapegoat for your problems — a quick trip to the Saturn electronics stores (sort of like Circuit City on steroids) was enlightening. As soon as you walked in, there were massive displays of blank CDs, including budget packs of CD-RWs that listed for about $1 each. While nowhere near as prevalent as blank CDs, blank MDs are also huge compared to the U.S. Either the Germans generate a lot of data and are very, very conscientious about backing it up, or there's a ton of copying going on. Everything I heard points to the latter.

One record company owner, Alex Merck, stated that he believes one reason companies are putting an emphasis on secure downloads is to try to put the CD genie back in the bottle. CDs are easy to clone; secure downloads would be much harder to duplicate, at least initially.

Why is piracy so much bigger over there than in the U.S.? It's hard to say, but the real question is how long it will take for the U.S. to reach similar piracy levels. If it no longer becomes possible to make a living from selling music, what does that mean to musicians and recording engineers? Will live performance and merchandising be the only real way to make money, with recordings being more of a promo device? Will secure downloads

continued on page 41

Whatever Happened to Air?

PHOTO BY JONATHAN POSTAL

In which our fearless reporter searches for dynamics in home mastering trends

BY AL KOOPER

One of the initial sonic advantages of CDs was the headroom afforded by digital mastering. That is, the dynamic range increased tremendously. If a project was recorded digitally, the silences were majestic; no lingering tape hiss — just gorgeous, empty silence.

Despite this increase in dynamic range, professional mastering engineers in the year 2000 are *pulling their hair out* over the amount of compressed and limited two-track mixes that are dumped in their laps. They're forced to spend most of their time trying to get the compression *out* of your second-guessed mix. Back in the '70s and '80s, as a producer, one would valiantly attempt to reproduce exactly what was on tape (analog) and get *that* on the mastering lathe. Major compression and limiting of the two-track mix was frowned upon; *it sucked the dynamics right out of the program material.* It was only employed to control a spontaneous flash peak of volume — and we would still agonize over having to keep it in the signal path through the entire song just for one or two peaks.

Nowadays, with the advent of DAWs, one just takes the cursor and calmly taps down any peaks without cluttering the signal path with any hardware (virtual or otherwise) at all. So why are some of you geniuses slapping compression and limiting onto your final mixes? To make them competitively louder? Why not just raise the bleedin' volume as high as you can and tap down the five or six highest peaks? Ahhhh, it's not your fault...and that's what I'm ranting about here. All these companies are bundling compressors and limiters in with their systems and defining in their manuals to employ them on the two-track masters instead of supporting the tap-down method of digital mastering or some other alternative that preserves dynamics. Is it any wonder that all the hit music on the radio has a certain slavish sameness to it? *Guess what that sameness is?*

> *All these companies are bundling compressors and limiters in with their systems and defining in their manuals to employ them on the two-track mas-*

You got it — compression and limiting for the sake of compressing and limiting, not because it's actually *needed.* And I'm not saying there isn't program material out there that doesn't benefit from this sort of tweaking, but now it's become *rote.* You finish your mix, you EQ it, you compress it, you limit it — it's the year 2000, after all, and most people have the wherewithal to do all these things right in their own home studios. We, as a country, have the hydrogen bomb at our disposal. The President doesn't whip it out (the H-bomb, that is) every time there's a problem with foreign policy just because it's at his disposal. It's too drastic a measure. Maybe we need to start thinking of our audio accoutrements as H-bombs. Don't destroy a two-track mix just because you *can.* Negotiate some simpler way to solve your problems instead of throwing the baby out with the bathwater. Perhaps then your mix won't sound like a cookie-cutter replica of every other mix out there that's equally devoid of dynamics. Try and come up with an original solution.

I had a student in one of my classes at Berklee who had cut a song, and the acoustic guitar strings on one track were squeaking every time the player slid his fingers. It was too late to recut the track and the squeaking level was cutting into the vocal. He asked me for help. I suggested using a notch filter to roll-off the frequency of the squeak on the guitar track. Off he went to joust with sonic windmills. The next week he came back to class with nary a squeak on his track. I asked if he employed the notch filter. He said he got out Pro Tools instead and isolated the waveform of the offending squeak. He then told the computer to remove that waveform every time it appeared in the track. He pushed a button and all the squeaks went to doggie heaven.

Now that's an example of using current technology to your advantage.

And he thought this up himself, God bless him. So please don't think I don't appreciate new wrinkles in the technology. I just love hearing the air and ambience in a digital track, that's all. Don't suck it out just 'cause you can or someone suggests it — 'nuff said! EQ

William Wittman is a multi-platinum Producer/Engineer, former Staff Producer/ A&R Vice President (RCA / BMG Records and Columbia / Sony Records), Musician and Songwriter. His career truly covers all the bases.

"I'll tell you a secret; I've always had a love hate relationship with near-field monitors. But these LSR's have changed all that. First, they're just easy to listen to. They've got plenty of full, real bottom, great stereo imaging, and they go loud enough to feel right. Plus, they translate incredibly well to the rest of the world. They're just *musical.* Wow; good sounding speakers I can trust! It's love-love."

LSR. Profiles

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Since its introduction in 1997, the system-engineered JBL LSR Series has become a favorite choice of engineers, producers and performers, many of whom have also become its most loyal advocates. More important, this acceptance is found in every major geographic area of the recording industry; from Los Angeles and New York to Nashville and London.

LSR 32 LSR 28P LSR 12P

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During a half century of building the most technically advanced studio monitors, JBL has developed a long list of working relationships with key recording professionals around the globe. As a direct result of this unique collaboration, these industry leaders have chosen JBL monitors more often than any other brand. Not once or twice, but consistently for decades. In fact, JBL monitors are a part of the history of recording itself. Consider as examples, the now fabled JBL 4200 and 4400 Series that, at their launch, actually defined an entirely new standard and new category of monitor. Such is the case now with the entire LSR line.

JBL PROFESSIONAL

Joel Jaffe is an award winning Engineer/ Producer / Composer and co-owner of Studio D Recording, Inc., home to a long list of platinum and Grammy Award winning albums and artists. Currently, Joel is working on DVD surround mixes for some of the industry's top touring acts. LSR surround systems are his choice for stereo and 5.1 channel multimedia projects.

"The THX Approved 5.1 JBL LSR28P with the JBL LSR12P subwoofer provide an extremely linear response, great transients and full-frequency monitoring in a near-field set up. In addition, the LSR speakers allow us to be able to go between stereo mixing and multi-speaker formats, which is absolutely necessary today in a state-of-the-art studio."

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Kevin Shirley

PHOTO BY ED FREEMAN

A caveman in Paris

BY MR. BONZAI

Mr. Bonzai: So why are you known as the Caveman?

Kevin Shirley: Because I've always looked like this, I guess. Big, hulking Neanderthal with long hair.

How do you like working here in Paris? What's different?

The language is not the same, and the women are magnificent! I feel like I'm a mute at a runway show when tramping along the Champs de Elysees. You see a beauty in purple skintight low hips pants, and long blonde hair, just two yards in front of you. You say to yourself, "Hey, it's Sunday, I've no plans. I don't have anything to do, and I'm all grown up — which way do I go?"

But, seriously, the studio here is phenomenal. The vibe is great, I don't have to deal with business for four hours before work; I just get up, have a *cafe noir*, and a *pain* — and off to work, which is a 40-minute walk along the Bois de Boulogne (a big woods on the outskirts of Paris). Also, the English folk living here don't have to pay tax back 'ome!

What's the most important thing to remember when you are mixing?

To stop when it feels good.

What are your main recording tools?

Neve 8068 console, Studer A800, Urei 1176 compressor, ATR 102 1/2 inch mastering machine — and I hate to admit it, but I just spent $50k on a rocking Pro Tools system, which is awesome when used correctly. I've been using Grace preamps, which blow my mind. Guy Charbonneau turned me onto them when we recorded the Page/Crowes thing live at the Greek Theatre.

Suspect: Kevin "Caveman" Shirley

Ancestry: Born in South Africa, grandparents Irish and English. "I moved to Australia, where my son was born to a Dutch woman. I have U.S. permanent residence, an Irish passport, EE residence, Australian residence — and South African birthright — so I can pretty much live and work anywhere!"

Occupation: Producer, Engineer, Mixer

Residence: New York City

Vehicle: Saab 9-3 convertible

Diet: "I'm the Caveman, so give me a big hairy steak! No, anything really, but I'm not crazy about ingesting French preparations of assorted internal organs."

Identifying Mark: Laughter lines.

Credits: Suspect has produced/engineered/mixed Aerosmith's *Nine Lives*; Silverchair's *Frogstomp*; Black Crowes's *By Your Side*; Jimmy Page and The Black Crowes *Live at the Greek Theatre* (Internet only at www.musicmaker.com); Journey, Our Lady Peace, and Jon Bon Jovi for *Armageddon*; Billy Joel for *Runaway Bride*; plus Iron Maiden, Divinyls, Cold Chisel, Dream Theater, etc. See suspect's Web site: www.cavemanproductions.com.

Notes: Meeting took place in Paris at Studios Guillaume Tell, a member of the World Studio Group, during sessions for Jimmy Page and the Black Crowes, as well as for Iron Maiden.

Where'd you get started as an engineer and why?

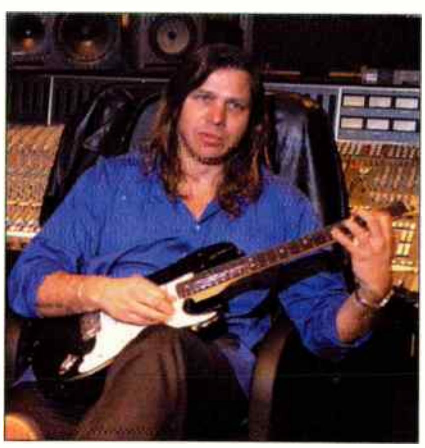
I started out in radio at the South African Broadcasting Corporation (SABC). I have always, and still do, love sonics — the sound of records! Television was just beginning there then, so everyone was saying, "Why radio," but I have always been interested in sonics — and I still dislike the way the video medium takes away from the audio experience. MTV has changed, and I think destroyed, the way we "hear" music.

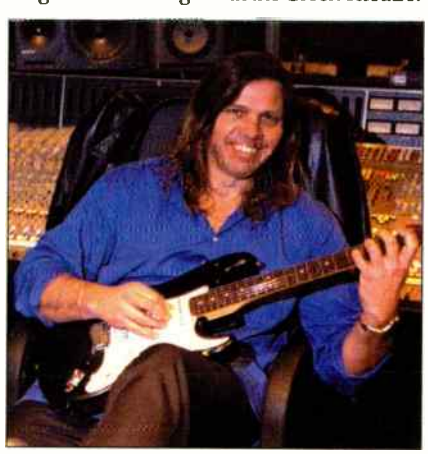
If you were a musical instrument, which would you be?

French horn.

What's wrong with the music industry?

Wow, don't get me started — "bottom line" is the biggest problem. Quick bucks, big bucks! I see great career bands, entire genres — like the Black Crowes — get f*cked over because it's easier to milk "Living La Vida Loca" (which is a good song) to death. "Let's bury them





KEVIN SHIRLEY PHOTOS BY MR. BONZAI

because radio is not a sure thing" — like this is the only way to run the industry. Also, label execs who like to present themselves as some untouchable, holier-than-thou, abusive — in every sense of the word — mafioso. I better stop, because this is/was my career.

What music would you like played at your funeral?

"You Can Leave Your Hat On," as sung by Joe Cocker.

What do you listen to while you're driving?

Whatever was the second to last thing I worked on, newly mastered — so there's been some time and space since I last heard it. Then I listen objectively for about the first time.

If you could go back in time, before the birth of recording, what would you like to hear?

I think Beethoven's Pastoral Symphony #6 in F major — with original instruments, and him conducting. I'm sure you'd be able to hear the brook, the picnic, the storm, and the birds and insects of the meadow — his vision in music.

What is the first music you remember hearing?

Simon and Garfunkel, hammer and a nail song.

What did you learn from Iron Maiden?

So soon, Mr. B! I learned how to use Pro Tools and bite my tongue. I also learned that not everyone hears things the same way, and it's all valid.

Who were your musical heroes when you were getting started?

Jimmy Page, Trevor Rabin, James Patrick Page, Pete Townshend, Richie Blackmore.

Who do you respect and admire today?
Mutt Lange, Bob Clearmountain, George Massenburg, John Kalodner.
Is there anyone in the world you would like to meet?
Some of those girls on Howard Stern's E! night show! I saw one the other day, Jaime someone, who's starring in *Son of a Beach.* My God, she was unbelievably gorgeous. I'm a window shopper — just lookin'.
What is your strangest characteristic as a human being?
Well, I'm a dad — and I love being single, but want more kids.
How would you like to be remembered in history?
Only as honest, diligent, and a friend. I don't need to be a hero.
What was your most ridiculous experience in a recording studio?
Aerosmith, Journey, Iron Maiden — I have lots of stories. Every band has something amazing and then something ridiculous about them, but Aerosmith takes the cake. You know, truthfully, I love the guys — but I inherited a big problem with *Nine Lives*, which has been written about a lot. Steven loved the record he did with Glen Ballard, but he was *alone*! All the label people, and the rest of the band were unhappy with the record — and it was not "*Pump 2*"! So, every time we needed to tackle a situation that Steven had already spent two years on, it could get ugly — and did! But other times were just mind-blowing. All of those guys just jamming on nothing for 30 minutes together was unforgettable. And Steven is the greatest talent I've ever worked with, just a monster musician!
What old saying do you hate the most?
"Man, this is going to be huge — we're gonna take it to the top!"
What animal do you identify with?
Dolphin — all that swimming and, well, you have to read Tom Robbins's *Skinny Legs and All.* My all-time favorite book.
Who is the most amazing artist you've worked with?
Well, definitely doing the Page/Crowes album, I kept soloing Jimmy's guitar and going, "Listen to this motherf*cker play!"
What makes a great producer?
I think it is someone who can capture and enhance the essence of a performer and performance. I cannot judge greatness by the quantity of airplay.
Have you ever witnessed a miracle?
The birth of my son Josh is as close to being overwhelmed by the magnificence of life and the lifeforce.
Any advice for getting a good start in the music business?
Sleep with the studio owners!
What would you like Santa to bring you this year?
More than anything, I'd like another child — so guess what I need? EQ

Mic Preamp Extravaganza

What do you call $75,000 worth of preamps in a single room? A party!

Some people might consider Lynn Fuston a bit of a maniac. After all, what else would you call someone who spends two months planning and arranging a weekend-long listening comparison between 33 microphone preamps?

Fuston is the owner of 3D Audio, a mastering and mixing facility in Franklin, Tennessee. He claims to have been crazy about mic preamps since he bought his Focusrite ISA-110's in 1987. Given the myriad preamps on the market these days, he was curious as to just how much sonic difference there really was between the various offerings. Were the expensive preamps really worth that much more than the inexpensive? What was the truth behind all the hype and opinion out there?

Lynn set out to examine these questions with the help of some friends. Here's how the 3D Audio Preamp Listening Party went down....

EQ: So what was the premise of the 3D Audio Preamp Listening Party?
Lynn Fuston: Some think it was a shootout, but it started out as a party; a bunch of engineers getting their preamps together in a single room to listen to all of them with a single source. We weren't trying to pick winners and losers. We wanted to find out once and for all what differences, if any, we could hear when the preamps were accurately level-matched. Some manufacturers insist that the differences people hear between preamps are volume-related, due to inaccurate calibration. I couldn't disagree until I had scientifically set up a listening test that absolutely ruled out that variable. That's what I wanted to accomplish with this party.

How many preamps did you have?
I was able to get 33 preamps in one control room (at George Cumbee's Classic Recording) for two days. We had a price range from the Mackie 1604 VLZ Pro at $65 per channel, all the way up to Neve 1081 modules at $4,000 per channel. All told, it was over $75,000 worth of preamps.

That's quite a lineup. What acoustic source did you use? Or did you listen to several?
Originally, I considered just auditioning female voice. I do a lot of female artist records, and I've found the preamp makes a huge difference. But, after getting all these preamps together, I wanted to hear as many sources as possible. I added acoustic guitar (miked in stereo), male voice, and also snare drum. Those are the primary instruments I use external mic pres for in the studio.

How long did you spend on this project? How many days did it take in the studio?
The planning stage took about two months. Once all the preamps arrived, we spent one day setting up and two days listening, broken into three-hour sessions per source. Add in a BBQ dinner, lots of tech talk, a few hours of looking "under the hood" at the preamps, then several hours boxing them up, and we had a very long weekend. I called it Preamp Boot Camp. Very tiring, but very rewarding. And very revealing.

Revealing? In what way?
We all thought we "knew" certain things, like the way our personal preamps sounded. Like the sound of a tube preamp versus a solid-state design, or the sonic difference between a discrete component design versus an all IC design. These are things you learn after years in the studio. Like the "sound" of a Neve preamp versus a Mackie or a Manley or an API. The listeners, ranging from professionals to weekend recordists, had dozens of years of experience with many of these preamps.

What we discovered is that, if you wipe away all the preconceived notions about brand names, designs, and circuitry, and just let your ears decide, you might be surprised at what you prefer.

How could you listen without those biases?
All the tests were completely blind. No one but the tech staff knew which was which until after the listening was complete. The listening order was completely random, determined by drawing names out of a coffee mug.

How could you keep their identities secret while sitting in front of the rack?
We stacked them all up and hung fabric in front of them while we listened so we couldn't see the meters or peak LEDs. It was absolutely fair in that regard. Dan Kennedy of Great River

PHOTO BY HARRY BUTLER

WALL MONITOR: 3D Audio's Lynn Fuston in front of the "wall" of 33 preamps.

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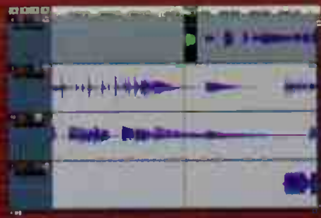

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WALL OF SOUND: A portion of the preamps that were auditioned.

supervised the patching and calibration.

How did you calibrate them?

We used an Auratone speaker duct-taped to a boom stand that we would swing into position in front of the mic. We used a pencil taped to the speaker to keep the mic-to-speaker distance consistent. We connected each mic to one preamp at a time, fed a test tone into the speaker, and then adjusted the gain on each preamp to within .02 dB. That was close enough to make sure the "volume difference" argument wouldn't come up.

Two hundredths of a dB? Seriously?

Yes. Dan Kennedy used a Tektronix scope to check for polarity and a Fluke meter to calibrate the gain. He also built a very minimal circuitry trimmer box to allow calibration that accurate.

What mics did you use?

I wanted a variety of mics ranging from expensive down to a common mic that everyone owns. I chose a Manley Reference for the high end, then a modified AKG C-414B, a Neumann KM 84, an Audio-Technica AT-4033, and the standard Shure SM57.

So how did you conduct the listening tests?

I would decide on the mic and position for each instrument, the female voice, for instance. We listened to solo voice with no processing or reverb. The signal path was mic > preamp > trimmer > console. We also recorded the results to hard disk so we could listen back later.

We sat at the console and listened as she sang through, say, preamp #1, then we wrote down our impressions. We each had a list numbered from 1 to 37 (some of the 33 pres had alternate settings), so we would repatch, re-sing, and write down our impressions of each one, until we had heard them all.

Could you possibly make meaningful judgments after listening to the same thing 37 times?

Absolutely. It was amazing how much difference each one made. The comments would range from "nasal, yuk" to "sublime." Sometimes several in a row would sound similar, but then I would hear one that just blew me away.

Each of us picked our favorites, and, among the six listeners, every person had a preference. Usually we each chose different favorites. On snare drum we had a more uniform consensus.

Did you rank the preamps by preference or make a list of favorites?

I didn't do any tabulation of the results to determine a winner or most-preferred mic pre. That would be missing the point. There are lots of written reviews that will give "one man's opinion" about the sound of a pre. I don't agree with making sonic decisions based on written opinions. I wanted this event to let listeners decide for themselves using their own ears. When it comes to preamp differences, one man's "subtle" is another man's "sublime." You've got to hear them to decide.

So tell us what you learned. Were there any startling revelations?

I learned that you should rely on your ears alone to judge preamps. That sounds obvious, but there are so many biases and prejudices out there — unless you listen blindly, as we did, you can't be truly impartial.

The big surprises? Some "revered" preamps didn't blow us away. Some that we expected to hate, we actually liked. Some we thought we would recognize, we didn't. Some we had never heard before jumped to the top of our "must-have" lists. I think everyone who was there discovered that there is no one "best preamp" for recording every-

PHOTO BY MATT ENGLE

THE MAN BEHIND THE RACK: Great River's Dan Kennedy switching preamps. Output levels were matched to within .02 dB.

thing. We all became more aware that we need to invest in more preamps!

I think that the CD of this event, the 3D Pre CD, will be very valuable in helping people decide just what they can or can't hear. It also comes with an encoded "key" so people can listen and make their notes before seeing which pre was which. Some people may not be able to hear the differences. I think most engineers will. Then they'll recognize the limitations of their current preamp lineup.

How can someone get a copy of the CD?
The ordering info is at my Web site, www.3daudioinc.com. There are also pictures and a complete list of the preamps. I had to break the recordings into two volumes just because we had so many preamps. Volume 1, which is shipping now, has female voice and acoustic guitar. You should hear the acoustic guitar tracks. It is absolutely amazing the recording you can make with just two mics, absolutely flat, with the right preamp. You need to hear it. EQ

LISTEN UP: The preamp party was not designed as a head-to-head comparison, but instead as an exercise in listening for the subtle differences between many different models.

Sounds Like a Plan

Want to succeed in the project studio business? A solid plan is essential.

BY JIM BORDNER

Maybe you're an established studio owner. Maybe (like me) you're a one-man commercial music factory. Or maybe you're a serious recording hobbyist preparing for that Glory Day when you can tell the boss to take this job and, well...give it to someone who wants it. But no matter how you're involved, you gotta love this crazy business of creating music.

I suspect that many of us are in this unpredictable business because we are free-spirited, creative people. And if there's one thing free-spirited, creative people aren't good at, it's planning. There's always something more interesting to think about: "Dude...why would I want to plan for next year when I can fool around with this cool new synth my buddy brought over?"

Hey, I'm not accusing you. Planning doesn't come easily to me, either. When I was preparing to open Gravity Music and some guy in a suit intoned the words "business plan," I just about abandoned the whole idea.

Fortunately, I found out that a business plan is both simpler and more necessary than I thought. First, a business plan forces you to ask yourself the hard questions, to dig for information, and to be honest with yourself. It's a roadmap, a course of study, a seminar that prepares you to make intelligent decisions and avoid pitfalls.

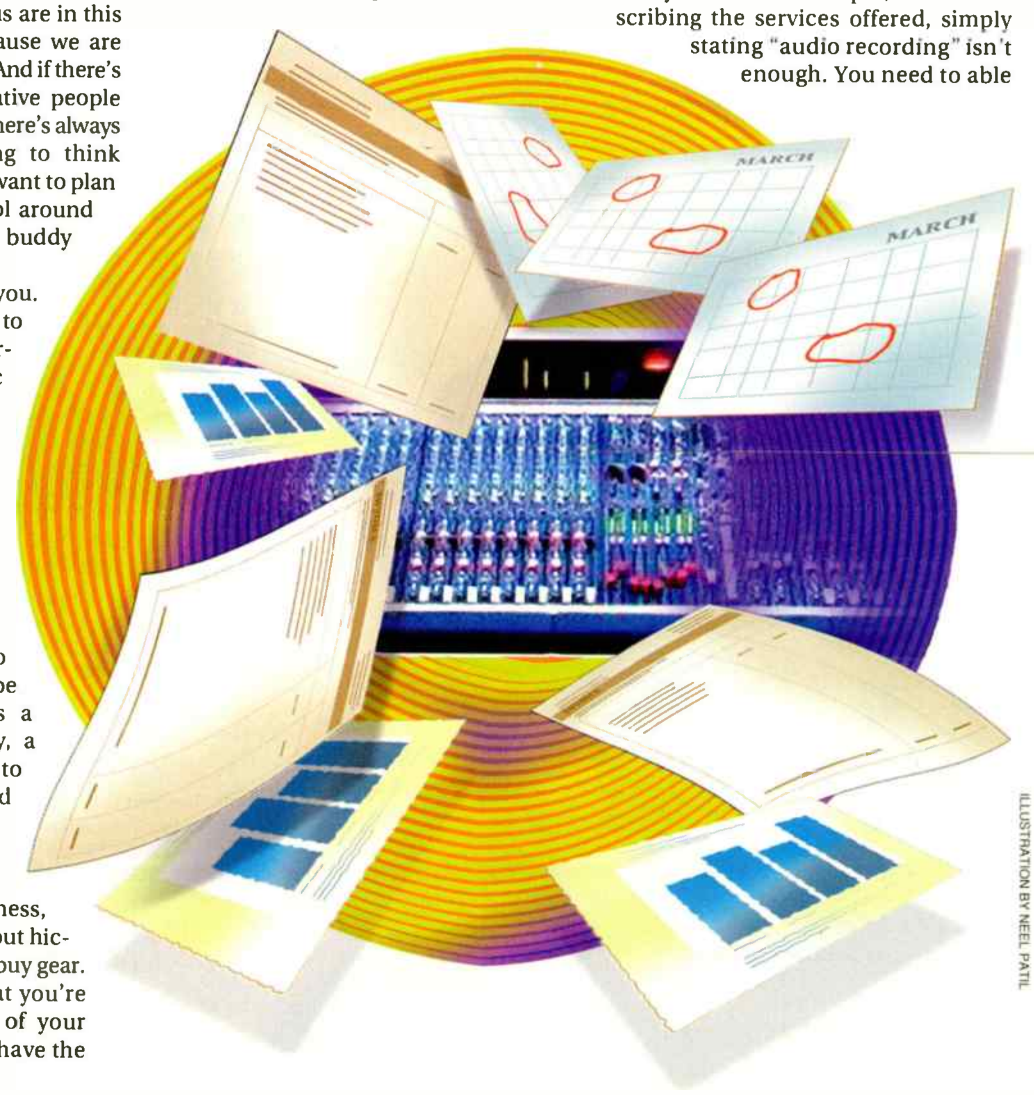
Second, a bank won't loan you money without one. If you're a new business, you'll need credit to smooth out hiccups in your cash flow and to buy gear. A bank needs assurances that you're not just flying by the seat of your pants, and that you actually have the good sense to make a go of it. You've probably heard the old saw, "Banks only lend money to people who don't need it." This is one of the lamest pieces of folk wisdom ever. If they only loaned money to people who didn't need it, they'd cease to exist. They *always* loan money to people who *need* it. They *don't* lend money to people who look like they can't pay it back.

It's not as hard to create a business plan as you might think. It's just answers to a series of questions. If you ask the right questions and answer them honestly, you'll know everything you need to know about your business' future, and so will your bank.

So let's talk about the questions that form a business plan, and what the answers mean to you and your banker. If you're just starting out, I encourage you to research the planning process (there are many books on the subject, and your banker may even give you one free). If you're an established studio, I urge you to dust off that old plan you wrote when you started. Updating your plan every year or two can be a powerful tool for keeping your business on track.

WHAT IS YOUR BUSINESS?

The answer to this question isn't "a recording studio." The right answer is a very detailed description of what you intend to do: type of structure (sole-proprietorship or C class corporation?), location, services offered — the whole nine yards. For example, when describing the services offered, simply stating "audio recording" isn't enough. You need to able



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to say, "Ajax Recording will provide tracking, mixing, and mastering services to independent recording artists throughout the Tri-Lakes region," or "Dead Cat Productions will provide creative and audio services, primarily for radio spot production, to ad agencies in Minneapolis and the surrounding cities." Your description may encompass several services, but don't throw everything you can think of in this dumpster. What you're trying to do is focus your marketing efforts.

What it means to you: Answering this question honestly, keeping in mind your personal strengths and weaknesses, will help you go after the work you really want, and point the business where you want it to go.

What it means to your banker: You're a realist, not a starry-eyed dreamer.

WHO'S RUNNING THE PLACE?

Is it just you, or do you have partners? Who are the principals of the business, and what experience, education, or special expertise do they bring to the work? Have you been working in a studio for 10 years, are you a graduate of a specialized school, do you have a gold record or an MBA on the wall?

What it means to you: You can identify the aforementioned strengths and weaknesses by taking this exercise seriously. You might also realize that, while your drummer is a wonderful friend, he's going to be a boat-anchor as a partner. Be honest. Business is tough enough without dragging dead weight around.

What it means to your banker: That you have the skills to make a product of sufficient quality to turn a buck.

WHAT'S THE AVAILABLE MARKET?

If you want to record bands and can only name three good ones around town, you might not be in the right location or be addressing the right market. As an extension of figuring out what your studio will provide as services, you need to figure out how much demand exists for those services. Get on the horn and network everybody you know, ask them who they work with now, what they like and don't like about their current supplier, and how you could get their business. You'll be using this detective work to answer the next few questions, so don't skimp; keep investigating until you've exhausted your resources, and then dig up more resources.

A side note: If the prospect of going through this exercise makes your stomach feel gurgley, quit now, because this is what your initial marketing effort is going to be like, and if you don't have the nerve for it, you're going to fail. Cold calling, following up on leads, and networking your pals is just part of being a good capitalist.

What it means to you: You have to know if there's a market for your studio. If there isn't, no amount of faith and hard work is going to help you succeed. If there is, answering the next couple questions will show you how to get your share.

What it means to your banker: That you've done your homework: You know how many customers are out there and what percentage you think you can attract.

WHO IS THE COMPETITION, AND HOW WILL YOU BEAT THEM?

If you've found a wide-open market with no competitors in sight, then call Ripley, honey, because you're the first one. From the day you open your doors, you will be embroiled in constant and ceaselessly changing competition. To succeed, you need to know who your competitors are, what they're good at, and what they're not so good at. In this section of your plan, you'll list every competitor you can think of or discover, list their strengths and weaknesses, and plan how you will sell against them.

Again, dig deep and be honest. Telling yourself, "We'll beat those guys on quality because they stink," or "Everybody I know will come here because nobody likes the engineers at Acme Studio" isn't good enough. I've said this in print before and I'll say it again: Any competitor who has been around for more than a couple years is doing something right, and you better know what it is. By the same token, nobody is so good that they can't be beaten in fair competition. Make your research in this area exhaustive.

This is another good reason for established studios to update their plan every once in a while. Your competition has probably changed significantly in the last two years (I know mine has), and it's good to take the time to re-evaluate their weaknesses and your own.

What it means to you: A clear approach to selling against the other guys. You have to know the battlefield to win the war.

What it means to your banker: Your banker will not be afraid of competition — they know it comes with the territory. They will be afraid of a business that hasn't taken a realistic look at how to win.

ARE YOU MAKING (OR ARE YOU GOING TO MAKE) MONEY?

This is the dreaded cash-flow analysis; a chart showing how much money you figure you'll take in and how much will go out. I'd recommend getting an accountant to help you here. They've done it thousands of times and know how to create a realistic analysis.

If you're a new business, you'll obviously have to guess at some things. But make educated guesses based on what you've learned about the market and the competition. A banker will recognize the difference between realistic numbers and pure graphite.

If you're an established business and haven't examined this for a while, you might be surprised to learn how your profit is being eaten by something simple such as supply costs or downtime. The cash-flow analysis can help you identify problem areas and stop the leaks.

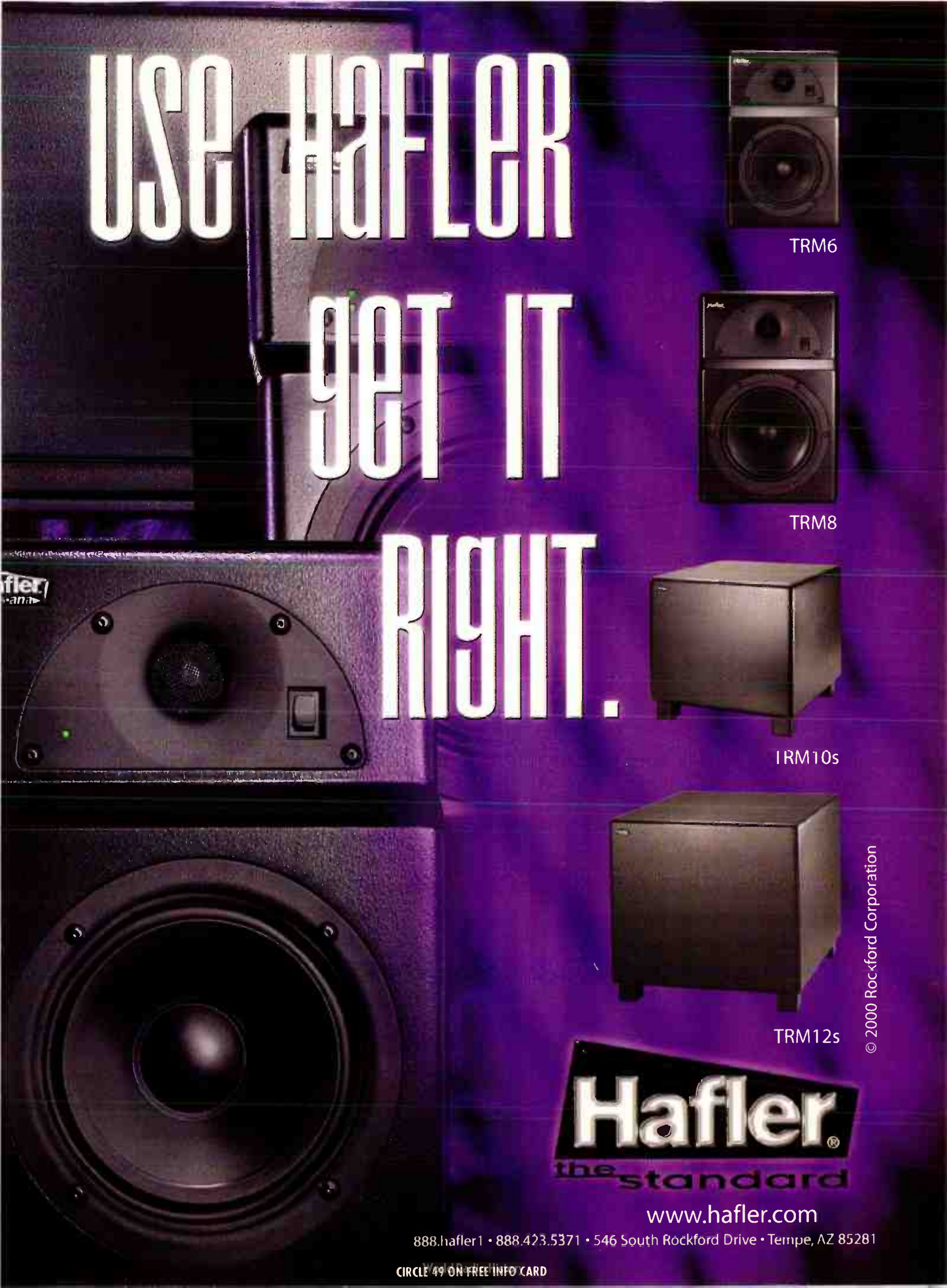
What it means to you: If you're a start-up studio, it will help you examine how much it's going to cost to stay open, and give you a figure to shoot for. If you've been around for a while, it can help you keep more of the money that flows through your facility.

What it means to your banker: You have an accountant and a solid idea of your business' cash flow.

Finally, you write a summary of all that goes before: a few paragraphs for the executive committee that say, in effect, "There's a need for this service, and I'm just the guy to provide it. Please lend me the money so I can make us all rich." Or words to that effect.

There's more to know about business planning, and you can learn most of it from your banker and accountant. But no matter how wild 'n' crazy or free-spirited you may be, don't overlook this important business tool. I hate corporate clichés, but stick this one on the bulletin board: "Failing to plan is planning to fail."

Jim Bordner is the sole proprietor of Gravity Music in Fort Wayne, Indiana, where the client list ranges from several Fortune 100 companies to the guy who put the furnace in his house. Now in his fifth year in business, he's rewriting his business plan for the third time. You can reach him at jim@gravitymusic.com.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Resurrecting Audio

DGM's David Singleton shares his tricks for remastering and remixing historic live recordings

BY JOYCE THOMPSON

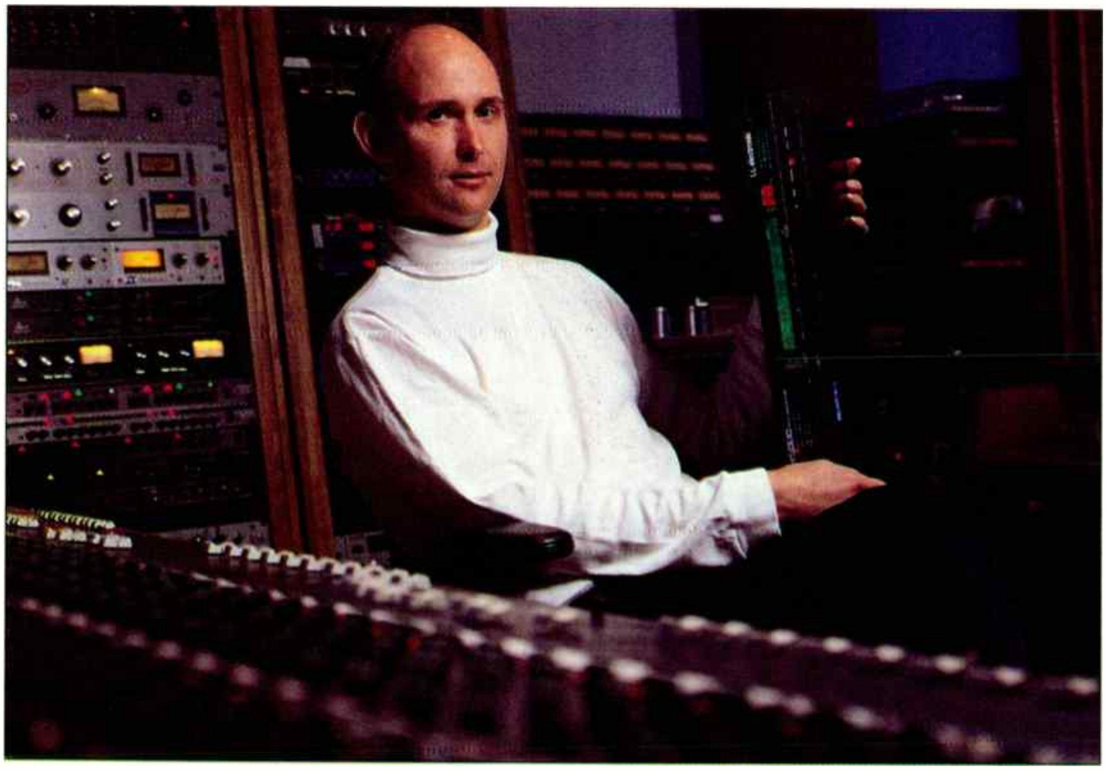
Had David Singleton's choirboy falsetto not cracked with the onset of puberty, he might have pursued a career as a musician and never turned to the profession of record producer. Had he not traveled about England in a truck that was also a studio on wheels — the Mobile — recording hundreds of live classical and pop performances, he might never have crossed paths with prog-rock legend and artists' rights advocate Robert Fripp, nor become label manager for Discipline Global Mobile (DGM). Certainly, he would never have developed his present esoteric skill at turning nasty old tapes of historic rock concerts into eminently listenable CDs. He would not be president of Internet startup BootlegTV, and he most certainly would not be in Seattle's Jungle Studios, teaching audio engineer Tom Betz, former sound supervisor at Warner Brothers, the tricks of the audio resurrection trade.

The studio is fitted with a multitrack board, but that's overkill when you're dealing with tapes recorded in mono — at best, in stereo — up to thirty years ago. Singleton's tool of choice, his "magic box," is TC Electronic's M5000 digital effects processor, a piece of equipment he first learned about from the mastering engineers at London's Metropolis back in 1996. "They were using it incredibly subtly, as mastering houses do, half a dB of compression," Singleton says. "It's quite wonderful. You go in there and sit for half an hour discussing whether this half dB sounds better or worse and whether it's the right thing to do."

Metropolis was just one of many places where David Singleton sought technical mentoring when he decided to undertake the restoration of at least some of the 385 live King Crimson shows that guitarist Fripp and other band members had on hand. Their tape archive existed in almost every known recording format and was in every sort of deterioration and disarray. Surely, Singleton assumed, sound engineers already knew how to do the kind of rescue he had in mind. In fact, the mixing and mastering engineers from whom he sought advice appeared to think he was quite mad for even considering dealing with the old tape trove. Why not just toss it in the dumpster and start fresh?

Encountering the TC Electronic M5000 was his first real bit of luck. "It was one of the very first high-end complete digital mainframes. It's a digital reverb, digital compressor, digital anything, and it sounded very good. So I got in touch with someone who could lend me one for a month." Singleton grins. "The way to sell anything is to give it to you for a month. I shouldn't say this in print, because I'll get lent a lot of things, but I don't think I've ever taken anything back."

The project Singleton was working on when the M5000 arrived for its trial run was the board recording from a King Crimson concert in the late 1960s. King Crimson has always had an avid, almost rabid following, which includes a couple of thousand people who will pay almost anything for almost anything that has to do with the band, making it one of the most frequently, and most lucratively, bootlegged bands in rock history. The



PHOTOS BY MICHAEL MATISSE

DO YOU BELIEVE IN MAGIC?: David Singleton and his "magic box" for restoration — the TC Electronic M5000.

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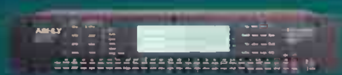

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band's fans are the sort known today as "early technology adopters" — gearheads who even years ago were willing to smuggle personal recording rigs into clubs or concert halls, copyrights be damned. Singleton and Fripp were hoping to offer legitimate recordings of the same concerts through the DGM label.

"I remember exactly what I was working on," Singleton says, "a board recording that was all drums and vocals, very loud, up front. I could hardly hear the bass and the guitars. We tried EQ'ing, and it didn't work. There really wasn't a spectrum. We tried everything. We even tried compressing, but we didn't try digital compressors. The problem with a slower analog compressor is that it doesn't gain you anything. The vocal comes in, but the compressor hasn't managed to turn up the track again in time so that whatever was in the gap gets turned up."

THE M5000 MAGIC

With the digital processor, Singleton discovered he could get the attack time down fast enough, with a five-second lead time, that whatever was happening in the interval when the vocal or drum wasn't present was turned up in time.

"What you can do with a digital processor is anticipate, so it knows the drum's going to arrive, and it actually shuts down as, or just before, it hits. You can actually push snare drums from way too loud, right back into the mix. You can do that with a very fast attack and release time if you want to, so you're basically doing an automated mix in real time. Suddenly, the guitar and bass that were completely inaudible before are perfectly loud. You end up with a uniform mix."

With live shows, it's a winning technique. At 15 dB of compression, a rock concert sounds just the way it's supposed to, splendidly noisy. "You get less and less definition, but it becomes increasingly powerful, with powerful noise, which is what most people want out of a live show."

MIXING BY FREQUENCY, NOT BY TRACK

Since, when he's doing an audio rescue, Singleton only has a stereo signal to start with, he uses the M5000's three bands to mix by frequency rather than by track. "I tend to set the low one at 20 Hz, the middle around 250 Hz, and the high one about 3 kHz," he says.

"By carefully defining where you put your crossovers, you can keep the very high end — most of the metal work and the cymbals and things — out, as well as the high-end definition of the voice. And, on the low end, you separate out the bass guitar and the bass drums."

"Often before I do anything to a recording in the middle," Singleton says, "I'll go through an EQ module. If the tape wants sweetening or it's muddy or it's wrong, I'd go through an EQ module. Within these toys [the M5000], you've got an exact digital replica of reverbs. You can correct the phase in here as well, and you can delay one set against the others."

Singleton twiddles the M5000's dials to select a very particular slice of the sound spectrum. "These notch filters are absolutely amazing — I think the smallest it would go was, like, .02 of an octave. It's like an incredibly thin slice of bread, and you say, 'I don't want it to touch what's here, and I don't want it to touch what's there, I just want it to do that bit,' and it will. These sorts of utilities tend to be absolutely stunning in digital EQs. I first discovered what you could do with [digital] notch filters to cure problems on these old recordings back in 1996, when I was trying to remaster audience recordings for the CD we released as *Epitaph*."

A TIME-INTENSIVE TIME MACHINE

It takes three to four months of hard daily work to produce a saleable CD from old live recordings. "What you realize with these archival things is, it never stops. It's purely incremental. As you begin to hear changes, you think, 'My God, I can transform this. It used to sound like this, and, now, doesn't it sound great?' Then you come in the next day, and you say, 'Okay, it sounds like this now, but it could sound like this. I could make it better.' "

Singleton's single-mindedness was abetted by Robert Fripp, who'd turn up once a day to act as executive producer, bringing an objective pair of ears to the day's labors. "Robert is wonderful," Singleton says. "Somehow he can come in and persuade me to go home, put the children to bed, and then return for the evening shift."

Because they were handling the audio so much, Fripp and Singleton went to some pains to choose the DAW used for editing the projects, settling on the Sadie digital editing system, which is widely used in Europe. Repeated digital passes through other DAWs produced artifacts, while it seemed to them that Sadie left the sound unchanged.

By the time the *Epitaph* project was complete, the duo had revitalized a four-CD set of live performances. Since then, Singleton has produced two or three historic live CDs each year, all of which have sold well to the King Crimson Collectors Club and other fans.

BOOTLEGGING THE BOOTLEGGERS

The best of the restoration business comes about in those cases when a bootleg recording of a concert suddenly surfaces and Singleton gets to mix it and the existing board recording together. Each version has its strengths. The board tracks are reliable, but tend to sound flat, dry, and rather sterile — in Singleton's words, "un-live." The bootleg, recorded from somewhere in the crowd, has the aural excitement and ambient noise the official version lacks, but it has its quirks and blemishes — those moments when somebody walked in front of

continued on page 134

Cram Sessions

Producer Steve Levine explains how his tiny project studio is good for his health and the importance of strong artist/producer relationships

BY HOWARD MASSEY

The tiny project studio in Steve Levine's London home is literally no bigger than a moderately sized walk-in closet, crammed from floor to ceiling with equipment. The vestibule outside serves as a makeshift vocal booth/overdub area, with the added advantage of glass doors leading to a small English garden.

Best-known for his work with Boy George and Culture Club — including their massive international hit, "Do You Really Want To Hurt Me?," Levine has kept up an active career that has seen him work with a broad range of artists, from rock guitarist Gary Moore to chanteuse Deniece Williams to the legendary Beach Boys. On the soggy afternoon when we met, we were both under severe time constraints, but we soldiered on, shoehorning ourselves into his cramped control room for a brief but revealing interview.

EQ: Tell me about your studio — I've never seen so much equipment packed into such a small space!

Steve Levine: Originally, when I first moved to this house, I thought I'd just set up a little editing/copying room. But I was working with an act in a big studio — we'd tracked the drums and everything, and we'd run out of time with the budget we had, so the singer said, "Why don't we just try doing the vocals back at your place?" When the A&R guy heard the tracks, he said, "Those are great vocals!" I think it's just down to the fact that the singer felt really relaxed.

From that moment, I started doing more and more work here. I had already been doing programming and routining here, establishing things like keys, tempos, and song sections. But once you get something going, you just want to add more and more to it and finish it off. Normally, I do recording here and mix outside, but I have done some mixing here, too.

It's incredible that you can mix in here.

That's because the speakers [PMCs] are fantastic. They're phenomenally accurate within one small area, and their bass response is extraordinary. There's no room treatment, either, other than the fact that there's so much gear packed in here that there are no parallel surfaces anywhere! [*Laughs.*]

Such a large proportion of the time spent on records is now spent on the vocals, so working here gives you the opportunity to develop a great relationship with the singers. You've got time on your side, so they can really listen to what they're doing. Sometimes people experiment with things on the run-through takes; they may not necessarily hit the right note, but they're going in the right direction. As soon as they hear it back in context, it's much easier for them to get a picture of what needs to be done. Sometimes they'll take a tape home, go away for the weekend, come back on Monday, and they're ready to sing.

Socially, working in the house is good, too. You eat well — you stop and you have proper food. There are also hygiene issues — you tend to get ill less because the cups go in the dishwasher and they're

COMPACT AND COMFORTABLE: Steve Levine's gear-crammed project studio allows him to get better performances out of vocalists, who find the tight space less formal than a commercial facility.

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*> H. D. Wells, Bromley, England, 1899**

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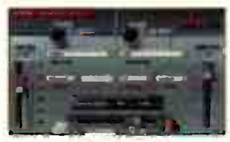

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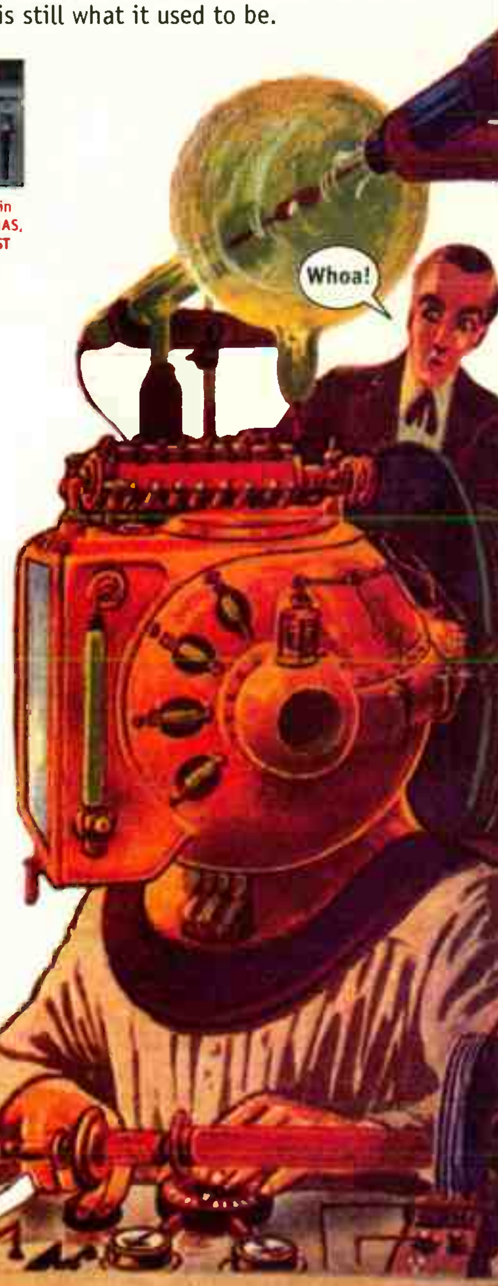

H. D. Wells**, *H. G. Wells's little-known older brother, shared his more famous siblings's visionary acumen but, due largely to his futile desire to be a rock star fully 50 years before the arrival of rock, lived most of his life in obscurity, playing in a succession of Gilbert & Sullivan cover bands in pubs in and around Bromley.

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WHERE THE FUTURE'S STILL WHAT IT USED TO BE

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02

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clean, things like that. When you're working in a commercial studio, you often get a cold or a sore throat because you're mixing with a lot of people. Especially when you're working on a vocal, you need the artist in peak condition. If it's a nice, relaxed atmosphere, everybody feels at ease and you get the best creative results.

How do you check the validity of your mixes?

Generally, I burn a CD and go play it in my car. But if it sounds good on the PMCs, it generally sounds good everywhere else. It's when you have speakers that are tricking you that you run into problems. Normally, it's the bass end that lets people down — you end up with not enough or too much because of a problem in the room. I knew there might be acoustic problems, so I chose speakers specifically for this room, auditioned lots of different ones right here.

What sort of things do you listen for on a demo that make you want to work with an artist?

The song is always the most important thing. I've learned my lesson over the years — if you don't have a great song, it doesn't matter how good the production is. But the word "song" is quite broad; it can also mean a good idea for a song. For example, take "Bohemian Rhapsody." That's not a normal, traditional song — God knows what the demo of that sounded like — but sometimes I can hear the concept in a demo that I'm sent; I can hear what they intend to do. So it doesn't have to be a traditional three-minute song, but it's got to be a good idea or concept for a song.

For me, a demo needs to be as bare as possible. The more work that's done on the demo, the harder it is for me, because people are set in their ways; they can't see the forest for the trees.

What sort of common mistakes are people making in their project studio recordings?

Spending too long on things, and using things like Finalizers and squeezing the daylights of the music. I don't mean to be derogatory — they are fabulous products — but they need to be used in the right hands. Still, so many people don't know what compression is, or don't know how to use it properly. By "properly," I mean either intentionally abusing the compression to get the snare drum to kind of smack you around the head, or using it for subtle gain riding, where you want to control the vocal. I have no problem with abusing equipment if you're specifically using it as an effect.

A producer is a bit like an orchestra conductor in that you need to know how to do many different things. What's the most important skill to master?

Being a diplomat and having a good working relationship with the artists. That will rise above everything else.

That's more important than having golden ears or strong musical sensibilities?

Absolutely. When I was an engineer, I worked with lots of producers who knew absolutely nothing about making a record in the normal sense. They would spend their time taking the artist to lunch, doing those sorts of things. But invariably they got good vocal performances because the artist felt so comfortable around them. They were almost like maitre'ds. I have no problem with that at all, because if that's the strength that producer has, then fine.

When you're making a record, do you normally have a vision of the end product in mind?

Absolutely, yeah. You never achieve 100 percent of it; I never have in my entire life. Sometimes you go down a road that meanders, but you do need to have a picture of the end result because you do need to work to a goal. But I'm totally flexible with regard to how I get there. Sometimes you'll make a mistake with a part, and all those lucky mistakes sometimes do things in the best sense.

Overproducing is a common problem. How do you know when to stop? What criteria do you use for deciding when a track is done?

It's like sawing legs off a table, particularly when you start manipulating too much stuff in hard-disk recording. You start moving one thing, then that shows off the ugliness of something else, so you move another thing.

"If you don't have a great song, it doesn't matter how good the production is."

Nowadays, making a record is almost like making a film, where the director shoots far more than he actually uses. The singers and musicians are like the actors, the producer is like the director, and you just over-record stuff and see what works. I'm all for trying out guitar parts — they can play as many passes as they want and I'm happy to go through it later and sift through for the nuggets. When you come back to it in the cold light of day, sometimes the bit that you didn't even think was important suddenly becomes the hook.

How do you approach mixing?

You've got to look at the big picture, so I find it easier to quickly get a rough balance and then solo things and put them back in. Especially with digital desks, you have to listen as you EQ — you can't just dial in certain frequencies because you know they've worked before. Sometimes you're selecting the most bizarre frequencies; they happen to work really well in the mix because digital EQs behave in a very different way. With analog EQs, you can often dial in the values almost without thinking, without even listening, because you know they have a certain color. But with digital EQ, you really have to listen.

When we were recording on analog all those years ago, certain things would be bounced many generations down, so they would attain, by default, a different sound. Therefore, things sat in the track differently, because, for example, the top had gone off of it, or because the backing vocals had been EQ'd and bounced so many times, they ended up being a very processed sound. These days, we have open tracks for everything and nothing is second generation, so everything is full range. You can't easily find a hole in the track, so now it's more important than ever before to put a few things — not everything, or you're back to square one again — through a valve [tube] compressor. You have to mix and match, filter a few things, put a few things through a synth, use different EQs just for the sake of it, run signals through different pieces of outboard gear in bypass mode...just to change the way it sits in the track.

That would also serve to change the phase relationships.

Sure, changing the phase relationships can make a huge difference. Putting the drum overheads through outboard boxes can help separate the drum kit, even if you don't add much EQ.

Do you tend to use a lot of compression?

Over the top if I specifically want that. Sometimes on bass or vocal, I'll go completely over the top, but for an effect. There are other things I record completely flat, from the mic through the preamp and straight to tape. Very often, tube preamps and DIs can add a color on the way in. Some mics or preamps can enhance the top end, almost like a fake EQ. But I work to get a different tone in each track, just to get each one to sit a little differently in the mix. The big problem with overusing all this stuff is that you end up with demos that are flat, with no life to them, no emotion.

This interview is excerpted from Howard Massey's new book Behind The Glass, *soon to be available from Miller-Freeman Books.*

Attention to Detail

Mastering engineer Bob Katz on testing your room, choosing speakers, and why not to normalize your mixes

BY BOBBY OWSINSKI

Co-owner of Orlando-based Digital Domain, Bob Katz specializes in mastering audiophile recordings of acoustic music, from folk to classical. The former technical director of the widely acclaimed Chesky Records, Bob's recordings have received disc of the month in Stereophile *and other magazines numerous times, and his recording of "Portraits of Cuba" by Paquito D'Rivera, won the 1997 Grammy for Best Latin-Jazz Recording. Bob's mastering clients include major labels EMI, WEA-Latina, BMG, and Sony Classical, as well as numerous independent labels.*

EQ: What's your approach to mastering?
Bob Katz: I started very differently from many recording engineers that I know. Number one, I was an audiophile, and, number two, I did a lot of recording direct to two-track. I work well with rock 'n' roll and heavy metal, but the sound and tonal balance of a naturally recorded vocal or naturally recorded instrument is always where my head turns back to. I find that my clients, while they don't necessarily recognize naturalistic reproduction as much as I do, love it when I finally EQ a project and make it sound what I think to be more natural.

What do you think makes a great mastering engineer? What differentiates somebody who's great as opposed to somebody who's merely competent?
Attention to detail. I will bend over backwards to get something right, even if I have to do it off the clock. Not to say that I don't charge for my time, but if I make a mistake or I feel that I could've done it better, the client will always get my best results. I think the answer is great attention to detail and extreme persnicketiness, stick-to-itiveness, and discipline. The desire to just keep working at it until it's as good as the sound that you have in your mind, and to keep trying different things if you're not satisfied.

What makes your job easier?
This is almost becoming a ubiquitous answer, but I have to say that if I get the highest resolution, highest sample rate, earliest generation, uncut, unedited by anyone (or if they do cut it, leave the heads and tails alone) version, then things are easier. Unfortunately, I get more and more chopped up material these days.

The bottom line is, send us the loose material. If a client has a real good idea on the fade out that they want to do, fine. Then send us both versions; the faded and the nonfaded. That way, if it proves to be a problem in context, we can still use the unfaded version.

Is most of your processing done prior to the workstation?
I think that there are two different types of engineers. There are the engineers who like to process during load-in, and there are the engineers who like to process on load-out. Many engineers will set up an entire chain, either analog or digital or a hybrid of both, and process on load-in, and then, if it doesn't work in context, they'll go back and reprocess and then load it in again.

I find that to be a very inefficient way of working, so I'm really puzzled why they put themselves through this. The most I will do with the analog tape is go through this great EQ on load-in only because I don't want to go through another conversion again. After that, I favor having as many processors automated as possible.

Do you ever normalize?
"Normalize" is very dangerous term. I think it should be destroyed as a word because it's so ambiguous. If you mean do I ever use the Sonic Solutions normalize functions so that all the tracks get set to the highest peak level? The answer is no. Or do I ever use TC Electronics Finalizer *normalize* function to find the highest peak and bring it up to 0 dB? No, I never do that. Do I use my ears and adjust the levels from track to track so that they

MASTERING SECRETS REVEALED: Bob Katz offers advice on premastering your tracks and explains what a mastering engineer does to them.

fit from one to the other, then use compressors and limiters and expanders and equalizers and other devices to make sure that the highest peak on the album hits 0 dB FS? Yes, I do. I don't call that normalizing, though.

Tell me why you don't do it.

I advise my clients not to do it and I've written about it extensively on my Web site (www.digido.com). I'll give you two reasons: The first one has to do with just good old-fashioned signal deterioration. Every DSP operation costs something in terms of sound quality. It gets grainier, it gets colder, it gets narrower, it gets harsher; and normalize is just changing gain. If you've already encoded and recorded something and you want to send me the earliest generation, adding a generation of normalization is just taking it down one generation.

The second reason is that normalization doesn't accomplish anything. The biggest problem with normalization aesthetically is that it's a fallacy. The ear responds to average level, not peak levels, and there is no machine that can read peak levels and judge when something is equally loud.

> *"The first thing you should ever do as an engineer is to familiarize yourself with the LEDR test."*

Tell me how you came about choosing your monitors. And, then, how you would suggest someone else go about it.

Let's start with the first question, which is a lot easier to answer. A great monitor in a bad room does absolutely nothing for you, so if you don't start with a terrific room and a plan for how it will integrate with the monitors, you can forget about it. No matter what you do, they will still suck and you will still have problems. So let's just say that, first of all, I started out by designing a great room.

The first test that anyone should do for a system is called the LEDR test. It stands for Listening Environment Diagnostic Recording and was invented by Doug Jones of Northeastern University. Basically, he determined the frequency response of the ear from different angles and heights. Then he simulated the frequency response of a cabasa if it's over your head, to your left, behind you, beside you, in the middle, and also beyond the speakers. In other words, from at least a foot to the left of the left speaker, over to at least a foot to the right speaker, all done with comb filtering that simulates the response of what the ears would hear.

The LEDR test is a substitute for about $30 to $40,000 worth of test equipment. If the sound for the up image doesn't go straight up from your loudspeaker, six feet in the air as you sit there in your position, then you've got a problem with your crossover or with reflections above the loudspeaker. If the sound doesn't travel from left to right evenly and smoothly with the left to right test, then you've got problems with objects between your loudspeakers.

So the first thing you should ever do as an engineer is to familiarize yourself with the LEDR test, which is available on Chesky Test CD, JD-37 and also on the ProSonus Test CD, which is about $50 more. Just test your speakers and room with the LEDR test. And, believe me, if you ever want to know how bad it can sound, just take a pair of cheap bookshelf loudspeakers and play the LEDR test through it and see what happens. It also shows how bad the lateral image is if you take a pair of monitors and put them on their sides with the tweeter and the woofer to the left and right of each other.

My room passes the LEDR test impeccably, so then it comes to the choice of loudspeakers. The speakers I chose are made in Switzerland by a man named Daniel Dehay. They're called Reference 3A's (www.reference3a.com), and they are your classic two-way high-quality audiophile loudspeakers. These do not have a crossover per se; the woofer is directly connected to a pair of terminals in back of the speaker and the tweeter goes through a simple RC crossover. They're wired to my Hafler amplifier. The woofer is an eight-inch speaker and it's ported in the back. The speaker has a really tight clean response down to about 50 Hz.

These speakers play loudly and cleanly without a problem since they have a 93 dB sensitivity. To top it off, I have a pair of Genesis Servo subwoofers and they have their own crossover amplifier. There is no separate high pass or bass management type of device on these speakers. I let the main speakers roll off with their own natural roll off and then I carefully adjust the subs to meet seamlessly with them. I could go on, but I think that covers it. EQ

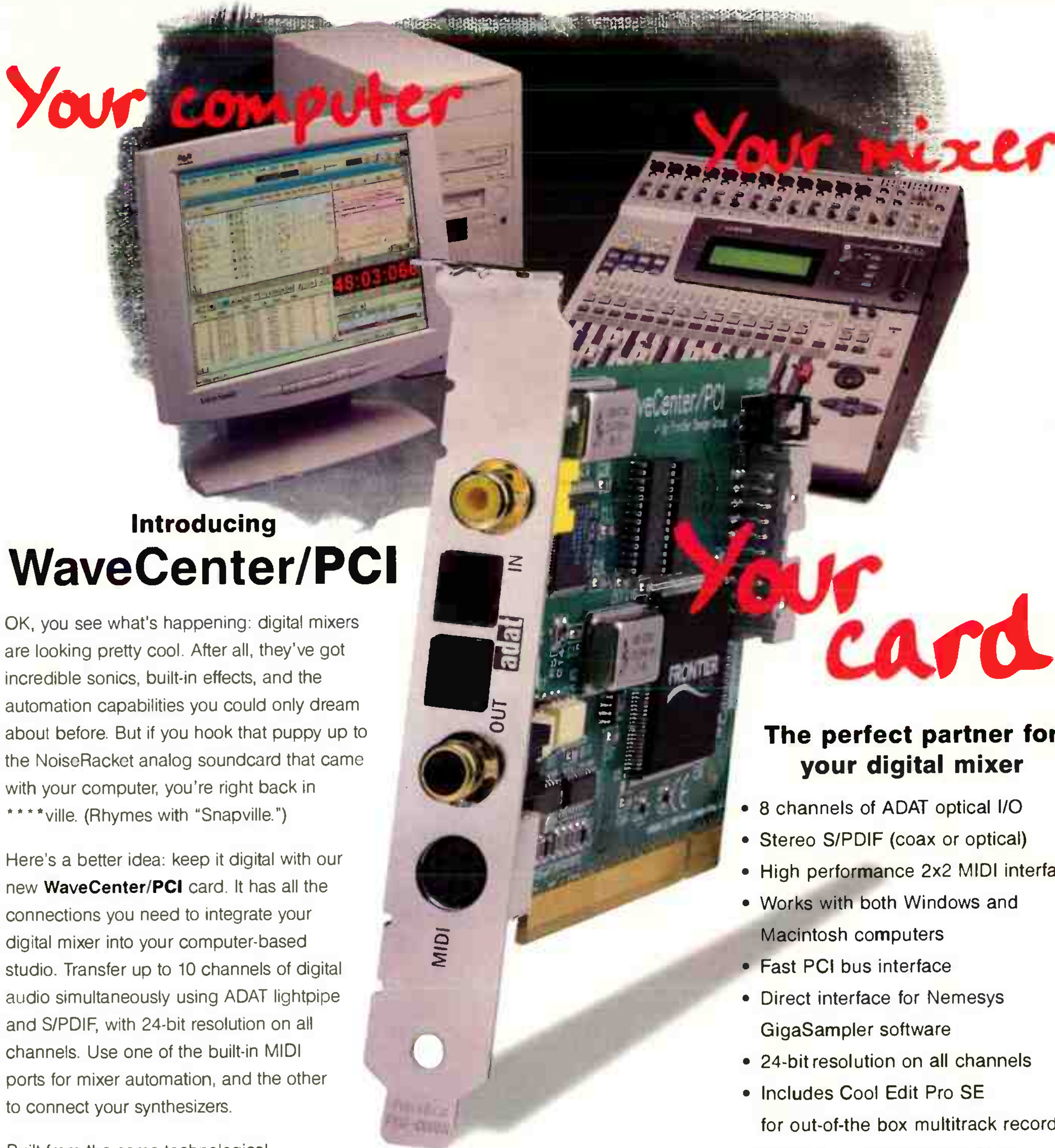

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PHOTO BY JACOB ROSENBERG

up in the suburbs of Los Angeles and first emerged from that city's fertile early-'80s punk scene, they've long since scattered across North America. Graffin lives in Ithaca, New York; drummer Bobby Schayer is in Seattle; bassist Jay Bentley makes Vancouver his home; guitarist Brian Baker dwells in Washington D.C.; and guitarist Greg Hetson has recently forsaken L.A. for Austin, Texas. This geographic isolation has given rise to a unique way of working: Every Bad Religion album since 1988's *Suffer* has been extensively demoed at Graffin's home studio in Ithaca. Other band members may drop in on Graffin to see how songs are developing, but the group generally only gets together to record or tour.

"Rehearsal, what's that?" laughs Brian Baker, Bad Religion's youngest and newest member, who joined the group in 1994. "My stock answer is, 'If you don't know three chords by now, you have no business doing this.' But also the majority of the people in this band have been playing together for at least 15 years. There's a certain rhythm that develops, and it doesn't take us very long to figure out what we're supposed to do."

At first, Graffin shared songwriting duties with Bad Religion co-founder Brett Gurewitz, a practice that continued even after Gurewitz left the band to focus full time on running Epitaph Records, his L.A.-based punk label. But, as Gurewitz lapsed into a dysfunctional phase due to drug addiction, songwriting for Bad Religion became Graffin's sole responsibility. He soldiered on admirably, but opened the writing process up to group input on Bad Religion's last album, 1998's *No Substance*. For that album, Graffin's home demo studio became a project studio where master guitar and vocal tracks were recorded.

But the situation reverted to "normal" for *The New America*. A recovering Gurewitz contributed to one song, "Believe It," but the remaining 12 tracks were composed by Graffin, working alone at home.

"It was more of a private affair, putting the songs together for this album," he says. "I used my home studio more in the way I usually do, as a project environment for writing songs. I spent more time tweaking in my laboratory, so we wouldn't have to assemble the album during the recording process."

Graffin had separate, "upstairs/downstairs" recording spaces in his home for the making of *No Substance*. But now, he says, his studio is "substantially different than when *No Substance* was made. I've moved everything downstairs and renovated considerably. So I now have a live room and a control room. I spent some money on infrastructure."

"WITH A BAND, THE LOUDEST THING IS THE DRUMMER REALLY, ESPECIALLY WHEN YOU HAVE A DRUMMER LIKE BOBBY. SO WE HAD TO FIND A PLACE WHERE WE WOULDN'T GET A LOT OF COMPLAINTS ABOUT THE NOISE."

The studio, Graffin continues, is based around "a Pro Tools rig and some 20-bit ADAT machines all synced together with a Yamaha 02R [console] via an Aardvark Aardsync. I've got a lot of good microphones and expensive outboard gear. I have two Focusrite EQs and two of their two-channel compressors. I also have some Avalon compressors and mic pres."

Outboard effects at the studio include Lexicon's LXP-15 and MPX-1, and a Yamaha SPX990. Graffin monitors on JBL LSR-32's powered by a Hafler Trans-nova 7000. He says he uses his Pro Tools setup very sparingly. "I use it for one purpose only, and that is to assemble songs. I use it as a tool, as it was intended, for looping parts and building songs. I generally don't use the plug-in effects. I have the real Focusrite outboard stuff, so I don't need the Focusrite plug-ins. Talking with people, I think there's a consensus: as long as you have good D-to-A

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and A-to-D conversion systems, it's better just to send the signal out to some outboard unit and return it back to your console. That's not to take away from Digidesign or any of the people making plug-ins. That has a purpose, too, but I think it's got a way to go yet."

NEXT STOP, HAWAII

Greg Graffin has been a Todd Rundgren fan ever since age nine, when he purchased *Another Live* by Rundgren's band Utopia. "I liked the idea that it was pop songs, but it was presented as Todd being more of an underground artist," Graffin says. "I think that had a great influence on the way I always wanted Bad Religion to be as well: a band that had a lot of pop potential but was always appealing on an underground level."

Graffin and Rundgren began communicating around the time of Graffin's 1997 solo album, *American Lesion*. But they didn't get together until it came time to make *The New America*. On reviewing Graffin's demos, Rundgren's input mainly had to do with the lyrics. He suggested what he calls "a more proactive lyrical direction: Instead of the typical Bad Religion, 'what a bunch of crap the world is' kind of thing, there's more of a sense of 'there's work to be done and let's organize to do it.' At this point, most of the band members have children and things like that. They can't go around acting like they're post-pubescent troubled young men."

Graffin did some rewriting and Bad Religion went to record with Rundgren on the island of Kauai, Hawaii. They set up in a rented space that Bad Religion guitarist Brian Baker describes as, "an old, non-climate-controlled barn that I believe was initially built for the drying and curing of marijuana sometime in the '70s. The current owners are professional photographers, a wonderful couple who thought it would really be fun to have a band come and set up in their garage. But it was completely primitive."

"The owners have this really nice house and gourmet kitchen," adds guitarist Greg Hetson. "They're food photographers. They make gourmet food and take pictures of it. So we ate pretty well. But there was very little ventilation in the barn, so we had to keep the door open. A lot of mosquito candles were burned. [Drummer] Bobby [Schayer] got bit up the worst."

"You should have seen my face when I showed up at the barn the first day," Baker laughs. "I'm like, '*How* much are we paying for this?' But Greg Graffin said, 'No, no, no. Just remember, Todd's a genius. Utopia. He's a genius. He knows what he's doing.' "

"Well, there are no studios of sufficient size and soundproofing out here anyway," Rundgren shrugs. "With a band, the loudest thing is the drummer really, especially when you have a drummer like Bobby. So we had to find a place where we wouldn't get a lot of complaints about the noise."

The producer wasn't after room acoustics. All he needed was to get clean signals into his Pro Tools Mix|24 system, which he was running on a Mac G4. "I only have 16 channels of I/O," says Rundgren. "But I haven't needed more than that so far. Two guitars, bass, drums, and scratch vocal is pretty much where you start from. With most of the projects I do, I never need to record more than 16 tracks live at one time. For the most part, I close-mic the drums to try and eliminate the room as a factor. If you keep everything fairly isolated, you can add whatever kind of ambience you want afterwards."

"GREG [GRAFFIN] WAS THERE THE WHOLE TIME, AND WE HAD A LOT OF FUN," CLEARMOUNTAIN SAYS. "HE LET ME DO WHAT I DO, BUT HIS COMMENTS WERE REALLY HELPFUL. HE HAD AN EXTREMELY OBJECTIVE VIEWPOINT. PLUS WE HAD SOME REALLY INTERESTING CONVERSATIONS ABOUT PHYSICS."

The entire project was recorded direct to Pro Tools. "Nothing ever left the computer," says Rundgren, "and I never used any outboard gear." All signal routing and levels were set via a Digidesign Pro Control console. "We called it the $50,000 mouse," Graffin quips. Basic tracks were recorded live in the studio. "Todd really helped us with tempos," says Baker. "We do have a tendency to play everything at 150 mph, even if a song shouldn't be played that fast."

Although initially skeptical, the band's two guitarists wound up going 100 percent Amp Farm for the sessions. "We shipped all our equipment out and a bunch of spare amps," says Greg Hetson. "And Todd goes, 'Check this out: I got the Amp Farm built into my Pro Tools. Give it a try.' And we were like, 'This is gonna sound like sh*t. It's fake. It ain't the real thing.' But we plugged in and said, 'Hey, this sounds pretty damned good!' "

"I had my Les Paul plugged into the back of a G4, and I could hear the air pushing off speakers that weren't really there," Brian Baker marvels. "I said, 'Okay, the man does know what he's doing. And he's got good hair. So I'll sit here and play.' I just wish we hadn't shipped all the amps out."

While some of the basic rhythm guitar tracks were kept, a substantial number were replaced, according to Hetson. "We were in a barn, we had the mosquito

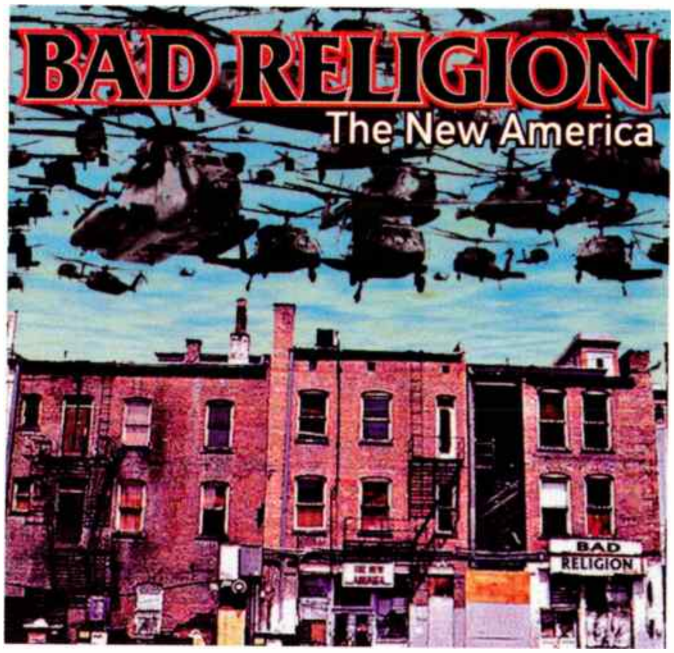

Bob Clearmountain (left) and Greg Graffin at Clearmountain's Mix This! studio

candles going, the headphone mix was terrible, and it was loud. So we pretty much went and redid the guitars and bass. Fix it up. We moved the computer over to Todd's house and did a lot of guitar overdubs in his living room. But most of the rhythms were done in the barn."

Although the Bad Religion rhythm guitar sound is massive, it's generally just two tracks: Hetson in one channel and Baker in another. Lead guitar embellishments and solos are added via a democratic overdubbing process. Baker explains: "Greg and I just go in separately and play whatever we feel like. We both tend to have set ideas of what we want to do on any given song. And then the poor guy who's mixing has to figure out what to use — take the cream off. On this record, as on many past ones, Greg and I didn't really realize what was going to be kept until we were at the mix. It's very egalitarian."

"We were essentially plugging the guitars right in flat," says Rundgren. "The guitarists were hearing an Amp Farm algorithm in their headphones, but if you were to listen to the signal flat, it was just completely dry, direct guitars. That allowed us to adjust the sound of the guitars at any point we wanted. We'd only print the sound at the very last moment. This way, the guitarists could just concentrate on doing overdubs and solos without having to constantly move mics around and plug into different things. Everyone knew we could spend time making adjustments later on and not have it impinge on the performance."

Graffin's vocals were also recorded straight into Rundgren's Pro Tools rig, via an old Neumann U 87, the singer's usual mic preference. "When I'm recording a vocal," says Graffin, "I sometimes like to add the Focusrite compressor, set to a soft, 1.5:1 compression ratio. But, on this record, Todd used a plug-in for a lot of that. It was in my headphones, but he didn't print it. It was just a raw vocal signal. When we got to Bob Clearmountain's house for the mix, we just processed it again."



Like all Bad Religion albums, *The New America* is well fortified with Graffin's solid baritone choral harmony style. "There's a lot less to it than you might think," the singer says. "If it's a three-part harmony, the high and middle parts are generally doubled — one in each speaker. And the low one is usually mono. So the most you'll ever hear on a harmony is five voices. People think there's some kind of trick to it, but the only trick is technical perfection in your vocal execution. What makes a chorus harmony sound enormous is that the harmony note is precise in relation to the melody note. That's all it's ever been on Bad Religion records. I have really high standards of vocal execution. And Todd does, too. Where a lot of heavy metal bands used to fail was layering vocal upon vocal and coming up with a sort of pseudo-grand chorus background."

For all instruments and voices, Pro Tools was used primarily as a recording medium. There wasn't a lot of editing or track manipulation involved. "There were only occasional things — places where it became invaluable," says Rundgren. "For example, Brett Gurewitz's solo on 'Believe It.' Because of the meaning it would have for fans, the band wanted him to make a cameo appearance on the record — without having to fly to Hawaii. So I sent him a mix of the song on CD. He went into some project studio, played along to the CD audio, and then sent me back a CD of the solo, which I then — using Pro Tools — laid into the song. It saved everyone a lot of misery, I think."

Pro Tools was also used to lay in a few small elements from Graffin's original home demos: a keyboard and guitar loop in the intro to "A World Without Melody" and some sound effects in "I Love My Computer." But situations like these were the exception rather than the rule, as Bob Clearmountain remarked when the tracks arrived at his studio for mixdown.

"I've done a lot of projects where people bring in Pro Tools sessions and when you look at it there's an edit on just about every beat. With this, there was practically none of that. Looking at the session, they basically just used Pro Tools as a tape recorder."

CHEZ CLEARMOUNTAIN

Mix This!, Bob Clearmountain's studio, is located in his home in scenic Pacific Palisades, CA. The engineer's first act on receiving the Bad Religion masters was to transfer them from Pro Tools onto digital tape. "I have a Sony 3348HR with six Apogee AD-8000SE converters," he details. "I have a Pro Tools rig here, but I had to rent in another one because I didn't have all the plug-ins that Todd did. So we rented one, hooked it up, and bounced everything over. It took a couple of hours."

This enabled Clearmountain to mix the record on his modified 72-input SSL 4000 G+ console and deep collection of outboard effects. "Bob mixes analog," says Graffin. "Even though the Sony is a digital machine, he pipes it all through his SSL console. I think that gave the record a much more familiar feel — made it really crackle."

Band members were present for various parts of the mixing process. "We were a little bit worried that we would lose something when we tried to transfer all this digital information from Pro Tools onto a medium Bob could work with," says Baker. "So I wanted to be there. I did wind up fixing a couple of little things I could have done better. I don't think we actually lost any tracks, but I was basically there as a clean-up man, to put a few finishing touches on. Bob's a huge Todd fan, which is nice. He'd come across something and rather than saying, 'Well this is digital distortion because the mic was recorded too hot,' he'd say, 'Hmm, I wonder what Todd was going for here.' "

By all accounts, Clearmountain and the band got on very well. "They're huge Dead Boys fans," Clearmountain relates. "And one of the songs on the album, '1000 Memories,' starts off with sort of a tribute to the Dead Boys' 'Sonic Reducer.' They didn't know that I

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PHOTO BY JACOB ROSENBERG

played bass on that first Dead Boys album. Most people don't. When I told them they were like, 'Wow, really? We got the right guy!' "

Graffin ended up working most closely with Clearmountain. "Bob preferred to have the songwriter present," Graffin explains. "He says that's an important element. And he really is great to work with, because he looks for input and ideas."

"Greg was there the whole time, and we had a lot of fun," Clearmountain adds. "He let me do what I do, but his comments were really helpful. He had an extremely objective viewpoint. Plus we had some really interesting conversations about physics."

The transfer from Pro Tools to the Sony 3348 included both the dry, unprocessed guitar tracks and the final "Amp Farmed" tracks that Rundgren had printed. This situation afforded Clearmountain considerable flexibility in the mix. "I could actually go in and change the sound of a guitar track if it wasn't working," he says. "Which I didn't do very often, because most of the sound selections Todd had made were right on the money. But there were a few places where I was able to substitute a different amp sound, or augment what he had with another amp in Amp Farm. To me, that was a real advantage."

"One curious thing I noticed," Clearmountain continues, "is that all the guitar tracks needed the same sort of EQ. Usually when you do a rock album, the different types of amps used by the band have such divergent sonic characteristics that you have to EQ and process them differently. But even though Todd and the band used different settings in Amp Farm, the relative brightness was very similar. Looking back at my EQ settings, there was a similar shelf at around 5 kHz, just to brighten things up generally before I started doing any other EQ. I'd do that and then maybe I'd plug in a Pultec for a specific type of character."

Unlike Rundgren, Clearmountain used plenty of outboard processing gear. A Motion Sound rotating speaker simulator was used in a few spots, including the breakdown section in the song "The New America." "It's actually a rotating horn in a rackmounted box," says Clearmountain of the device. "There's a couple of controls on the front. And it's got four mics in it. It has a little tube circuit and a nice little overdrive sound."

Clearmountain tends to prefer delay over reverb as a means of pumping up rock guitar tracks. "I use delays that are in time with the tempo of the song," he says. "So you don't really hear it as delay. I'm sure I used some reverb for the guitars, because I have two live chambers here. I might have used that a little bit, but not a lot. I had an old AMS DMX80 that I use for guitars, and an old Echoplex that I use once in a while. The coolest delay gadget is this thing that nobody knows about, called the Yamaha D5000, which is just the best digital delay on the market."

"WE WANTED IT TO BE A BIG ALBUM AND SOUND LIKE THE GUITARS ARE LARGER-THAN-LIFE," SAYS GRAFFIN. "BUT WHAT I LIKE ABOUT IT IS IT DOESN'T HAVE THE CONVENTIONAL TRAPPINGS OF THOSE VAIN HEAVY METAL ALBUMS THAT PRETEND TO HAVE THOSE GRANDIOSE GUITARS."

"Old stuff," is how Clearmountain characterizes his preferences in outboard gear: "I don't mean super old stuff. But things like the Roland SDE 3000, which I use for vocals. They're so easy to work with and there's something about the sound of them I really like. They've got a warmth — almost an analog kind of sound to them. As far as EQ, I have three Pultec EQP183's that I really like. It's a cool thing for guitars. And the Focusrite Red 3 is also good for guitars. The [Empirical Labs] Distressor is good, too."

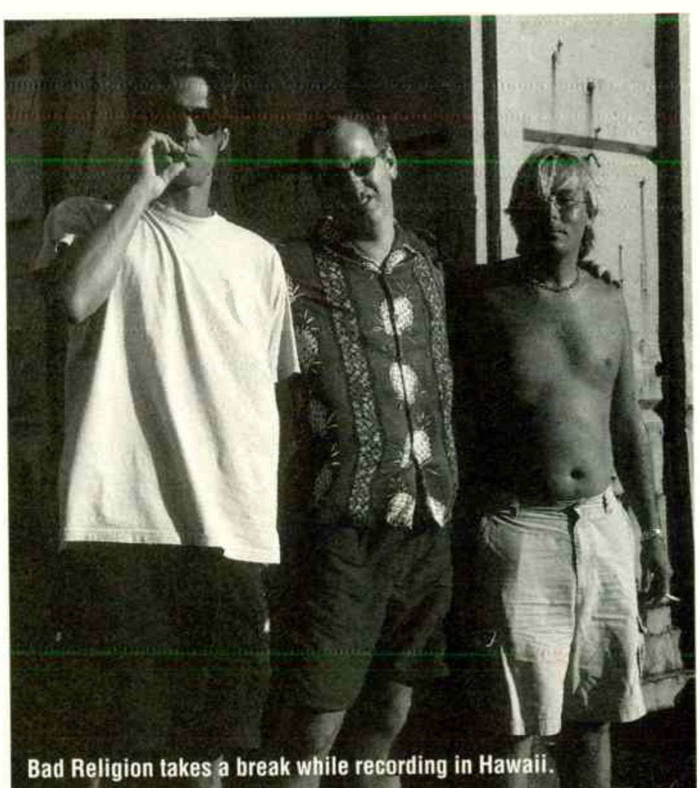
As for *The New America*'s profoundly punchy rhythm sound, Clearmountain gives maximum kudos to Rundgren's recording and drummer Bobby Schayer's playing. But Clearmountain does admit to triggering a bass drum sample from the original bass drum track for a lot of the album. "We used Sound Replacer, which is a Digidesign plug-in," he says. "But that's the only thing I replaced. And I don't think I actually replaced it. I just sort of augmented what was there, to give it a bit more punch."

In the main control room at Mix This!, Clearmountain monitors on Yamaha NS-10M, KRK E7, and Audix M5 speakers. "I switch back and forth between monitors. The Audix's are better to listen to, but the E7's are a bit better to mix on. Mixes I do on them tend to sound better when I take them out into the real world. Which is why the NS-10M's are still my main thing. You take the mix outside and it generally sounds good. There are no big surprises. But really my favorite monitors right now are these little self-powered Apple computer speakers that I have over at the side of the control room. Also, because the studio's in my house, I have the whole place wired up. There's a little lounge with some speakers and another little room upstairs with some consumer Tannoys that I'll also use to check out a mix."

Not that much chin-scratching was required to mix *The New America*. "The whole thing took less than two weeks," says Clearmountain. "We were mixing one to two songs a day. It's real simple. You can hear what's on there. Those guys knew what they were doing. They didn't waste a ton of tracks trying to make something out of nothing, as a lot of people do."

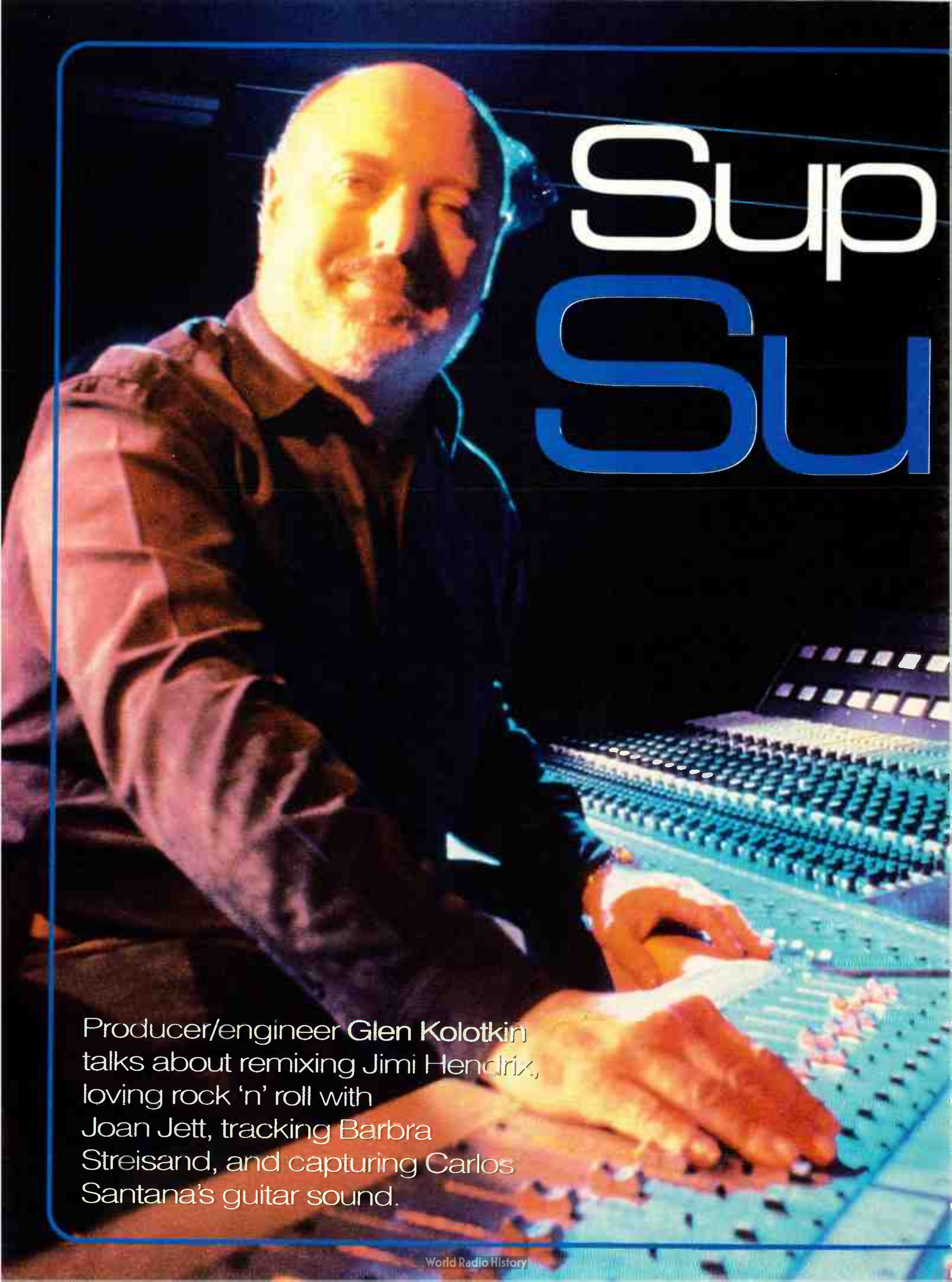
With *The New America*, Bad Religion feel they have attained their goal of making a punk-rock album for the masses. "We wanted it to be a big album and sound like the guitars are larger-than-life," says Greg Graffin. "But what I like about it is it doesn't have the conventional trappings of those vain heavy metal albums that pretend to have those grandiose guitars. This is much more crisp. It has much more clarity."

"What I like," adds Brian Baker, "is that every single thing is too loud. But nothing steps on anything else." EQ



Bad Religion takes a break while recording in Hawaii.

PHOTO BY JACOB ROSENBERG



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Producer/engineer **Glen Kolotkin** talks about remixing Jimi Hendrix, loving rock 'n' roll with Joan Jett, tracking Barbra Streisand, and capturing Carlos Santana's guitar sound.

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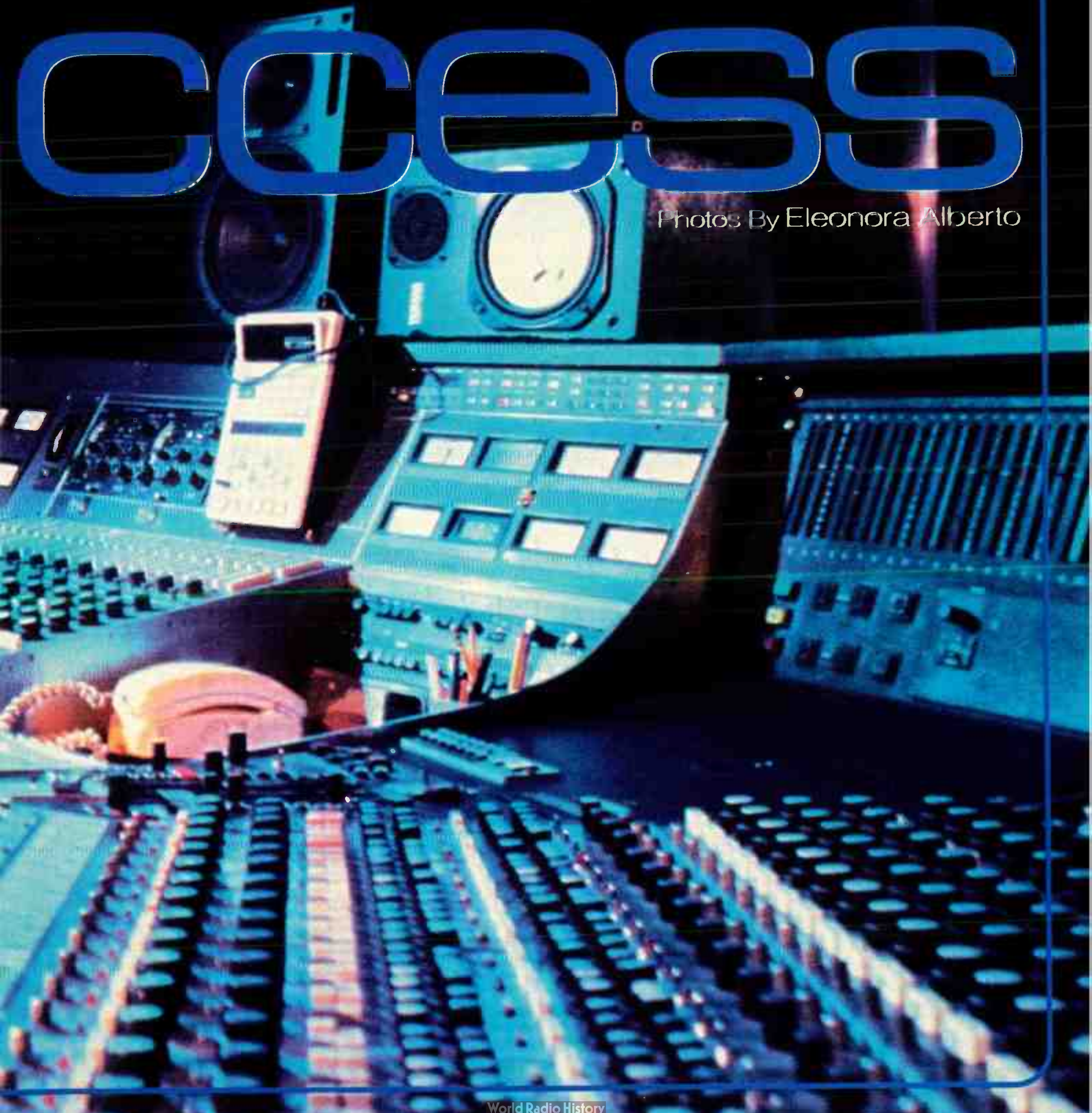

How would you like to have these projects listed on your resume? *Love, Peace, and Happiness* from the Chambers Brothers; the Byrds, *Byrdmaniax; Subterranean Jungle, Ramones Mania,* and *Hey! Ho! Let's Go: The Anthology* from The Ramones; and *Pearl* from Janis Joplin. If that's not enough, we could add in some remix work on *Electric Ladyland* by Jimi Hendrix, *Black Brown and Beige* from Louie Bellson, *I Love Rock 'n' Roll, Flashback,* and *Fit To Be Tied* from Joan Jett, *Stoney End* from Barbra Streisand, and *Ridin' High* from Robert Palmer. While most producer/engineers would be happy with a credit list like that, Glen Kolotkin's work also includes projects with Greg Kihn, Paul Winter, Lionel Hampton, and a series of records with Carlos Santana, including *Caravanserai, Santana III, Borboletta,* and *Moonflower.* Most recently, Glen came full-circle with work on Santana's Grammy-award-winning *Supernatural. EQ* had the pleasure of speaking with Glen regarding Santana then and now, as well as some of the highlight projects of his flourishing career.

EQ: You've worked with some very diverse artists over the years. What were some of the most memorable thus far?

Glen Kolotkin: I guess it started in the late 1960s when I was a new engineer at Columbia Records. The staff producers there were working with the well-known artists over and over again, so I never got a shot to do anything notable. One day I was told to set up the room for a remix, but they didn't tell me who was coming down. Who walks in with a pile of tape in his arms? Jimi Hendrix! He says, "Hi, I'm Jimi." I said "Oh, I know who *you* are." Jimi told me, "I've been working on this album for a year, it's been remixed, and I'm still not happy with anything that's been done. This is my last stop for this album. They're screaming at the record label because they need to put it out." And we proceeded to remix *Electric Ladyland.*

That's something most people don't know about because my name isn't on the album. The art for the record had already been printed up a year before we remixed it. At the time I wasn't even a fan of his, but I became a fan after hearing tunes like "Voodoo Chile" and "Crosstown Traffic." Jimi asked me to leave Columbia and join him to do records, and also live sound. At the time I was with Columbia for only about six months or so, and I liked having a steady job, so I declined. I always wondered what would have happened if I had joined him. I wonder if he still might be alive today. I was never into drugs, and we were becoming very good friends during the period I worked with him. It was quite an experience.

What do you remember technically speaking about those sessions?

Obviously, Jimi was really into special effects, and so was I. I would try to come up with a special effect out of my "bag of tricks" and I'd say, "What do you think of this?" He loved them. If you get the original version of the record, you'll hear sounds that appear to come from off the walls or behind you. There's a phasing thing I came up with at the time using three tape machines. Of course, now there are effect boxes that'll do this, but we had to make it back then. I'd take the track I wanted to have the effect on, split it, send it to two different tape recorders, and vary their speeds. They could be stereo decks or whatever was available. I'd bring the tape decks back to two separate channels in the desk and put one out of phase with the other. Pan them extreme left and right, and then vary the level of the faders until you hear the sound start bouncing off the walls. [*Laughs.*] You can get it just right so that the sound appears to come from left-of-left or right-of-right. It drove people crazy back then. I'd call people into the room and they'd swear I had a speaker behind me. No, it was just an effect I came up with by fooling around. At Columbia back then we had plenty of tape recorders available, but there wasn't much for special effects. It really was fun working there and being able to experiment.

Were there other odd things you did with extra tape recorders?

Sure. Things like backwards echo.

Exactly how does one produce a backwards echo?

Let me think about it...I haven't done

it in quite a while! We'd take the multi-track tape off the machine, turn it upside down, and put it back on the machine so it was running backwards. Then we'd record the echo on an open track while the tape was running backwards. When you flipped the tape back over to play it forwards, the echo would play backwards, and it would happen *before* the actual sound occurred. The whole psychedelic thing was coming in and everyone wanted special effects. Clive Davis had taken over as president of the company; he wanted younger engineers and he wanted rock 'n' roll. The older engineers weren't interested in doing rock 'n' roll. I was like the new kid in town and I was all into rock 'n' roll. I had a clear field.

What were some of the technical challenges you faced during those times that you're not worried about now?

One time I got called to do a two-track session at a big studio in Manhattan called Nola Penthouse. Tom Nola (who owned the Penthouse) couldn't make it, so he asked me to do the session for him. It was about 25 pieces, live-to-two-track. I'll never forget it because it was my first big session and I was just thrown into it. He had given me the setup for the studio, the way he'd do it. So I did it the way Tom would, and it was amazing. I started opening the faders and all of a sudden this *record* came out of the speakers. It was a total record because we did the whole thing live: lead singers, background singers, strings, horns — the whole bit.

It must have been a bit scary....

I think I was more scared afterward because there was no remixing — after the session we cut the tape off the reel and mastered it! I don't do much live-to-two-track anymore, but I recently worked on an album with Wallace Roney who was Miles Davis's prodigy. He brought in all these CDs that he liked and wanted to get those kinds of sounds. Basically those records had been done live on four-track, so we recorded the whole band live (about eight pieces). It came out great. We compared it to some of those early recordings and we blew them away, but, you know, we really haven't come that long a way in the recording industry. I think the biggest thing that's happened is the invention of the CD. We're still using microphones from the 1940s because the designs haven't changed that much.

How do you approach miking Wallace's trumpet?

I would use different mics depending upon whether it's a live recording or an overdub situation. When it's a live recording, the mic's directionality is most important — for keeping the leakage down. I'd set the band up in a circle or a semi-circle, so that the backs of the mics are facing the other band members — this helps keep leakage to a minimum, but also allows everybody to see each other. An [Electro-Voice] RE20 would be a good choice for a session like that where I need the isolation. What we're really trying to do is get a great *performance*. A lot of people forget that this is what it's all about. The technical thing takes over and they forget that these musicians have to play together. Musicians need to get a feel, and, with everyone in a box, it's hard to accomplish that.

> If you record with Dolby SR, there's really no tape hiss. It's as quiet as digital, plus you have that nice warm sound. You can get away with running half-inch at 15 ips with SR, which is nice, or at 30 ips without SR.

Especially with the drums. How do you maintain some isolation of the drums without making the drummer feel musically isolated?

Use directional microphones, and put low baffles around the kit so the drummer can still see over them and feel [like] part of the band. Some studios have a room for the drummer with a lot of glass. That works sometimes, but, if it's a jazz session, the players like to be in the same room with each other.

How do you mic a jazz kit as opposed to a rock or pop session?

For a jazz kit, you're trying to get the true sound. All jazz drummers have "their sound." You try to capture that. For a rock kit, there's going to be a lot of processing to get a certain sound, especially on the bass drum and toms. You might use noise gates and parametric equalizers, which can be extreme.

Do you tend to use less mics on a jazz kit?

In the early days we would, but then the drummer might say, "Gee, I really don't hear enough of the hihat." So we'd put up more microphones to get the control back. There usually weren't a lot of good mics in the early studios, and the consoles didn't have a lot of positions [channels]. We had a system called the "Detroit Sound" with three mics for the drums: it was basically one overhead, one on the snare, and one on the bass drum.

Do you still use that technique?

It's gone by the wayside. We tried it for Wallace because he wanted that original sound, and it worked pretty well. But then we tried putting up mics for the toms. The sound did get better, so we ended up using those microphones as well.

You mentioned taking a different approach in an overdub situation for recording the trumpet....

Yes. I'd use a higher-quality microphone and place it farther away to get a better sound. If you move the mic too far away in the live situation, you start picking up the other instruments. You don't want to do that too much.

How close can you get with a trumpet mic in the live situation before you lose the overall tone of the instrument?

It depends on the microphone. If you use an [Electro-Voice] RE20, you can get really close. It has a built-in pop filter and it can take loud sounds. You can use a pop filter on another type of mic, like an [Neumann] '87. Most trumpet players tend to move their instrument right up to the mic. The problem is keeping them far enough away from it. Sometimes I put up two microphones: one for them to aim at and one that's really doing the recording! [*Laughs.*]

So there's a bit of psychological switching going on....

Yes. Just to get that full sound. We try to protect people from themselves. [*Laughs.*]

Do you generally use compression when recording trumpet or horns?

No, I don't like it because you can hear it working, especially when they stop playing. All of a sudden the mic starts sucking in sounds from other instruments nearby.

In mastering, I would use compression. In the early days I'd do some disc mastering after I finished a project. But mastering for vinyl has a different set of problems. Physically getting the sound on the disc was a problem — just to get the record to track correctly. With things like radius EQ; as the stylus moves toward the center of the record, you have to EQ differently to make it sound good. The better-sounding bands were always at the edge of the record. With CDs it doesn't matter. You can just go for great sound, which I prefer — to get great sounds and not have to worry about physically fitting it onto the disc.

So, as you move through the songs on a side of vinyl, you need to adjust your EQ?

Yes. They call it radius EQ. Most people don't even know about that.

I was aware of the fact that distortion tends to increase as the stylus and tonearm near the end of the record, but I didn't know about the EQ.

We had a measuring device that would tell you how far the cutting head was from the edge. At so many inches you'd adjust the EQ by a click and then, after another couple of inches, you'd adjust it another click, as you got toward the center of the record. Most people wouldn't notice it because you're compensating, but the records always suffered. For instance, you'd have to roll off bass just to get the mix to fit on the record. That's not necessarily for the benefit of good sound, it's just physics. Now you can get it to sound great.

That must have been a real drag. You go through all this effort to make the record sound good and then physically you can't fit the bottom-end on the disc.

Right. It was a problem, especially on longer records. The shorter the record the better, because you had more room to spread out the grooves. We had a thing called variable pitch where the grooves would be spread out over so much space. The lathe would range from about 70 lines per inch to about 320 lines per inch.

Is that why a lot of dance records were cut as 12-inch 45s?

Yes. You have more room so you can cut more level on them.

Did that affect your recording process? Were you conscious of it during tracking or overdubbing?

You had to be conscious of it when you were mixing and placing instruments. You'd put bass in the middle. Now you can pan it to one side and still get as much level as you want. But, back then, there would be a problem in cutting if you placed things radically like that.

Is this where our tendency to place kick drum and bass in the center came from?

I think so. I know I did it that way because it made for a better-sounding record and a louder record.

Having said you're optimistic about CDs, what's your feeling about analog versus digital recording media?

Well, of course, when digital recording came out everyone wanted to use it, including myself. And the complaint seemed to be "it doesn't sound as warm as analog." I went back to analog for the recording process. The only thing "wrong" with analog was that it added tape hiss. Nowadays, tape is better, so hiss is less of an issue. If you record with Dolby SR, there's really no tape hiss. It's as quiet as digital, plus you have that nice warm sound. You can get away with running half-inch at 15 ips with SR, which is nice, or at 30 ips without SR.

Do you track with SR?

When available. It's a shame studios charge extra for SR. You should be able to make the best possible record without being "held up": "SR is going to cost [so much] more per day." It depends on the budget. For some groups it matters — especially with jazz acts where the budgets are small to begin with. So it's a trade off. You record at 30 ips and you're saving money on SR, but you're using twice as much tape!

No matter how you look at it, you're paying for it.

Yes, but in some cases I have been recording at 15 ips without SR. With the new tape you still don't hear much tape hiss and you can print more level.

You won a Grammy recently for *Supernatural* from Carlos Santana. How did you become involved with the new record?
That's an interesting story. I had previously done about nine Santana records with the original group. Eventually, the group stayed in San Francisco and I moved back to New York. I recall being home with the flu when my wife Carol told me to turn the television on because Santana was on. I turned on the TV and there was the original band (with Greg Rollie and Mike Shrieve) being inducted into the Rock and Roll Hall of Fame. I told Carol some of the stories about the records I did with these guys, and she suggested I write a letter to Carlos congratulating him — which I did. Then I got a call from Carlos. He had just been signed with Arista and asked me to come out and work on the next album. Before I knew it, I was on the next plane to San Francisco.
How did *Supernatural* differ from the previous records?
On the older projects, the band would go into the studio and basically jam. We'd record the whole jam session on a roll of tape and then play it back. Out of, say, 45 minutes or whatever, maybe there'd be 15 or 20 seconds that were really magical, and the band would work that into a song. A lot of that creative work was done in the studio. Sometimes an idea would be brought in and we'd work off that, modifying it for the band's needs. Now the songs are brought in by songwriters. It's Carlos's band, but it's a different situation. He's the band leader and he has a lot of input.
As opposed to the collective of the older group.
Yes. Everybody had say — like a family. It was fun that way.
What was the setup for recording Carlos's guitar on *Supernatural*?
Carlos had several guitar amplifiers, which we put into an isolation booth. Then he set the sound to what he loved. In the beginning, he was in the iso booth working on the sound, and then he came into the control room. By mixing several microphones from the different amplifiers, I tried to get the same sound in the control room as it was in the isolation

booth because that's what he liked. Then we tailored it a bit to fit into the recording. The amps were all in one small room, maybe 10 x 15 feet. I think we had a couple of Mesa Boogie amps (Carlos prefers those) and maybe a Fender Super Reverb...possibly a Peavey as well. By putting them in the booth I didn't have to place the mics right on top of them. I used [Electro-Voice] RE20's maybe eight inches to a foot away from the cabinets, toward the edge of the speaker, not the center. We left them open — no baffles — because we were also trying to get the overall sound coming from the booth. We even had a couple of room mics in the booth itself. Those were [Neumann] U 87's, as far away as you could get in a small room.

In a situation like that, do you use the U 87's in a directional pattern, or set to omni?

Cardioid, with the pad on, and no rolloff.

Were the amps recorded on different tracks?

I mixed them to a stereo pair and tried to get the sound naturally without using a lot of EQ. If it sounded good from the amp, with the quality of mics I had, it wasn't that hard to translate. It was "his sound." I tried to duplicate that and maybe improve it a little, but you can't always improve it because you can put mics where the ear can't go.

When you're listening to an instrument coming through an amplifier, do you put your ear where a mic might go?

Yes, definitely, unless it's something that's just too loud. Like Billy Cobham's drums. I wouldn't get too close to those or I'd be deaf! [*Laughs.*] But especially with some of the Indian instruments...who knows where the sound comes from?

When you record an acoustic guitar, you have a pretty good idea of where the sound comes from — you know it's not coming off the headstock. More like somewhere in the vicinity of the sound hole. But what about a sitar?

That's the thing — there's hardly any sound coming off a sitar. I usually have a couple of very sensitive microphones — maybe [Neumann] KM 84's — one where the player is actually plucking, and one off the body to give the instrument a full sound. Hardly any sound comes off a sitar. Outside noise is a problem, like trucks passing the studio. Record at night when it's quiet and use a booth!

What's the most challenging session you've done?

I think the most challenging sessions were with Paul Winter. Whenever possible, Paul likes to record at the Cathedral of St. John the Divine [New York City]. He's a musician-in-residence there. The reverb time is something like 14 seconds! [*Laughs.*] Trying to get definition on Paul's soprano sax in that room really was difficult. Of course, those recordings were done at night after all of the services were finished. And working in the winter time, it was extremely cold. Everyone wore gloves whenever they could. It was freezing! Doing big orchestras there is a huge challenge.

What was the instrumentation you recorded there?

It was mostly the Consort [The Paul Winter Consort], an ensemble with drums, percussion, cello, and electric bass. We used the tremendous organ in the Cathedral, which is tough to record. If you get too close, it doesn't sound right. You have to be at least 20 feet back, so we had mics on very tall booms up in the air, aimed at the pipes. It's a loud instrument, and the reverb time in that room is really long. Again, I tried using fairly directional mics. In some cases, the less-expensive mics work better. Mics like the Shure SM57 are wonderful for live use because they are so directional, and the sound quality is pretty good. Even if you sacrifice a bit of sound quality, it's worth it as opposed to having so much leakage wash out the definition of your sound. It's a bit of a trade off. Fortunately, we have some very directional microphones. That's really about all you can do, especially in a church. For Paul's soprano sax, I probably used a [Sennheiser] '421, maybe with a U 87.

Do you use automation when mixing?

I've been using automation probably since it first came out, but I'd rather have my hands on the faders. I remember mixing a record for about three or four hours with automation, and, in the end, I wasn't quite happy. I mentioned to the associate producer that I'd like to try it again with a totally different approach, so I turned off the automation, pulled all the patches, and did a mix in about 15 minutes. It was completely different than the mixes we had done in the previous three hours. Then I called the other two producers in and asked what they thought. They liked it, but they also like the other ones as well. So we put my 15-minute mix in the middle of the reel with the other four or five automated mixes that took several hours. We called the band in and let them listen to all of the mixes so they could pick out the one they wanted to use. When we got to the 15-minute mix, everyone jumped up and down going, "That's the one!"

Now, this record did real well. It wasn't a perfect mix, but it had the excitement we lost in the automated mix, which technically was probably perfect. That record was "I Love Rock 'n' Roll" [Joan Jett], which sold four times platinum on the album. That was a 15-minute mix. It kind of tells you that you're going on the emotion. What's a perfect mix? To hear everything at every time? Not really. To get the excitement. To get the chill on the back of your neck. Ever since then I try to record things in such a way that if you open up all the faders, the mix is basically there. It's just a matter of polishing it a little, which you can do manually without a problem. If it's not recorded properly, you have to depend on the automation, and it's always a disaster. You end up doing it for hours and hours. There are so many tools available that a lot of people overuse them or use them just because they are there. It's like a carpenter's toolbox. Just because you have all these tools doesn't mean you need them to build a house.

Do you compress the entire mix during the mixdown process?

Sometimes, especially if I will not be involved in the mastering. I remember taking things to the mastering room and the guy would say, "I don't have to do much on it. It's there." When we were doing records, we'd do the radius EQ, a little touch up, no problem. Not like re-making the record in the mastering. It should sound good on the first playback. The rough mixes at the end of the day are probably the best mixes. You've been working on it all day. Put that one on tape and save it for reference. A lot of times they become final mixes because they sound the best. I remember remixing an eight-track date for 12 hours and we thought we had great mix. The next day we listened, and it sounded awful. We re-did it in an

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Hardware Mastering Processor

Roundup

Mastering processors are those much sought-after units that put the "finishing touches" on a mix by optimizing its dynamics, equalization, normalization, and other parameters — optimization that is usually done in the post-mix, "mastering" part of recording production.

The first unit offering multi-band dynamics, EQ, and other helpful mastering tools in a single piece of digital gear, was the M5000 multieffects unit by TC Electronic, originally introduced in the early 1990s. There was a software option for the M5000, called the MD2, which enabled it to perform multi-band dynamics processing, equalization, limiting, and several other unique functions, in a user-selectable signal chain.

This ability proved popular with many mixing and mastering engineers, so TC decided to release a standalone hardware unit, the now famous Finalizer, which was introduced in 1996. Since then, several other companies have offered their own variations on the "mastering processor" concept in both hardware and software versions.

What's a Mastering Processor?

Mastering processors were designed to add most of the functionality of the signal processing that's typically found in a mastering house to a studio setup — all wrapped up in one multi-functional box. This allows the user to polish up and tweak the sound of his finished mixes, preparing them for release on CD (or other format).

Mastering processors are, by and large, digital signal processors, and offer processing that was difficult to achieve outside of a high-end mastering house, such as multi-band dynamics manipulation. Multi-band dynamics processing takes the signal and splits it up into two, three, or more frequency bands that can each have their own amount of compression, limiting, and expansion/noise gating. When full-band compression is used on a mix, it can sometimes have negative effects on the signal. For example, the bass frequencies could affect the amount of compression on the vocals, the vocal levels could affect the cymbals, and so on. Using multi-band compression to process the full mix provides a type of flexible equalization that can respond to the frequency spectrum of a mix in a dynamic, responsive way.

Prior to the development of mastering processors, the only way to achieve frequency-conscious dynamic gain control of a signal was to use the "sidechain" capability of a compressor, although that was generally used for "de-essing" sibilant vocal tracks, not for "EQ'ing" an entire mix.

Of course, with a lot of effort and a bunch of equipment, one could set up three (or more) compressors and a crossover to get multiple bands of compression; one for the high frequencies, one for the mid frequencies, one for the lower frequencies, and so on. But this is a time-consuming process, often providing less than ideal results.

With the advent of mastering processors, all that has changed. Three or more bands of compression, limiting, and expansion/noise gating are now common, and using them is as simple as finding the correct page in the menu of the unit being used.

Equalization is another mastering tool that's generally contained within a mastering processor. Anywhere from two to six bands of EQ are simultaneously available for use in addition to the dynamic control described above. The parameters provided are usually comprehensive, allowing accurate control over peak and shelving curves as well as high- and low-pass filters.

Real-time normalization is often offered, allowing maximum signal levels to be reached without clipping. Prior to the introduction of these units, this process necessitated a DAW, or computer software dedicated to performing this function.

And, depending on the manufacturer, a number of proprietary effects may be included, such as "stereo enhancement," "Digital Radiance," "tube emulation," "single-ended noise reduction," and so on. Also, some of the units offer a patch point that allows you to add an outboard signal processor to the chain.

The analog-to-digital and digital-to-analog converters, AES/EBU or S/PDIF digital ports, analog, and various other inputs and outputs found on most mastering processors are useful for a variety of applications. They're often superior in quality to those found in DAT machines and DAWs, for example. Or they can sometimes be used to translate one digital format or sample rate to another.

All of these processes are available simultaneously, in one unit, giving a full complement of mastering options previously only obtainable using many dedicated pieces of outboard gear, at a much greater cost. Using these processors, an engineer can now "master" his mixes in a studio, without need for a mastering house, with one piece of equipment!

Well, almost. Although this type of processing is certainly a major part of mastering, there are also several others that must be performed to truly *master* a project. These include editing the tracks, "cleaning up" the beginning and ending of the songs, fading tracks in and out, making the average levels consistent, and setting the timing and sequencing of the tracks, including the length of the spaces or crossfades between the tunes. Finally, there is the adding of "PQ" codes, a vital part of mastering for CD production.

So, for as many things as a mastering processor *can* do, you may still need two-track digital editing software (and maybe a CD-R burner with a CD burning program) for your computer to perform these other functions. You can also have them done at a CD duplication house prior to the mass production of your CD.

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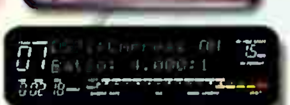

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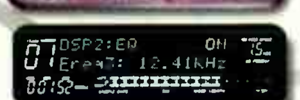

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Hardware Mastering Processor

Roundup

In this roundup, we'll be looking at the dedicated hardware models that are currently available. There are other units that offer all the features of mastering processors, while including additional digital effects such as reverbs, delays, flangers, and other time-based patches, but we've limited this collection to the products that are dedicated to mastering processing.

With that said, let's take a look at the units that will help you put that "mastering polish" on your mixes.

BEHRINGER ULTRA-DYNE PRO DSP9024

The Behringer Ultra-Dyne is a six-band, stereo mastering processor with dual high-performance DSPs. It employs 24-bit A/D and D/A converters from Crystal, and offers an optional 24-bit AES/EBU digital interface. Up to 600 msec of delay allows six bands of "look-ahead" compression, limiting, and noise gating, with each band being separately adjustable. The limiter and gating sections are independently adjustable for all six bands as well.

The Ultra-Dyne also features a multi-band exciter, with an adjustable ratio of odd and even harmonics. The signal can be further enhanced with tube simulation, which includes a selection of popular tube types from which to choose.

A multi-band Denoiser, Berhinger's trademarked name for their single-ended noise reduction, works along with the six bands of noise gating to remove any existing hiss from the material to be processed. Included is an Ultramizer function that provides Behringer's proprietary normalization for maximum signal loudness.

A large, high-resolution LCD monitors the signal level, and the MIDI implementation in the device is extensive. The Ultra-Dyne has an automatic parameter-setting function, called the Virtuoso, to automatically set the parameters of the unit based on the program material's content.

A hard-wired relay-controlled bypass, which has an auto-bypass function for safety during power loss or failure, is included. The inputs are servo-balanced, with XLR and 1/4-inch TRS inputs and outputs. Behringer says they will have a free downloadable Windows-compatible editor for remote MIDI control available soon from their Web site. List price is $579.

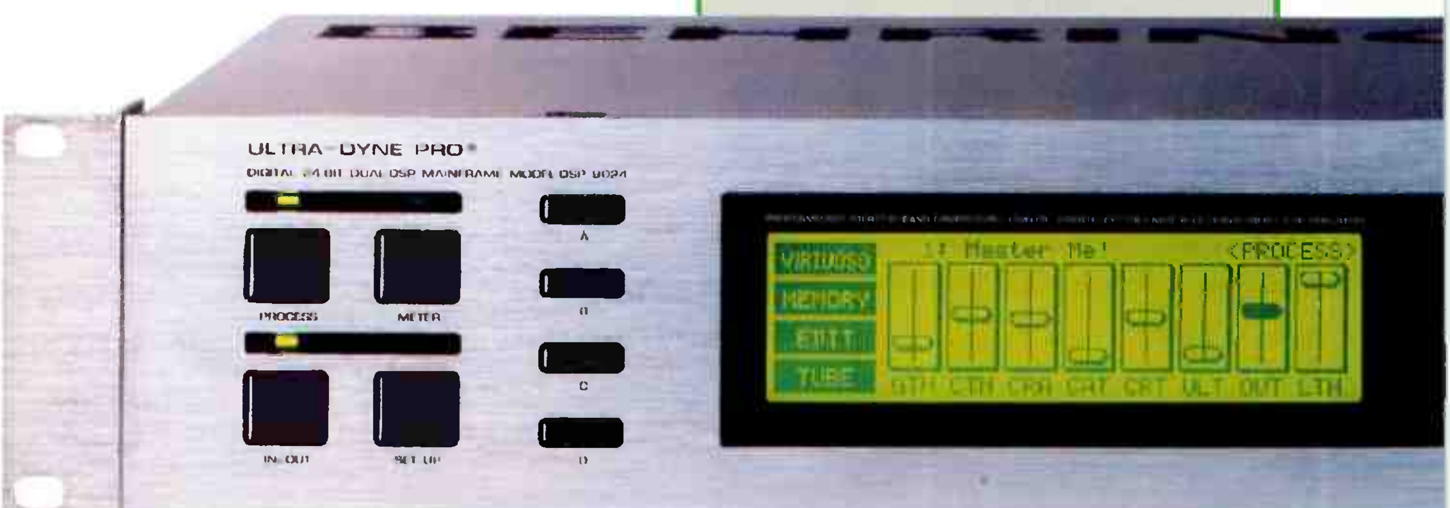
BEHRINGER ULTRAMIZER PRO

Behringer's Ultramizer Pro is a 24-bit digital compressor/leveler/loudness processor with two-band control. Besides the intelligent, program-driven compression, the Ultramizer Pro contains an exciter, stereo surround functions, and a Denoiser, Behringer's single-ended noise reduction system.

The inputs and outputs are servo-balanced analog XLR and 1/4-inch TRS connections, with 20-bit A/D and D/A converters and 24-bit internal processing. MIDI control is also available.

There are 50 user-definable programs that can be saved to a card in the PCMCIA slot, and a free

Behringer Ultra-Dyne Pro DSP9024: A "bang for the buck" leader, the Behringer Ultra-Dyne offers six bands of dynamic signal processing.



Behringer Ultramizer Pro: The lowest cost dynamics processor in our roundup, the Behringer Ultramizer offers two-band dynamics control plus additional signal processing for under $300.

saved to a card in the PCMCIA slot, and a free Windows-compatible software-editing program is available for downloading from the Behringer Web site. List price is $229.

DBX QUANTUM

The dbx Quantum will support sampling rates up to 96 kHz, with a true 48-bit signal path. The available multi-band dynamics are comprised of four bands, and each band can be individually compressed, limited, and gated. The Quantum's four-band crossover lets you choose crossover points for each band with independent slope selection. True stereo or independent mono operation is available within the unit. There's also a stereo five-band parametric EQ with variable, adaptive, or constant equalization, and high and low shelving ability.

The Quantum is said to preserve analog warmth and fullness while still achieving clarity by using dbx's trademarked Type IV conversion system. Additionally,

Ten Tips for Getting More Out of Your Mastering Processor

1. Don't overuse it — You can't "undo" a process, especially if it's compression. You might be able to re-equalize if you excessively EQ, but you'll run the risk of messing with the phase of the signal and causing other problems. It's better not to overdo it. Remember, all this processing is done in real time, so what you hear is what you get.

2. Check your mix in mono — Even though you may think there isn't anything but stereo anymore, you'd be surprised to find out how many times your mix may be listened to in mono. If you have a mastering processor that provides "stereo enhancement," or "stereo expansion," or something similar, be careful. This effect is often achieved by phase manipulation, which might not translate very well if the final signal is monitored in mono — you don't want to find instruments missing or canceling each other out, or any other weird stuff happening. It's better to check out mono compatibility early in the process, so you aren't surprised later, when it's too late.

3. Watch your headroom — Even though you're probably trying to get as hot a signal as possible *out* of the unit, don't slam the input too hard. Remember that it's a digital device. While you want to get as high an input level as possible for the best resolution, you *don't* want to clip the thing.

4. Save your settings — If you save the settings of all your parameters (if the model you use supports this) and you mix in a fairly consistent style, you may find that you only have to do some minor tweaking for different songs. And if you are working on a few projects at once, you can easily get back to where you were without all the fumbling through parameters and menus.

5. Use the presets — These will usually get you off to a good start. Once you're in the ballpark, you can then adjust to your heart's content, without laboriously starting from scratch.

6. Use the "Wizard," "Virtuoso," or similar "automatic" functions — Although these may seem bland to begin with, the "automatic" parameter functions they implement may help facilitate and speed up your operation of the box.

7. Automate — Some mastering processors will let you change parameters using automation, usually via MIDI. If you have the unit set up where it sounds great except for the bridge of the song where things get quiet and need different settings, don't be afraid to automate the processor. This will give you results more like those you can achieve with a DAW, where you can have different amounts of signal processing for different parts of the tune.

8. Put the A/D or D/A converters to use — The analog-to-digital or digital-to-analog converters on your mastering processors may be better than the ones you currently have on your DAT machine or any other converter built into a piece of equipment. Since most have a 96 kHz/24-bit resolution, they may outperform the usual A/D or D/A converter you normally use, and some models give you additional dithering options. You can use the device for this function only, or maybe with a bit of normalization, tube emulation, or whatever else you want when converting from the analog to digital, or vice-versa.

9. Insert the mastering processor while you're mixing the tracks — In some cases it works well to set up an overall EQ and compression scheme using a mastering processor across your stereo bus *before* you start to add those processes to individual tracks. Same with any "enhancement" processes you choose, whether it's "stereo width," "tube saturation," or "Digital Radiance." This way you can tailor your individual tracks to the sound that the mastering processor is adding to the entire mix.

10. Constantly A-B — It's very easy for your ears to get used to the processing you're adding — particularly high-end EQ — and you may find yourself cranking parameters up further and further just to continue hearing the effect. By constantly bypassing the unit and comparing the dry and processed sound, you can keep a better perspective on what you're *really* doing to the mix.

Hardware Mastering Processor

Roundup

Focusrite Platinum 4 MixMaster: The Focusrite Mixmaster differs from the rest of the mastering processors in our roundup, using analog Class A and "opto" electronics rather than all-digital processing.

New features in the Finalizer 96K include the ability to enter and exit at different sample rates, allowing sample-rate conversion between two supported rates. The Finalizer 96K has 24-bit/96 kHz resolution analog-to-digital and digital-to-analog converters, un-correlated stereo dithering, correlated mono dithering, and even inverse dithering for interfacing between various types of other digital audio equipment.

A new three-band stereo width control has been added, and the Finalizer 96K now has an enhanced dynamic range. The 96K provides TC's "Wizard" function, which automatically helps set up the parameters of the unit based on the program material.

Also available are word clock input via BNC, a stable digital clock for jitter-free digital input and output (AES/EBU or S/PDIF inputs may also act as the master clock), and an optical port for analog-to-Lightpipe stereo conversion. (You can also use this I/O port to process any two source ADAT tracks and send them out to any two destination ADAT tracks).

TC Electronic Finalizer 96K: The Finalizer 96K is the newest version of TC Electronic's Finalizer, the original stand-alone mastering processor.

Feature Chart

	Behringer Ultra-Dyne DSP 9024	Behringer Ultramizer DSP 1400P	DBX Quantum	Drowmer DC 2476	Focusrite MixMaster	TC Electronic Finalizer 96K	TC Electronic Finalizer Express
Multi-band compression	6-band	2-band	4-band	3-band	3-band	3-band	3-band
5-Band EQ	No	No	5-band	5-band	3-band	5-band	No
Digital enhancement	Yes	Yes	Yes	Yes	No	Yes	No
De-esser	Yes	No	No	Yes	No	Yes	No
Stereo width adjust	No	Yes	Yes	Yes	Yes	Yes	No
Expander/ gate	Yes	No	Yes	Yes	Yes	Yes	No
Storable presets	Yes	Yes	Yes	Yes	No	Yes	No
Normalizer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Dithering	No	No	Yes	Yes	Yes	Yes	Yes
Choice of dither type	No	No	Yes	Yes	No	Yes	No
A/D, D/A converters	24-bit	20-bit	24-bit	24-bit	Optional	24-bit	24-bit
High-res metering	Yes	No	Yes	Yes	Yes	Yes	Yes
PCMCIA card slot	No	No	No	Yes	No	Yes	Yes
MIDI in/out/thru	Yes	Yes	Yes	Yes	No	Yes	Yes
Sample rate conversion	No	No	Yes	No	No	Yes	No
Adjustable limiting	6-band	No	3-band	3-band	No	3-band	No
"Look ahead" delay*	6-Band	No	Yes	Yes	No	3-band	3-band
List price	$579	$229	$1,995.95	$3,100	$1,395	$2,995	$1,599

*Most of the units provide at least some look-ahead delay. In some cases, more extensive look-ahead capabilities are provided

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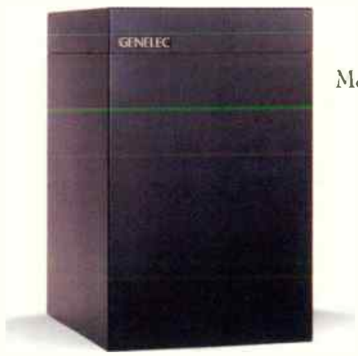
Model 1032A



Model 1094AP Subwoofer

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Hardware Mastering Processor

Roundup

TC Electronic Finalizer Express: The TC Electronic Finalizer Express is a scaled-down version of the TC's top-of-the-line Finalizer 96K, with an easy-to-use interface and many of its big brother's features.

Dynamic filtering and multiple simultaneous inserts are available, allowing you to run more than one insert effect at a time. The inserts include a five-band parametric EQ, dynamic EQ, de-essing, MS encode/decode, "Digital Radiance Generator," stereo adjust, and an external device insert, which gives you the ability to patch an outboard piece of gear into the signal path.

The unit has a variable-slope three-band expander/gate, a three-band compressor, and a three-band limiter, as well as real-time normalization. There's a L-R balance control in the input menu, and you can do fades in the unit itself or with an optional Master Fader. List price is $1,995.

Special thanks to Richard V. Wielgosz for compiling the manufacturer contact information and feature chart.

Manufacturer Contacts

Behringer, www.behringer.com, support@behringer.de, 425-672-0816. Circle EQ free lit. #120.
dbx, www.dbxpro.com, customer@dbxpro.com, 801-568-7660. Circle EQ free lit. #121.
Drawmer, www.drawmer.com, tech@drawmer.co.uk, 805-241-4443. Circle EQ free lit. #122.
Focusrite, www.focusriteusa.com, prodinfo@digidesign.com, 800-333-2137. Circle EQ free lit. #123.
TC Electronic, www.tcelectronic.com, info@tcelectronic.com, 805-373-1828. Circle EQ free lit. #124.

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Joemeek Pro Channel VC3Q
Mic Preamp/Equalizer/Compressor

Need a channel strip to drive your recordings? Joemeek has a solution.

BY MITCH GALLAGHER

More and more, engineers are routing input signals through the shortest signal path they possibly can. In most cases, this means bypassing the mixer and using an external preamp, especially for mic-level sources. Joemeek is one manufacturer that has addressed this trend with products such as the VC1Q, VC2 (tube), VC6Q, and others. Now the Devon, England-based company has introduced a compact, more affordable version of their input-channel technology, the VC3Q Pro Channel, which carries a retail price of $399.

The VC3Q combines a mic/line preamp with a compressor and a three-band equalizer. A number of convenience features are included such as dual outputs, an insert for connecting external gear, and a "mix" input for routing auxiliary signals into the VC3Q and combining it with signal flowing through the main signal path. The box is half-rack width; a rackmount kit is available for $24.95 list. Joemeek says that they are also planning a dual-rack kit for mounting two units in a single rackspace for the future. It's my bet that most people will use the VC3Q as a freestanding box; at roughly 8.25" wide by 6" deep by 1.75" high, it can be placed just about anywhere.

GETTING STARTED

The VC3Q uses an external wallwart 12VAC power supply. The power jack on the back of the unit pokes through a hole in the back panel, but isn't secured to the case. Given that the box will likely be used stand-alone, and probably moved around a bit, the power jack didn't strike me as particularly sturdy. The plug also had a tendency to drop out when the unit was moved. No power switch is provided. In fairness, there really isn't room for one on either the front or back panels.

All of the connections on the VC3Q are found on the back panel. These include an XLR input for mic-level signals, balanced 1/4-inch input for instrument- and line-level signals (plugging into it disables the XLR input), Mix In and Insert jacks (we'll come back to these), and two balanced 1/4-inch outputs. The dual outputs are especially cool for those tracking signals to sound cards; one jack can be used to feed the sound card input, while the other is monitored. This eliminates the latency problem common to monitoring through most sound cards and audio interfaces. Think of it as a built-in output splitter.

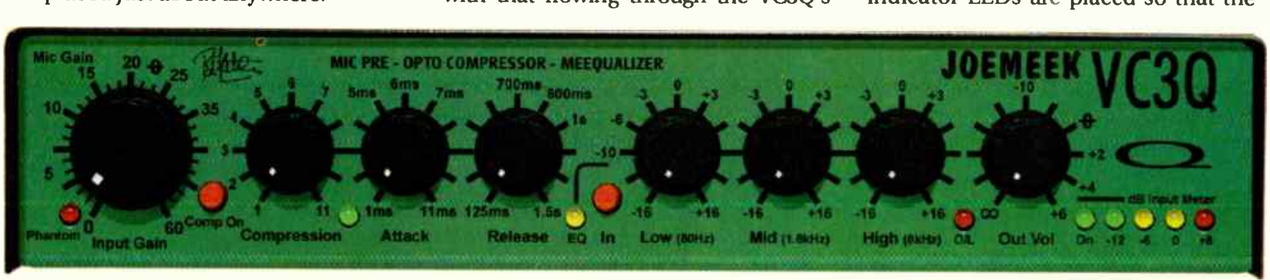
As mentioned above, the back panel sports two more connectors: the Mix In and the Insert. The TRS Insert jack functions as you would expect; signal is sent on the "tip" connection and returned on the "ring." The primary application for this connection is to insert an external processor into the Pro Channel's signal path; the insert point is located post-EQ/pre-compressor.

The line-level Mix In jack is intended to route external signals into the VC3Q. Signal at the Mix In is combined with that flowing through the VC3Q's main signal path. Strangely, the Mix In point in the signal path is also post-EQ/pre-compressor. This means that any signal coming in the Mix In will be compressed along with the main signal. But it also means that, if the main signal isn't loud enough to cross the compressor's threshold but the signal coming in the Mix In is hot enough to cross the threshold, the whole works will be compressed. The manual suggests that a use for the Mix In might be to cascade multiple VC3Qs when overdubbing vocals. If you do decide to use it that way, be careful not to compress at each individual VC3Q; only switch the compression on for the last VC3Q in the chain. Otherwise, you'll be compressing various signals multiple times. Not something you'd normally want. I'm not sure how many people will regularly use the Mix In jack; I consider it more of a convenience feature than a "must-have." Still, I would have preferred to have seen the Mix In routed into the signal path at the final output stage.

Rounding out the back-panel features is the phantom power switch, which provides 48V power for condenser mics. I have a pet peeve against back-panel phantom switches, although, as mentioned earlier, panel real estate on the VC3Q is at a premium; there's just no other place for it. A bigger quibble is that the switch is almost flush with the panel when depressed. If you're a fat-fingered engineer like me, you may find it hard to turn off.

AROUND FRONT

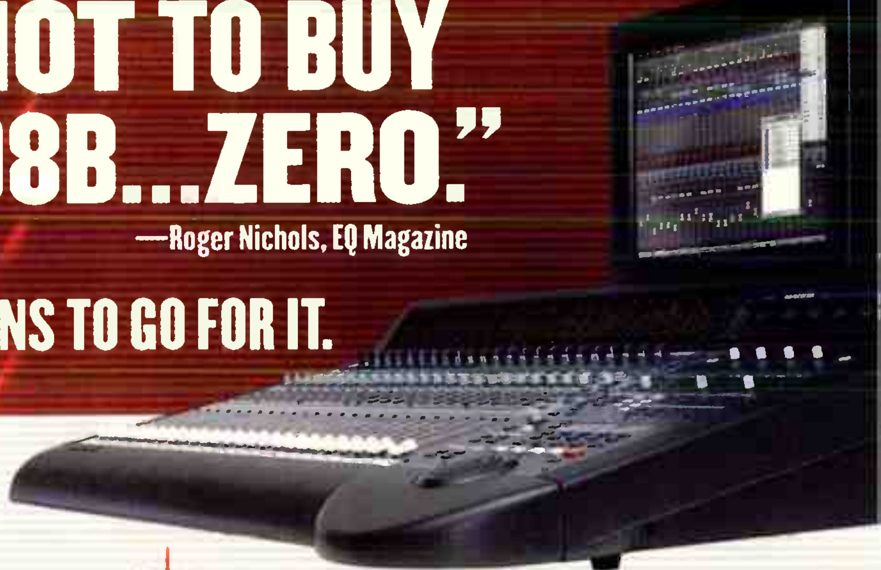
The front panel of the VC3Q is clear and easy-to-understand. The markings are readable, and the knobs, switches, and indicator LEDs are placed so that the



"REASONS NOT TO BUY A MACKIE D8B...ZERO."

—Roger Nichols, EQ Magazine

PLUS 3 MORE REASONS TO GO FOR IT.



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- **Updated GUI** including 48-channel fader bank view screen.
- **Time Offset** (delay) adds a delay of up to 999 samples to the signal at the pre-DSP (dynamics / EQ) point in the signal path.
- **New surround capabilities** including depth-of-center control (LCR mixing with divergence), multiple surround panner window, individual LFE channel level control.
- **Multiple direct outs** per channel.
- **Optional level to tape** fader control.
- **Assignable, bidirectional MIDI control** of all parameters.
- **Cross patching** allows substitution of channels between various banks.

2 DSP PLUG-INS!

Antares' Auto-Tune for the D8B uses advanced DSP algorithms to detect the incoming pitch of a voice or solo instrument as it's being tracked and instantly pitch-correct it without introducing distortion or artifacts. Fully automatable.

Massenburg Parametric EQ. MDW 2x2 High-Resolution Parametric Equalizer plug-in from Grammy-winning engineer/ producer George Massenburg. Mono/stereo EQ at 96kHz sample rate for unprecedented clarity and high frequency smoothness.

Drawmer offers two dynamics packages for the D8B: ADX100 includes their industry standard frequency conscious gating, plus compression and limiting; ADX200 adds variable "Peak Punch" and further Drawmer innovations.

IVL Technologies' VocalStudio provides real time vocal doubling, multi-part harmonies and pitch correction in an easy-to-use interface. A free demo is built into the Digital 8 • Bus. Just add a second MFX card to own this innovative plug-in from a world leader in vocal processing.

TC Electronic Reverb (bundled with the D8B UFX card) provides Reverb 1 and Reverb 2 algorithms from the renowned TC Electronic M2000 Studio Effects Processor. TC FX upgrade package contains an expanded set of M2000 reverbs plus Delay, Chorus, and Pitch. TC 2000 adds the TC M2000's Reverb 3, de-essing, tremolo, phasing, and panning.

3 1999 TEC AWARD WINNER!

Normally we don't name competitors in our ads. But in this case, Mix Magazine published the other nominees for the 1999 TEC Award for Outstanding Technical Achievement in Small Format Consoles: Allen & Heath's GS-3000, Digidesign's ProControl, Panasonic's WR-DA7, Spirit's Digital 328 and Yamaha's 01V. Thanks to all who helped us win this prestigious award.

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status of each is visible, even if you're not looking straight-on at the unit.

I did find the layout a bit strange. There's no signal flow chart for the VC3Q provided in the manual, but during my tests it became apparent that the flow is from preamp to EQ to compressor. But the panel is laid out (left-to-right) preamp to compressor to EQ. When you factor in that the Insert and Mix In are post-EQ/pre-compressor, this is even more confusing.

Layout concerns aside, first up, you're given control over the preamp gain. Plenty of gain and headroom is available for dealing with mic, instrument, and line signals. Next comes the compression knob, which is actually another gain control. Turning up this control pushes the signal past the VC3Q's threshold into compression. The VC3Q is set up to compress harder and harder as you increase level, so setting this knob is important for achieving the results you want. Following the compression knob are the compressor attack and release controls. You're given a reasonable range for each, although I occasionally wished for faster release times (125 ms is the shortest available). A small switch turns the compressor on and off. No "compressor on" indicator is provided, but an LED does indicate when the threshold has been crossed and compression is being applied. The light fades out as the release stage passes, giving you a good indication of what's happening inside the box.

Following the compressor is the EQ, which is a three-band, fixed-frequency design (80 Hz, 1.8 kHz, and 8 kHz). The VC3Q provides ±16 dB of gain for each band — which should be enough for most applications! In most cases, I found that I was overloading things downstream before I had reached the limits of the EQ's gain. With so much range across the rotation of the knobs, a small setting change can make a big difference in the sound; let your ears be your guide. The frequencies are well chosen, and the sound of the EQ is warm and musical. Even at extreme mid- and high-band settings, I didn't find it harsh. I found the low-frequency control especially effective for aggressive synth basses. You can use it to really dial in the thump and drive. Like the compressor, the EQ features an on/off switch; there's an "EQ on" indicator LED provided.

Up last is the output gain control. The VC3Q has enough output range to cover –10 to +4 level requirements. An LED shows overload at the output of the equalizer section and a five-stage LED meter shows input levels. Strangely, this input level meter is located near the output knob, rather than the input knob.

IN USE

I put the VC3Q to work in a variety of situations. If you read my Bomb Factory SansAmp plug-in review last month, you know that I used it to feed my Strat-style guitar directly into Pro Tools. I also checked it as an input channel for fretless and fretted basses, for keyboards, with a Shure SM57 and Audio-Technica AT4050 on vocals, and as an inline processor for pre-recorded tracks.

In every situation, the VC3Q had plenty of gain. When plugging a direct guitar straight in, there was no need to use a direct box, although you could, of course, use one if you desired. There's enough headroom in the preamp and range in the input gain control to deal with hot signals as well. I would characterize the sound of the preamp as warm and round. Definition on the top end is good, but it's not "crystalline." Don't take this to mean there isn't clarity on the top end, rather, it's a softer-sounding top than you may find on other designs. I compared several preamps with the VC3Q, and found that it stacked up well against units in all price ranges. The box offers quiet, clean operation, although there's enough gain that if you want to grunge something up or even outright overdrive it, you certainly can.

The VC3Q compressor is easy to dial in. Set the attack time to allow as much of the

EQ LAB REPORT

MANUFACTURER: Joemeek, distributed in the U.S. by PMI Audio, 23773 Madison St., Torrance, CA 90505. Tel: 877-563-6335. Web: www.joemeek-uk.com, www.pmiaudio.com.

SUMMARY: A clean, quiet, compact, good-sounding channel-strip-in-a-box with that signature Joemeek sound.

STRENGTHS: Round, warm tone. Cool price. Effective tone-shaping EQ. Aggressive compressor. Compact package. Easy-to-use. Insert jack. Dual outputs.

WEAKNESSES: Mix Input is pre-compressor. Power input jack feels less than sturdy; power connector falls out easily. No sidechain or link capabilities.

PRICE: $399 **EQ FREE LIT. #:** 130

SPECIFICATIONS

Inputs: Mic-level XLR, balanced 1/4-inch instrument/line, unbalanced 1/4-inch Mix Input
Outputs: Balanced 1/4-inch [2]
Insert: TRS 1/4-inch, tip is send, ring is return
Phantom Power: Yes, switchable
Input Gain: Up to 60 dB
Mic input Stage Noise: –125.5 dB, 20 Hz to 20 kHz @ 200 ohms
Output Noise: –80 dB
Frequency Response: +0/–.5 dB 10 Hz to 50 kHz
Equalizer Bands:
Low: shelving @ 80 Hz, ±16 dB
Mid: peaking @1.8 kHz, ±16 dB
High: shelving @ 8 kHz, ±16 dB
Compressor Attack Time: 1 ms to 11 ms
Compressor Release Time: 125 ms to 1.5 sec
Compression Ratios: Soft threshold, 1.2:1 to 7:1

transients through as you want, then set the release. Next, dial up the compression knob, which, as mentioned before, is actually a gain control. Crank it up until the signal crosses the threshold, and compression sets in. Once you start moving past the threshold, compression quickly becomes evident. As with all Joemeek compressors I've used, I wouldn't characterize the VC3Q as overly "subtle," although you can set it up so that it's just "touching the peaks" of the signal if you want. But this processor seems happier when it can grab your signal and put a sonic signature on it — great for aggressive rock sounds, adding punch to a signal, and giving the sound some excitement. About the only additional thing I'd ask for is the ability to be able to link a pair of these babies for stereo operation.

The equalizer was useful for tailoring the tone of signals. I especially found it good for guitars and basses. Note that you're not going to be able to do "sonic surgery" with this EQ. It's designed as a tone-shaping device and does this job very well.

BOTTOM LINE

So what's the final verdict? In a word: Cool. I had a few quibbles, but overall I found that the VC3Q was effective in every situation. No one will tell you it's the most uncolored preamp or that the compressor is subtle, but that's the beauty of it. As with other Joemeek offerings, it sports a bit of an attitude, pure and simple. Being able to buy into that attitude for only $399 is great for those looking to beef up the preamp, EQ, and compression capabilities in their studio and live rigs. EQ

TC Electronic M•One Dual Effects Processor and D•Two Multitap Rhythm Delay

A pair of reasonably priced processors from one of the "audio elite"

BY STEVE LA CERRA

When TC Electronic introduced their M•One and D•Two processors, *EQ* reacted with a bit of a raised eyebrow. Here was a company we consider one of the "audio elite" introducing a pair of effect devices with a list price well under a grand each; processors that would appeal to the MI and home studio market as well as the pro audio market. Surely they must be stripped-down boxes with limited editing facilities, and with user interfaces that would send any tech-head running for the hills, right? Not by a long shot. TC Electronic has, in fact, produced a pair of processors that may leave you scratching your head and wondering, "How do they do it?"

NUTS AND BOLTS

Since the M•One and the D•Two share many physical characteristics, we'll take a look at those first. Each unit is packaged in a single-rackspace chassis with rear-panel, stereo analog, and balanced TRS inputs and outputs. Analog-to digital conversion and digital-to analog conversion is 24-bit with 128x oversampling, and sample rate may be set to 44.1 kHz or 48 kHz.

Additional rear-panel features include a pedal input for bypass or tap functions, MIDI in, out, and thru ports, S/PDIF input and output, and an IEC power receptacle (no wall wart, thank you!). Front-panel controls include a power switch, input level and mix controls, rotary encoders for data entry, and a very informative, multicolor LCD (the M•One also has an "effect balance" control). In addition to left and right channel input levels, the display also shows program name/number and indicates when a preset set has been edited. TC furnishes the M•One and the D•Two with detailed manuals that provide operational tips and examples; quick start guides are also included for the impatient toy aficionado.

THE M•ONE

The M•One Dual Effect Processor contains two stereo processors or "engines," making it possible to simultaneously use any two of the following algorithms: Hall, Room, Plate 1, Plate 2, Spring, Live, Ambience, Delay One Tap, Delay Two Tap, Chorus Classic and 4-voice, Flange Classic and 4-voice, Pitch Detune and Pitch Shift, Parametric EQ, Compressor/Limiter, Gate/Expander, De-esser, Tremolo, and Phaser. Complementing this variety of algorithms is the M•One's ability to perform multiple I/O routing schemes.

Pressing the front-panel Routing button allows access to six different I/O options:

• Dual S/R allows use of the unit as two separate stereo effect devices. Input to engine one is on the left input, input to engine two is on the right. The stereo output from engine one is combined with the stereo out from engine two, and sent out of the rear-panel TRS jacks. A front-panel "balance" pot allows you to adjust the relative level of the two different effects.

• Parallel sums the left and right inputs, sending both inputs to both engines. This is useful for putting two different effects on the same source.

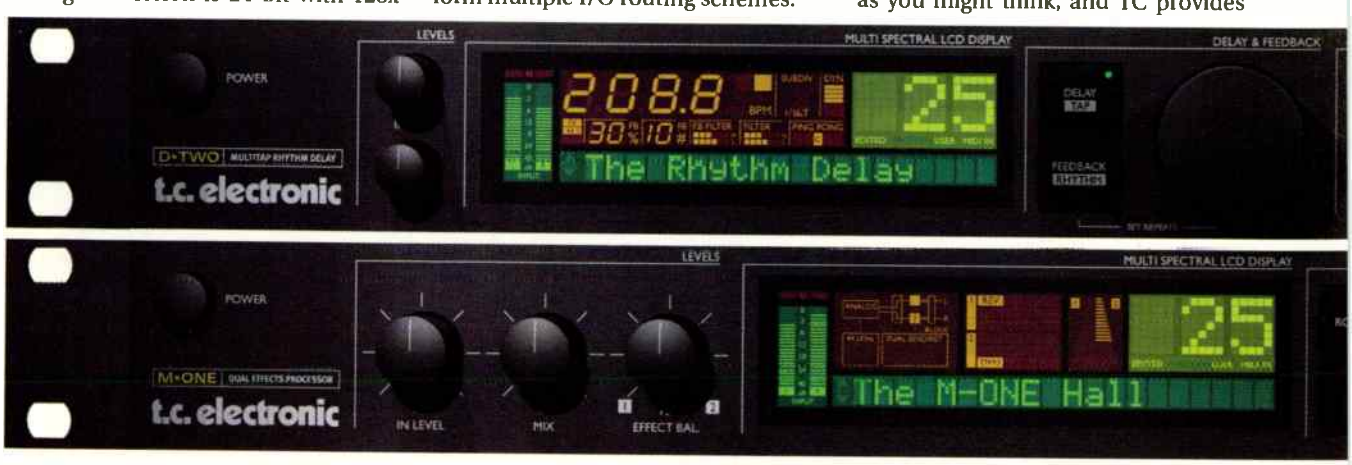
• Parallel/Serial is similar to Dual with one addition: the output of engine one can also be fed back to the input of engine two, enabling you to have engine two process the output of engine one (adding reverb to the repeats of a delay, for example).

• Serial routes the input through engine one and then through engine two.

• Stereo-linked runs both engines with the same algorithm. Changing a parameter on either engine will result in both engines being modified (this would be good for a true stereo compressor).

• Dual Mono keeps the two engines completely separate, as might be appropriate for insertion on channels.

These options are not as confusing as you might think, and TC provides



excellent tips for the various routings in the brief (but useful) manual. Routing may be stored as a parameter with a particular program, but there's also a "manual" global routing setting that takes precedence over the program. Part of the front-panel display indicates which routing scheme is active and also shows a graphic of the signal flow.

During operation, the front panel shows the name of the preset currently loaded. A twist of the Encoder knob scrolls through the factory presets while the Enter button flashes. When you find one you want to hear, press the Enter button. It stops flashing and the new preset is active. To edit engine one, simply press "algo/edit 1" (press "algo/edit 2" for engine 2). The display shows "Fx1" and the name of the currently loaded algorithm; the algorithm may be changed by turning the Encoder. To edit parameters, use the Up and Down buttons next to the Encoder to select the parameter, and turn the Encoder to change the value. The instant you change a parameter, "edited" lights underneath the preset number, making it clear that you have changed something. TC's designers have carefully chosen the available parameters for each algorithm, providing access to the ones most likely to be tweaked, yet not confusing the user by providing too many. As an example, the Hall algorithm includes delay time, predelay, room size, high cut, high color, low color, reflection level, reverb level, modulation (type, speed, and depth), and overall level. Definitely enough to keep tweakers happy without overwhelming someone that just wants a little more decay time. The "color" parameters on the Hall effect adjust decay time in the high frequency or low frequency range (I found this quite useful).

If you decide to store your edited program, press Store and the M•One will set you up to write the program into the first available user bank (you can choose to write it into any user bank). This was where I found the only awkward part of the M•One's interface: After pressing Store, you name the program by pushing the Down button. Once you do this, a cursor blinks underneath the first letter of the name, indicating that you can change the letter. At some angles, this cursor can be hard to see. Other than that, the interface is a breeze to get around. Bypass buttons for each engine do exactly that, and if you feel like you're getting lost, the Recall button takes you back to the last stored version of your program.

EQ LAB REPORT

MANUFACTURER: TC Electronic, Inc., 742-A Hampshire Road, Westlake Village, CA 91361. Tel: 805-373-1828. Web: www.tcelectronic.com.

SUMMARY: Dual-engine effect processor with true stereo operation (M•One); programmable multitap delay (D•Two)

APPLICATIONS: Studio recording and mixing, live sound.

STRENGTHS: Extremely powerful processors. Quiet operation. Excellent-sounding effects. Flexible routing schemes. Reasonable prices.

WEAKNESSES: Some parameters have coarse adjustment (M•One). LCD cursors can be difficult to see under certain lighting conditions.

PRICE: $699 each **EQ FREE LIT #:** 102

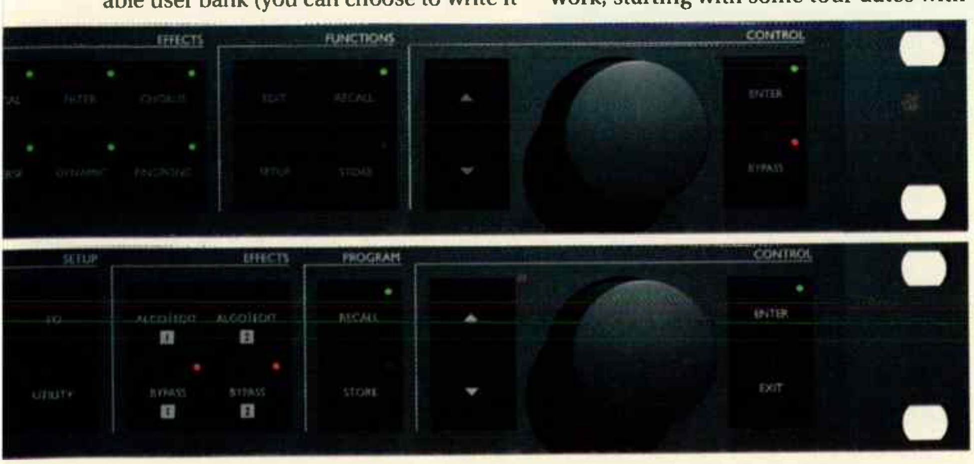
The M•One has a dedicated Tap button on the front panel, which may be used to tap-enter delay time for the delay algorithms. Although the manual states that this may also be used to set chorus rate, I couldn't seem to get the tap function to affect parameters other than delay time. In any event, the tap function is a very useful addition. A note value may be set so that the M•One recognizes what sort of beat you're entering (e.g., quarter-note, eighth-note, sixteenth-note). The delay time range is from one millisecond up to five seconds, so the unit should be able to accommodate most delay needs.

M•ONE IN USE

I wasted no time in putting the M•One to work, starting with some tour dates with Blue Öyster Cult. At the time, I hadn't cracked the manual open, so I crossed my fingers for a friendly welcome from TC. I wasn't disappointed. After scrolling through the various presets, the LED on the Enter button began flashing, prompting me to press it to call up the program. The "algo/edit" buttons were self-explanatory, as were most of the front-panel controls. For the first show at House of Blues in Orlando, Florida, I used preset 1 ("M-One Halls") on Bobby Rondinelli's drums and it was wonderful, to say the least. It sounded as if I placed him in a room, as opposed to adding an effect. Decay time was easy to tweak during the show, and the reverb decay was smooth with none of the zippy "boing" you might hear from a lesser unit. It even sounded good when kick drum was added into the 'verb — a feat that most reverbs can't accomplish without sounding "fake." Reverb quality is consistent regardless of the algorithm used, and the plate actually sounds like a plate (though I admit it's been a while since I used one). No doubt, this is a TC Electronic 'verb.

Over the next few nights I used the M•One on more Blue Öyster Cult shows as well as on a mix session for an upcoming CD from the alt-rock band GodSalad. To put it bluntly, the M•One rocks. Producing good reverb, delay, and chorus sounds would have been enough (in particular preset 86, "Delay and Chorus," was stunning). But I was pleasantly surprised by the fact that just about every sound that came out of it was useful. The phaser algorithm brought us back to the garage days of Electro-Harmonix stomp boxes, sounding both smooth and dirty at the same time. On a 12-string acoustic guitar, I added just a touch of phaser for a subtle sweep that was wonderful in the mix. Even the compressor sounded good. (Which I didn't expect — I figured it was just there because it could be.)

With routing set to Stereo Link and the compressor patched insert-style on a



pair of drum overhead mics, the kit sounded killer. It'd be nice if the compressor attack time was adjustable, and I found the minimum release time to be a bit on the long side (10 dB per second). Available release times for the expander were also somewhat coarse, ranging from 20, 30, 50, 70, 100, 140, 200, 300, 500, and 700 milliseconds, continuing to 1.0, 1.4 2, 3, 5, and 7 seconds; I'd like to have finer steps. When using any of the dynamics algorithms, a portion of the LCD indicates gain reduction per channel, letting you see what's happening to the signal.

Overall, the M•One was very quiet, adding no discernible noise to mixes. Interestingly, I found that, when the mix control favored the dry signal, output was slightly more noisy than when set to 100 percent wet. Long reverb decay times of 10 to 20 seconds are a bit hissy in the tail, but still far better than the average bear — besides, when was the last time you used a 20-second reverb?

M•ONE CONCLUSIONS

Much more than "just" a high-quality reverb and effect processor, the TC Electronic M•One is a cool creative tool with a lot of useful sounds. If they want it back, they're gonna have to come and get it.

M•ONE AND D•TWO SPECIFICATIONS

Memory locations: 100 factory, 100 user (M•One); 50 factory, 100 user (D•Two)
Analog I/O: stereo 1/4-inch balanced
Converter resolution: 24-bit, 128x oversampled, 44.1 or 48 kHz
Frequency response, analog I/O: 20 Hz to 20 kHz, +0/–0.5 dB @ 48 kHz sample rate
Digital I/O: stereo S/PDIF
Frequency response, digital I/O: DC to 23.9 kHz, ±0.01 dB @ 48 kHz sample rate
Crosstalk: less than 95 dB, 20 Hz to 20 kHz
Maximum input level: +24 dBu
Total harmonic distortion: less than 0.002% at 1 kHz

THE D•TWO DIGITAL MULTITAP DELAY

Since the TC Electronic M•One and D•Two were designed to complement one another, I simultaneously reviewed the D•Two multitap digital delay. Capable of delay times up to 10 seconds in mono or five seconds in stereo, the D•Two provides tap-tempo entry for delay time, as well as dedicated dynamic, spatial, filter, chorus, and ping-pong effects. The D•Two's appearance differs from that of the M•One in that the D•Two has a second rotary encoder for quick access to delay time and feedback parameters without the need to scroll through a bunch of menu pages. Presets are loaded by turning the right encoder to scroll through the programs, and then hitting the Enter or Recall button. As soon as you adjust a parameter, "edited" lights underneath the program number.

The D•Two offers the option of three different operational modes:

• Traditional Delay Mode, where the unit works like a typical delay and offers access to delay time and feedback parameters. This mode is entered by holding down the Feedback/Rhythm button while turning the delay encoder counterclockwise until "Feedback #" in the display is set to "–."

• Straight Delay works like Traditional but adds a parameter for specifically adjusting the number of repeats. This mode is entered by setting the feedback parameter to any value between one and ten.

• Rhythm Mode, which is unique to the D•Two. A rhythm is tapped into the front-panel Rhythm button, and the unit automatically sets up delays to create the rhythmic pattern tapped. This mode is automatically entered when the rhythm is tapped.

All three modes may be used in conjunction with spatial, filter, chorus, dynamic, and ping-pong effects, all of which can be turned on or off instantly via dedicated pushbutton (the LED in the respective button lights up to indicate that the effect is active). A double-click on any of these buttons instantly switches the D•Two to the edit page for that effect, negating the need to search through menu pages looking for a parameter.

The ability to instantly turn the effects on or off — as well as the ability to double-click directly to the edit page — is extremely useful in live sound applications, where there's no time to spare in calling up an effect. You want a filter on the delay? Press the Filter button.

D•TWO IN USE

As with the M•One, I wasted no time digging into the D•Two, programming it the night before a string of shows with Blue Öyster Cult. Since the show requires about a dozen song-specific delays, I thought writing some programs would be a good way to evaluate the "friendliness" of the D•Two. I turned the unit on, set the delay time with the encoder, and pressed Store. The D•Two took us to <EMPTY PRESET> 1, with the "1" blinking to indicate this was where I'd be storing the program. When I pressed the store button a second time, I thought that nothing happened, but, in fact, a cursor appeared under the "E" (it was difficult to see at our viewing angle, plus I wasn't exactly looking for it). Turning the right encoder scrolls through the alphabet for naming the preset. The Up and Down buttons move the cursor to the next (or previous) character. It took no more than about 10 minutes to write 15 programs for the show — without opening the manual. I'd say this unit was easy to program.

A push of the Dynamic button reveals that the D•Two shares some characteristics with its well-established older sibling, TC's 2290 delay. The dynamic function is equivalent to "ducking" the delay via compressor with a sidechain signal. As an example, let's suppose you're adding a delay to a lead vocal. To keep the mix from getting muddy, you might want to duck the delay down when the voice is singing, and bring it up a bit after the vocal line has ended. A typical way to do this in the past was to run the delay through a compressor and use the dry vocal to trigger the compressor's sidechain. When the vocal is happening, the compressor kicks in, reducing or ducking the level of the delay. As the vocal line ends, the compressor lets go, raising the level of the delay and making it more apparent in the mix. Certainly this is easy enough to do in the studio, but it's a pain in the you-know-what under live circumstances.

Press the Dynamic button and the D•Two does it for you, providing adjustment for threshold, release time, and

continued on page 136

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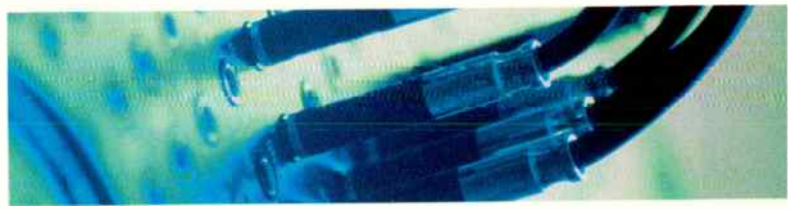

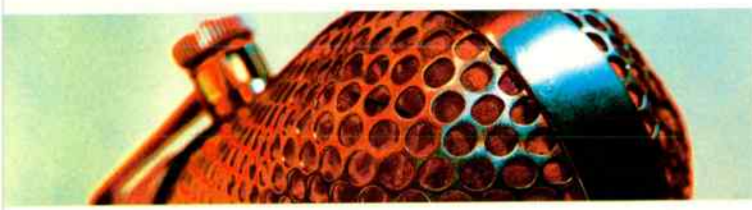

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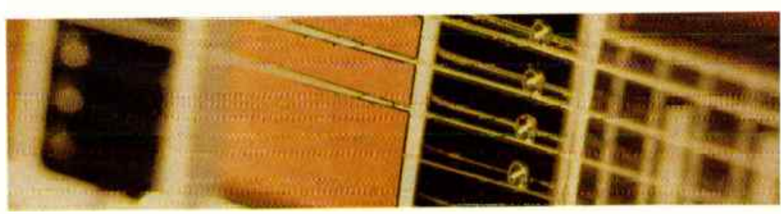

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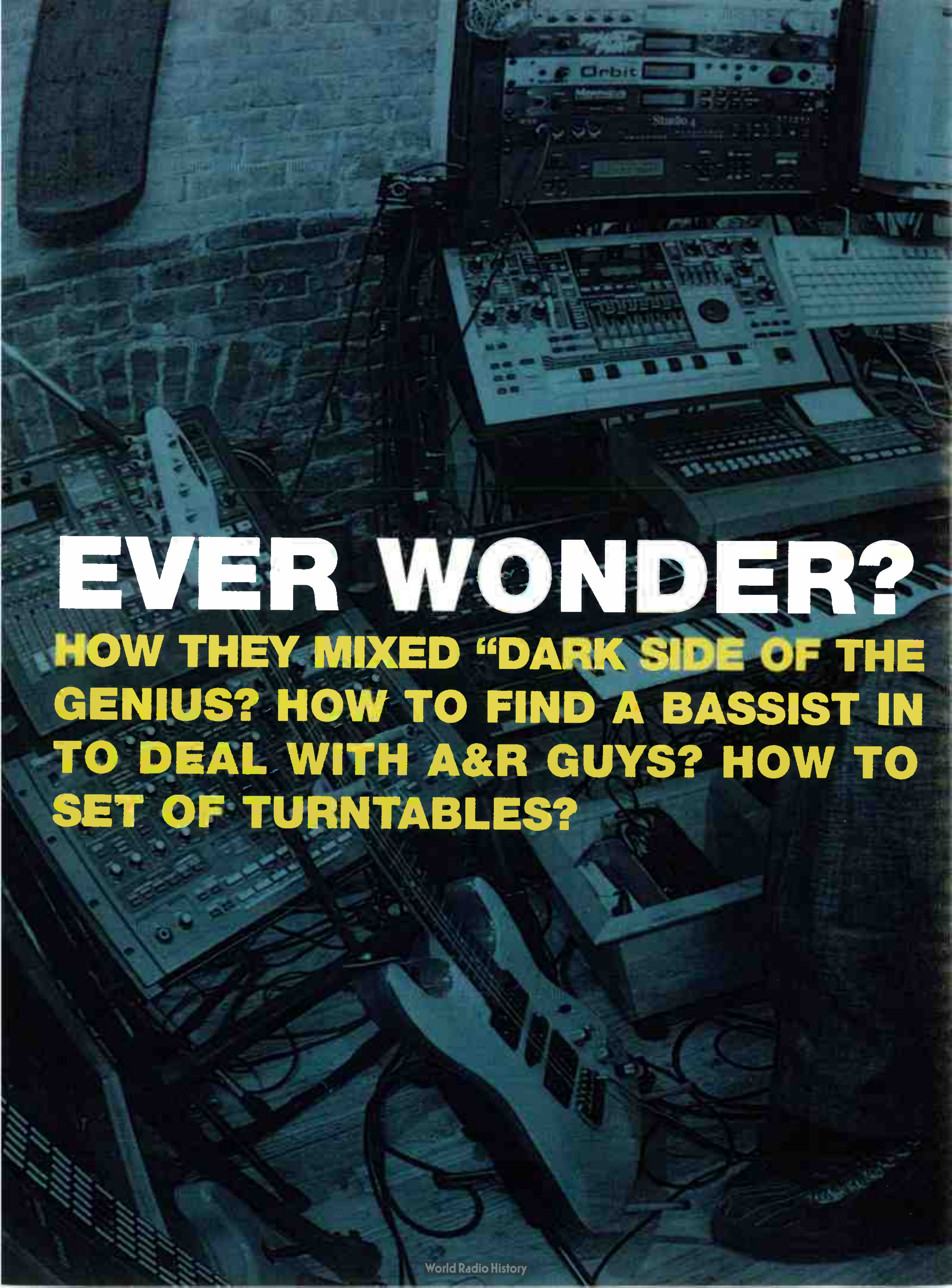

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Steinberg LM-4 and Model-E 24-bit Virtual Drum Machine and Virtual Synth

Put your Mac or PC to work as a synth and drum machine

BY MITCH GALLAGHER

The move toward virtual computer-based studios marches inexorably along. Given the current burst of new model releases from companies such as BitHeadz, Native Instruments, Koblo, Access, and others, it's now possible to potentially do away with a variety of hardware-based units, or to at least acquire reasonably priced recreations of the sounds that live in those units. With their new VST 2.0 spec and the release of several virtual synths plug-ins of their own, Steinberg have clearly made the ongoing development of this part of their music-production system a priority. In this review we'll be taking a look at two of the virtual synths recently introduced by the company, the LM-4 drum machine and the Model-E analog-modeled synth.

LM-4

Let's clear up one thing: Calling the LM-4 a "drum machine" is a bit of misnomer, since the plug-in doesn't actually contain a sequencer of its own. It's actually more of a drum sound module, modeled to look familiar to drum machine users. The sequencing power necessary to drive the sounds in LM-4 comes from whatever application is its host. But that doesn't stop LM-4 from being very cool. You're given 18 "pads," under which can live 16- or 24-bit sampled sounds. The output of the pads can be routed to stereo outputs or to one of four individual outs. This lets you create a stereo submix of hihats and cymbals while still bringing your kick and snare out separately for independent EQ and effects processing. The six outputs show up as Virtual Instrument channels in Cubase's VST Channel Mixer, where they can be automated and processed just like any other VST audio channel. In Logic Audio, only the main stereo outs are available.

One thing to be aware of: LM-4 shares the memory of whatever host program you're using to drive it, so on the Mac you'll need to kick up the amount of RAM assigned to your sequencer. The included basic Steinberg drum kits will work with a minimal amount of extra RAM. But also included on the install disc are ten 24-bit multi-sampled kits created by the sound designers at Wizoo. For these, you'll want more RAM; each of the Wizoo sets can require up to 50 MB. If you're loading multiple instances of LM-4, each requires its own RAM for loading kits. The point is this: The minimum free RAM requirement of 64 MB is just that, a bare minimum. Even the recommended free RAM of 96 MB probably isn't enough if you're going to push LM-4

The Model-E "analog" polyphonic synth requires a VST2.0-compatible host program to operate. You're given three oscillators with six waveforms each, a noise source, a dual-mode resonant filter, two ADSR envelope generators, MIDI control, and more.

Steinberg's LM-4 creates a 24-bit sample-playback drum machine within any VST2.0-compatible digital audio sequencer, such as Cubase VST or Logic Audio. LM-4 provides 18 velocity-sensitive drum "pads," each of which can contain a velocity-switching multi-sample.

hard. If insufficient RAM is available, LM-4 will load as samples into as many pads as it can, then leave the rest empty. There's no top-end limit to the amount of RAM LM-4 can use; I created kits that required up to 300 MB of RAM before giving up.

The kits included with LM-4 comprise a variety of styles, from Steinberg's basic GM Real Drumset to Wizoo's Hip-Hop, Latin, and Acid Jazz sets. The sound quality of the kits is uniformly high; the Wizoo kits, in particular, sport excellent detail and nicely done velocity switching.

You can trigger LM-4 using a MIDI controller, from a MIDI track in your sequencer, or using the mouse to "play" the pads on its front panel. By clicking on the LM-4 logo at the bottom of the plug-in's window, you can play the pads with the mouse and record the results as MIDI data in a sequencer track. I tried LM-4 in both Steinberg's Cubase VST/24 v4.1r2 and Emagic's Logic Audio v4.2.2 using all three trigger methods with uniformly excellent results.

CREATING YOUR OWN KITS

Creating your own LM-4 drum kits is a bit non-intuitive. Rather than some sort of menu system or the clever drag-and-drop kit-design interface you might be expecting, you're forced to create a text file listing a string of parameter values and sample names. LM-4 interprets this text file as instructions for loading a drum kit. In practice, it's not that difficult to do, but it's also not much fun. The easiest way to get started is to open up one of the supplied drum kit files in your word processor or text editor and use it as a starting point. A "read me" file on the install CD-ROM describes the various parameters in very brief fashion; there's little in the manual on this process. After experimenting a bit, I was soon rolling my own kits using both AIFF and WAV files.

As mentioned above, samples can be at either 16- or 24-bit resolution, at any sample rate. It's best to set the sample's rate to the one being used for your song before loading it into a kit, otherwise LM-4 will attempt to sample-rate convert it, often with mixed results. Note that, in Logic Audio, you're only able to load in banks of kits; this makes it impossible to load in a single kit of your own sounds. If you're only going to use the factory kits, this isn't a problem. But if you do want to dig in and make your own drum sets, beware!

Even after learning your way around the text files, creating your own kits is still a bit of a pain. You have to have your host application open with LM-4 running, then switch off to a text editor to modify the kit, save the text file, switch back to the host app, reload the kit into LM-4, listen to the results, switch back to the text editor, and so on. Reloading the kit after each change is what can take time; if you're using small kits, this won't be much of an issue, but if you're setting up large kits with lots of velocity switching, then it's a slower process.

One alternative is to download Wizoo's free Drum Kit Editor utility from www.cubase.net. While it's clearly not intended to be super-pretty or flashy commercial release software, the Wizoo editor is graphically oriented, and simple enough to use to put kits together. (It better be — no documentation is provided.) Still, at the price, who can complain?

Within a kit's text file, you're given control over a number of sample parameters:

• Each pad can be named, and can hold up to 127 velocity-switching samples.

• You can assign a key range to each pad, but at this time the sample pitch doesn't change as you play across the key range. The "read me" file on the install CD hints that this may change in the future.

• Stereo samples can be loaded, and the volume of each side can be adjusted.

• Sample Start Offset (determines where in the sample file the sounds starts playing back) and Sample Length parameters are included, but don't seem to operate. It would be nice to see these implemented, and for MIDI velocity control to be allowed over Sample Start Offset. LM-4 plays back in "one-shot" mode, where each triggered sample plays all the way through. With no sample-length parameter, you're forced to either destructively edit the sample to the length you want or use another pad to cut it off (see below).

• Pads can be grouped and their polyphony can be limited. This lets you cut off the sound of one pad with another. For example, a closed hihat pad can be used to cut off an open hihat pad.

Ultimately, this may all be academic: One wonders how many people will want to bother creating their own kits at all. But if you do have the desire, the capability is there, which is cool. If not,

continued on page 136

EQ LAB REPORT

MANUFACTURER: Steinberg, 21354 Nordhoff St. #110, Chatsworth, CA 91311. Tel: 818-678-5100. Web: www.us.steinberg.net, www.cubase.net.

LM-4

SUMMARY: Twenty-four-bit virtual sampling drum machine plug-in for Mac and Windows VST 2.0-compatible host applications.

SYSTEM REQUIREMENTS: Mac: 200 MHz 604e or faster PowerPC, System 8.0 or higher, 64 MB or more of free RAM, VST 2.0-compatible host application. PC: 233 MHz Pentium II or faster, Windows 95 or 98, 64 MB or more of free RAM, VST 2.0-compatible host application.

STRENGTHS: Comes with a number of good-sounding kits. Easy to use. Up to 128-way velocity switching. Can load and create your own drum kits.

WEAKNESSES: Creating your own drum kits is less than intuitive.

PRICE: $99 **EQ FREE LIT. #:** 103

MODEL-E

SUMMARY: Modeled polyphonic analog-style synthesizer plug-in for Mac and Windows VST 2.0-compatible host applications.

SYSTEM REQUIREMENTS: Mac: 250 MHz 604e or faster PowerPC, System 8.0 or higher, 64 MB or more of free RAM, VST 2.0-compatible host application. PC: 266 MHz Pentium II or faster, Windows 95 or 98, 64 MB or more of free RAM, VST 2.0-compatible host application.

STRENGTHS: Good sounds. Familiar sound programming/user interface. No latency. Multitimbral or instantiate multiple copies.

WEAKNESSES: None to speak of.

PRICE: $199 **EQ FREE LIT. #.:** 104

VST 2.0, REWIRE, AND VIRTUAL INSTRUMENTS

The advent of Steinberg's VST 2.0 specification provides (among other things) a powerful new capability for users of compatible applications: virtual instrument support. Cubase VST ships with two bundled virtual synths: Neon, a "simple" analog-style synth, and VB-1, a bass synthesizer with a cool visual interface. In addition, Steinberg and third-party developers are offering a number of optional synths, such as the LM-4 and Model E reviewed here, but also including new Waldorf PPG and Native Instruments' Prophet V models, and others. In addition, Emagic's Logic Audio v4.1 and higher includes support for VST 2.0.

Note that VST 2.0 is different from Propellerheads' existing ReWire spec, which supports separate synth programs such as Propellerheads' ReBirth and upcoming Reason (see First Look, May, '00 issue for more on Reason), as well as BitHeadz Retro AS-1 virtual synth (a VST 2.0-compatible version of Retro has been announced) and Unity DS-1 virtual sampler, and others. ReWire provides a virtual patching system for communicating audio and control information between the synth programs and the sequencer. With VST 2.0, the VST synths are actually running as plug-ins within the host program itself.

I loaded version 4.1r2 of Cubase VST/24 and Emagic's Logic Audio v4.2.2 (which also includes ReWire support) into my 400 MHz blue-and-white Apple G3 PowerMac along with LM-4 and Model-E, and set about exploring the limits of the system. First of all, let me say that working with the virtual synths within Cubase and Logic is a pleasure. The synths live within the host environment itself, so there's no switching to separate applications as when running synths under ReWire. The "feel" difference is noticeable when editing synths, loading new drum kits into LM-4, and so on. Because the VST 2.0 plug-ins run inside the host, you'll need to make sure that the program has plenty of RAM assigned to it — especially if you're using LM-4 to load large drum kits. I found that, in most situations, 128 MB of RAM (above that required for the operating system, etc.) was a comfortable amount; more is even better.

On my system using Cubase, I was able to create a song consisting of an LM-4 drum machine playing a complex pattern (using the stereo and all four individual outs), two VB-1 synth basses playing octave-apart versions of the same bass line, two Model-E synths playing multi-note pads each, and one Neon synth playing a monophonic lead line. I also applied three VST effects plug-ins (stereo delay, reverb, and stereo chorus) as well as EQ on several tracks. This pretty much maxed out the machine; adding even one more note of polyphony on any of the synths overloaded the CPU, although I could still add more EQ, etc., without difficulty.

Don't take this as too much of a benchmark for system performance, though. Your mileage will surely vary with the combination of synths you're using, how you're using their sounds, your computer's horsepower, and so on. Still, I felt this was respectable, usable performance; at no time (up to overload) did the computer feel "bogged down" or labored. I also experienced no noticeable latency as I played the sounds using my MIDI keyboard controller.

—Mitch Gallagher

MOTU Digital Performer 2.7
Mac Digital Audio Sequencer

The new version of the venerable sequencer offers cool features and enhancements

BY MITCH GALLAGHER

Mark of the Unicorn's Performer was one of the first "pro" MIDI sequencers for the Macintosh. The program steadily evolved over the years, until finally, with the addition of audio support, it morphed into Digital Performer. (Performer remains available.) Recently, MOTU has been ramping up development efforts, adding a number of impressive features with each rev. The current release, version 2.7, follows this trend by adding major enhancements and a number of "small" improvements to the program.

THE DRUM EDITOR

The most "visible" of the new features is the Drum Editor (see fig. 1). In the Drum Editor, each pitch represents an instrument; C1 might be a kick drum, D1 a snare, and so on. You can customize these names, or, if you recall a patch that contains a name list, they'll automatically show up. Each pitch can be independently and non-destructively quantized and/or offset in time (by ticks).

Numerous cool features are included such as the nifty "Rhythm Brush," which lets you graphically "paint" a rhythmic pattern. You can easily create, name, store, and recall patterns: Just select the pattern you want, then tell the brush to "learn" it. Another nice touch is the "alternating stroke" velocity feature that simulates hitting a drum at slightly different levels with the right and left hands.

You can graphically edit velocity and continuous controllers from within this window. I find the tiny icons MOTU uses to represent continuous data to be difficult to see, especially when the data is dense. I'd prefer some sort of continuous curve representing the data. This small complaint aside, the new Drum Editor window is a solid success.

REWIRE

Version 2.7 adds support for ReWire, the Propellerheads-developed standard for communication between MIDI/audio software. I used Propellerheads's ReBirth, as well as BitHeadz' Unity DS-1 and Retro AS-1 extensively under ReWire with Digital Performer without difficulty. The respective programs' audio outputs show up as input sources in Digital Performer mixer; once there, they can be processed with plug-ins, bussed, and automated freely.

Note that this isn't the only way to incorporate virtual synths into Digital Performer. BitHeadz support FreeMIDI and MAS directly, as does Koblo, and I'm told that Native Instruments' Reaktor will have this feature soon.

There's no VST support in Digital Performer. If this is a concern for you, there are several solutions: TC Works Spark includes an MAS-compatible version of the Fxmachine plug-in matrix that can load VST plug-ins (see the Spark review in the April issue). And AudioEase VST Wrapper and Cycling '74 Pluggo support both VST plug-ins and instruments from within Digital Performer.

MIX AUTOMATION

The new automation features in Digital Performer are excellent. Not only can mixing board parameters be automated (with sample-accuracy, no less), but MAS plug-in parameters are also now automatable.

You can graphically view automation curves for multiple parameters simultaneously in the audio track-editing window. The parameter you're currently working on is shown as a bold black line, the others are shown as thin grey-blue lines. It would be nice if each parameter could be assigned a different color.

Numerous modes are provided for writing and editing automation using the mixer's controls or an external MIDI control surface. If you punch automation in and out, you can set how long it takes to "ramp" to the new data, assuring a smooth transition. Ramps are calculated with sample-accurate 32-bit floating point processing.

Also provided is tempo-based automation, allowing you to lock parameters to the tempo of the song. This is especially useful for controlling plug-in parameters such as LFO speeds and delay times.

PLUG-INS

Digital Performer includes 19 audio processing plug-ins, although this number is a bit misleading since some of the plug-ins are multi-purpose. With version 2.7, the plug-ins have been given a face-lift. Some plug-ins,

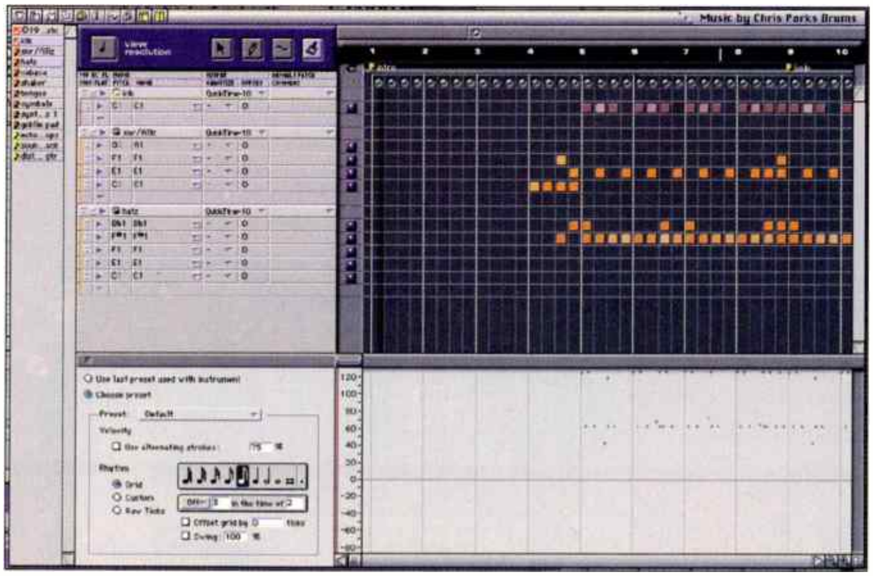

FIGURE 1. Digital Performer 2.7's new Drum Editor window provides a grid-based visual approach to data editing. Cool features such as the "Rhythm Brush" (which lets you "paint" in rhythmic patterns) help make this a powerful tool for MIDI work.

such as Sonic Modulator, now offer two views: the full view and a "condensed" view that takes up less screen real estate. Since v2.6, three new plug-ins have been added: a tempo-syncable stereo delay, a multimode filter, and a ring modulator.

Other plug-in enhancements include support for sidechain inputs, parameter automation (including tempo-based automation), and the ability to compare a modified preset with the original. An appreciated enhancement is support for plug-in folders, which lets you better organize your plug-in insert menu.

IS THAT IT?

MOTU has also seen fit to include a number of other enhancements and features, including full support for Mackie's HUI, an automatic Find Tempo command (matches tempo to a list of markers), scrubbing while trimming regions, audio region time-stamping, graphic time-stretching of audio regions, and much more — we've really just scratched the surface.

Two more items bear mentioning. The first is the "Adjustable PPQ" feature, which increases internal MIDI timing resolution up to around two trillion PPQ. This enhanced resolution is always active, but you can set the view resolution to between two and 10,000 PPQ. As a long-time user, I still found myself viewing data at 480 PPQ; old habits die hard.

The second is MIDI Time Stamping (MTS). MTS is hardware-based timing technology; when Digital Performer is used with a rackmountable MOTU USB MIDI interface (such as the USB MIDI Timepiece AV), MIDI timing is said to achieve one-third of a millisecond accuracy.

WRAP UP

At the upgrade price (free to registered users), version 2.7 is more than a sure winner; it's a must-have. MOTU should be commended for not settling for just releasing a bug-fix/maintenance upgrade, instead choosing to release a new version well stocked with new features and enhancements. EQ

EQ LAB REPORT

MANUFACTURER: Mark Of The Unicorn, 1280 Massachusetts Ave., Cambridge, MA 02138. Tel: 617-576-2760. Web: www.motu.com.

SUMMARY: A more-than-worthy upgrade to the respected digital audio sequencer adds new functionality, enhancements, and new plug-ins.

MINIMUM SYSTEM REQUIREMENTS: Power Mac with 32 MB of RAM (64 MB recommended) and System 7.5.5 or higher. System 7.6.1 or later recommended.

STRENGTHS: Enhancements too numerous to list. Highlights include: Powerful drum editor. ReWire support. Excellent automation features, including plug-in automation. Audio region "sync points." Nineteen bundled plug-ins. Version 5 DAE/TDM-compatible. Full HUI support. Free to registered users.

WEAKNESSES: Tiny icons used to represent continuous controller data. Can't group audio tracks for editing.

PRICE: $795. Free upgrade for registered users. Competitive cross-grade, $395

EQ FREE LIT. #: 131


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Kind Of Loud Technologies Reverb

Point and click your way to great-sounding reverb processing

BY RICH TOZZOLI

EQ LAB REPORT

MANUFACTURER: Kind of Loud Technologies, P.O. Box 3800, Santa Cruz, CA 95063-3800. Tel: 831-466-3737. Web: www.kindofloud.com.

SUMMARY: A highly flexible and great-sounding room simulator/reverb with full automation and morphing capabilities.

SYSTEM REQUIREMENTS: Macintosh-based Pro Tools Mix TDM, or an RTAS- or MAS-based DAW system. CD-ROM and floppy drives for installation.

STRENGTHS: High-quality reverb. Excellent graphic interface. Artifact-free morphing capability. Well-written manual.

WEAKNESSES: Current version uses a full DSP chip per instance.

PRICE: $695, TDM; $295, RTAS/MAS **EQ FREE LIT. #:** 105

RealVerb is a new stereo reverb plug-in from Kind Of Loud Technologies, makers of the SmartPan Pro and RealVerb 5.1 surround plug-ins. With a flexible graphic interface allowing such options as real-time morphing of room shapes and sizes, 36 different surface materials for the rooms, and a three-band equalizer, this is a powerful and efficient reverb design tool for music and post applications.

Users of RealVerb will need a Digidesign Pro Tools Mix or Mix Plus system running under Mac OS, a CD-ROM drive for installation, and a floppy drive for authorization. A well-written manual is included as a PDF file, which features an extensive and informative technical profile section. RealVerb can be inserted directly into audio tracks, or you can bus to it on a mono or stereo aux input.

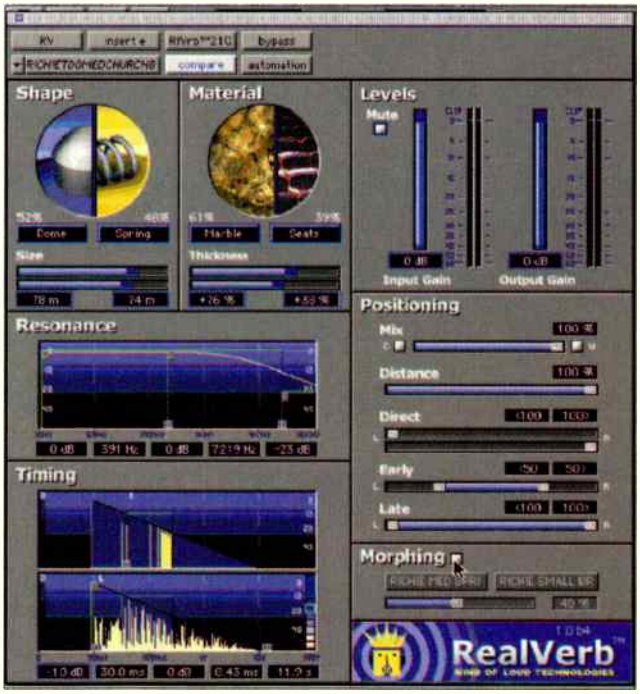
RealVerb lets you control the major characteristics of the room model, including room shape and size and surface material composition and thickness. Working with the Shape window, you access a pull-down menu allowing selection of one of 12 different shapes, including such choices as plates, springs, and classic rooms. You can set up two completely different rooms, then blend between them by sliding the vertical bar inside the graphic. Room sizes can be adjusted from 0.5 meters to 99 meters. The rooms, shape, size, and blend are used by RealVerb to compute the early reflection patterns, which are shown in the upper window of the Timing panel. Each early reflection is represented by a yellow vertical line, which gives you a visual representation of relative arrival times and levels.

The Materials window allows users to control the way different frequency components of the reverb decay over time, based on the material on the surface of the room. You're given a choice of 36 different material types. Examples include brick, hardwood, and marble, as well as artificial materials. RealVerb uses measurements of the absorptive properties of actual materials to model the reverb decay characteristics, and, as you would expect, a brick room sounds totally different than a marble room.

The effects of the material can actually be exaggerated, suppressed, or even inverted by using the Thickness sliders located underneath the graphic display. At the default thickness of 100 percent, the materials affect the reverb decay rates as they would in an actual room constructed of the chosen materials. As you increase the thickness, the frequency response is exaggerated, and as you decrease thickness, the response is suppressed. When choosing a negative thickness, the material is "inverted" — materials that normally absorb high frequencies quicker than low ones will do just the opposite. This may sound complex, but it's actually quite easy to use. I found it simple to quickly create and dial-in different room characteristics.

The Resonance section features a pre-reverberator three-band "paragraphic" equalizer, which can be used to shape the overall sound of the reverb. Values can be entered directly into the text boxes, or by dragging the handles on the graphic control. I found this useful for taming the highs of the brighter reverbs, and also for reducing excessive low frequencies.

Next is the Timing window, which allows control over the early reflections and late reverberation. The top, or Early Reflection window, provides control over the amplitude of the early reflections, along with delay between the arrival of the direct sound and the onset of those early reflections. The two text boxes at the bottom



of the Timing window can be used to enter in early reflection values, or you can simply drag the handles inside the graphic. The Late Reverberation window controls the decay of the reverb tail. You can modify the reverb time, reverb tail onset time, and tail amplitude. On the far right side of the Late Reverberation panel is the Diffusion control. Raising the diffusion value allows for a denser-sounding reverb. This feature came in handy when I needed to create a high-diffusion hall for a classical recording I was mixing.

The Morphing panel allows you to smoothly transition between any two presets, which are selected in the corresponding pull-down boxes. Using the slider, you can dynamically automate a morph between, say, a large concert hall and a small room — all with no zipper noise. When you're in morph mode (accessed by clicking on the morph box), all other controls are grayed out and cannot be used. Post mixers will love this feature, but I also used it to automate a real-time change on a snare drum from a small room to a large cathedral.

The Positioning panel includes the Mix slider, which controls the wet/dry mix. It should be set to 100 percent wet (far right) if RealVerb is being used on an aux input, and set for the desired wet/dry mix when RealVerb is used as a direct insert on an audio track. Below that is the Distance slider, used to control the distance of the perceived sound source from the listener. Also controlling the perceived positioning of the sound is the Direct panner, located below the Distance slider. This controls the panning width of the direct (dry) signal, with hard left and right representing the widest spread, and center representing a mono spread. Below that are the Early and Late panners, which control the spatial width of the reverb. The Levels control input and output gain, and feature stereo meters.

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REALVERB 5.1

Kind of Loud's RealVerb 5.1 ($1,495) for Pro Tools Mix TDM is a surround version of RealVerb — the first surround reverb to be released. It behaves similarly to the company's SmartPan Pro surround panning plug-in. There are two parts to the RealVerb 5.1 package: "return" plug-ins and the main plug-in, which serves as a graphic interface for controlling the reverb itself. Three RealVerb 5.1 return plug-ins are placed across three stereo aux inputs, creating a six-channel (L/R, Ls/Rs, C/S) output stem. The RealVerb 5.1 plug-in, like its stereo brother, can be inserted directly on an audio track, or on a mono or stereo aux input. As with SmartPan Pro, the bussing is handled "behind the scenes," returning the six outputs of the reverb back into the Pro Tools environment in stereo pairs.

RealVerb 5.1 features a similar layout to RealVerb with Shape and Material graphic displays, as well as the same Resonance and Timing sections. Also featured in this plug-in are six channel meters, which can be turned on or off, an output slider, and a stereo input meter.

The Spatial control section is quite different than RealVerb (stereo), and features four circular joystick displays that should look familiar to the users of SmartPan Pro. The Direct Path joystick control determines the panning of the dry (direct) sound, and the width of the sound around the speaker array is displayed in the perimeter of the circle. When RealVerb 5.1 is inserted on a stereo track, the graphic changes to a stereo display on the perimeter, thus allowing surround panning of the stereo signal. The Position joystick allows the user to control the distance and direction of the perceived sound, or source, from the listener. Below the Position panner is a Mix slider for control over the wet/dry mix.

The last two joystick panners control the early reflections and late reverberant fields. The width and position of these can be controlled by using the panner or by entering values into the text boxes. When mixing most material, I found it very effective to send the early reflections to the front speakers in a tight pattern, and to spread the late reverb out into the rears.

Like RealVerb, RealVerb 5.1 features great sound and amazing flexibility. If you're using your Pro Tools rig for surround mixing, you must check it out. *—Rich Tozzoli*

Ad Index *For fast and easy information use the* ***reader response card*** *in this issue*

BIAS Peak 2.1 Mac Digital Audio Editor

Still easy to use, the latest version of Peak takes the editor much further

BY CRAIG ANDERTON

Peak 1.0 put the Mac back on the map for digital audio editing after Alchemy disappeared, then Digidesign focused on Pro Tools instead of Sound Designer. Now, what was once a fairly basic sample editing tool with mastering options has evolved to the more sophisticated, and multimedia-oriented, Peak 2.1 (a free upgrade for v2.0.X owners).

INSTALLATION

Peak comes on CD; a floppy key disk registers/authorizes the program. Floppyless Macs can do the "challenge/response" shuffle (*i.e.*, when you boot the program, it generates a "challenge" phrase; submit this to BIAS, and they issue a "response" that unlocks the program).

The CD also includes Adaptec's Toast V3.5.6 along with program demos from Fireball, Qdesign, Groovemaker, and Waves, and demo sounds from Rarefaction and Sweetwater. There's also Acrobat-compatible documentation in French and Japanese, drivers for Digidesign, Lucid, Korg, and USB interfaces, demos of other BIAS programs, and QuickTime 3.0 and Sound Manager 3.3 (required for operation).

UPGRADE GOODIES

Probably the biggest change in v2.1 is support for ASIO 1 drivers. This circumvents Sound Manager's inability to handle bit resolutions or sample rates higher than standard CDs. With an appropriate sound card, Peak can not only record and play back 24/96 material, but supposedly allows sampling rates up to 10 MHz (!) at 32-bit resolution. (I don't have any cards I could use to test these capabilities, but, come to think of it, you probably don't either — so we'll take their word for it.)

ASIO worked fine with the SeaSound Solo, but there was some noise at 96 kHz. This seems related to the Solo Mac drivers (which require system 8.5.1 or higher: I'm running 8.1) rather than Peak. Other sample rates supported by the Solo worked fine.

There are some enhanced recording functions such as specifying a duration for recording (*e.g.*, record exactly two seconds of audio if you need something that will loop for exactly one measure at 120 BPM). You can insert new audio into a file at the insertion point, or append if the insertion point is at the end. Peak also handles dual mono (non-interleaved stereo) files a little better, as they can now be saved as stereo interleaved files.

When importing audio CD tracks, you can name them first. This makes it much easier when importing multiple tracks from an audio sampling CD.

There are many other minor enhancements, such as vertical zooming (and spacebar playback audition) in the loop tuner, a linear blend option for crossfades, the ability to export regions (not just files) through the batch processor and specify a suffix for batch-processed files, and SMPTE sync support.

EDITING AND PLUG-INS

Editing with Peak is fast and efficient, because you can edit while the audio plays, as well as do tape-style scrubbing and "dynamic scrubbing," where a piece of audio loops "in front of" the cursor. There are nine available dynamic scrub loop values, extending from 10 to 600 ms.

All edits are non-destructive until you save the file, and any undo/redo operations are stored in an event edit list. You can't undo just a particular step (except the last one), but you can go back in time to a particular point in the edit list, and carry on from there.

Peak includes the usual cut, copy, paste, delete, and fade in/fade out (with customizable curves) functions. But you can also crossfade between pieces of audio (again with a customizable curve), and there's a fine collection of DSP (see sidebar, "Peak's DSP Functions").

As for plug-ins, Peak unfortunately does not support the VST format, but does handle Adobe Premiere Audio plug-ins such as those from BIAS, Arboretum, Waves, DUY, and Intelligent Devices. [BIAS says that version 2.5, scheduled to be shipping by the time you read this, will have VST support.] You can record through these plug-ins, but the best feature is the ability to add an envelope that fades in the plug-in effect over time.

For Digidesign-compatible hardware, the DAE support that started with

FIGURE 1: The Loop Surfer feature makes it easy to check out how loops would sound at different places in a sample. Also note the cool-looking toolbar.

v2.0 allows using AudioSuite and TDM plug-ins, and offers 24-bit operation in a non-ASIO context. Adobe plug-ins work only under Sound Manager (preferably with ASIO Sound Manager); Digi-based setups should use the Digidesign Sound Manager driver if compatibility is a must.

When it comes to importing (and exporting) files, there's support for the "old school" standards (AIFF, WAV, QuickTime, Red Book audio, AU, SND, System 7 sounds, SDII, and dual-mono Pro Tools-type files), but Peak now supports PAF (Ensoniq PARIS cross-platform format), RealAudio 5.0, MP3, and Shockwave Internet sound files. You can also compress AIFF or QuickTime files with algorithms such as MACE 3:1, MACE 6:1, µ-Law 2:1, Qualcomm PureVoice, Alaw 2:1, Qdesign Music, IMA 4:1, and others.

EQ LAB REPORT

MANUFACTURER: BIAS, 1370 Industrial Ave., Suite A, Petaluma, CA 94952. Tel: 707-782-1866. Web: www.bias-inc.com.

APPLICATION: Digital audio editor for samplers and mastering, with the ability to burn CDs and save in Web-friendly formats.

SUMMARY: The latest version manages to enhance the original program, while retaining its ease of use.

MINIMUM SYSTEM REQUIREMENTS: Power PC, System 7.6.1 or later (8.5 recommended), QuickTime 3.0 and Apple Sound Manager 3.3 (provided), 64 MB RAM minimum (add 10 MB for DAE support), 5 MB available hard disk space for program and online help. (Tested with PowerComputing 604e/200 MHz processor and SeaSound Solo interface.)

STRENGTHS: Simple to learn and use. Excellent sampler support and looping tools. Runs fast, even on non-"G" machines. Supports Premiere, AudioSuite, and (with appropriate hardware) TDM plug-ins. CD-burning via Adaptec Toast. Batch processor. Exports in multiple formats. Supports ASIO. Imports QuickTime videos for audio editing. Imports Red Book audio. Good scrubbing.

WEAKNESSES: Doesn't accept VST plug-ins yet. No spectrum analysis view, signal synthesis, or signal analysis tools. Operates on single-file rather than project basis. Lacks traditional mastering DSP (equalization, dynamics).

PRICE: $499 **EQ FREE LIT. #:** 125

SAMPLER TIME

Peak's handling of sampling and loop-oriented applications is outstanding. For example, with crossfade looping, you can call up any of nine "blend" envelopes, or edit your own. This is particularly wonderful when what you really want is a butt splice, but need just a bit of fade time to smooth out the transition. A loop tuner shows how the loop end segues into the loop beginning and can adjust this, but my favorite loop-oriented function is Loop Surfer (fig. 1). This lets you specify a particular tempo and number of beats; Peak then automatically sets up a loop based on where you specified the loop start. (It can also calculate tempo if you click/drag to specify a range and enter the number of beats this represents, as well as "guess" the tempo based on observing the relationship of amplitude to a user-settable threshold.) After initiating Loop Surfer, you can still adjust the start/end points, but, more interestingly, you can move them in tandem. This allows trying out different loop placements, while retaining the same number of beats.

I checked out SMDI transfers with the Ensoniq ASR-X, and it generally had no problem getting samples to and from the target. Furthermore, the manual does an excellent job of explaining the intricacies of SCSI sample transfers, including proper power-on order for particular samplers. As far as I'm concerned, Peak is at the top of the heap for dealing with samplers.

VARIATIONS ON A THEME

Peak comes in many permutations (see table). While not included in the table, "lite" versions are available for as little as $99. Of particular interest are the

PEAK'S DSP FUNCTIONS

Add: Mixes clipboard contents into audio at the insertion point; clipping is possible.
Amplitude Fit: A loudness maximization-type algorithm.
Change Duration: High-quality time compress/expand without pitch shift.
Change Gain: Includes a defeatable "clip guard" function.
Change Pitch: Pitch up/down, with or without duration change — it even sounds good.
Convert Sample Rate: Converts rate without changing pitch.
Convolution: Multiplies the selected audio by an impulse stored in the clipboard; takes a while and uses up lots of RAM if the impulse is big, but can give really otherworldly sounds.
Crossfade Loop: See the review section "Sampler Time."
Invert: Changes signal polarity.
Fade In or Fade Out: Either one lets you edit the fade curve.
Gain Envelope: Apply an envelope that can attenuate and/or boost level.
Find Peak: Places the insertion point at the highest signal level.
Loop Tuner: See the review section "Sampler Time."
Mix: Mixes clipboard contents into audio at the insertion point; automatic scaling avoids clipping.
Modulate: Provides ring modulation effects.
Normalize: Boosts signal level so that peaks use up the maximum available dynamic range.
Panner: Rubber-band envelope panning function.
Phase Vocoder: Takes even more time than convolve, but allows modifying pitch and duration simultaneously — only problem is it's not very clean.
Rappify: Lo-fi effect that can make percussive material sound more "spiky."
Repair Clicks: Automates the process of finding and removing clicks; works well with obvious clicks, but subtle clicks require drawing out with the pencil.
Remove DC Offset: Makes sure zero-crossings are truly at zero.
Reverse Boomerang: Reverses the audio and mixes in a user-settable amount with the original.
Reverse: Your basic backwards Satanic-message generator.
Threshold: Splits up a file into smaller segments; when applied to drum loops, for example, you can sequence individual beats via MIDI to change tempo without affecting fidelity.

WHAT YOU GET

Product includes	Peak 2.1	Peak 2.1 TDM	Peak 2.1 SFX Machine Bundle	BIAS Studio	Peak 2.1 PowerBook Edition	BIAS Studio PowerBook Edition
Peak 2.1	•		•	•	•	•
Peak 2.1 TDM		•				
SFX Machine			•	•		•
SFX Machine Lite	•	•			•	
Deck				•		•
Adaptec Toast	•	•		•	•	•
Digigram PCMCIA card					•	•
Price	$499	$699	$599	$749	$999	$1,179

PowerBook bundles for portable recording. They are based around the Digigram VX-pocket PCMCIA card, which provides stereo analog and digital I/O with 24-bit resolution. Throw in BIAS Deck, and you can do multitrack recording as well as portable field recording.

PEAK VERSUS SPARK

Although both products are in the same general category, each has its own bias (no pun intended). TC Works' Spark is outstanding for mastering, but at the expense of high-end sample editing functions. Peak's sampler support is superior, and its looping options — coupled with the fast undos for destructive edits and available optional set of weird-ass plug-ins — make it the clear choice for those who are more into sample editing and mutation. But these divisions are somewhat artificial; thanks to Spark's brilliant FX Machine matrix you can do some pretty severe warping with suitable plug-ins, and Peak can do some tasty mastering.

Although both Peak and Spark cost $499, Spark includes a batch of plug-ins; add the SFX Machine to Peak, and you're up to $599 — but you get a ton of creative plug-ins that go way beyond the normal compressor/EQ/loudness optimizer type of routines. Where Peak will top out is with plug-in synths, which the Premiere format isn't designed to handle; TC Works is already showing synth modules for Spark's effects matrix. Their advantage may not last long, however, as VST support is slated for Peak release 2.5.

Peak isn't just a cool program written for the Mac; it's a truly Macintosh program that has come a long way since its intro. And if past history is any indication, it will go even further in the future. EQ

SUPPORTED SAMPLERS

The following supported samplers have customized dialog boxes. There are also generic SDS and SMDI dialog boxes for other samplers.

Akai S1000/1100/2000/2800/3000/3000/3200/3200, CD3000 (including XL versions)
Emu E-IV, ESI-32, ESI4000, E-64
Ensoniq ASR-10, EPS 16+, ASR-X
Kurzweil K2000/2500
Peavey SP/SX
Roland S-760
Yamaha A3000

Arboretum Systems Ionizer
Mac Audio Processing Plug-In

An affordable and effective way to rid unwanted noise from your tracks

BY MIKAIL GRAHAM

What do you do when a track has too much noise due to poor recording techniques? How about when working with tracks recorded on older analog tape-based systems? Or tracks with AC hum, or excessive background noise?

Arboretum System's Ionizer offers an affordable and effective means to correct almost any noise or hum issue you might come across and much more. But that's not all; in version 1.3 you also get a 512-band vocoder and a frequency morph feature that allows you to capture the EQ characteristics of any file. Other functions include upward compression, limiting, expansion, pop/click removal, frequency morphing, and spectral keying, to name just a few. Ionizer is available for Premiere, MAS, and AudioSuite formats, or it can run stand-alone within Arboretum's free HyperEngine.

I tested version 1.3 of Ionizer on a Mac 9600/300 with 256 MB of RAM running OS 8.6. I used HyperEngine, BIAS Peak 2.1, Digidesign's Pro Tools 4.3.2 and 5.0, and also Emagic's Logic Audio 4.1 as my host programs. Happily, I have no glitches to report using any of the programs.

CURVES & ZONES

To say that this plug-in is deep would be putting it mildly. This is not to say that Ionizer is hard to use once you grasp the concepts. Ionizer has a graphic "green grid" display, which shows red, blue, and black lines, or, as Arboretum calls them, "curves." Understanding these curves is critical to understanding Ionizer.

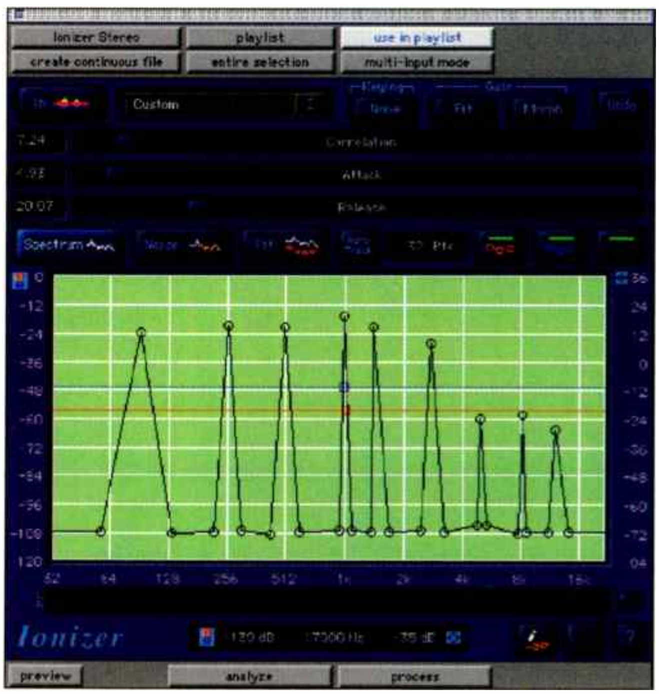
Ionizer's 512 bands of gated EQ each cover a range of 43 Hz, and are manipulated with a series of "zones" determined by the positions of the Red, Blue, and Black Curves. The Red Curve sets the threshold of the 512 EQ gates. The gates can be set to activate via a positive or negative signal level compared to the Red Curve setting. This is determined by the position of the Blue Curve, which also defines the range of the active zone. If the Blue Curve is above the Red Curve, a gate will become active when its band's energy exceeds the Red Curve. If the Blue Curve is below the Red Curve, a gate will become active when its band's energy drops below the current Red Curve threshold setting.

Last, we have the Black Curve. This determines the overall gain that will be applied to a band when an audio file's energy level falls within the "gate transition zone" — the area between the Red and Blue Curve settings. The important thing to remember here is that the Black Curve's gain only takes partial effect when a band's energy level is between the Red and Blue Curves. The Black Curve's gain takes full effect when a band's energy goes beyond the Blue Curve, into the "active" zone.

Just remember that the Red and Blue curves are thresholds for where processing is active. The Black Curve determines the amount of cut or boost that's applied to each of the 512 bands. Ionizer's ability to do its job, as well as what job it does, comes down to how you set these three curves.

TOURING THE GUI

Ionizer's window sports a number of interesting features: The Keying button lets you extract both the spectral and dynamic content from either the left or right channel of a stereo audio file and use this data to process the other channel. Very useful for locking, say, the attack of a bass line to the pulse of a hi-hat or even more impressive as a real-time vocoder. The only caveats here are that both the source and destination or "target" audio must be within the same stereo file, and the fact that the final output must be a mono file. Being able to use a separate audio file for keying would be a nice addition.



Next we have the Spectrum, Noise, Fit, AutoTrack, and Fit Points buttons. Spectrum performs a spectral analysis on the currently selected audio file and plots a "frequency profile" envelope that the Fit button uses to automatically calculate Red and Blue Curves.

The Noise function looks across the current audio file and tries to make an "educated guess" about where it thinks there's noise. When it's finished, it displays what it thinks is the frequency profile of the noise. This function certainly does make the process easier than having to find the noise yourself. Of note, this algorithm is the same one that makes Arboretum's Ray Gun so easy to use. (See the Ray Gun review in the April '00 issue.)

The AutoTrack function follows the noise level in your signal, raising or lowering thresholds to match the changing amplitude of the noise signature. Arboretum says this results in a smoother, more pleasing-sounding noise reduction, and I'd have to agree, although I did find that AutoTrack could sometimes cause all audio to mute, depending on how the other controls were set.

EQ LAB REPORT

MANUFACTURER: Arboretum Systems Inc., 75 Aura Vista, Pacifica, CA 94044. Tel: 650-738-4750. Web: www.arboretum.com.

SUMMARY: An effective and flexible noise reduction, EQ, and audio processing plug-in.

SYSTEM REQUIREMENTS: 120 MHz or faster PowerPC with Mac OS 7.6 or higher and 10 MB of free RAM minimum. Premiere-, MAS-, or AudioSuite-compatible host application.

STRENGTHS: Diverse palette of audio tools perfect for complicated noise reduction, EQ, vocoding, limiting, and expansion. Frequency morphing.

WEAKNESSES: Bit of a learning curve. Keying only works within a stereo file, can't be applied to a separate file.

PRICE: $399 **EQ FREE LIT. #:** 106

Located to the right of the AutoTrack button is the Fit Points setting that determines how many individual breakpoints for the Red and Blue Curves will be created when you click the Fit button. The default setting is 32 with a maximum number of 512 and minimum of one point. This parameter takes a bit of tweaking: too many points and things can get complicated, too few and you may not have enough "resolution" to make the desired adjustments.

The Gain Fit and Morph buttons enable the remapping of a sound's frequency profile. Gain Fit works much like the Fit feature, except that it conforms the Black Curve to the contours of the current frequency profile. Gain Fit is a great way to create customized gain curves, which brings us to the Morph button. Morph lets you map the spectral profile of one piece of audio onto another. Arboretum calls this process "Frequency Morphing."

Sliders for Correlation, Attack, and Release are our next stop. The Correlation slider controls the amount that adjacent bands will follow each other. The Attack slider controls how fast the bands become active, and the Release slider determines how fast the band gates close.

The Pencil tool is used to make adjustments to or add/delete points to the various Curves, and the Zoom tool allows for magnifying a range of these points. A cool feature of the Zoom functions is that you can use the Command key while dragging the mouse to fine tune the area you're selecting. Another Zoom tool plus: You can select a specific section of the displayed frequency profile, and then use the Fit button to create points only within this area, rather than for the entire frequency range.

The Mood Bar gives a real-time graphic overview of Ionizer's current gate and gain activity. Red indicates a no gain state, as all energy is below the threshold curve (passive zone). Purple indicates that some gain is being applied showing that the energy is in between the Red and Blue Curves (gate transition zone). Finally, blue indicates full gain is being applied to a band and that it's in the active zone.

THE FINAL VERDICT

In the course of my Ionizer tests, I worked with various cassette and DAT mixes of bands I have produced or played with over the last 25 years or so to see how I might improve upon their vintage status (oops, age-watch alert!). In all cases I was able to reduce both tape hiss as well as add more clarity and punch to the mixes. Throughout my time with the program, I found it to be stable, to be flexible, and to offer some great creative possibilities.

True, there is a learning curve. But when it comes to manuals, Arboretum has some of the best that I've seen. Even if you already know a lot about the various processes, there's a lot of useful background information you can learn from the manual. Before you purchase, you can check out the manual at Arboretum's Web site.

There are few competitors on the market that can truly rival Ionizer's overall list of features. And having worked with noise-reduction software since it first hit the market, I can honestly say that, for me, Ionizer is definitely the new king of Mac-based noise-reduction and hum-removal processors.

Mikail Graham currently finds himself intensely studying the aftereffects of overexposure to transcontinental travel and how it may or may not interfere with his ever-expanding DSP addictions.

USER TIPS: PUTTING IONIZER TO WORK

Ionizer can be used to create a variety of processors, including an Expander.

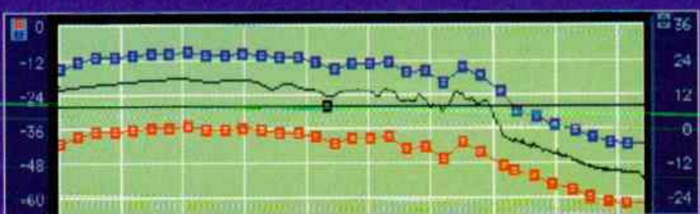
Graphic EQ: Making a graphic equalizer in Ionizer is as easy as deciding what frequencies you want to control and clicking on the Black Curve to add in the desired adjustment points.

Upward Compressor: To make an upward compressor, simply select the audio, click the Spectrum button, then the Fit button, be sure the Red Curve is above the Blue Curve, and then move the Black Curve into a +dB area of the grid.

Limiting: A limiter can be made by reversing the upward compressor settings so that the Blue Curve is above the Red, and then using the Black Curve to control the amount of attenuation.

Notch Filtering: To make a notch filter, start by disabling the Red and Blue Curves, set the Black Curve to a value of 0 dB, use the pencil tool to set the frequency to whatever value you need; then just click and drag this point downward.

Vocoding: Ionizer can function as a 512-band vocoder by using the Keying function. A cool feature: You can choose which channel is processed by toggling the Keying button to its Left to Right or Right to Left setting. By adjusting the Threshold and Gain curves and the Correlation, Attack, and Release sliders, you can create an almost limitless range of vocoding effects.



Guitars and Geeks


PHOTO BY STEVE JENNINGS

Two approaches to a new relationship

BY JON LUINI AND ALLEN WHITMAN

Established artists are finally getting a handle on the true interactive nature of the Internet. A look at the latest version of Joe Satriani's Web site provides a good example. (Disclaimer: This section about the Satriani site is written by FezGuy #1 because the other FezGuy #1 had a hand in the site's development.) This is the second major overhaul of the popular guitar player's site and features new ways for fans to contribute feedback (including what songs fans want to hear in Mr. Satriani's live sets), share information, and learn about the artist.

The opening splash page is simple and loads quickly. Mr. Satriani gets familiar with his people through a streamed welcome message in RealVideo (at multiple bitrates) or downloadable as a QuickTime movie. There's a listing for the next live show (updated every day) that links to the complete itinerary for the current tour. On the itinerary, links lead to fan comments about each specific show. Even shows in the future have fans chatting back and forth about hooking up at the venue, creating a pleasing sense of sociability. Some shows have photographs. Subscribed members can mark their attendance at a show, entering them in a random drawing for after-show passes.

The community has a host of ways for fans to interact. A "lookup" function allows fans access to other specific fan nicknames' real names, locations, and e-mail (this can be limited if the user has chosen not to share that info). There's a well-organized contest, the winner of which was announced during a QuickTime Webcast of a live show in Seattle. Using his resources within the industry to benefit his fans, Mr. Satriani's sponsors (Ibanez Guitars, Guitar.com, Epic Records, and Apple Computer) award one winner a guitar, an amp, a complete videography, and discography of Joe's material and, to top it all off, transportation from anywhere in the U.S. for a guitar lesson with Joe. What fan could resist?

You must become a member to get full use of site (only registered members can post to the comments page or talk in the chat room) but, thankfully, the sign-up process is very simple. Unlike a typical corporate Web site, only a nickname and valid e-mail address are required. The e-mail address is necessary so the site can automatically return a password. Unwanted spam can be de-optioned by not signing up for the mailing list. The site doesn't require a phone number, address, or your real name. That's refreshing.

The site also features streaming and downloadable music and video, tons of information (including an electronic press kit or "EPK"), and many photographs. Really, this site has to be seen to be believed. Almost every major-label owned band Web site designer could benefit by spending time studying the methods of this artist's online presence. Let's face it: Joe really groks the Web. And, of course, he groks his fans, too, knowing their supreme importance to his life and work.

GOOD COP/BAD COP

Speaking of using the Internet to connect with your fans, Metallica made a big splash recently by attempting to pull the plug on unauthorized online sharing of their music. Hundreds of thousands of Metallica fans have been using Napster (see last month's FezGuys column) technology to trade MP3 files of the band's songs. The band wants to put a stop to that, citing the reasonable position of: it's okay to share, but the band should decide what and to whom. It makes sense, but Lars Ulrich, the drummer, takes it into the realm of the absurd by saying he doesn't like to see Metallica songs traded as a "commodity." Gosh, Lars, what do you call the sale of compact discs? Thankfully, Metallica hasn't sunk too deep into the typical 100 percent anti-copying stance of the corporate music industry. The band points out that it's only their released album material they are concerned about, not the bootlegs, interviews, and other related underground materials. We respect their desire to be in control of how the music from their recorded masters is sold. Any musician who isn't concerned about such issues will be hard-pressed to make a living.

The wild popularity of Napster begs the question: How totally will the existing

IUMA'S MUSIC-O-MANIA

The scruffy indie music Web site washed behind its ears and reached a moment of clarity last month. Rock headliner Primus played at the Fillmore Auditorium in San Francisco, CA, with four IUMA-sponsored college battle-of-the-bands finalists opening. As part of the eligibility, each band had to have at least one member enrolled as a college-level student.

This is a fine example of the tangible results Web companies are producing these days for independent musicians. The finalists were undoubtedly excited to be at a venue with as much rock 'n' roll history as the Fillmore. And any rock music devotee can tell you the value in getting your music introduced to fans of a band like Primus. And, hey, $10,000, seven days in a recording studio, a 5,000-CD pressing, and promotion on IUMA and EMusic's Web sites for the winner ain't exactly chump change!

The event Webcast is archived at **www.musicomania2000.com**.

commerce structure of the music industry get hammered? Some Napster supporters say music should always be free to trade and artists should make their money through live performance and other merchandise sales. Others claim that free trade of music (bootlegs or released recordings) only increases overall sales for the artist in the long-term because true fans will patronize the band. The FezGuys feel that this is beside the point.

Technology is merely a tool. This particular technology is about moving bits around, and it should serve the needs of both those wanting the bits and those making the bits. It should be easy for fans who want to support a band to support them. Napster could benefit everybody by re-inventing itself to serve the needs of artists making reasonable decisions about the distribution of their music. Perhaps an artist wants to release one song within the context of a set of other related songs. Maybe they want to charge $10 for the whole thing or give it away for free. It's the artist's choice.

Interestingly, the Grateful Dead, forever known as a shining example of the positive effect of sanctioned live bootleg trading, doesn't allow their released recordings to be traded. It's interesting that Metallica and the Grateful Dead are now in the same camp.

In a recent staged media event, Metallica's drummer and lawyer arrived at the office of Napster, in San Mateo, CA, to deliver a whopping 60,000-page printout containing over 350,000 user names purported to be actively trading the band's music using the Napster network. Though the music *per se* doesn't live on the Napster servers, the band wants Napster held responsible. During a mediated online chat with band members (www.metallica.com/news/2000/000503.htm), it was suggested by a fan that criminal charges for "aiding and abetting piracy" would be more appropriate than a lawsuit.

So what does the lawsuit specifically say? Filed in the middle of April, the suit claims that Napster has violated three different areas: copyright infringements, unlawful use of a digital audio interface device, and the Racketeering Influenced & Corrupt Organizations Act (RICO). Specifically, it alleges Napster has infringed by "encouraging and enabling visitors to its Web site to unlawfully exchange with others copyrighted songs and sound recordings without the knowledge or permission of Metallica." In a related lawsuit, popular rapper Dr. Dre seeks damages from Napster amounting to $100,000 per illegally copied song. It's not clear if the same holds true for Metallica's suit.

The situation is complex because it wraps multiple issues into one big confusing ball. One thing the lawsuits are *not* about is MP3 as a format. Rather, the focus is on how MP3 is used (similar to piracy not being about CDs, but about how they are illegally trafficked). At the heart of this tempest in a teapot is frustration and fear over the increasingly easy methods people use

continued on page 134

MY.MP3.COM SLAPPED DOWN

The RIAA has been granted summary judgement in its suit against My.MP3.com (the service allowing users online access to MP3 files of CDs they already own). Though the judgement doesn't require MP3.com to shut down the My.MP3.com section, it has kicked the company into actively talking with major labels about an agreeable solution. That's a step beyond MP3.com's stonewall tactic of the last couple of years.

RESTORATION

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the mic, or the guy doing the bootlegging fell down drunk or ducked into the Sanican. The bootleg may be noisy or distorted, but it has a spatial dimension the board recording doesn't have.

Singleton uses Cedar to clean up tapes before he does anything else to them. "If you've got a cassette that's very hissy, and you can find a silent bit, one that's got no music at all, just the hiss, you can take a sample of that and you can say, 'I don't like that, take it away.' " Then he'll add compression and reverb to the bootleg, put it in another track, and use it as ambience. The difference between the three versions — the board mix, the bootleg, and the Singleton recombinant — is quite remarkable.

He also shows us a four-CD set of bootleg recordings that showed up for sale in Japan, attractively packaged and sporting the credit, "Produced by Tommy." It features shows King Crimson did as the opening act for the Rolling Stones in 1969. Included is one concert where the two bands played to a crowd of 500,000 in London's Hyde Park, a show for which Robert Fripp has had the video footage for years. Until now, it's been nothing more than a silent film. Laughing, Singleton expresses his gratitude to the felonious Tommy: "It was really very kind of him. Now I can put the audio back on my video. They're only very slightly out of phase." As much as he enjoys the electronic jigsaw puzzles he's assembling from bits and pieces of the aural past, David Singleton equally enjoys the double-reverse Robin Hood spin of beating the bootleggers at their own game. EQ

GLENN KOLOTKIN

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hour and it turned out to be the mix we used for "Time Has Come Today" by the Chambers Brothers. Less is more!

Is there a way to know when you've hit the point of no return on a mix? Can you know when you're at the peak, where you can work another day and not get it any better?

Always. When I feel I've really got it, I'll put it right down on tape. Usually there's other people involved and everyone wants to try something. I don't have any problem with trying other things, as long as I've recorded the one I felt comfortable with. If we can beat that, then fine, I'm in favor of it. A mix changes very gradually on each pass...it's almost unnoticeable. Over the course of several hours, it changes a lot and you can lose it. But if you refer to that first mix, a lot of times that's it. I think every song has a natural balance that's easy enough to get to, but people think that if it's mixed too quickly then it can't be good. That's not true. The only way to prove it is to play the earlier mixes and let your ears judge.

What song or songs did you know were hits when you were working on them?

"Stoney End" by Barbra Streisand. Right out of the box. We recorded her live but, after the dust settled, she wanted to make changes, so we overdubbed. She's great! The first time you open up the microphone — there it is: Barbra Streisand. Her sound is wonderful. Maybe put on a bit of echo and that's it. She might want to do it over for performance reasons or things I might not hear. Her pitch is perfect and she's very exacting. It's a pleasure working with her because she's a real talent. You need to use a good-quality microphone because she has such a quality voice. I think we used an [Neumann] M 49.

What's next for you?

Right now, I'm working with an artist that my managers Kenny Ketcham and Marc Steinberg of Ketcham and Steinberg Entertainment discovered — an artist called Mixx. This guy is a talented singer and guitar player who writes really good songs. We're looking to place him with a label, and I hope that will be my next project. He looks like a superstar and I'm really excited. I plan to be involved as producer and engineer.

Is that your preferred role — producer and engineer?

Definitely. It's really interesting. I won't mention the name in this case, but I was working with someone as a co-producer for a period of time. At one point he came to me and said, "Let me ask you a question: Why is it that *your* records sound better than *our* records?" It's because my records are made the way I like them, but *our* records are a combination of what we both think. If we're co-producing, then naturally we're going to try ideas from either of us. So he said, "Okay, from now on, you take care of the sound and I'll go into the studio with the band and work with the arrangements." That's what we did from that point on. But I got into the business because I wanted to make records sound the way I heard them. Most engineers don't get that chance because there are so many other people involved. If I had a choice, I would produce and engineer on all my projects. EQ

FEZGUYS

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to pirate music. Napster (to quote a link on their Web site about the legal issues) is "Ground Zero" for this hot debate.

Bottom line? The FezGuys feel (as usual!) that litigation isn't the only answer. We hope the press fallout (which has been considerable) will raise awareness of the real issues and set the stage for musicians to continue to make a living in this emerging landscape. Piracy will always exist in some form. Lawyers don't have to worry, they will always have work.

LET'S GET PHILOSOPHICAL, BABY!

All this brouhaha about litigation actually points to a much deeper issue: decentralization and control of intellectual property (*i.e.*, your music).

Typically, thousands or even millions of fans flock to a handful of large Web sites for music. Napster and Gnutella (see last month's FezGuys column) eliminate the need for this giant central location by allowing fans to become small servers, trading with each other. Any fan can call up the availability and location of another fan's music files and swap. *Voila!* An authentic network is born. The Internet actually acts like a web.

Now imagine a near future where home connectivity continues to expand. Almost everyone has broadband access. Data flows from one place to another, and no one knows where it resides at any one time. When you want it, it can be found. Home users become common carriers (or small servers) and can't be sued for content they aren't even aware they have (and may not have moments later). How, in the name of all that's equitable, do you create a practical commerce model out of a giant beehive where all the bees make their own choices?

The FezGuys hold these axioms to be self-evident:

1. Musicians deserve to get paid for their work.
2. Musicians have the right to give their work away.
3. There will always be piracy and theft.

Technology's role is to make the first two possible. The user's role is to use it in a responsible fashion. This is better achieved through education and collaboration than litigation. Perhaps Napster should stop playing the role of middleman. Perhaps it should get involved in monitoring its traffic and facilitating the sale of protected works. Perhaps successful musicians can set a positive example by adopting new technology, leading the way in educating fans, and acting as positive role models. Let's not leave it to the lawyers! EQ


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STEINBERG REVIEW

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you can still enjoy the kits that are included with LM-4; hopefully others will be released in the future. The first of these, BitBeats's *XXL Compilation*, began shipping just as this issue went to press.

MODEL-E

If the screenshot of Model-E looks vaguely (or more than vaguely) familiar, there's a reason: The look of the plug-in synth is more than just a bit reminiscent of the classic Minimoog synth, although the plug-in brings things squarely into this century with features including up to 64-note polyphony, 16-part multitimbral operation, storable presets, MIDI control, and more. Still, you can't deny the similarities: the "analog" sound, three oscillators with one usable as an LFO, the "voltage-controlled" filter, dual ADSR's, and so on.

Like its cohort, the LM-4 drum machine, Model-E is a VST 2.0-compatible plug-in virtual instrument. You can open multiple instances in your host program; each can be multitimbral. As with LM-4, the audio output of Model-E shows up in the host's audio mixer and can then be processed with plug-ins, EQ'd, etc. Four sets of stereo audio outs are available for bringing multiple sounds into Cubase's mixer, if you desire. One caution: As with LM-4, in Logic Audio you can only access the first set of stereo outs.

As mentioned above, three oscillators are included, plus a noise generator (white or pink noise). Each oscillator offers a choice of six waveforms tuned to one of six octaves. Oscillators 2 and 3 can be detuned from Oscillator 1 for fatter sounds, and Oscillator 3 can be used as an LFO to modulate targets such as filter cut-off. A basic mixer is provided for combining the oscillators; its output is routed to the input of the resonant low-pass filter (two-pole or four-pole slope), which can be controlled with an ADSR envelope.

After leaving the filter, the signal enters a VCA (Voltage Controlled Amplifier, although in this case, the "voltage" is virtual) controlled by another ADSR envelope. You're given control over the Attack, Decay, and Sustain portions of the envelopes, but not Release, which can only be switched on and off — the release time setting matches the Decay setting. In addition, you can switch a channel of Model-E to monophonic operation, add polyphonic glide (portamento or glissando) between notes, and use MIDI note numbers to control modulation and filter modulation amounts. Extra controls below the main window can scale the effects of MIDI on the various parameters as well as "spread" or distribute the audio output across the stereo spectrum.

In practice, Model-E is simple to program; just twiddle the knobs until you're happy with the results. If you've used an analog synth in the past, it will feel very familiar to you. Besides being able to store and recall banks of 128 programs, you can use MIDI to automate parameters, either through sys-ex or using MIDI continuous controller messages. The installer disc includes five banks of preprogrammed sounds, although the banks aren't full of patches. Still, plenty of programs are provided either for use as is, or as a jumping-off point for creating your own sounds.

In use, I found Model-E to be very satisfying. I experienced no latency problems playing the plug-in from my MIDI controller through either my Mac's built-in Sound Manager outputs or when routing the output through my Pro Tools system using Direct I/O. Model-E's sound is fat and round. Does it sound like a Minimoog? Pretty close in some cases, maybe not as warm or as "deep" — but I've had similar complaints about most digital recreations of analog synths I've used, including those provided by expensive hardware models. True vintage freaks aren't going to be happy regardless — only the real thing will be good enough. But if what you're looking for is a fat, polyphonic, easy-to-program analog-style synth that can live in your computer and come up as a plug-in in Cubase or Logic Audio, Model-E definitely delivers the goods, especially given its $199 list price. EQ

TC ELECTRONIC REVIEW

continued from page 108

damping. Damping refers to how far down the signal is pushed when it's ducked, so you can have the delay ducked drastically for weird effects or just a few dB for a subtle reduction of the delay. It works, it sounds great, and it's easy to use.

Ditto for the filter and ping-pong effects. Since the filter function offers high-cut and low-cut for both the source signal and the delayed signal, a wide variety of timbres are available. Two front-panel bar graphs display the contour of the filters, and, by using the feedback filters, it's possible to get a reasonably good simulation of tape echo (check out preset #9, "Tape Echo"). While ping-pong does what you'd expect (pans the delays between left and right), there are a few twists due to the fact that three variations are provided. L-R mode pans the signal hard left and hard right, alternating repeats between speakers. L-C-R pans the signal from left to center to right. Dynamic adjusts the number of pan positions to correspond with the number of repeats. If you have seven repeats, the D•Two automatically assigns each repeat to a unique pan position. Very interesting indeed!

Some really bizarre effects can be achieved using the reverse function. Not content to just be able to produce backwards echo, TC Electronic has supplied several different styles of reverse. Though I could spend pages examining the reverse possibilities, some of the things that can be accomplished are playing all taps in reverse, playing the first or second tap in reverse while the rest are played forward, playing the last tap in reverse and the others forward, playing the odd-numbered taps in reverse, or playing the even-numbered taps in reverse (got that?!).

Our initial reaction to the "chorus" function was that it wasn't doing anything. I was expecting to hear that "traditional" chorus sound, but the "chorus" function adds chorusing to the repeats — not the dry signal. (Of course, you can also set up the unit to produce that great TC chorus sound.) "Spatial" employs a psychoacoustic effect to widen the stereo image; this was our least-favorite part of the D•Two, but it could be useful in certain applications where you want to place an instrument in a room without blatantly using delay or reverb.

D•TWO CONCLUSIONS

In producing the D•Two, TC Electronic has turned what could have been "just another delay" into a unique product that provides a lot of interesting options. With typical delay units, you'd have to add additional equipment (such as a compressor) to achieve some of these effects, but the D•Two has what's needed on board. The additional effects mean that you're not likely to get bored using the D•Two, and you might even find that it inspires sonic experimentation. The audio output of the unit is very clean and quiet. By using the filters, the timbral quality of the delayed signal can be tailored to your needs, ranging from ratty and muffled to sparkling-clear. Another winner from TC Electronic. Another one they're going to have to come and pull from my rack! EQ

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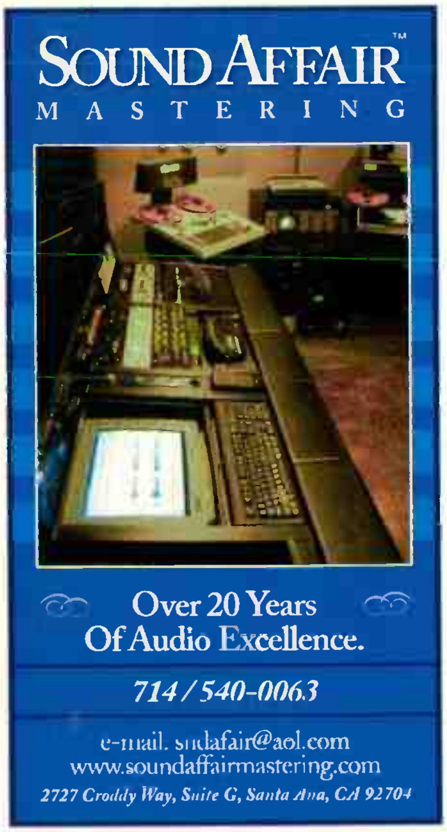
CIRCLE 67 ON INFO CARD

CIRCLE 70 ON INFO CARD



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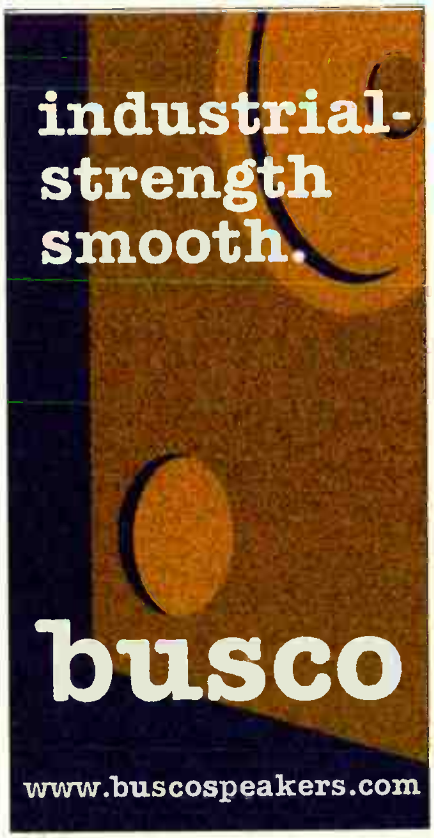

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HARD DISK RECORDING

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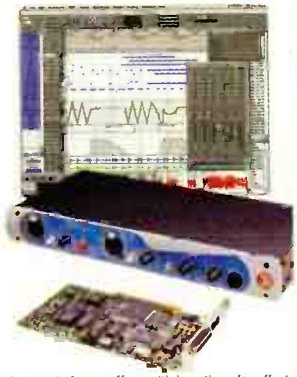
DIGI001 Digital Audio Workstation For Mac And PC



A completely integrated digital recording, mixing and editing environment for the Mac and PC, the DIGI-001 offers a 24-bit multi I/O breakout interface along with Pro Tools LE software— based on Digidesign's world renowned ProTools software. The DIGI-001 interface features 18 simultaneous I/Ds made up of 8 analog inputs and outputs— two of the inputs are full featured mic preamps with phantom power, and digital I/O including standard S/PDIF as well as an ADAT optical interface that can also be used as a S/PDIF I/D. ProTools LE supports 24 tracks of 16 or 24-bit audio and 128 MIDI tracks and also features RealTime AudioSuite (RTAS) effects plug-ins. For ease of use, MIDI and audio are editable within the same environment and all mixing parameters including effects processing can be fully automated.

FEATURES–
- 18 simultaneous, 24-bit ins and outs with support for 44.1 and 48 kHz sample rates
- 20Hz - 22kHz freq. response ± 0.3 dB
- 2 channel. XLR mic/1/4" line inputs with -26 dB pad, 48v phantom power, gain knob, and HP Filter at 60Hz
- 6 ch. line inputs (1/4") TRS balanced/ unbalanced w/ software controlled gain
- +4dB balanced 1/4-inch Main outputs
- Balanced 1/4" monitor outs with front panel gain knob
- 1/4-inch unbalanced line outputs channels 3-8
- Headphone output with independent gain control knob
- 2 channel S/PDIF coaxial digital I/O
- 8 channel ADAT optical I/O can also be used as 2 channel optical S/PDIF

Pro Tools LE
- Supports 24 tracks of 16 or 24-bit audio and 128 sequenced MIDI tracks
- Sample-accurate simultaneous editing of audio & MIDI
- Real-time digital mixing capabilities include recall of all mixing parameters, support for edit and mix groups and complete automation of all volume, panning, mutes and plug-ins.
- Route and mix outboard gear in realtime
- MP3 and RealAudio G2 file support (Mac)
- Two plug-in platforms offer multiple options for effects processing— Real-Time AudioSuite (RTAS) is a host-based architecture that allows an effect to change and be dynamically automated in realtime as the audio plays back. —AudioSuite is a file-based format, that renders a new file with the processed sound.
- Bundled RTAS plug-ins include, 1 and 4-band EQ; Dynamics II- compressor, limiter, gate and expander/gate; Mod Delay - short, slap, medium, and long delays with modulation capabilities for chorus or flange effects and dither. AudioSuite plug-ins include Time Compression/Expansion, Pitch Shift, Normalize, Reverse.

MIDI Functions
- MIDI functions include graphic controller editing, piano roll display, up to 128 MIDI tracks and editing options like quantization, transpose, split notes, change velocity and change duration.
- MIDI data can be edited on the fly

MOTU AUDIO Hard Disk Recording Systems

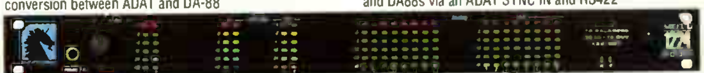
MOTU PCI-324 AUDIO

The MOTU Audio System is a PCI based hard recording solution for the Mac and PC platforms. At the heart of the system is the PCI-324 PCI card that can connect up to three audio interfaces and allows up to 72 channels of simultaneous I/O. Audio interfaces are available with a wide range of I/O configurations including multiple analog I/O with the latest 24-bit A/D/A converters and/or multi channel digital I/O such as ADAT optical and TDIF I/O as well as standard S/PDIF and AES/EBU I/O. Each interface can be purchased separately or with a PCI-324 card allowing you to build a system to suit your needs. Includes drivers for all of today's hottest audio software and AudioDesk, multitrack recording and editing software for the Mac.

THEY ALL FEATURE–
- Mac OS and Windows compatible • Includes software drivers for compatibility with all of today's popular audio software plus AudioDesk, MOTU's sample-accurate audio workstation software for Mac OS • Host computer determines the number of tracks that the software can record and play simultaneously, as well as the amount of real-time effects processing it can support • Front panels display metering for all inputs and outputs
- AudioDesk Audio Workstation Software for Mac OS features 24-bit recording, multi-channel waveform editing, automated virtual mixing, graphic editing of ramp automation, real-time effects plug-ins with 32-bit floating point processing, crossfades, support for third-party audio plug-ins (in the MOTU Audio System and Adobe Premiere formats), background processing of file-based operations, sample-accurate editing and placement of audio, and more

1296 24-bit/96kHz Interface Features–
- 24-bit, enhanced multi-bit 128x oversampling 96kHz converters • A-weighted signal-to-noise ratio of 117 dB
- 12 Balanced XLR inputs and outputs can support two simultaneous 5.1 mixes • AES/EBU I/O with sample rate conversion both in and out • Compatible with existing PCI-324 cards (requires new PCI-324 driver)• Connect up to 3 1296 interfaces to one PCI-324 card for a total of 36 inputs and outputs or mix and match the 1296 interface with up to three of the other MOTU audio interfaces

2408 mkII FEATURES–
- 7 banks of 8 channel I/O: 1 bank of analog, 3 banks of ADAT optical, 3 banks of Tascam TDIF, plus stereo S/PDIF.
- Custom VLSI chip for amazing I/O capabilities • • Format conversion between ADAT and DA-88
- 8x 24-bit 1/4" balanced analog I/Ds • 24-bit internal data bus for full 24-bit recording via digital inputs • Standard S/PDIF I/O for digital plus an additional S/PDIF I/O for the main mix • Sample-accurate synchronization with ADATs and DA88s via an ADAT SYNC IN and RS422



1224 FEATURES–
- 24-bit analog audio interface • State-of-the-art 24-bit A/D/A • Simultaneously record and play back 8 channels of balanced (TRS), +4 dB audio • 24-bit balanced +4 XLR main outputs • Stereo AES/EBU digital I/O • Word clock in/out • Dynamic range of 116 dB (A weighted) • Front panel displays six-segment metering for all inputs and outputs • Headphone jack with volume knob

CD RECORDING/MASTERING

ALESIS

Masterlink ML-9600
High-Resolution Master Disk Recorder

The Alesis MasterLink ML-9600 is a 2-track 24-bit recorder that combines hard disk recording, CD burning, digital signal processing, and mastering functions to create compact discs in the standard "Red Book" 16-bit/44.1kHz format, or high resolution CDs that utilize Alesis' revolutionary CD24 AIFF-compatible technology. MasterLink is capable of recording and playing up to 24-bit/96kHz resolution CDs using the inexpensive, readily available CD-R media. The amazing sonic quality, powerful built-in tools and CD24 technology offers a uniquely versatile and affordable solution for everyone from large commercial audio facilities to project studios and recording musicians.

FEATURES–
- 24-bit 128x oversampling analog to digital and digital to analog converters
- Supports 44.1, 48, 88.2, 96 kHz sample rates and word lengths of 16-, 20- and 24-bit
- 20Hz-20kHz frequency response at 44.1/48 kHz sample rates
- 20Hz-40kHz, frequency response at 88.2/96 kHz sample rates
- 113dB signal-to-noise ratio (A-weighted)
- Matsushita ATAPI CD-RDM drive allows up to 4x CD burning using standard CD-R discs.
- Built-in sample rate conversion & noise shaping to change sample rates & bit resolution as needed
- Reads and Writes 16-bit 44.1kHz Red Book Audio CDs
- Alesis' exclusive CD24 is a high-resolution mastering format that reads/writes files up to 24-bit 96kHz in the ISD 9660 disc format. AIFF caompatible file format that can be read by MacOS, Windows and Unix computer platforms.
- Built-in 3.2GB IDE hard drive
- Hard disk max recording times 95 min. @ 24-bit/96kHz 310 min. @ 16-bit/44.1kHz
- Create and store up to 16 playlists containing as many as 99 tracks

Analog Inputs and Outputs
- Balanced XLR connectors (+4dBu input and +19dBu max. output)
- Unbalanced phono (RCA) connectors (-10dBV input and +5dBV max. output)
- 1/4-inch TRS headphone output with level control

Digital Inputs and Outputs
- AES/EBU balanced XLR inputs and outputs
- S/PDIF unbalanced phono (RCA) inputs and outputs

Editing
- Gain control
- Cropping allows adjusting start and end points.
- Join and Split features allow combining and separating song sections.

DSP Finishing Tools
- Equalization, Compression, Normalizing and Peak Limiting

Includes
- Infra red remote control and rackmount brackets

marantz

CDR-631 Professional CD Recorder

The CDR631 offer all the features and functions of the CDR630, its popular predecessor, but adds many features and functions that were previously unavailable. Its full complement of digital and analog connections lets you record your own CDs from audio sources such as CDs, LPs, cassettes, DAT, or even a computer.

Features–
- Pro and consumer CD-R and CD-RW compatible
- Track titles can be saved and edited in CD-TEXT format that can be read on CD-TEXT compatible CD players
- Memory buffer that prevents the beginning of tracks from getting cut off
- Menu selectable SCMS copy protection
- Digital and analog record level and balance control
- XLR-Balanced and RCA unbalanced analog inputs
- AES/EBU (XLR), Coaxial, and Optical digital inputs
- Unbalanced (RCA) analog and Coaxial digital outputs including Coaxial loop-out for unprocessed connection to other digital equipment
- IR remote control included

MICROBOARDS

StartREC Digital Audio Editing/ CD Duplication System

The Microboards StartREC is the first digital audio editing system combined with a multidrive CD recordable duplication system for professionals. Audio is recorded to the internal 6.2 GB IDE hard drive using analog or digital inputs. Sample rate conversion is automatic. Tracks can be edited and sequenced using the StartREC's user friendly interface and up to 4 CDs can be recorded simultaneously. StartREC is the ideal solution for studio recording, mastering, post production or any pro audio environment requiring digital audio editing and short run CD-R duplication.

Features–
- 2X, 4X, or 8X recording speeds
- 6.2GB IDE hard drive
- Editing functions include move, divide, combine or delete audio tracks, add or drop any index or sub index, and create track fade in or fade out
- Coaxial SP/DIF or AES/EBU digital input plus optical S/PDIF I/O
- XLR balanced and RCA Line inputs and outputs
- Automatic sample rate conversion from 32 and 48kHz
- Automatic CD Format Detection feature and user friendly interface provide one touch button operation
- Front panel trim pot and LCD display provide accurate input signal and time lapse metering
- SCMS (Serial Copy Management System) is supported, regardless of the source disc copy protection status
- **StartREC Models Include:** ST2000 (2) 8x writers. ST3000 (3) 8x writers and ST4000 (4) 8x writers

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DIGITAL MULTI-TRACK RECORDERS

TASCAM MX-2424 24-Bit 24-Track Hard Disk Recorder

Co-designed by TASCAM and TimeLine Inc., the MX-2424 is an affordable 24-bit, 24-track hard disk recorder that also has the editing power of a digital audio workstation. A 9GB internal hard drive comes standard as well as a SCSI Wide port that supports external LVD (Low Voltage Drives) hard drives from up to 40 feet away. An optional analog and several digital I/O cards are available so the MX-2424 can be configured to suit your work environment. SMPTE synchronization, Word Clock, MIDI Time Code and MIDI Machine Control are all built in for seamless integration into any studio.

- Records 24 tracks of 24-bit audio at 44.1 or 48 kHz, or 12 tracks at 88.2 or 96 kHz. Up to 24 tracks can be recorded simultaneously using any combination of digital and analog I/O.
- Supplied 9GB internal drive allows 45 minutes of audio across all 24 tracks
- Wide SCSI port on the back panel allows you to add multiple drives. A front 5-1/2" bay available for installing an additional drive, or an approved DVD-RAM drive for back-up.
- ViewNet MX, a Java-based software suite for Mac and PC offers DAW style editing of audio regions, dedicated system set-up screens that make set-up quicker and easier and track load screens that make virtual track management a snap. Connects to a computer via a standard Ethernet line.
- Can record to Mac (SDII) or PC (.WAV) formatted drives, allowing later export to the computer. The Open TL format allows compatible software to recognize virtual tracks without have to load, reposition and trim each digital file.

Transport Controls-
- Jog/scrub wheel
- MIDI In, Out, and Thru ports are built-in for MIDI Machine Control.

Editing-
- Built-in editing capabilities include cut, copy, paste, split and ripple or overwrite
- 100 levels of undo
- Supports destructive loop recording and non-destructive loop recording which continuously records new takes without erasing the previous version.

Build-In Synchronization-
- TBUS protocol can sample accurately lock 32 machines together for 384 tracks at 96kHz, or 768 tracks at 48kHz.
- Can generate or chase SMPTE timecode or MIDI Time Code.
- Word Clock In, Out, and Thru ports

I/O Options-
- Optional analog and digital cards all provide 24 channels of I/O. There is one slot for analog and one for digital.
- IF-TD24- T/DIF module
- IF-AD24- ADAT Lightpipe module
- IF-AE24- AES/EBU module
- IF- AN24- A-D, D-A I/O module with DB-25 connectors

Software Updates-
- System updates are made available through a front panel Smart Card slot or via computer directly from the TASCAM web site.

DA-78HR Modular Digital Multitrack

The DA-78HR is the first true 24-bit tape based 8-track modular digital multitrack recorder. Based on the DTRS (Digital Tape Recording System) it provides up to 108 minutes of pristine 24-bit or 16-bit digital audio on a single 120 Hi-8 video tape. Designed for project and commercial recording studios as well as video post and field production, the DA-78HR offers a host of standard features including built-in SMPTE Time Code Reader/Generator, MIDI Time Code synchronization and a digital mixer with pan and level controls. A coaxial S/PDIF digital I/O allows pre-mixed digital bouncing within a single unit, or externally to another recorder or even a DAT or CD recorder. Up to 16 DTRS machines can be synchronized together for simultaneous, sample accurate control of 128 tracks of digital audio.

Features-
- Selectable 16 bit or 24 bit High Resolution audio
- 24 bit A/D and D/A converters
- >104dB Dynamic range
- 20Hz - 20kHz frequency response ±.5dB
- 1 hr. 48 min. recording time on a single 120 tape
- On-Board SMPTE synchronizer - chase or generate timecode
- On-Board support for MIDI Machine Control
- Internal digital mixer with level and pan for internal bouncing, or for quick mixes
- Track slip from -200 to +7200 samples
- Expandable up to 128 tracks (16 machines)
- Word Sync In/Out/Thru
- Analog output on DB25 balanced or RCA unbalanced
- Digital output on TDIF or 2 channels of S/PDIF

A TO D CONVERTERS

APOGEE Rosetta 24-bit A to D Converter

The high-end quality analog to digital solution for the project studio. With support for both professional and consumer digital formats you can now record your audio at a higher resolution and with greater detail than standard converters found on MDM's, DAT's and DAW's. Ideal for mastering or tracking.

FEATURES–
- 24-bit, 44.1-48, 88.2-96 kHz Sample Rate (±10%)
- 116dB dynamic range (unweighted)
- Improved UV22HR for 16 and 20-bit A/D conversion

FRONT PANEL:
- Power switch • Sample Rate (44.1, 48, 88.2, 96kHZ)selector • 16-bit (UV22), 20-bit (UV22) and 24-bit resolution selector • S/PDIF-ADAT optical selector • Soft Limit on or off • 12-segment metering w/ over ondicator & Meter Clear switch • Level trim

REAR PANEL:
- XLR balanced inputs • 2 x AES/EBU for 88.2/96kHz 2 channel path, Coaxial S/PDIF, switchable S/PDIF or ADAT optical outputs • Wordclock out

LUCID AD 9624 24-bit A to D Converter

Transparent analog to digital conversion designed to bring your music to the next level. XLR balanced inputs feed true 24-bit converters for revealing all the detail of the analog source. 16-bit masters can take advantage of the AD9624's noise shaping function which enhances clarity of low level signals.

FEATURES–
- 24-bit precision A/D conversion • Support for 32, 44.1, 48, 88.2 & 96kHz sample rates • Wordclock sync input• Selectable 16-bit noise shaping • Simultaneous AES/EBU, coaxial and optical S/PDIF outputs • 20-segment LED meters w/ peak hold & clip indicators • **ALSO AVAILABLE:** DA9624 24-bit D/A converter

DIGITAL MIXERS

Roland VM Basic 72 Digital Mixing System

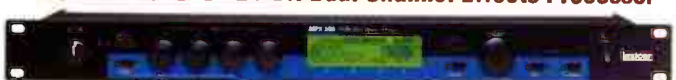
The all digital Roland V-Mixing System, when fully expanded, is capable of mixing up to 94 channels with 16 stereo (32 mono) onboard multi-effects including CDSM Speaker Modeling. Utilizing a separate-component design, comprised of the VM-C7200 console and VM-7200 rackmount processor, allows the V-Mixing System to be configured to suit your needs. Navigation is made easy via a friendly user interface, FlexBus and EZ routing capabilities as well as a large informative LCD and ultra-fast short cut keys.

- 94 channels of digital automated mixing (fully expanded)
- Up to 48 channels of ADAT/Tascam T-DIF digital audio I/O with optional expansion boards and interfaces
- Separate console/processor design
- Quiet motorized faders, transport controls, total recall of all parameters including input gain, onboard mixer dynamic automation and scene memory
- 24 fader groups, dual-channel delays, 4-band parametric channel EQ + channel HPF
- FlexBus and "virtual patchbay" for unparalleled routing flexibility
- Up to 16 stereo (or 32 mono) multi-effects processors using optional VS8F-2 Effects Expansion Boards (2 stereo effects processors standard)
- CDSM Speaker Modeling and Mic Simulation technology
- 5.1 Surround mixing capabilities
- EZ Routing allows mixer settings to be saved as templates
- Realtime Spectrum Analyzer checks room acoustics in conjunction with noise generator and oscillator
- Digital cables between processor and mixer can be up to 100 meters long- ideal for live sound reinforcement.

- **VS8F-2 Effects Expansion Board** -- Provides 2 stereo effects processors including CDSM Speaker Modeling. Up to 3 additional boards can be user-installed into the VM-7200 processor, for 8 stereo or 16 mono effects per processor
- **VM-24E I/O Expansion Board** -- Offers 3 R-Bus I/Os on a single board. Each R-Bus I/O provides 8-in/8-out 24-bit digital I/O, totalling 24 I/O per expansion board.
- **DIF-AT Interface Box for ADAT/Tascam** -- Converts signals between R-Bus (VM-24E expansion board required) and ADAT/Tascam T-DIF. Handles 8-in/8-out digital audio. 1/3 rackmount size.
- **VM-24C Cascade Kit** -- Connects two VM-Series processor units. Using two VM-7200 processors cascaded and fully expanded with R-Bus I/O, 94 channels of audio processing are available.

EFFECTS & PROCESSING

lexicon MPX-500 24-Bit Dual Channel Effects Processor



The MPX 500 is a true stereo 24-bit dual-channel processor and like the MPX100 is powered by Lexicon's proprietary Lexichip and offers dual-channel processing. However, the MPX 500 offers even greater control over effects parameters, has digital inputs and outputs as well as a large graphics display.

- 240 presets with classic, true stereo reverb programs as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, 5.5 second Delay and Echo
- Balanced analog and S/PDIF digital I/O
- 4 dedicated front panel knobs allow adjustment of effect parameters. Easy Learn mode allows MIDI patching of front panel controls.
- Tempo-controlled delays lock to Tap or MIDI clock

t.c. electronic M-One Dual Effects Processor

The M-One allows two reverbs or other effects to be run simultaneously, without compromising sound quality. The intuitive yet sophisticated interface gives you instant control of all vital parameters and allows you to create awesome effects programs quickly and easily.

- 20 incredible TC effects including, Reverb, Chorus Tremolo, Pitch, Delay and Dynamics
- Analog-style user interface
- 100 Factory/100 User presets
- Dual-Engine design
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

D-Two Multitap Rhythm Delay

Based on the Classic TC2290 Delay, the D-Two is the first unit that allows rhythm patterns to be tapped in directly or quantized to a specific tempo and subdivision.

- Multitap Rhythm Delay
- Absolute Repeat Control
- Up to 10 seconds of Delay
- 50 Factory/100 User presets
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

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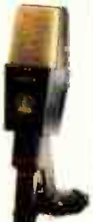
MICROPHONES

AKG C414 TLII "Vintage TL"



Combines the best of old and new: legendary C12 acoustics and the latest generation of C414 transformerless FET electronics. Although similar in design and shape to the C414BULS, the TLII features a capsule that is a faithful sonic recreation of the one used in the classic C12 tube mic combined with computer-aided manufacturing techniques that assure greater uniformity in response from microphone to microphone.

FEATURES-
- Cardioid, hypercardioid, omnidirectional and figure 8 polar patterns
- Warm, smooth microphone that is suitable for high-quality digital recording.
- **Frequency response 10Hz to 20kHz**

C4000B
ELECTERET CONDENSER

This new mic from AKG is a multi polar pattern condenser micropone using a unique electret dual large diaphram transducer. It is based on the AKG SolidTube deisgn, except that the tube has been replaced by a transistorized impedance converter/ preamp. The transformerless output stage offers the C4000B exceptional low frequency

FEATURES-
- Electret Dual Large Diaphram Transducer (1st of its kind) • Cardioid, hypercardioid & omnidirectional polar patterns • High Sensitivity
- Extremely low self-noise • Bass cut filter & Pad switches • Requires 12, 24 or 48 V phantom power
- Includes H-100 shockmount and wind/pop screen
- **Frequency response 20Hz to 20kHz**

RØDE NT-2 Condenser Mic

The RØDE NT2 is a large diaphragm true condenser studio mic that features both cardioid and omnidirectional polar patterns. The NT-2 offers superb sonic detail with a vintage flavor for vocal and instrument miking. Like all RØDE mics the NT-2 is hand-assembled in Australia and is available at a breakthrough price.

FEATURES-
- Dual pressure gradient transducer
- 1" capsule with gold-sputtered membranes • Low noise, transformerless circuitry • Omni and cardioid polar patterns • 135dB Max SPL • High pass filter switch and -10dB pad switch • Gold plated output connector and internal head pins
- Shockmount, Flight Case, and Pop Filter included
- **20Hz-20kHz frequency response**

audio-technica AT4047 Cardioid Condenser

The AT4047 is the latest 40 Series large diaphragm condenser mic from Audio Technica. It has the low self noise, wide dynamic range and high sound pressure level capacity demanded by recording studios and sound reinforcement professionals.

FEATURES-
- Side address cardioid condenser microphone for professional recording and critical applications in broadcast and live sound
- Low self noise, wide dynamic range and high SPL
- Switchable 80Hz Hi Pass Filter and 10dB pad
- **Includes AT8449 SV shockmount**

MICROPHONE PREAMPS

AVALON DESIGN VT-737SP Mono Class A, Vacuum Tube-Discrete Preamp-Opto-Compressor-Equalizer

The VT-737SP is a vacuum tube, Class A processor that combines a mic preamp, instrument DI, compressor and sweepable 4-band equalizer in a 2U rack space. Like all Avalon Design products the VT-737SP utilizes a minimum signal path design with 100% discrete, high-bias pure Class A audio amplifiers and the best active and passive components available. Used by renowned artists and studios world wide and the winner of the Electronic Musician 1999 Editors' Choice Award for Product Of The Year.

FEATURES-
- Combination of TUBE preamplifiers, opto-compressor, sweep equalizer, output level and VU metering in a 2U space
- Four dual triode vacuum tubes, high-voltage discrete Class A with a 10 Hz to 120kHz frequency response ±0.5dB
- **The Preamp** has three input selections- The first is a high performance XLR balanced mic input transformer with +48v phantom power, the second is a high impedance instrument DI with a 1/4" jack located on the front panel and the third is a discrete high-level Class A balanced line input.
- High gain switch boosts overall preamp gain and a passive- variable high pass filter, hardwire relay bypass and phase reverse relay is available for all three inputs
- **The Opto-Compressor** uses a minimum signal path design and features twin Class A vacuum tube triodes for gain matching. A passive optical attenuator serves as a simple level controller. Variable threshold, compression ratio and attack and release offer dynamics control from soft compression to hard-knee limiting.
- The dual sweep mid-EQ can be side chained to the compressor allowing a broad range of spectral control including de-essing. The EQ can be assigned pre and post compressor from the front panel to add even greater sonic possibilities.
- Two VT-737 SPs can be linked together via a rear panel link cable for stereo tracking
- **The Equalizer** utilizes 100% discrete, Class A-high-voltage transistors for optimum sonic performance.
- The low frequency passive shelving EQ is selectable between 15, 30 60 and 150HZ with a boost and cut of ±24dB
- The high frequency passive shelving EQ is selectable between 10, 15, 20 and 32 kHZ with a boost and cut of ±20dB
- The low-mid frequency is variable between 35 to 450 Hz while the high-mid frequency is variable from 220Hz to 2.8 kHz. Both mid-band frequencies offer a boost and cut of ±16 dB and a hi-Q/lo-Q switch.
- When the EQ to side chain is used, the low and high EQ is still available for tonal adjustment
- **The Output** level is continuously variable and utilizes an another dual triode vacuum tube driving a 100% Class A, high-current balanced and DC coupled low noise output amplifier.
- Sealed silver relay bypass switches are used for the most direct signal path

POWERED STUDIO MONITORS

VERGENCE A-20 Studio Reference Monitor System

Incorporating a pair of 2-way, acoustic suspension monitors and external, system-specific 250 watt per side control amplifier, the A-20 provides a precise, neutral studio reference monitoring system for project, commercial and post production studios. The A-20's control amplifier adapts to any production environment by offering control over monitoring depth (from near to far field), wall proximity and even input sensitivity while the speakers magnetic shielding allows seamless integration into today's computer based studios.

- Type Modular, self-powered near/mid/far-field monitor.
- 48Hz - 20kHz frequency response @ 1M
- Peak Acoustic Output 117dB SPL (100ms pink noise at 1M).
- XLR outputs from power amp to speakers
- Matched impedance output cables included.
- -6dB LF Cutoff 40Hz
- 5 position wall proximity control
- 5 position listening proximity control between near, mid and far-field monitoring
- Power, Overload; SPL Output, Line VAC and Output device temperature display.

Amplifier
- Amplifier Power 250W (continuous rms/ch), 400W (100ms peak).
- XLR, TRS input connectors
- Headphone output
- 5-position input sensitivity switch with settings

Speakers
- 2-way acoustic suspension with a 6.5-inch treated paper woofer and a 1-inch aluminum dome tweeter
- Fully magnetically Shielded with an 18-inch recommended working distance

EVENT ELECTRONICS PS-5 Bi-Amplified Project Studio Monitors

The PS-5s are small format, full-range, non-fatiguing project studio monitors that give you the same precise, accurate sound as the highly acclaimed 20/20 series studio monitors. The use of custom driver components, complimentary crossover and bi-amplified power design provides a wide dynamic range with excellent transient response and low intermodulation distortion.

FEATURES-
- 5-1/4-inch magnetically shielded mineral-filled polypropylene cone with 1-inch diameter high-temperature voice coil and damped rubber surround LF Driver
- Magnetically shielded 25mm diameter ferrofluid-cooled natural silk dome neodymium HF Driver
- 70 watt continuous LF and 30 watt continuous HF amplification per side
- XLR-balanced and 1/4-inch (balanced or unbalanced) inputs
- 52Hz-19kHz frequency response ±3dB
- 2.6kHz, active second order crossover
- Built-in RF interference, output current limiting, over temperature, turn-on transient, subsonic filter, internal fuse protection
- Combination Power On/Clip LED indicator
- 5/8" vinyl-laminated MDF cabinet

Hafler TRM-6 Bi-Amplified Studio Monitors

Offering honest, consistent sound from top to bottom, the TRM-6 bi-amplified studio monitors are the ideal reference monitors for any recording environment whether tracking, mixing and mastering. Supported by Hafler's legendary amplifier technology providing a more accurate sound field, in width, height and also depth.

FEATURES-
- 33 Watt HF & 50 Watt LF amplification
- 1-inch soft dome tweeter and 6.5-inch polypropylene woofer
- 55Hz - 21kHz Response
- Magnetically Shielded
- Electronically and Acoustically Matched

Also Available- TRM-8
- 1-inch soft dome tweeter and 8-inch polypropylene woofer
- 45Hz - 21kHz frequency response ±2dB
- 75 Watt HF, 150 Watt LF amplification

TRM-10s And TRM-12s Active Subwoofers

Combining Hafler's legendary amplifier technology with a proprietary woofer design, the TRM10s and TRM12s active subwoofers provide superb bass definition required in today's studio and surround sound environments.

TRM-10s
- 10-inch cellulose fibre cone down firing woofer.
- 200 watt low frequency amplifier
- 30Hz to 110Hz frequency response ±2dB
- 24dB/octave Linkwitz-Riley crossover variable (40Hz to 110Hz)

TRM-12s
- 12-inch cellulose fibre cone down firing woofer.
- 200 watt low frequency amplifier
- 25Hz to 110Hz frequency response ±2dB
- 24dB/octave Linkwitz-Riley crossover variable (40Hz to 110Hz)

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ACROSS THE BOARD

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can locate multiple machines to the same point in time. SMPTE timecode and ADAT sync provide both speed and position information at the same time. There may be one that I left out, but hey…I'm powering down free drinks at 33,000 feet.

When digital audio is transmitted from one device to another, it's done serially. That is, a 24-bit sample is taken of an analog signal at one point in time, taken apart into individual bits, sent down the wire one bit after the other, put back together at the other end, and then put out as an analog sample equal to the input. AES and S/PDIF signals contain some extra information that tells the receiver where the start and end of each sample is in the stream, so they're "self clocking." The word clock and bit clock are built into the signal.

In some digital audio interfaces, there is no reference built in, so the reference has to be carried externally. Enter word clock. The word clock signal is just a pulse that announces the start of the sample bits. Without word clock, the receiving device would not know which bit was supposed to be first, and you would hear hissing, tearing garbage. I am sure you are all familiar with that sound. If not, you must still be recording analog.

YOU SHALL HAVE BUT ONE MASTER

Digital audio must have been around for a long time before it was re-discovered, because even the Bible says there can only be one Master. This proves that Cleanliness in Digital Audio is next to Godliness.

Any time two audio devices are connected digitally, one must be the master, and the other must be the slave. Most of the time this is taken care of automatically. When you switch a DAT machine or a CD recorder to digital input, it automatically synchronizes to the external digital signal's clock. Since the clock is built into the AES or S/PDIF or optical or ADAT signals, the user is not aware of the changeover. Digital consoles and DAWs can have other sources besides these for digital input. TASCAM TDIF needs a separate word clock (TDIF-2 includes word clock). SDIF (Sony multi-track format) and Mitsubishi's ProDigi interface require word clock.

If two digital devices are connected and they are both selected to internal clock, then the receiving device will not know where the samples begin and we will again get the hairy hash sound we have grown to love so much.

You can never get hurt if you have word clock and AES cables connecting two digital devices. If the digital audio is going through the AES cable or ADAT Lightpipe, a word clock cable going out of the master into the slave, and you have word clock selected in the slave device, everything will work fine. The word clock signal is lined up with the clock built into the digital audio signal, so both devices should remain happy.

If you have a digital console, DAW, ADATs, DA-78's, and whatever else all connected together, remember to have one master and everything else as a slave. I use the Aardsync box for a master clock and connect it to everything. There is also an AES sync signal generated in the Aardsync. This can be connected to devices that do not have word clock inputs, but sync to an external AES signal. Once everything is connected properly, you will no longer have that "syncing feeling."

DONE FOR NOW

Well, I still have 1,234 hours left in this flight. Maybe I will go ahead and do all of my columns for the rest of the year. EQ

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MOTU

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BITHEADZ

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ChannelStrip

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The word is out...

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"I LOVE the MIXES I am getting with da ChannelStrip. It really makes mixing a pleasure. Hey – I got a $250,000 Euphonix just sitting there looking pretty; NOW WHAT !?!?!" —Giorgio Bertuccelli

"Wow, it really sounds great! Finally, real EQ in software. I love this thing!" —Jim Watson, FAT GROOVE Productions

"Sonically, I'm knocked out with ChannelStrip — I am really really critical of a lot of the plug-in stuff and work with some very demanding artists at some of the finest studios around, plus my own room which is about as good as it gets for overdubs and mixing. I would not think twice about using ChannelStrip on anything in front of anyone — in my limited time with it, I think it's that good. You're onto a gold mine with this. I've loaded up a few tracks only so far and used the automation to do a few things I normally have to do in real time. Unreal... Once again, it's the sound of ChannelStrip that I can't get over. The controls, layout, etc. are very cool, too." —Jack Hale

"The whole plug-in gives the impression you looked at an SSL pretty close. I've always been an SSL-man, but not any longer! Still can't believe my luck.... Nice work!" —Steve Rhodes, RME

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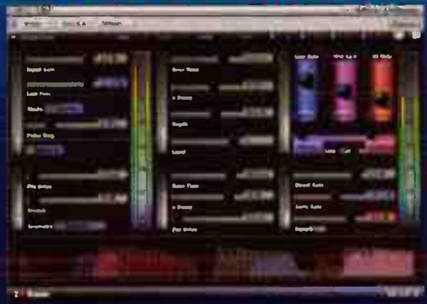

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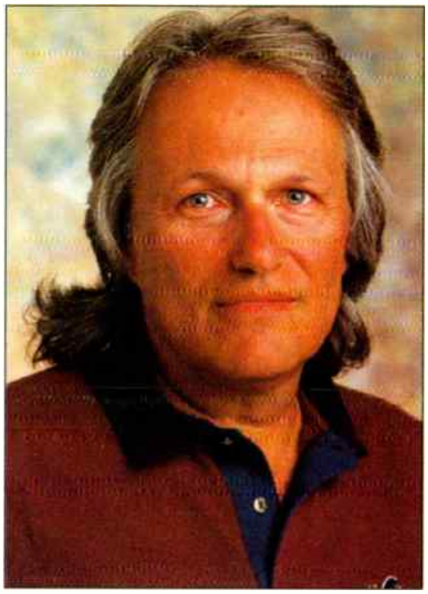

Everything you need to know about word clock

BY ROGER NICHOLS

As I write this column, I am looking down at the Pacific Ocean from 30,000 feet on my way to Japan for a Steely Dan tour. The second leg of the tour will be across the U.S. in June and July, with the third leg following in Europe this September. Maybe they should have called it the Steely Dan "Tripod Tour." Maybe not.

I have figured out the difference between First Class and Business Class. First Class gets warm peanuts and has 96 kHz word clock at every seat. Business Class gets cold peanuts and no word clock, only laptop power connectors.

COMPUTER CONNECTIONS & ROCKET NETWORK

More and more hotels are equipping their rooms with Ethernet computer connections to the Internet. Instead of seeing download speeds of 2k to 4k per second, the meter on my browser says 200k per second. I downloaded everything I could lay my hands on just to see how fast it was. I now need a bigger hard disk.

DSL is finally available in my section of Miami. I am putting in two DSL lines with teaming software that will give me 2.5 megabits per second. The advantage of DSL over cable modems is that you can communicate fast in both directions. I always have lots of graphics or audio files to send and receive, so modem speeds just won't hack it.

WHAT UP WIT DAT?

My need for speed comes directly from my introduction to the Rocket Network. Steinberg's Cubase VST with Rocket Power and Emagic's Logic Rocket support a new network whereby multiple sessions on computers connected to the Internet can share the same MIDI and audio sample data. In the past, there have been schemes that would allow distance collaboration, but they have failed. This is because they tried to collaborate in real time. The laws of time and space sent them packing. Rocket Network is different.

The guitar part I'm playing back is a clone of what was recorded 3,000 miles away. Think of it: No waiting for FedEx and no expensive plane tickets.

Each participant in a session has all of the audio and MIDI files associated with the session on his local computer. When a new participant joins the session, all of the files are downloaded to his computer. Each musician records his track on his own computer with his own word clock, playing back and recording on his own gear. When he is done with the overdub, the final results are uploaded to the master session computer and then downloaded to additional participants as needed.

As an example, let's say that I'm in Miami and have a song that I want Dean Parks to overdub a guitar on. Dean is in L.A. Dean is busy. Dean and I have been talking about "phoning in your parts" for 20 years. Now we can do it. I start a session with a mix of the track that he is to play on. I log onto the Rocket Network. Dean logs onto the Rocket Network and joins my session. The session is downloaded to his computer. I go about my business working on other stuff while Dean works on his guitar part. When he is done, the final guitar track is uploaded to my session. I play it back on my computer to listen. Great! I never have to look Dean in the face again. It's like e-mail for digital audio. You get to carry on involved conversations without ever hearing the person's voice.

Remember that this is digital audio. The guitar part I'm playing back is a clone of what was recorded 3,000 miles away. Think of it. No waiting for FedEx, no dual machine lockups over ISDN lines, no expensive plane tickets and hotels and escort services for out-of-town musicians, no "I didn't bring that guitar" excuses, just the ability to get the project done with the guys you want at a budget that is affordable.

Try it yourself. I think you'll like it. Now we just have to get more companies on the bandwagon to support the Rocket Network, like maybe Digidesign...then Steely Dan can work in New York while I edit and slide tracks around by my pool in Miami. Yeah, that's the ticket.

AT ONE WITH WORD CLOCK

I have talked in the past about word clock problems and sync issues, but many of the questions I get on the EQ Board (www.eqmag.com) are still sync-related. So, here comes a little word clock refresher course.

There are two types of synchronization signals; those that provide speed reference and those that provide position reference. Pilot tone, word clock, bit clock, AES and S/PDIF, Digidesign Superclock, tach pulses, bi-phase, and video sync provide speed information. This speed information is used to synchronize analog machines, digital machines, video machines, and DAWs so that they all run at the same speed. MIDI Timecode, VITC (Vertical Interval Timecode), and burned-in timecode provide positional reference so you

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