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EQ

PROFESSIONAL PROJECT RECORDING & SOUND

MAY • 2000

FIRST LOOK:

Propellerheads Reason — Who Needs Hardware?

REVIEWS:

Event Project Studio Series Monitors

TC Works Spark Audio Editor

dbx 386 Tube Preamp

and more

TODD RUNDGREN: BAD RELIGION, THE INTERNET, AND BEYOND

GREG DEBELLES PUTS SONG TO FILM

GATED REVERBS AND THE ANALOG SOUND WITH MICK GLOSSOP

Walter Afanasieff

The winner of the Producer of the Year Grammy reveals the secrets of his success

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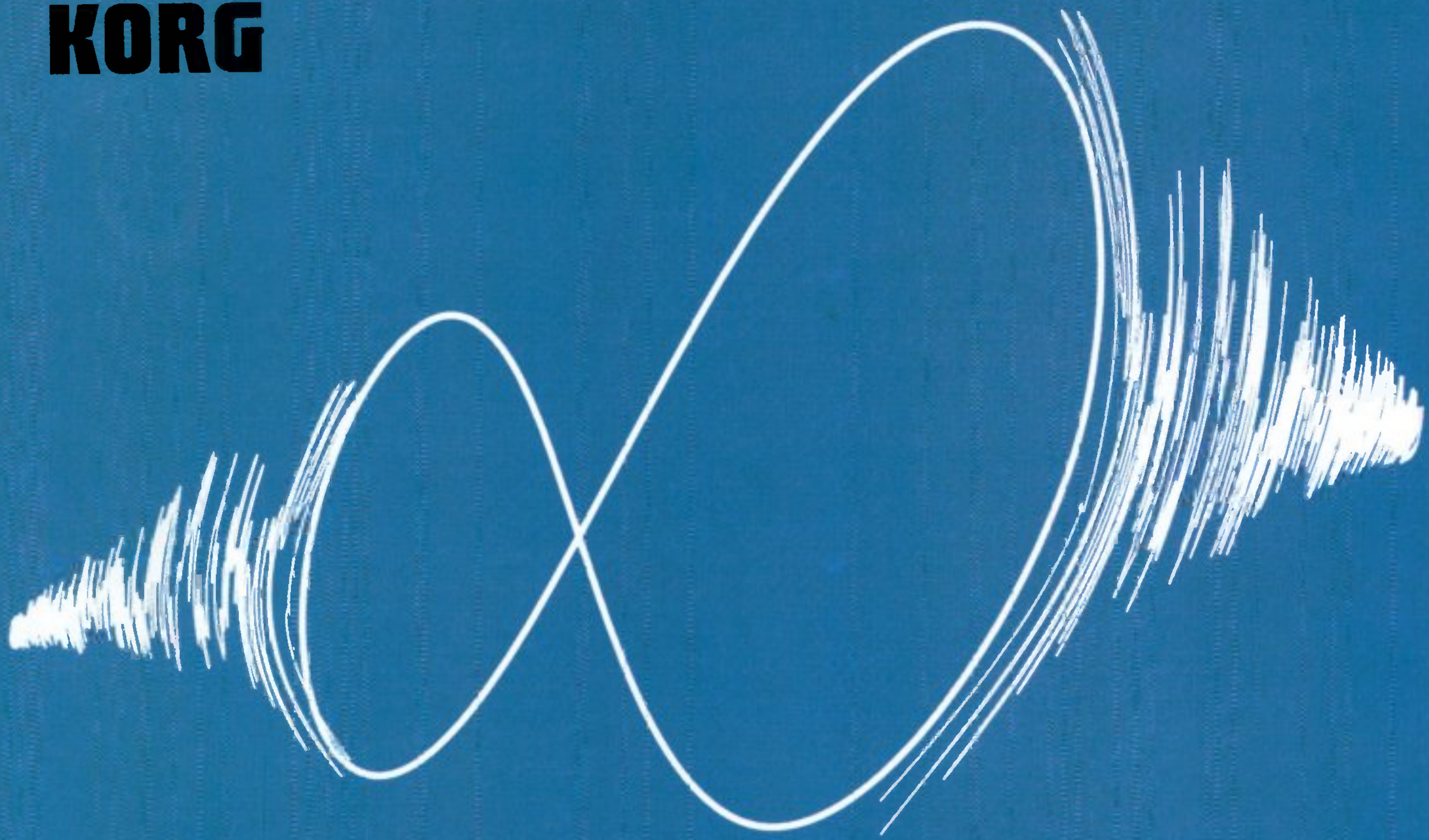

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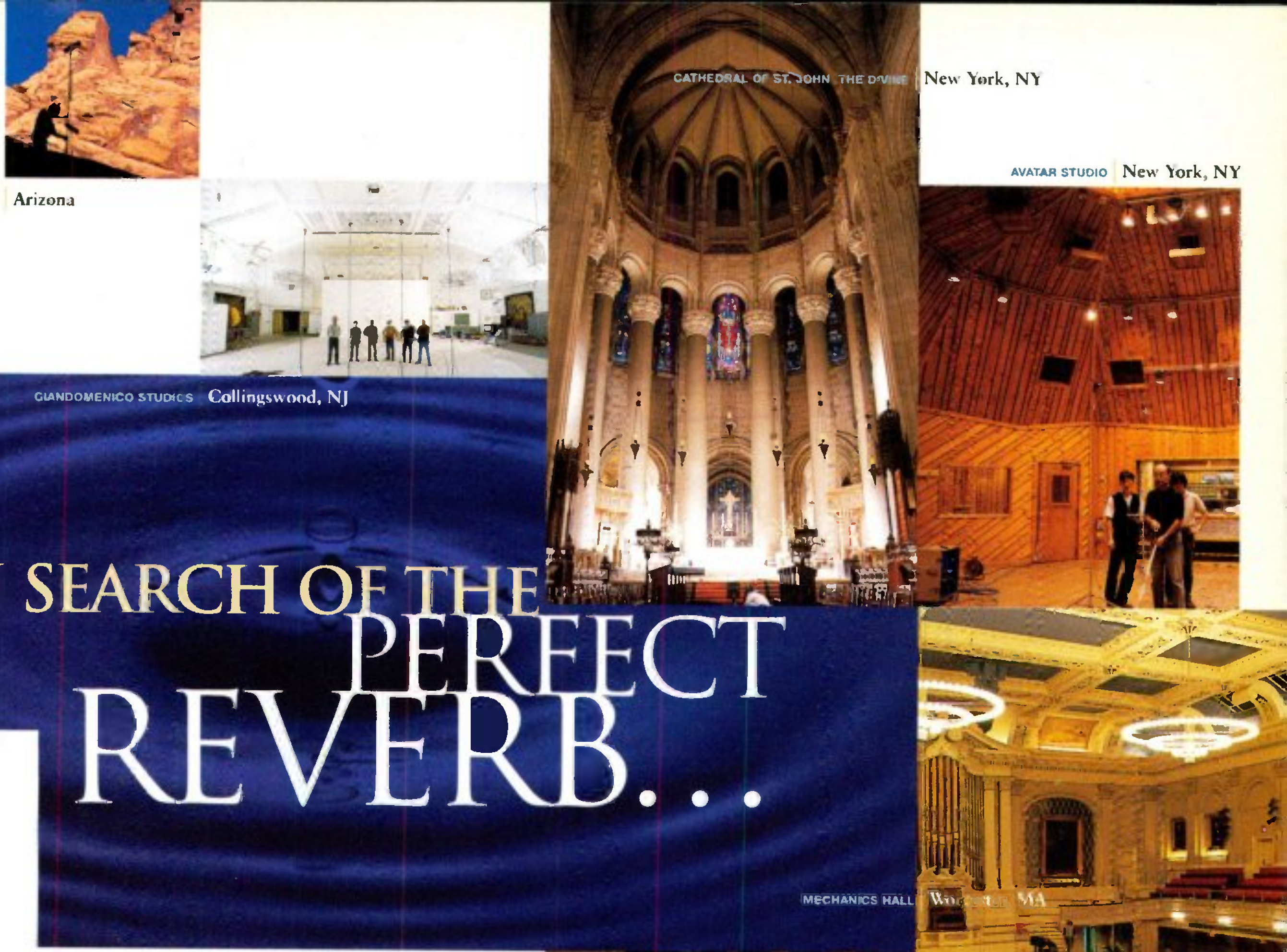

IN SEARCH OF THE PERFECT REVERB...

PRO AUDIO REVIEW 1999 REVIEWER'S PICK

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We've sampled and captured the actual reverberant characteristics of some of the finest acoustic environments in the world. Why? Because this is the underlying principle behind the Sony DRE-S777 Digital Sampling Reverberator.

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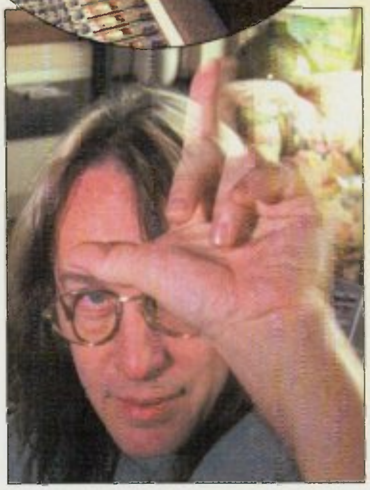
PROJECT RECORDING
& SOUND TECHNIQUES
VOLUME 11, ISSUE 5
MAY 2000



FEATURES

GROOVE

TECHNIQUES/WORKSHOPS

COLUMNS/DEPARTMENTS

ON THE COVER:
Walter Afanasieff in his project studio. Photo by Ross Pelton.

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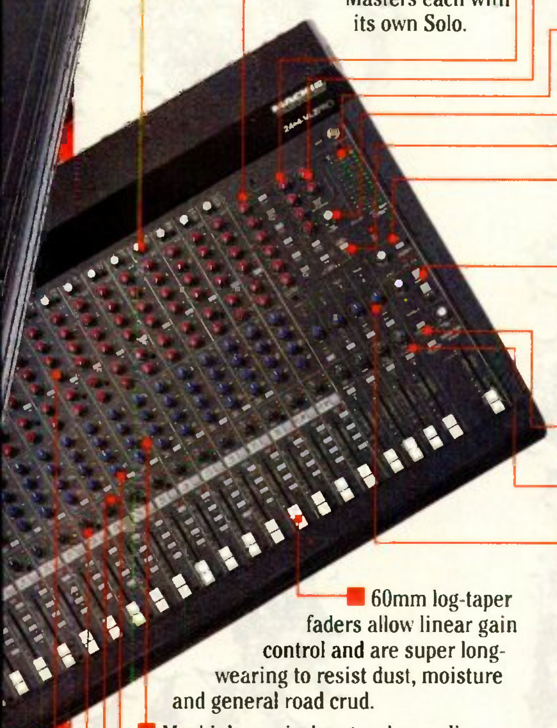
LIVE MUSIC TO YOUR EARS:
LOW $1599 USRP* ON THE 24•4-VLZ PRO,
AND A NEW LOW $1999 USRP*
ON THE 32•4-VLZ PRO
MACKIE
24•4-VLZ PRO

NEW 24•4 & 32•4-VLZ PRO™ SR MIXERS WITH PREMIUM XDR™ MIC PREAMPS.

XDR™

Why put ultra-precise, tweakazoid audiophile XDR™ mic preamps on sound reinforcement consoles? Because live performers deserve good sound, too. Especially when our new design also has the **best RFI protection of any mixer on the market**. It took several years of hard work to design a mixer preamp that could beat $2000-a-channel, esoteric outboard mic preamps in independent listening tests. But we did it. You'll enjoy more warmth, detail and headroom than has ever been possible with even the most prestigious mega-consoles. Plus less noise, and total freedom from potential hot-patching and short circuit damage, and flat frequency response regardless of mic/cable impedance.



- Trim control with 60dB mic gain & 10dB "virtual pad" handles anything from a timid vocalist to a rilly big kick drum.
- Six separate Aux Send Masters each with its own Solo.
- 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.
- Six aux sends per channel. 2 pre-fader, 2 post-fader and 2 pre/post switchable.
- Stereo Aux Return 4 Master can be assigned to Buses 1-2 or 3-4.
- 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.

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
- New high-performance 2068 op-amps
- Muted channels can be soloed!
- 6 individual aux sends per channel
- 4 master stereo aux returns
- Inserts on all mono mic/line channels
- 3-band EQ w/swept mid on mic/line chs.
- 4-band fixed EQ on stereo line channels
- 60mm long-life logarithmic-taper faders
- 6 aux send masters with individual solos
- 4 stereo aux returns w/EFX to Monitor
- 16kHz Air EQ, pan and solo sub buses
- Double-bussed subs for easy multi-tracking with 8-track recorders

LOADED WITH LIVE SOUND FEATURES.

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We'll send you our jumbo product brochure complete with hook-up diagrams — and a serious, graph-and-equation-loaded White Paper on why XDR technology beats the cables off anybody else's mic preamps.

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The 24•4 & 32•4 are made by Mackoids like these folks in scenic Woodinville, WA USA.

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CIRCLE 17 ON FREE INFO CARD

EQ

Vol. 11, No. 5 May 2000

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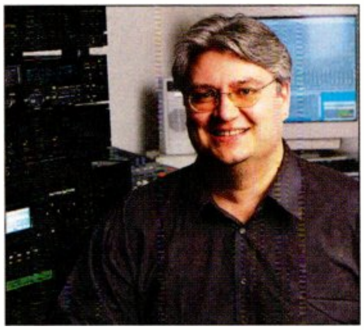
From the **Editor**



Gear, gear, and more gear. Is there an engineer or producer who doesn't love it? (Maybe love/hate it is more apropos!) Are there any *EQ* readers who don't obsess over how to improve the gear they've got or how to buy the new toy they *must* have to produce great audio?

Sure, it can be fun sitting around the studio, in a store, or online discussing the tweaky aspects of Product A versus Product B. But lately, I have to admit, I've been a bit bugged by the *level* of the gear obsession I'm seeing out there. I spend a great deal of time wandering through music stores, online visiting sites, and lurking in the dark corners of discussion forums. (Hey, I'm referring to music and audio sites and forums, get your mind out of the gutter....) Over and over again I see the same topics come up; "which is the best," "should I buy that," on and on. Clearly people are struggling to make decisions, which is understandable — there are so many choices out there. But, at the same time, we're fortunate to live in a time when the quality of even the lowest-end products is at an all-time high, and in most cases capable of producing excellent, if not pro-level, results. And often the tweaky things I see people concerned with and obsessing over are simply not going to make that much difference to their end productions.

It's not just those looking at new gear who obsess, of course. I recently read an article profiling virtuoso guitarist Eric Johnson's stage rig; Eric is famous for hearing the difference between various batteries in his effects pedals, brass versus chrome cable plugs, and so on. Now he's insisting that the distance between his pedals affects the sound, and, in one case, that whether the battery cover plate is held on with a screw or a rubber band affects a pedal's tone. Now, I happen to believe that Eric is one of the best guitarists around, but *wow.* I'm sorry, but even in the most hyper-critical studio situation (let alone at a live show), I defy *anyone* to listen blind to two guitar signals and tell me in which one the wah pedal was further from the delay pedal!

Okay, the example is extreme; and, honestly, if Eric feels he can hear the difference (and has *time* to explore the issue), then more power to him. Still, it's a good bet that we're all guilty of spending too much time worrying about gear minutia. I doubt that kicking the habit cold-turkey is in the cards for most of us, but consider this: Honestly take note of how much time you spend fretting over gear (I was going to say "instead of fretting your guitar," but I won't sink that low...). Be especially aware of the time spent on ridiculous trivia — I'll bet there's more of it than most of us would care to admit. Put it into a real-world perspective: How much is all this time, energy, and passion going to affect your music or productions? More importantly, how much does it *take away* from your music or productions?

Now take just a small amount of that time and passion, say 10 percent, and put it back toward the final results of your labors. Spend that time practicing your instrument, working on recording technique, placing a microphone more carefully, tweaking a mix, even listening to the work of one of the masters. Where are you getting the most benefit?

So am I suggesting that we should all stop worrying about gear, never think about what we should buy next, and not worry about specs and comparisons? Nah, what fun would that be? What I'm suggesting is that we try to keep these gear issues in perspective and that we put the focus back on what really matters: the music.

Let me know what you think; you'll find my direct e-mail address below. And, no matter what, keep pushing those faders!

—Mitch Gallagher
gallagher@psn.com

Since we introduced the MX-2424 hard disk recorder, there has been a lot of speculation about its price (which is so low it seems too good to be true).

So we get questions. Like...

"24 tracks is an upgrade?" (No, it's 24 tracks right out of the box.)

"24-bits is an upgrade?" (No, all the bits are there too.)

"Do I have to pay extra for inputs and outputs?" (No. At $3,999 estimated street price* you get a full set of 24 TDIF-1 *or* ADAT® optical digital inputs and outputs — plus an assignable stereo AES/ EBU - S/PDIF pair. For a little more you can get 24 channels of AES/EBU digital I/O, or analog — or both digital and analog!)

"Does it need an external computer?" (No. The MX-2424s front panel has a full set of professional transport, editing, and track assignment controls, including a shuttle/ scrub knob. So you don't have to have a computer to run it. But — if you happen to own a Mac or a PC, you can take advantage of the digital audio editing and control software that comes standard with each MX-2424 to do even more. Your choice.)

"Before I start recording do I need to buy a monitor, a keyboard, or a hard drive? Or anything else?" (No. Nyet. Nope. Not at all. Just hook up power and start recording.)

So let's make this as plain as we can: The MX-2424 is an amazing, full-featured professional 24-track digital recorder. And there's never been anything like it at this size or price.

Its sonic performance is outstanding. Lots of companies claim 24-bit 48k performance, but only the MX-2424 is part of TASCAM's M Series family of multitracks — the products chosen for their sonic performance by such discriminating facilities as Skywalker Sound, Universal Studios, and 20th Century Fox.

$3,999 ESP*.

Superior reliability is guaranteed. The MX-2424 was designed from the bottom up to be a great recorder, and nothing but a great recorder. Its processors and circuitry are fully optimized for audio - not video games, spreadsheet software, or surfing the web. And isn't that absolute focus and rock solid performance exactly what your music deserves? Over the last three decades we've designed and built literally millions of professional recorders and recording systems; the MX-2424 is the culmination of everything we've learned.

So easy to operate, you could do it blindfolded. Of course that way you'd miss the great light show from the 24 tracks of level metering and channel status displays... but the real point here is simplicity. When you want the MX-2424 to start recording, just reach over and press REC + PLAY (just like a traditional tape recorder). In a fast-paced production environment, you can record to hard drives that mount into standard Kingston® carriers and plug into the front panel drive bay. Just pop in a new drive at the start of each session. It doesn't get any simpler than that.

It's a complete professional hard disk multitrack in a portable, affordable, rack-mount box. You can plug it in, turn it on, and start recording.

Really.

The power to meet your needs. A standalone MX-2424 is an incredibly powerful unit, with enough internal hard disk capacity to hold about 45 minutes of 24-bit 24-track audio. The MX-2424's Fast/Wide SCSI port lets you connect up to 15 external drives and record directly to all of them. And if you need more than 24 simultaneous tracks, just add additional MX-2424's. Up to 32 MX-2424's can be locked together in *sample accurate sync* to act as a single recorder.

Professional recorders need to interface with increasingly complex systems.

✔ It provides video and time code lock capabilities as standard features, making it easy to integrate with external workstations.

✔ It resolves to AES/EBU, S/PDIF, word clock, TDIF-1, ADAT optical, SMPTE Time Code (LTC), and video, and chases MIDI Time Code.

✔ Available Input/Output modules include TDIF-1, AES/EBU, ADAT optical, and analog.

✔ Back panel ports include Fast/Wide SCSI, ethernet, MIDI, RC-2424 remote, and TL-BUS!

Extend your reach — Want a remote control? Get the one that's made to take advantage of the power in your MX-2424. The RC-2424 remote is a powerful, professional multi-machine controller with all of the MX-2424's front panel features, plus macros and more.

MX-2424 shipments are about to start, and there is already a waiting list. To get yours sooner instead of later, contact your authorized TASCAM dealer!

**So... what's this Estimated Street Price?* Instead of quoting you some meaningless "List Price," ESP is what we expect typical U.S. customers to actually pay for an item. It gives you a better way to compare value when you shop.

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CIRCLE 58 ON FREE INFO CARD

MINI-MASTERERS

As the proprietor of a small music production studio that offers digital domain mastering services, I read "Mastering By The Masters" (*EQ*, March 2000) with great interest.

The article contained lots of useful information. I was particularly gratified that Doug Sax felt that some of us little guys occasionally produce good work.

However, I feel that the article left an overall impression that the mastering services offered by smaller facilities are rarely an appropriate choice. In that regard, I must respectfully disagree.

Consider the huge amount of musical product that finds its way onto CDs: song, artist, and studio musician demos, client presentations, independently produced albums, product for subsequent use in TV and film scoring and production libraries — all of these are routinely delivered on audio CDs or CD-ROMs, and can benefit from mastering services.

In my case, I bring over a decade of digital domain editing and mastering experience to my work. I use two well-appointed, multi-kilobuck DAW systems, good A/D and D/A converters, good acoustic isolation, a respectably hi-fi monitoring system, and a carefully researched CD burner and CD-R media. And, yes, a Finalizer. Used carefully, it can sound really good!

The high-end mastering studios in my area charge five times my rates, to the dollar. There is no question that they are better staffed and equipped than I am. Their orientation is toward high-budget, large distribution product where the anticipated revenues can offset higher production costs. In that rarefied world, it makes sense to spare no expense to get the best possible master. But that is not everybody's reality: For the folk musician, who, as a labor of love, has made a purist record of limited commercial potential, mastering costs at a high-end studio could easily exceed his or her entire recording budget, and squash any hope of earning back his/her production costs. For a jingle house, that has to present the spot you are mixing at 11:00 AM on CD at a 1:00 PM meeting, there simply isn't time for a separate mastering session.

Of course, the increased availability of "desktop" mastering equipment does not mean that everybody knows how to use it. Anyone thinking of using a lesser-known mastering studio should carefully evaluate some of the studio's product before committing themselves.

However, there are a number of us out there working very hard to make a superior product at a reasonable price. I believe that we produce fully professional work and provide a valuable service to the appropriate clientele.

Jon Gordon
Jon Gordon Music Production
New York, NY

CLASSICAL KUDOS

Lots of brickbats are thrown at magazine articles, but Mike Sokol, in April 2000's classical instrument mic techniques, deserves a big bouquet. As a former touring FOH mixer, I have "graduated" to classical music, TV commercials, and lots of video EFP. For many years, I have cajoled, argued, and pleaded with fellow audio engineering types to study and understand the differences in the miking and recording between rock and classical.

At last, Mr. Sokol states what I have long urged in a clear and readable form. *Bravo!* As one with over 12 years of doing classical for bucks, I can say that he is absolutely right in every statement he makes. If any budding recordist wants the truth on classical mic techniques, read this, heed it, and save it for repeated reference.

Allan Soifer
via Internet

HELPING HANDS

After working as an audio engineer for Educational Television for 17 years, my husband, Woody, has learned the many areas of the music and recording industry; working on such projects as PBS' *Great Performances* and *American Playhouse Theatre*. As a result of post-polio problems, he took an early retirement.

Through the years, he has managed to assemble a fully equipped project studio in our basement.

Because Woody cannot do such things as yard work, every year he runs an ad in our local musicians' guide saying he will trade studio time for yard work. And every year we get a barrage of calls from musicians who are willing to do any work around our home because they have no other resources to get their work recorded.

This time of year our house is filled with extremely talented musicians who tell me how disappointed they are in the steep rates commercial recording studios charge. Many of them have also told me that, without Woody's graciousness, there would be no other way to break into the music industry.

Is it possible for recording studios to have compassion and come up with similar ideas to help starving musicians as they struggle to get started?

Anonymous
via Internet

FAME'S HISTORY CORRECTED

The March issue's cover story on Jimmy Johnson contained several inaccuracies. First and foremost, despite what was reported in the story, FAME Recording Studios of Muscle Shoals never "decided to close its doors." On the contrary, the world-renowned studio remains open and thrives, having produced a near endless stream of hit records that spans five decades. According to Jimmy Johnson, what actually happened is that, in 1968, FAME founder and owner Rick Hall decided to do only in-house productions after signing a label deal with Capital Records. He therefore closed his doors to outside work for a time.

Also, in the article, Rick Hall was referred to as a "local man," which could be taken as a slight toward him — that was certainly not the intention. The photos on pages 62 and 64 should have been credited to FAME Recording Studios. Finally, the caption for the group photo on page 64 was less than accurate: Those in the photo included Jimmy Johnson, Dan Penn, Spooner Oldham, Wilson Pickett, Rick Hall, Jerry Wexler, and Charles Chalmers, not the MSRS as was stated in the caption.

Our most sincere apologies to Jimmy Johnson, FAME Recording Studios, and the family of Rick Hall for these errors.

WRITE TO US

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JUST A PHASE

Q *I have a small studio that specializes in spoken-word product. I've been converting some files to MP3 in preparation for posting them to the World Wide Web. I've set up Audio-Grabber with Blade Encoder, and it seems to do a reasonable job when encoding at128 kbps and 44.1 kHz. I can also encode decently at 112 kbps/44.1 kHz. However, if I try any lower than this, I get what sounds like a phase shift. I am sure it's not, but this seems to be the best way to describe the way the files sound. Am I doing something wrong? The portable MP3 players all list 64 kbps on their specs as if it is a usable sampling rate. Surely the world must be able make this acceptable, but when I do it the files become unlistenable. Do I need different software, more lessons in MP3, or a swift kick in the posterior? I have a lot of files to encode over the next six months, and could use the help.*

Fred
via Internet

A If it sounds like your files encoded at 112 kbps have phase problems, then perhaps they do! Sometimes the simplest answer is the right one. Most MP3 encoders will automatically submix down to mono at a particular bit rate (typically around 48 or 64 kbps). If your raw audio has phase problems (God forbid!), it is possible you aren't noticing them until this happens. The easy way to test is to manually mix the raw audio to mono and see how it sounds. Good luck!

FezGuys
EQ Columnists

LOST IN SPACE...

Q *This question came up recently in a conversation with a client, and I was unable to answer it. If a minute of stereo recording at 44.1/16-bit requires 10 MB of disk space (I've been under this assumption since the dawn of the digital age), how can a 650 MB CD-R hold 74 minutes of music?*

Greg Hilfman
via Internet

A When CDs were first created, they weren't meant to be used for data — only for audio. The specification required that 74 minutes of 44 kHz stereo data be able to be placed on the disc. So if we look at the amount of space needed for this we can calculate it as follows:

2 (channels) x 2 bytes per sample x 44,100 samples/sec = 176,400 bytes per audio second

60 seconds x 176,400 bytes per audio second = 10.58 MB per audio minute

74 minutes x 10.58 = 783 MB of necessary space for audio data.

(There's actually even more than this data on a CD at a lower level.) If we're creating a disc that holds this much data, we can assume that there's no way we're going to have all this data be error free and that, over time, some of this data will be lost due to wear and tear and handling of the disc. Thus, there's additional data on the disc that is used for error detection and error correction. This redundant data will actually allow a CD player to correct errors on the disc if they are not too bad. If the errors aren't correctable, they will be detected and the audio player has a choice at this point: It can interpolate the audio if the detected errors are relatively small. Or, if the disc is seriously flawed, the player can mute the output so it doesn't blast the users' ears away by playing nasty error data.

So continuing forward, small errors on a disc are not a big deal for audio data since the player can interpolate or mute the outputs and, for the most part, the user can still enjoy the disc. The problem is, if we decide to use this same disc for computer data, we can't accept any errors in the data. Imagine if your computer decided to "interpolate" your accounts payable for you if it ran across an error. Suddenly you find that your electric bill went from $50 to $5,000 and you didn't even know it. When the specification for using audio discs as computer data discs was created, the need was apparent for additional error detection and correction to be added. So to accomplish this, part of the available audio data space was sacrificed.

So the question is how did we get from 783 MB to 650 MB? The answer is really quite simple: Each second on a CD audio disc is subdivided into 75 frames.

Thus the MM:SS:FF (minute-second-frame) terminology used when working on CDs. If we divide the amount of data needed for one second of audio (176,400 bytes) by 75, we see that each frame of audio is 2,352 bytes long. The number 2,352 is not actually a very nice number to deal with in the computer world, so the decision was made to use only 2,048 bytes of data in each frame for computer data discs. This left 304 bytes to be used for addition error correction/detection as well as information for seeking accurately to any given frame on the disc.

So for computer data:

2048 bytes x 75 frames/sec = 153,600 bytes per second

60 seconds x 153,600 = 9.0 MB per minute

74 minutes x 9.0 MB per minute = 650 MB

(When calculating this, use a MB as 1,048,576 bytes as opposed to 1,000,000)

This is probably more information than you wanted, but it is the "real story."

Monty Schmidt
Sonic Foundry
Madison, WI

ASK US

Send your thought-provoking questions to:
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THREAD OF THE MONTH

As seen on the EQ Forums at www.eqmag.com

I like most of the sounds I get in my studio, but am still struggling with the tight, "phat," and focused bass guitar that drives pop records.

I play a Yamaha active humbucking bass into a SansAmp Bass Driver DI and then direct into the converters on my 02R. I have at my disposal the Waves Gold Bundle (I use their Renaissance Comp for most compression chores), a Line 6 POD, and the comps and EQ in the board.

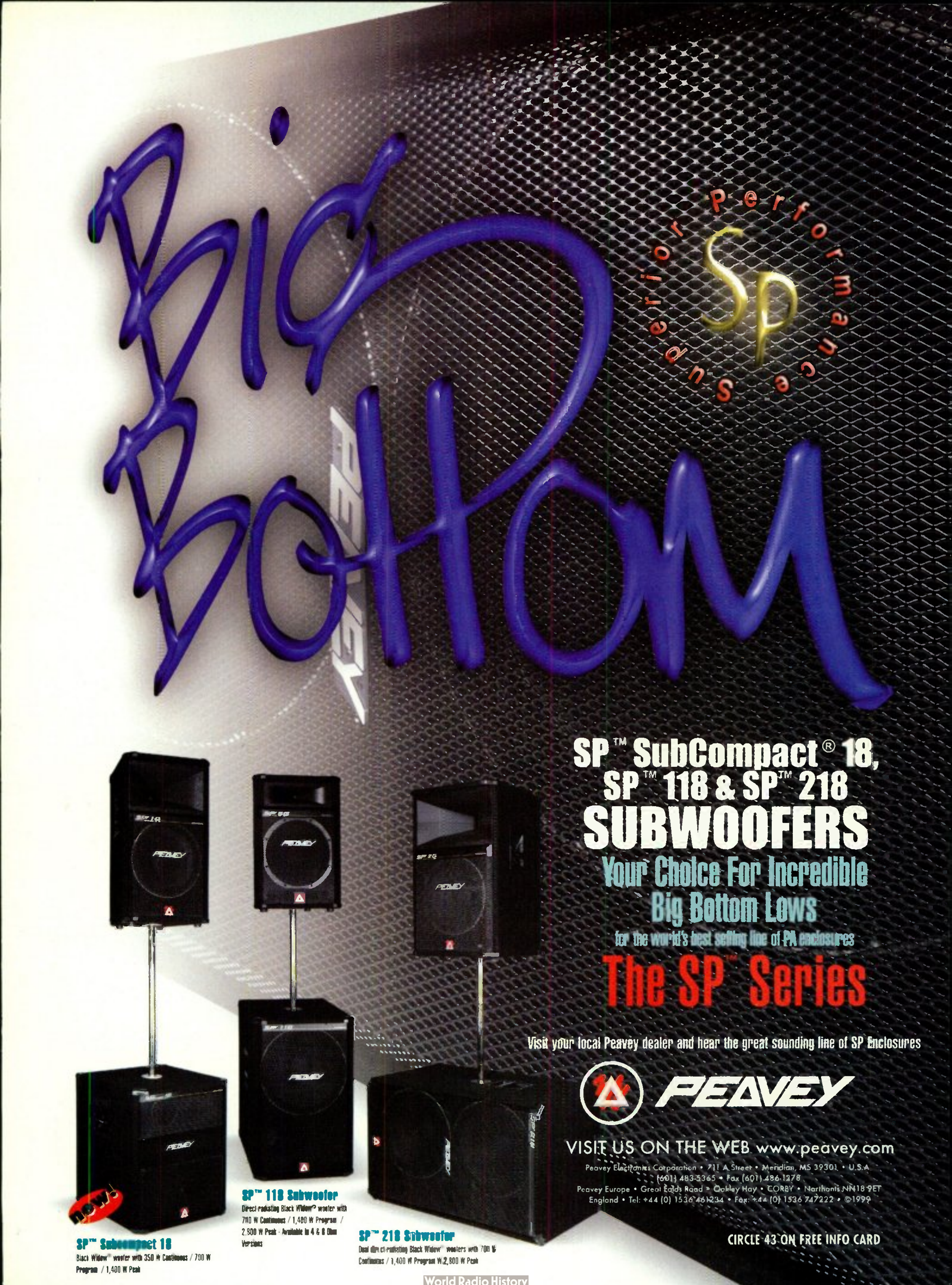
What am I doing wrong? Any suggestions as to gear, amps, mics, compression, and so forth? Can you even get that sound recording digitally (24-bit/44.1k)? Thanks in advance for any insight....

—Jim Bordner, Gravity Music

The bass is actually easier to record than most people think. Of course, it is imperative that you have speakers that have a fairly accurate low end and a listening environment that is not experiencing wierd standing waves. Most people work too hard on the bass before they have to. When recording the bass, I typically take it direct, maybe even routing into the recorder flat right off of the direct box (sometimes active, sometimes passive — depends on the instrument). Same thing with the amp — get a good clear, clean sound coming out of the cabinet (if that's what you are going for). I try to get it recorded as full as I can. Then I'll do some trimming with EQ, maybe some compression just to protect the tape from peaks, or to try and even it out if the player's E string, for example, is way too loud, or another string is way too soft.

Most people start EQing and compressing too hard and too soon. Start with the sound out of the axe, and get the amp sounding clear and full. The real truth is that the sound originates from the musician. If he (or she) is happening, you are. With most of the great players that I work with, you basically just have to push the fader up to set the level. Same axe, different player, the bass can be too boomy, scratchy, uneven, etc. It's funny, my engineering got a lot better as I worked with better musicians.

—Ed Cherney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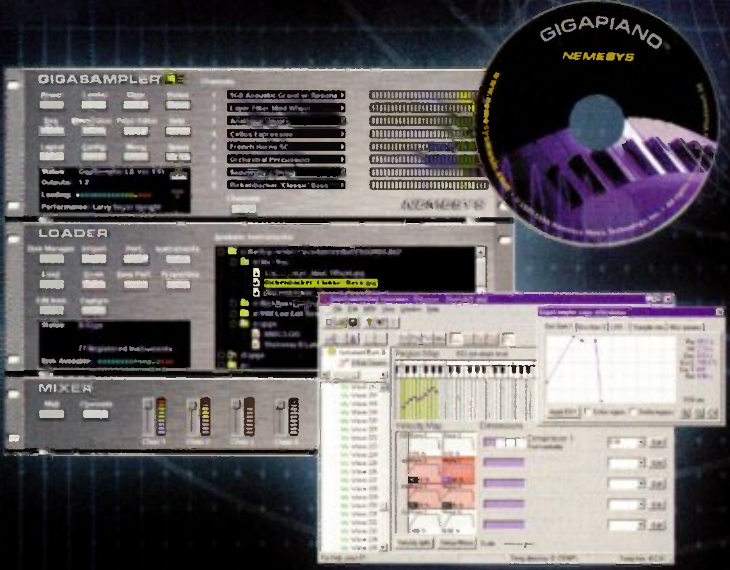

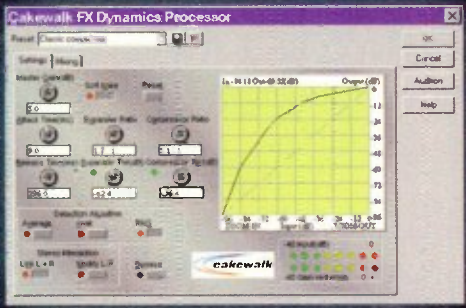

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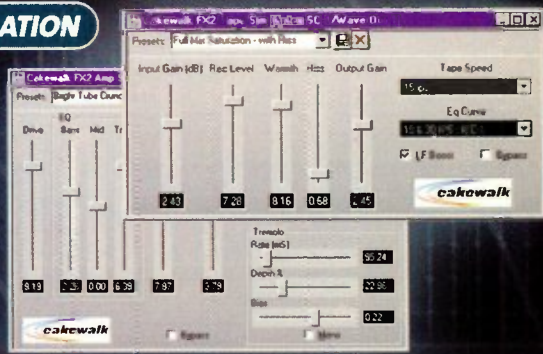

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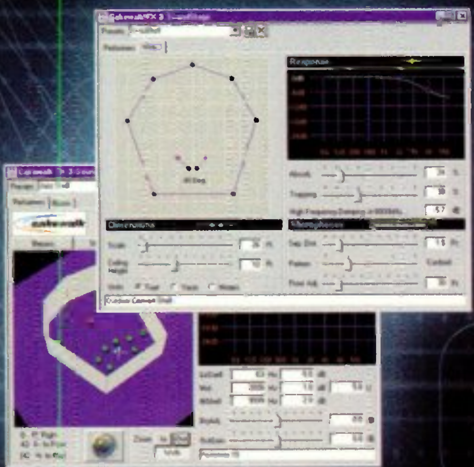

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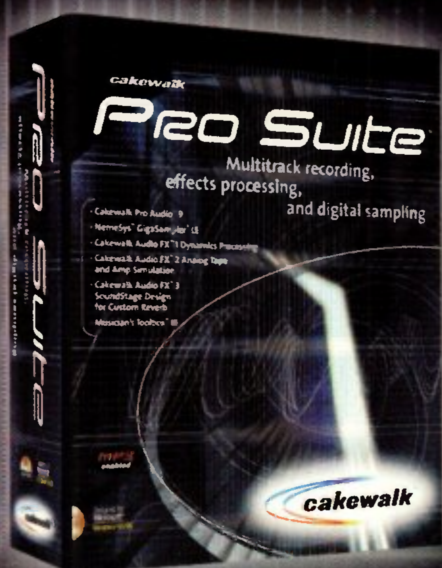

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CIRCLE 36 ON FREE INFO CARD

EQ NEWS

MAY • 2000

Dreamhire Announces its "Virtual" and "Real" Rack Rental Programs

Dreamhire Professional Audio Rentals has launched two new programs to better serve its ever-increasing clientele. "What we want to do is streamline the rental process for our high-end clients as well as offering them some new services," says David Olivier, sales and marketing manager at Dreamhire's New York City location.

Often, an engineer's or producer's sound is based on years of experience that usually results in a laundry list of specific outboard gear for certain session requirements, *i.e.*, tracking, overdubbing, or mixing. What the "Virtual" rack program does is keep a database of such requirements and applications so that it can be easily assembled, tested, and sent out to a session.

"If we know who uses what, then all you have to do is call us and say, 'Send so and so's 'overdub rack' to Right Track,' or 'Whomever's 'mixing rack' to Battery,' and so on. The discounts involved in signing on to this program instead of renting piecemeal every time can be substantial," notes Olivier.

The "Real" rack program is for an engineer or producer who has a rack they own and lug around to every session. Dreamhire will provide the cartage, storage, and, best of all, the maintenance, so, for example, if the rack comes back with a note, "VU light out on LA-2A," it will be fixed by the next time it goes out, plus if a piece of gear goes down on a session it can be replaced with a rental at a substantial discount while it's being repaired. "The fact that this will enable our customers to have their own gear impeccably maintained, stored, and carted makes this program an incredible value," adds Olivier.

For more details, call 212-824-1309.

DirectConnect Adds Sampling and Synth Power to Pro Tools Systems

Digidesign has announced the availability of DirectConnect, a new host-based audio streaming plug-in for Pro Tools|24 Mix systems. DirectConnect allows audio data from a software synthesizer or sampler to be routed directly into the Pro Tools mixer. Each input stream from the host application can be independently recorded, automated, and mixed within the Pro Tools environment.

With DirectConnect, Digidesign has enabled the DAW-based studio to become a creative instrument in its own right. A number of third-party applications support DirectConnect, including Koblo's Studio 9000, Reaktor by Native Instruments, Alkali by Audio Genetics, and Retro AS-1 and Unity DS-1 by Bitheadz, with ReBirth by Propellerheads coming soon.

DirectConnect features streaming of up to 32 channels of 24-bit audio directly into Pro Tools, integration of third-party applications, real-time automation and audio processing, and total recall of the entire DirectConnect setup (if supported by the third-party application).

DirectConnect requires Pro Tools software version 5.0 or later, and is available now as a free DigiRack plug-in for MacOS-based Pro Tools|24 Mix systems. It will be included with the Pro Tools 5.01 software release. Free downloads for v5.0 users are available on Digidesign's Web site (www.digidesign.com).

Mackie Completes EAW Acquisition

Mackie Designs, Inc. has completed the purchase of Eastern Acoustic Works, Inc. (EAW).

"We are excited about the opportunities afforded us by this acquisition," states Greg Mackie, founder and chairman of Mackie Designs. "EAW is a leading brand in the professional audio industry and is recognized as a world-class developer and manufacturer of high-end professional audio equipment."

Multitrack Software Free from recordLab

Internet TapeDeck, the digital audio recording software from recordLab Corp., is available as a free download from www.recordlab.com. Geared toward amateur musicians and home recordists, the application is said to emulate the basic recording functions of a hardware-based recorder costing several hundred dollars or more.

Internet TapeDeck features simultaneous playback and recording of up to eight tracks, punch-in/punch-out capability, and a three-band equalizer on each track. While the software includes recordLab's Reverb DirectX plug-in at no charge, another dozen or so plug-in effects will be available to users at a cost of $10-$30 each.

Internet TapeDeck allows users to link directly to over 300 instructional articles on the recordLab Web

site, many of which are accompanied by audio and video tutorials. And soon, recordLab plans to offer pay-per-view classes, the first of which is being taped now at the University of California at Los Angeles.

POD Pro from Line 6 Shipping

POD Pro features 32 amp models and 16 cabinet models that can be used as is, or mixed and matched to create custom sounds. Thirty-six programmable channels are provided for storage of custom amp/cabs along with 16 digital effect combinations including choruses, flangers, rotary speaker, delay, reverb, tremolo, and compressor.

With 24-bit AES/EBU, S/PDIF digital out, word clock, 44.1/48 kHz operation, unprocessed guitar output, and line level input, POD is well-equipped to perform in professional recording environments. POD Pro also features dual-mode XLR direct outs with AIR — Line 6's proprietary technique for simulating the complex interaction of microphones and speaker cabinets in both recording and live situations.

Equipped with ToneTransfer, POD Pro sounds can be transferred to any other POD Pro, POD, or Flextone II amp, allowing sounds to be ported easily from rehearsal to studio to stage and new sounds to be acquired through the Line 6 ToneTransfer Web Library. Retail price is $799.

For more information, contact Line 6 at 805-379-8900 or visit www.line6.com.

Telex Establishes Brand New State-of-the-Art Production Facility

Telex Communications, Inc. has purchased the former Arrow Automotive factory in Morrilton, AR and has plans to create a state-of-the-art production facility designed to optimize Telex's manufacturing capacities, realize the benefits of the latest production technologies, improve customer satisfaction, and fully support the continued growth of its Electro-Voice speaker business.

Currently, Telex manufactures speaker systems at two plants: one in Austin, TX and one in Newport, TN. These operations will be consolidated into the Morrilton site to reduce inventories, overhead, shipping costs and cycle times, eliminate duplications in plant operations, and to improve Telex's ability to react more quickly to changes in customer demand.

At the same time, wired microphone products currently manufactured at a facility in Sevierville, TN will be integrated into other existing facilities to make better use of manufacturing capacities and a streamlined supply chain, while eliminating unnecessary duplications and overhead.

The Morrilton facility, scheduled to begin initial production in June 2000, is expected to create approximately 225 new jobs in the Morrilton area and will sufficiently support the company's forecast of doubling its speaker system production over the next three to five years. The facility will handle all aspects of speaker production, including cabinet manufacturing, system assembly, and system testing.

For more information, call Telex at 1-800-667-3968 or visit www.telex.com.

Dick Swettenham of Helios: 1927-2000

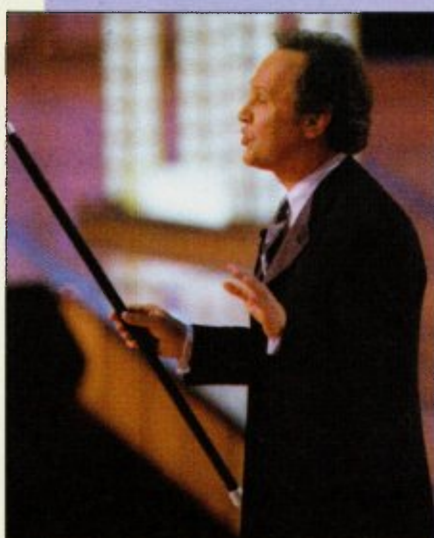
Dick Swettenham — audio pioneer and founder of custom console manufacturer Helios - succumbed to cancer on April 9, 2000 in a Bristol, UK hospice. He remained lucid to the end; and only a week before his death, he was sketching new designs.

The remaining members of the company will be continuing Swettenham's work, partly drawing on his old designs; but since he was always focused on the future and on bringing new ideas to the pro-audio field, Helios will do its best to carry Swettenham's vision and innovative concepts into the future. Swettenham was particularly keen on the idea of a digitally controlled analog console and wrote a number of magazine articles about the concept.

A funeral was held at Canford Crematorium, Bristol, UK on Friday, April 14. For more information, visit www.helios-audio.com.

David Was and Event Electronics at the Oscars

Renowned producer/composer David Was recently served as orchestrator and band member for the 2000 Academy Awards Oscar® telecast. Along with this year's musical directors, Burt Bacharach and Don Was, David composed and produced the musical underscore for the montage packages that accompanied the nominations for major awards. He recorded and mixed the music cues for the show exclusively on Event Electronics PS6 powered monitors, which are reviewed in this magazine on page 116.



Kind of Loud Technologies Announces Dolby Digital and DTS Encoder Software for Pro Tools

Kind of Loud Technologies has announced the development of two 5.1 encoder software packages for Digidesign's Pro Tools — SmartCode Pro will be available in both Dolby Digital and DTS versions.

Kind of Loud Technologies provides a complete suite of surround production tools that allow Pro Tools users to mix in 5.1 surround. SmartPan Pro is a panning plug-in that brings true discrete 5.1 surround panning to the Pro Tools platform. SmartCode Pro allows Pro Tools users to record, edit, and encode their surround mixes completely within the Pro Tools environment. Until now, Pro Tools users have had to rely on costly dedicated hardware to encode their 5.1 mixes; to date there are few software encoders and none designed for Pro Tools.

"Pro Tools users are known to be 'early adopters' of new technologies and, not surprisingly, they have sought 5.1 encoding/decoding software for some time," says Suz Howells, VP marketing and sales for Kind of Loud. "With SmartCode Pro, Digidesign's customers will have the technology to create and encode 5.1 mixes entirely within Pro Tools."

SmartCode Pro is an AudioSuite program that allows Pro Tools users to preview, in real time, 5.1 mixes created by the Pro Tools software and the SmartPan Pro plug-in; then encode and decode the mix to create a 6-channel surround master. SmartCode Pro/Dolby Digital ($995 MSRP) and SmartCode Pro/DTS ($1995 MSRP) are expected to ship Q3, 2000.

Both products will be sold by Kind of Loud Technologies through its worldwide resellers network and via its Web site, www.kindofloud.com.

Fabulous Thunderbirds Mixed to DVD-Audio

Engineer/producer and EQ Forum moderator Ed Cherney recently remixed a live Fabulous Thunderbirds concert recorded to a Euphonix R-1 disk-based digital multitrack with enhanced 24-bit/96 kHz fidelity. Cherney worked at L.A.'s Soundproof Studios with co-producer Kim Wilson and executive producer Tom Consolo to prepare stereo and 5.1-channel surround-sound mixes on the facility's System 5 high-performance digital console. Assisting on the session was engineer Martin Kloiber.

The all-digital material was tracked during the inaugural "One Night in L.A." concert at The Complex, West Los Angeles, which was simultaneously streamed to the Internet in high-definition audio and video. The February concert was also videotaped on HDTV for a DVD-Video release with a 5.1-channel soundtrack.

"We recorded not only the 60-minute live Web broadcast," says Cherney, "but also an additional 40-minute concert in front of the invited audience. At Soundproof Studios, we used the Euphonix System 5 to make a number of all-digital stereo and 5.1-channel mixes for a conventional CD in addition to a DTS-encoded 5.1-channel CD and a DVD-Audio release."

All material from the two sessions was tracked and mastered to R-1 at 96 kHz sample rate in full 24-bit enhanced resolution.

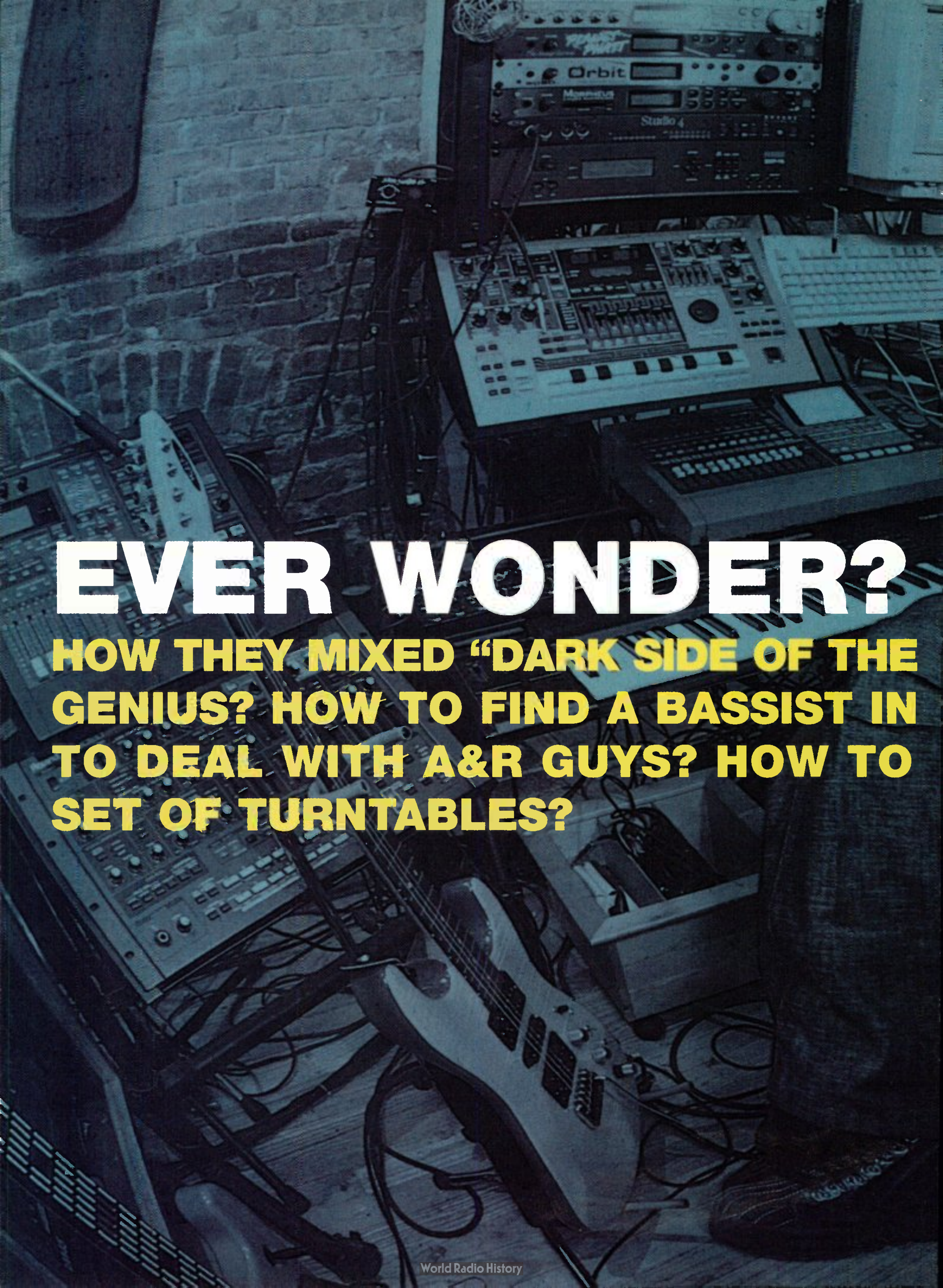
"We did no overdubs," he continues. "For safety, we recorded the various pre-concert rehearsals and run-throughs in 24/96. That allowed us to perform one or two minor touch-ups, either using the R-1 to transfer between hard drives in the digital domain, or with a 24-bit Pro Tools workstation."

Earjam Inks Two Dot-Com Deals

Earjam.com today announced a pair of partnerships that will involve both the company's music player software and its Internet guide to online music.

One alliance will link MP3.com's Retail Music Service Program (MSP) to Earjam's Internet Music Player (IMP) software. Set to launch May 1, MSP will deliver to subscribers music programming via the Internet. MP3.com said its service will cost up to 50 percent less than current business-to-business services that deliver programming on tapes, CDs, or over satellite.

Separately, Earjam and Listen.com said they will create a co-branded site providing access to Listen.com's guide to online music, featuring original reviews of more than 60,000 artists with legally available music from 1,000 Web sites.



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EQ PRODUCT VIEWS

MODEL T

Shure has re-launched its VHF wireless T Series microphones, combining receiver upgrades and a new design with an upgraded accessories package. The most significant update is the redesigned receiver, which now includes Shure's Predictive Diversity technology. Four application-specific systems comprise the T Series: the Vocal Artist ($380 for the BG3.1 non-diversity, $560 for the SM58 diversity); the Guitarist ($260 for non-diversity, $370 for diversity); the Presenter and Headset systems (sharing pricing at $360 for non-diversity setups, $480 for diversity); and the Body Pack (listing for $320 non-diversity and $440 diversity). Call Shure at 847-866-2200 or visit www. shure.com. Circle EQ free lit. #116.

TWIN WONDER

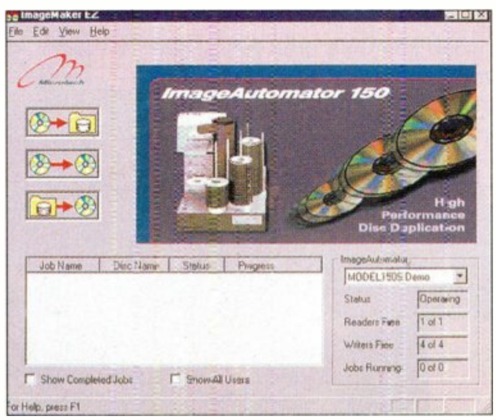
Denon's new DN-1800F dual CD player sports bright LEDs over each drive, Pitch Bend (±18 percent) which may be controlled via the jog wheel in addition to the standard dedicated buttons, and a "Plat" function that can be used to speed playback up to +40 percent when the jog wheel is turned clockwise and slow it down to –50 percent when turned counter-clockwise. The machine also boasts a new "Cue Stutter" mode, which plays momentary audio as the Cue button is pressed down and then recues to the original point when released. In addition to traditional analog outputs, digital S/PDIF outputs enable the dual CD player to be used for direct dubbing to a digital recorder (MD, DAT, CD-R, etc.) or for direct digital connection to an outboard processor or mixing console. For more information, call Denon at 973-396-0810 or visit www.del.denon.com. Circle EQ free lit. #117.

SMALL WONDERS

KRK has announced the introduction of two new additions to its V-Series line, and two additions to its S-Series subwoofer line. Responding to the needs of the professional recording community to develop a line of small monitors that are good enough to mix on, KRK has created the V4 mini monitor and S8 mini subwoofer. The V4 is a bi-amplified, two-way monitor that uses a 4-inch coated woofer and a 1-inch titanium dome tweeter. It includes an active crossover, which sends 30 watts to the woofer and 15 watts to the tweeter. The S8 is a 100-watt powered subwoofer that includes an 8-inch coated paper woofer, variable crossover, two-channel XLR balanced inputs, and two-channel XLR line level outputs with fixed 80 Hz 4th order highpass. For more information, call KRK Systems at 714-373-4600 or visit www.krksys.com. Circle EQ free lit. #118.

QUICK BURN

Microtech Systems has enhanced throughput and cut handling time on its ImageAutomator CD-R production systems through a simplified user interface and the addition of asynchronous duplication mode of operation. Called ImageMaker EZ, the new software release accounts for as much as an 80 percent increase in production speed, depending on the number of drives in the system. Now each drive on a multi-drive Microtech system can start a duplication process independent of other drives, as opposed to batch mode or synchronous operation where a drive is loaded with a blank CD-R and waits until the rest of the drives are loaded before the job can begin. Users can also choose between CD-to-CD copying, mastering an image to the hard drive for later copying, or making copies from a hard drive image. For more information, call Microtech Systems at 800-223-3693 or visit www.microtech.com. Circle EQ free lit. #119.

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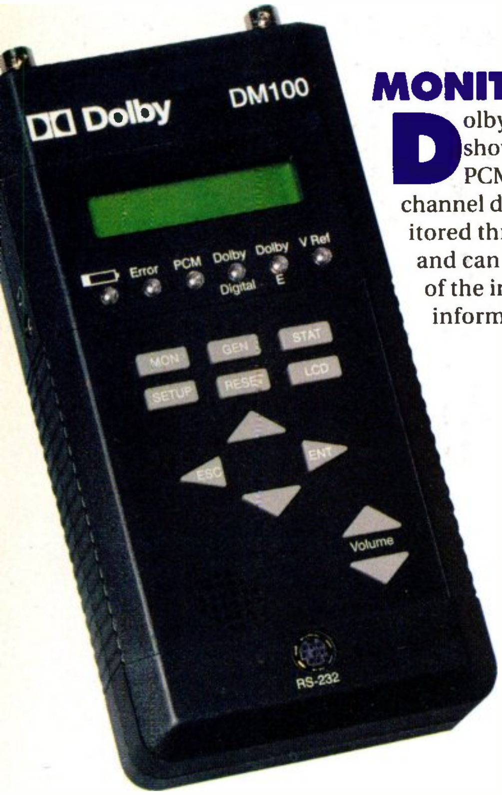
EQ NAB VIEWS

Hidden amidst the overwhelming array of video displays at the recent **National Association of Broadcasters (NAB)** convention in Las Vegas, some notable audio products made their debut. Here are a few of the audio highlights.

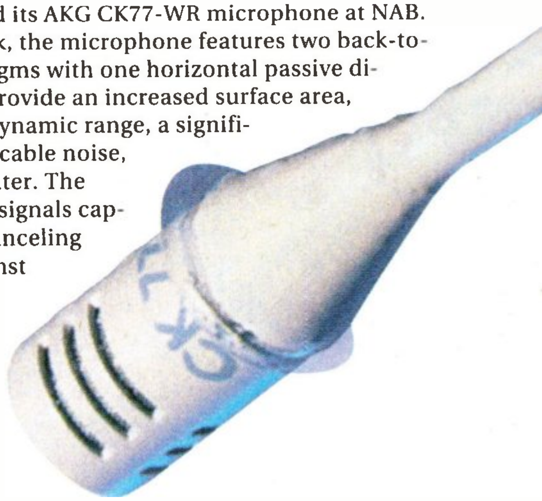


MONITOR IN YOUR PALM

Dolby Laboratories debuted and demonstrated its new DM100 portable audio monitor at the show. The DM100 is a hand-held diagnostic tool for monitoring Dolby Digital, Dolby E, and PCM bitstreams. Through headphones, users can monitor individual channel pairs or a two-channel down-mix of any multi-channel program, while the sum of the two channels can be monitored through a small built-in speaker. Test bitstreams are stored in internal non-volatile RAM, and can be changed in the field via software update. A pass-through mode allows modification of the input signal's AES channel status bits before passing to the output connectors. For more information, call Dolby at 415-558-0200 or visit www.dolby.com. Circle EQ free lit. #120.

MULTIPLE MULTICOLORED DIAPHRAGMS

In a move that reinforces AKG Acoustic's advancements in sound technology, the company launched its AKG CK77-WR microphone at NAB. Available in beige, white, and black, the microphone features two back-to-back vertically aligned active diaphragms with one horizontal passive diaphragm. The multiple diaphragms provide an increased surface area, which provides lower noise, greater dynamic range, a significant improvement in the rejection of cable noise, and resistance to perspiration and water. The dual-diaphragm transducer adds the signals captured by the two diaphragms while canceling noise caused by clothes rubbing against the microphone. For more information, call AKG at 615-360-0499 or visit www.akg-acoustics.com. Circle EQ free lit. #121.



PURELY PROCESSING

AMEK unveiled its Pure Path range of rack-mounting signal processors at NAB 2000. Leading the range is the Channel-in-a-Box or CIB, essentially the input channel of the AMEK 9098I audio mixing console, combining microphone preamplifier, line input amplifier, high- and lowpass filters, four-band equalization, and a fully-featured compressor. DIB comprises eight separate line-in/line-out transformer-coupled line amplifiers. Completing the range is the Stem Compressor, a multichannel compressor/limiter designed for multi-stem mixing applications and which includes eight digitally controlled analog processors. For more information, call AMEK at 818-973-1618. Circle EQ free lit. #122.

We've done the formatting so you don't have to!

BASF introduces a Formatted DTRS Master that saves you time and head wear.

Now you can get the world's best-performing DTRS Master already formatted - saving you time in the studio and wear and tear on your DTRS recorder head. BASF's new Formatted DTRS Master lets you record to DTRS immediately, without having to format the master tape.

FORMATTED

PLUS, get all the benefits of a BASF DTRS Master tape that's designed for all of the rigors of digital multitrack recording.

- consistently delivers lower error rates than any other major brand on the market;
- specially constructed ABS shell providing precision tracking, virtually eliminating dropouts caused by static or dirt;
- convenient sliding erase-lock tab offering a simple means to safeguard your masters.

Available in 44.1 kHz, 113-minute length only. Compatible with Tascam's DA-38, DA-88, DA-98 and Sony's PCM-800.

EMTEC Magnetics

EMTEC

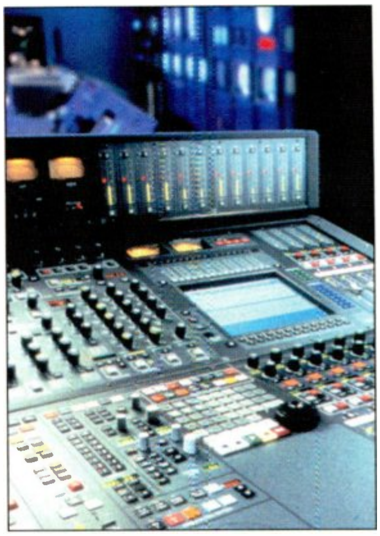
EMTEC Pro Media, Inc. US 1-888-295-5551 Canada 1-800-817-4478 or visit our web site at www.emtec-usa.com

CIRCLE ... ON FREE INFO CARD

EQ NAB VIEWS

LIBRA LAS VEGAS

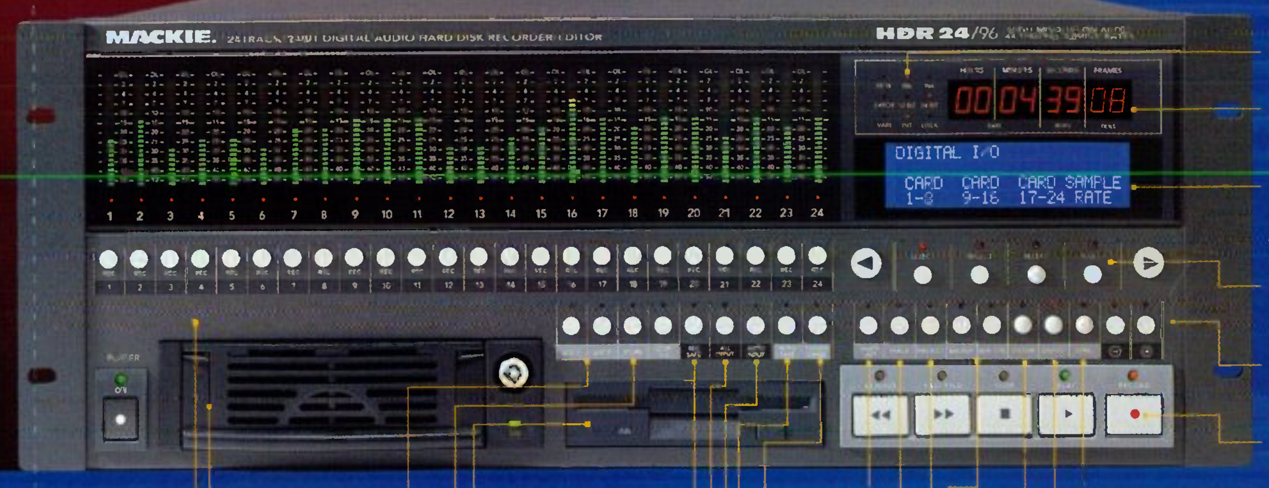
AMS Neve launched the Libra Live Series II digital broadcast console at this year's NAB Convention in Las Vegas. Designed for versatile operation in a broadcast production environment, it provides the advantages of digital control and a digital signal path with ease of use of a conventional analog console. Features of the Libra Live II include: mix-minus, GPI, and other broadcast-specific facilities, full processing in every channel, rapid recovery control system, multi-format surround sound options, 24-bit analog and digital interfacing, and suitability for live and multitrack production in the studio or OB truck. For more information, call AMS Neve at 44/128-245-711 or visit www.ams-neve.com. Circle EQ free lit. #123.



PORTADISC & THE FAT MAN

NAB 2000 also saw the launch of the HHB Portadisc portable MiniDisc recorder, and the Radius 3 "Fat Man" stereo tube compressor. The Portadisc adds a full complement of professional features including balanced XLR microphone inputs with switchable 48V phantom power and limiter, and a USB port. The new Radius "Fat Man" makes it easy to add real tube warmth while tracking or mixing by serving up 15 preset compression programs. Presets include settings for vocals, keyboards, bass, acoustic and electric guitars, snare, kick and kit drums, plus pop, rock, and dance mixes. For more information, call HHB at 310-319-3111 or visit www.hhb.co.uk.com. Circle EQ free lit. #124.

HDR24/96 Interchangeable media. 24-bit accuracy. *This* is the future of digital recording.



External/internal clock indicators.

LCD display shows hr : min : sec : frames or bars : beats : ticks : fleas.

24x4 LCD display shows operation information, prompts and selections options via paging buttons.

SELECT buttons correspond to options on the screen above them.

+ and – (up and down) buttons scroll through parameter options.

Good 'ole familiar analog-type transport controls.

LOCate 1 & 2 buttons select cue or loop points.

Drive bay for Mackie Media™.

(inside) Internal hard drive with recording time equivalent to M•60 Mackie Media.

STORE saves commands.

3.5" disk bay imports tempo maps & upgrades.

LOOP 1-2 selects loop start and end.

RECord SAFE prohibits track arming.

ALL INPUT arms all tracks.

AUTO INPUT—On monitors playback in Play, and Input in Record/Stop/FF/REW for armed tracks. Off monitors input for armed tracks.

DELETE LAST.

TIME CODE chase.

AUTO TAKE.

TRACK selects options for slide and nudge, etc.

PROJECT saves and loads project information.

DISK UTIL & BACKUP control disk operations.

SYNC selects sync flavor (internal, word clock etc.).

DIGI I/O configures I/O cards.

SYSTEM is the zillion-and-one-options button for accessing preferences and system setup details.

Right out of the box, you can start using the HDR24/96 just by hitting REC + PLAY.

You can instantly punch-in as many as 24 tracks at once without the momentary delay that plagues many computer-based recording systems.

You can turn it into a fullblown digital audio workstation just by plugging in a monitor, keyboard and mouse.

You can use it with any mixer, not just with the Digital 8•Bus.

It uses media so affordable that you can keep one around for each project and treat 'em just like tapes[1]. And so rugged that they can withstand heavy impacts that would doom a creampuff SCSI drive.

The HDR24/96 syncs with video, SMPTE and MIDI so you can quickly integrate it with existing post production workstations.

It's sonically impeccable with 144dB dynamic range[2], 0.00001% THD and 2Hz to 22kHz response ±0.5dB[3].

[1] i.e., leave stacks of them out of their cases to gather dust, carry them around in a backpack, use them as doorstops, etc.

[2] Internal processor measurement

It ships with what we think is the most functional, and best-looking software interface in the pro audio industry. Software so easy to use that HDR24/96 beta testers tell us they don't even have to "crack the manual."

Multiple HDR24/96s can be linked with *sample accuracy*.

Its built-in 100kB/sec Ethernet "HDR Bridge" lets you send files to any computer desktop at blazing speed.

And you have your choice of two remotes: our compact "engineer in a box" Remote 24, or the mondo Remote 48 PRO that runs two HDR24/96s at a time (and impresses the heck out of clients).

In short, the HDR24/96 can turn any console, from an analog 24•8 to a Digital 8•Bus, into a serious hit factory.

And, equally as important, it comes with Mackie's proven track-record for bullet-proof reliability and superb customer support.

[3] At 48kHz sample rate.

MSRPs, ESPs and TtTESPs*.

FIGURING THE TRUE COST OF A HARD DISK RECORDER.

As with cars, the base cost of a hard disk recorder represents a *starting* point for comparison.

But it doesn't reflect two critical pieces of information: *1)* what you're going to *really* pay to get it up and working in your studio and; *2)* the long term cost of ownership.

Consider these factors when pricing hard disk systems…

I/O cards. You're going to need three 8-channel I/O cards to interface a hard disk recorder with your analog or digital console. Mackie's full line of I/O cards starts at just $99 each.

Media recording capacity. The HDR24/96 ships with over NINETY MINUTES of 24-channel recording capacity. This is a major consideration; some recorders ship with as little as 45 minutes of recording media.

Compatibility and connectivity. Our recorder's built-in Ethernet "HDR Bridge" lets you FTP data from the recorder to your computer's desktop, existing SCSI disks and the outside world at near realtime speed (100kBs/second).

The cost of digital editing. Our hard disk recorder is a lot more than just a box that records. It's your gateway to creative non-linear editing without having to lay out big bucks for a full-blown workstation. Mackie's HDR24/96 is an *integrated system* that requires as little as $300 worth of standard PC peripherals to become a $10,000 workstation. Some recorders require adding a complete new computer system (with the interfacing issues that inevitably result).

Long term cost of ownership. What's it going to cost to run the recorder through project after project? Mackie's M•60 media are significantly less expensive than our competition's external SCSI pull-out disks. And they eliminate the need for extra backup devices like Jaz® or M/O drives.

Do your homework. Then decide. We think you'll see that, as usual, Mackie offers the best combination of value and performance. Experience the HDR24/96 at a Mackie dealer soon.

*Tony the Transhipper's Extra Special Price with Bolivian Warranty

We were gonna encrust the HDR24/96's on-screen TOOL panel with our typical tangle of call-out lines and captions until we realized that the darned thing is so intuitive and analog-like (analog-esque?) that it pretty much explains itself.

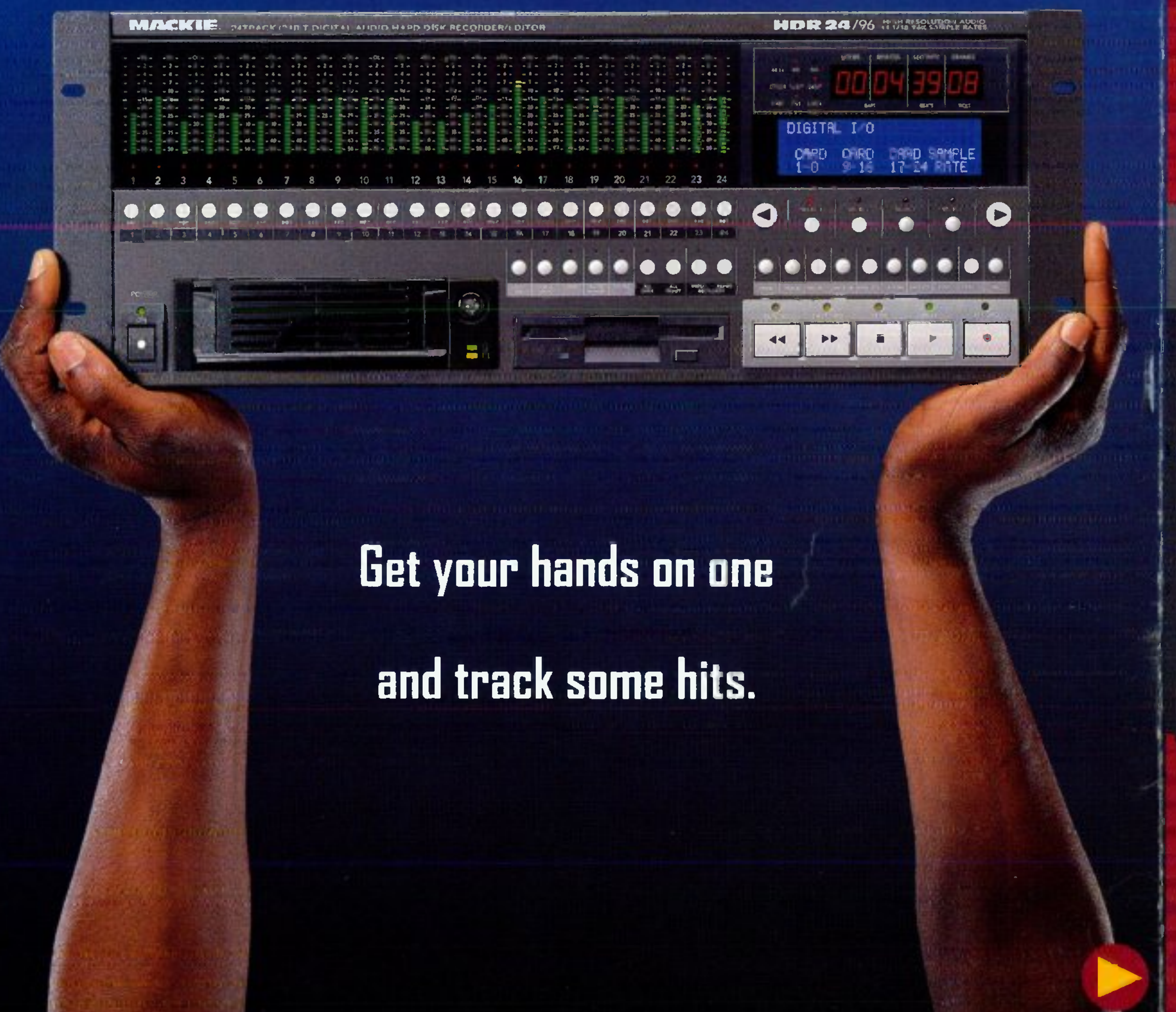
It makes beautiful music with the Digital 8•Bus, too.

The HDR24/96 works great with mixers as small as our analog 24•8. But, as you might imagine, it really comes into its own when you sync it with a Mackie Digital 8•Bus. From the D8B you can arm recorder tracks, operate the transport, select cue points and move or scrub back and forth with that big jog shuttle wheel.

MACKIE. Digital Systems

HDR24/96

Mackie's new standalone 24-track digital hard disk recorder with affordable pull-out media.



Get your hands on one and track some hits.

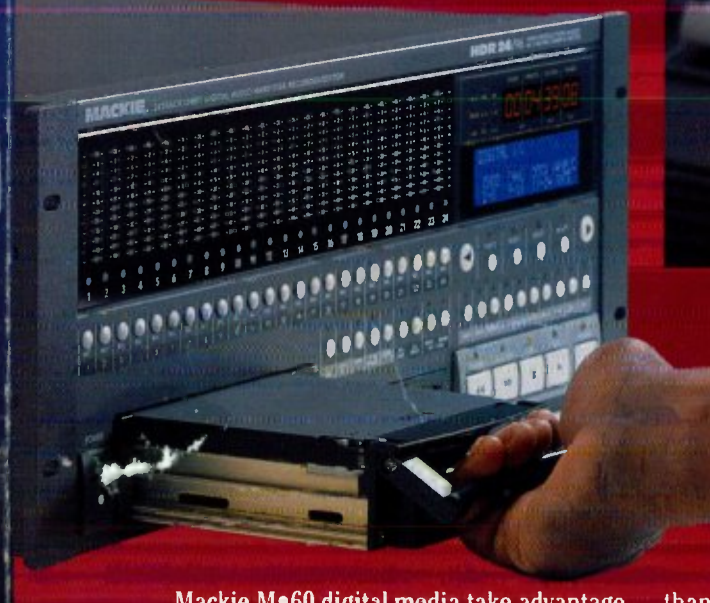
Use it with any analog or digital mixer. Record more than 90 minutes on each M•60 pull-out cartridge.

The HDR24/96 redefines hard disk recording by combining Mackie value and ergonomics with a ton of hot features:

- Intuitive analog recorder interface
- Built in internal hard disk plus external pull-out M•60 cartridge bay
- 24 tracks with 192 virtual tracks
- 24-bits – 44.1 & 48 kHz sample rates*
- Large and small remotes available
- Sample-accurate sync lets you slave unlimited numbers of HDR24/96s together for gazillion-track recording
- Analog, ADAT™, TDIF and AES/EBU I/O via 8-channel mix-and-match cards
- Built-in sync to all flavors of SMPTE, NTSC & PAL video black burst, MIDI or internal work clock
- 100BaseT Ethernet port standard

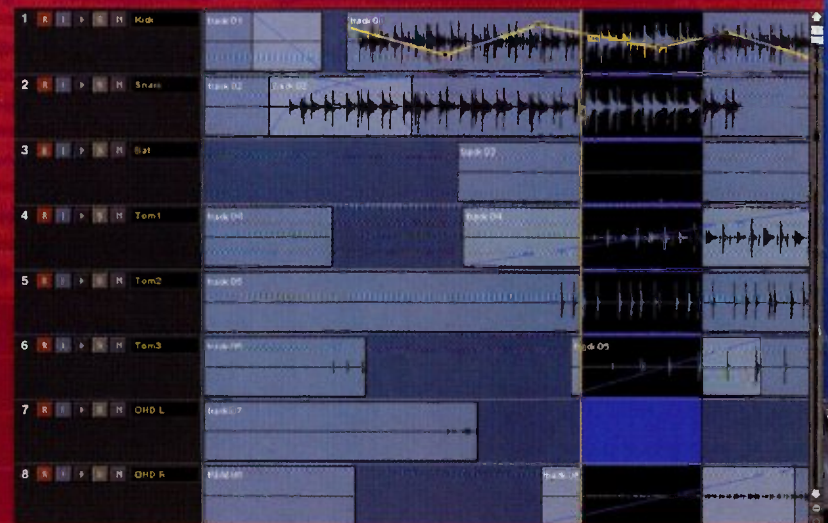
* plus 96 kHz via software & I/O card upgrade

Optional Remote 24



Mackie M•60 digital media take advantage of a major hard disk technology breakthrough called Ultra-DMA. So fast and reliable that they're used in streaming video servers and Apple Macintosh® G4, Ultra-DMA drives are significantly less expensive than SCSI drives. They can withstand impacts of up to 100Gs without damage. Mackie Media are available from any dealer who carries the HDR24/96.

Free $10,000 bonus: The HDR24/96 is also a digital audio workstation with waveform editing. Simply add a monitor, keyboard and mouse!



The unspeakably cool part of hard disk recording is being able to manipulate tracks. Slide, chop, group and cross fade them. Copy the chorus. Tweak the verse. And then go back and do it all again…because every action is nondestructive and you have 999 levels of "undo." You'll find that your creativity and productivity are boosted by a factor of ten.

Gone are the days of sweating over on-the-fly punch-ins and punch-outs that you have to live with once they're made. Now you can zoom right in to the waveform level, scrub back and forth and find the *exact* point you want for the punch-in or edit.

This level of precision used to mean spending $10,000+ on a computer-based workstation (or $50,000+ on a dedicated system). Now you can have it without even having to tie up your computer. Built right into the HDR24/96 are all the features you need for music production including…

Regions and Super Regions. You can audition or modify their start and end points instantly, capture a region and turn it into another track, or replace multiple regions with a new one. They can be cross-faded, grouped, sliced and diced.

Unlimited cue points with looping and auto-punch-in mode. Cue points are visible on screen or can be accessed from the sidelist that includes History, Groups and Regions.

Track grouping. Lock related tracks (such as drum or background vocals) together and manipulate them as one track.

Quantization. Define a time line and then "snap" your edits, moves and inserts precisely to these points.

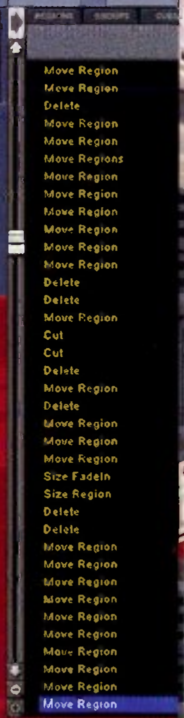
Time Bar with user-defined resolution (sample, 1/4 frame, msec, second, BBT); Punch, Loop, Cue and Tempo Change markers; Snap function; Zoom arrows; Locate; Punch In/Out, AutoPunch, Open End, Pre-Loop and Post-Loop functions.

Okay, we'll stop with the jargon. You get the idea. The HDR24/96 is a serious digital workstation disguised as a mild-mannered hard disk recorder. With a rich, but non-garish visual interface that you won't mind staring at for hours on end.

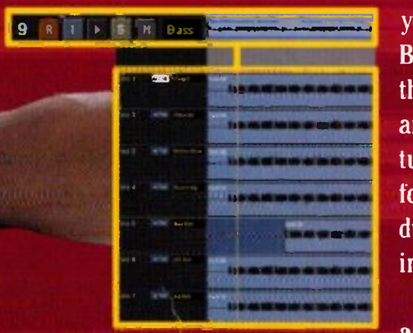
You don't need an extra computer. You don't need to spend megabucks. Visit your Mackie dealer for a demonstration of how the HDR24/96 can boost your creativity.

Mark a track segment and name it as a Region *(continuous segment) or* Super Region *(non-adjacent parts of a track). Then cut, copy and paste these segments anywhere… onto blank tracks or right into the middle of an existing track (the part after the insert just "slides" down to make room).*

Nondestructive drag-and-drop cross-fades, fade-ins and fade-outs with your choice of nine different styles.



History display shows every editing and recording activity from the time you turn on the HDR24/96. Unlimited levels of undo mean you can experiment to your heart's content.



Each HDR24/96 track "expands" into eight virtual tracks. Record multiple takes and then comp them into a perfect final. Virtual tracks can also be converted into completely new tracks or divided up into Regions and distributed into existing ones.

Editing aside, the HDR24/96's detailed SVGA display is a fantastic tool in itself. It duplicates every front panel control and features true scrolling tracks that you can view in groups of 24, 12, 8, 4 or 2. Transport section includes Rehearse, AutoPlay from locate recall, Scrub, Shuttle at 2x to 8x and 1/2 to 1/8 speed, All Input, Repro, Input and All Safe modes. Toggling meter bridge display shows 24 tracks with peak hold and dBf/dBu options.

2x waveform display with Tool bar selected.

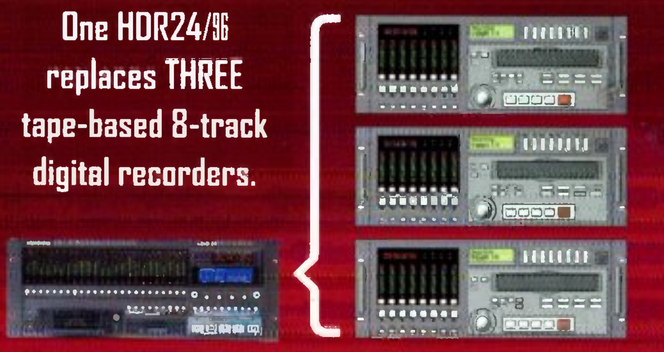
24x waveform display with meter bridge selected.

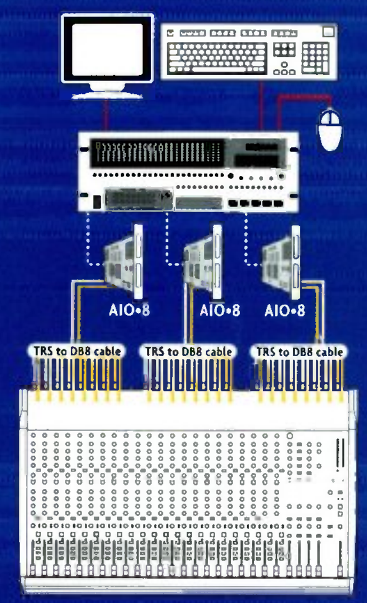
Until now, whatever you recorded on a hard disk recorder or workstation was usually stuck inside. Storing, transporting or just changing projects meant a tedious transfer to Jaz® or magneto-optical disk.

With the HDR24/96 you simply pop in a Mackie Media M•60 cartridge and start recording. When you change projects, just change cartridges and put the first one back in its case and back on the shelf (or record to the internal drive and use the pull-out for ultra-fast back-up).

Convenient pull-out 24-track media also beats the oxide off of having to deal with three separate tape cassettes.

One HDR24/96 replaces THREE tape-based 8-track digital recorders.

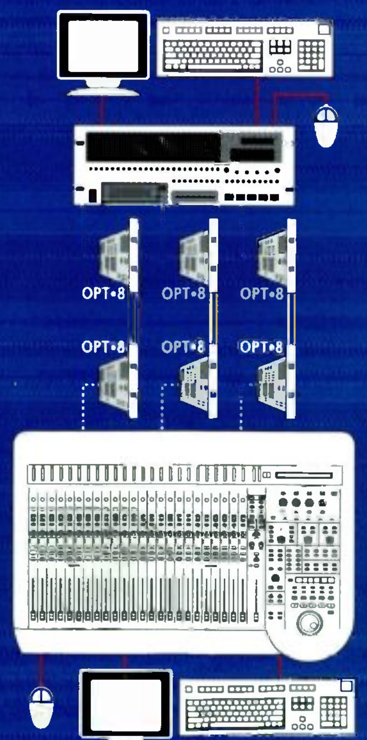




TYPICAL ANALOG CONSOLE HOOK-UP

TO HDR24/96 VIA AIO•8 I/O CARDS

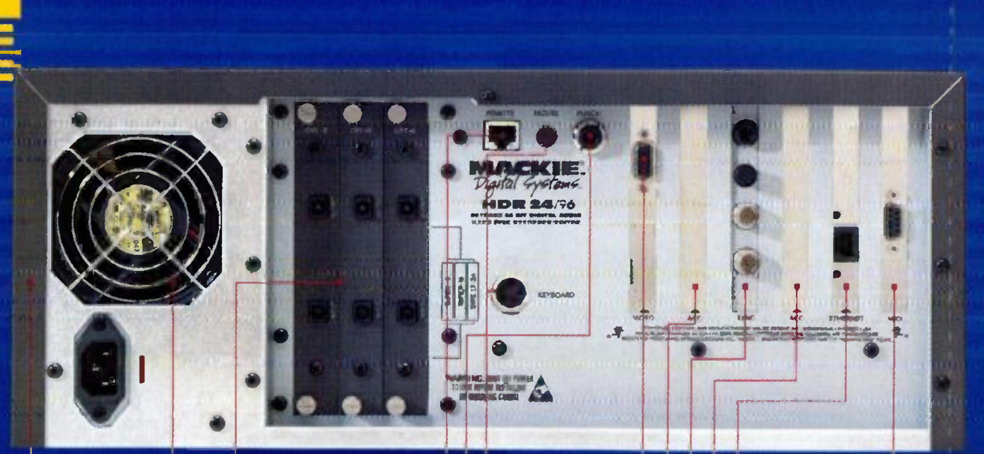
Sends to recorder can be patched from console bus outs or from channel direct outs (if you have a 24•4 or 32•4, you have at least 24 of them). Returns are patched to channel line inputs. Keyboard, monitor and mouse are not required



TYPICAL DIGITAL CONSOLE HOOK-UP

TO HDR24/96 VIA OPT•8 I/O CARDS

OPT•8 cards have ADAT optical inputs and outputs. DIO•8 cards (ADAT optical & TDIF with 16-to-24-bit conversion and format conversion) or PDI•O (AES/EBU with sample rate converter) may also be used in any combination.



4 RU configuration.
Ultra-quiet cooling fan.
3-slot card cage for any combination of AIO•8, DIO•8, PDI•8 and new OPT•8 24-bit ADAT optical I/O.
Remote 24 or Remote 48 PRO input.
Jack for punch-in/punch-out foot switch.
Mouse and keyboard port take standard PC peripherals.
SVGA video output.
Accessory card slot.
Sync ports for Word Clock, SMPTE, NTSC and PAL video black burst.
MIDI port (no extra black box needed).
100BaseT Ethernet "HDR Bridge" out.
Accessory card slot.

HDR24/96 SPECIFICATIONS

Electronic[1]

Frequency response[2]	2 Hz to 22 kHz±0.5dB
Harmonic distortion	0.00001%
Dynamic range	144dB digital / 106dB analog
Adjacent channel crosstalk[3]	–90dBu

Digital

Quantization	24-bit
Sample rates	44.1 kHz / 48 kHz / 96 kHz[4]
Internal CPU	433 MHz Intel® Celeron® w/ 128k internal L2 cache memory
SVGA out	ATI® Rage™ Pro AGP w/8Mb RAM 1024x758 pixels, 72Hz refresh rate
Internal RAM	64 Mb PC66 SDRAM
Ethernet throughput	100kB/sec.

Hard Disk

Internal Capacity	<90 minutes[5]
Mackie M•60 Media Capacity	<90 minutes[5]
Hard disk seek time[6]	9.5 millisecond
Hard disk throughput[6]	295 megabits/sec.

I/O

Mouse	PS/2 style, 6-pin mini-DIN
Keyboard	IBM®-style, 5-pin DIN
Video	SVGA High density 15-pin D-Sub
MIDI	MIDI IN / MIDI OUT, 15-pin D-Sub
I/O cards	OPT•8 *ADAT optical* DIO•8 *ADAT optical* & TDIF™ PDI•8 Digital AES/EBU AIO•8 Analog
Nominal / Maximum digital input[7]	–18dB*fs* / 0dB*fs*
Nominal / Maximum analog input[8]	+4dBu / +22dBu
AIO•8 A/D Converters	24-bit, 128X oversampling

Physical

Height	7" / 178 mm (4 RU)
Width	19" / 483 mm
Depth	13.25" / 337 mm

Electrical

Operating voltage	120V / 240V / 100 V, 50 /60 Hz
Power consumption	50W

Optional

Remotes	Remote 24, Remote 48 PRO
Future	96 kHz I/O, accessory cards

Irrelevant

Weight of Corporate Chihuahua	3.5lb / 1.6 kg

[1] with 24-bit digital I/O; [2] 48 kHz sample rate; [3] 0dBu at 1 kHz; [4] 96 kHz via software upgrade and 96 kHz I/O card, [5] 24 tracks at 48kHz sample rate, your actual time will vary with number of virtual tracks and amount of non-linear editing; [6] same for internal and external M•60 media; [7] OPT•8, DIO•8 and PDI•8 I/O cards; [8] AIO•8 analog I/O card

MACKIE. Digital Systems

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MOTU 1296 Audio Interface

Bring 12 channels of high-resolution audio into your Mac or Windows workstation

BY STEVE LA CERRA

The latest entry into Mark of the Unicorn's (MOTU) growing line of audio interfaces designed for hard disk recording is the 1296. As you might guess from its name, the 1296 is a 12-channel audio workstation I/O capable of recording audio at 44.1, 48, 88.2, or 96 kHz sampling rates, with resolution of 16, 20, or 24 bits. MOTU has designed the 1296 for sample-accurate, 24-bit multichannel recording with a variety of audio software on both the Mac OS and Windows 95/98 through use of ASIO-2 drivers. MOTU has also engineered the 1296 to integrate seamlessly into its current product line. In addition to using the same PCI-324 card as MOTU's 24i, 2408, 308, and 1224 for connection to the host CPU, the 1296 can be mixed and matched with any of these interfaces. Up to three devices of any combination may be used on a single PCI-324 card; a single-card system with three 1296's would support a total of 36 analog inputs and 36 analog outputs, plus three AES/EBU digital I/Os.

Front-panel features of the 1296 include 19-segment LED meters for each of the input and output channels, clock status LEDs, and a power switch. Interestingly, MOTU has built all of their audio interfaces to be hot-swappable, so you can turn off a 1296 (or any of the other interfaces), disconnect it, connect a different MOTU interface to the same PCI-324 interface, and power up the new interface without needing to shutdown and restart the computer. On the rear of the 1296's two-rackspace chassis are 24 XLR connectors (12 male, 12 female) for the 96 kHz/24-bit, +4 analog I/O. Dynamic range is claimed to be 117 dB (A-weighted) on the analog inputs and 116 dB on the outputs. To ensure that the 1296's internal power supply does not cause interference with the audio I/O or produce distracting acoustic hum, MOTU has employed low-noise R/CORE transformers in the 1296's power supply.

AES/EBU digital I/O is on XLR connectors, while word clock input and output connectors are provided on BNC jacks; the 1296 can slave to external clock or serve as a master to other devices. A third, independent word clock connector drives clock data to the AES/EBU I/O, allowing this input to slave to a sample rate different from the sample rate at which the 1296 is recording. As a result, the 1296 can record at 96 kHz while using digital signals of a different sample rate as part of the recording. For example, let's say you were recording a snare drum using the 96 kHz rate. You could simultaneously record the 48 kHz digital output from a reverb unit, mix it in with the snare, and record the resulting audio at 96 kHz.

The rear panel also includes MOTU's proprietary Audio Wire digital input/output interface. Introduced with MOTU's 2408 (the first audio interface in the line), Audio Wire links any MOTU interface with the PCI-324 card. Audio Wire is capable of simultaneously carrying 24 channels of 48 kHz/24-bit audio input and output, or 12 channels of 96 kHz/24-bit audio in and out. MOTU's various interfaces all connect to the PCI-324 using the same type of cable. The 1296's software driver provides adjustable buffer memory and MOTU's Cue Mix feature, which (the company claims) provides the lowest host-based latency available with any software.

The 1296 (and all MOTU audio interfaces) is compatible with both Mac OS and Windows. Macintosh system requirements are any 604-based processor or faster (G3 recommended), 64 MB RAM minimum (128 MB recommended), Mac OS 7.6.1 or later, and a 9 GB hard drive. Requirements for use with Windows are a 200 MHz Pentium CPU or faster (Pentium II preferred), 64 MB RAM

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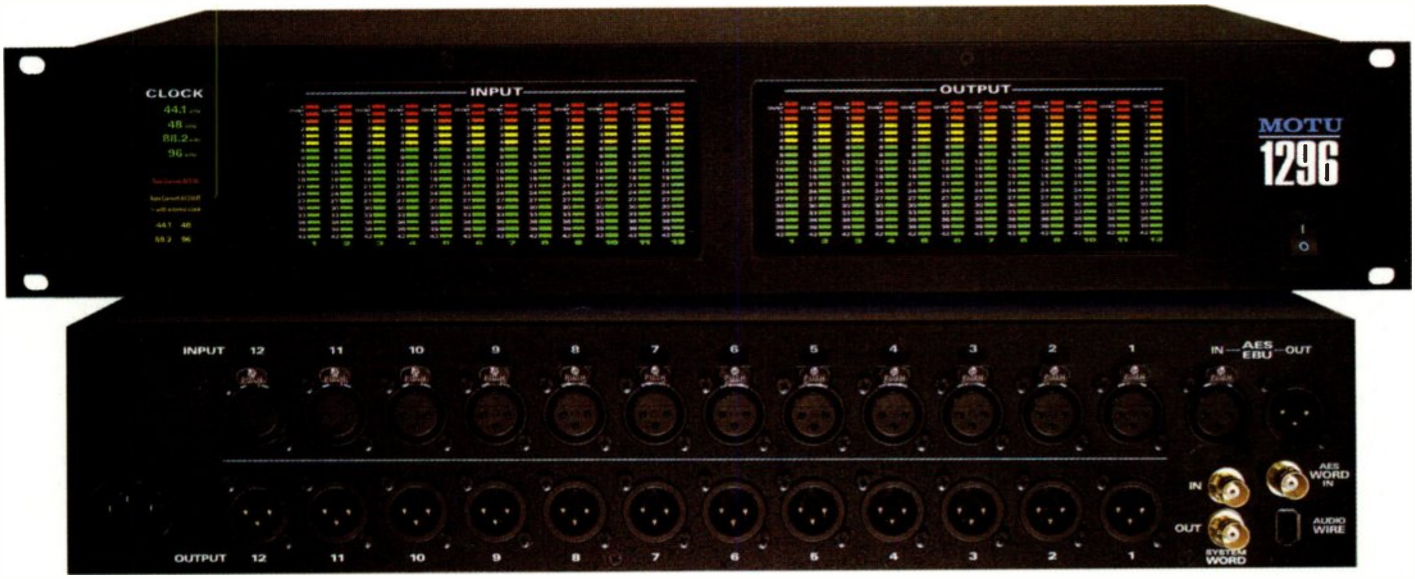

C2 Stereo Compressor **$399.99**

Electronic Musician 2000 Editors Choice

SC2.2 Stereo Compressor **$799.99**

SC2 The Stereo Compressor
The classic Analog Photo-Optical Stereo Compressor that started it all! **$1,999.99**

SC4 Digital Stereo Compressor
A Joemeek SC2 Compressor with M/S features, Dual Meters, Sample Rate Converter and optional 24bit AES/EBU Digital Card (DC4) **$1,999.99**
Optional DC4 $299.99

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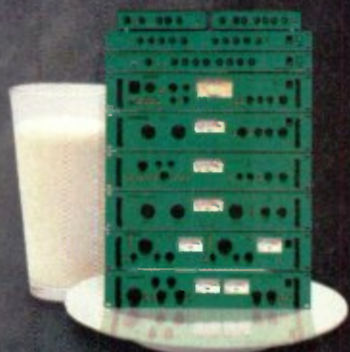
JOEMEEK

When it comes to color, Joemeek set the standards. From the early "vintage" designs by the master himself, to the latest Class A analog circuitry. Joemeek sets the standard for quality and value. No other stereo compressor can match the

Color Does Matter!

silky smooth colored sound of the Joemeek stereo compressors, when laying your tracks down or going to two track mixdown. So while other stereo compressors claim to reproduce the warm colored Joemeek vintage sound you're looking for, wouldn't you rather own one that does!

got meek?



CIRCLE 14 ON FREE INFO CARD

Propellerheads **Reason**

Turn your computer into a self-contained synth-based studio

BY STEVE LA CERRA

Studio-in-a-box? Well, not exactly, but Reason from Propellerheads could be looked upon as a studio-on-a-CD-ROM. The most recent offering from the company that brought us ReBirth and ReCycle!, Reason is best described as an expandable, software-based MIDI studio, the limitation of which is defined only by the amount of horsepower in your computer. Reason includes a variety of audio "devices," including a 14-channel mixer with four sends, the Subtractor (an analog synthesizer), a MIDI note-to-gate converter, the NN19 digital sampler, the Re-Drum drum machine, a ReCycle!-based loop player, a sequencer, and a plethora of effect units such as digital delay (which syncs to MIDI clock), reverb, parametric equalizer, compressor/limiter, chorus/flanger, envelope-controlled filter, and phaser. Multiples of any of these "devices" may be created, and you can mix and match devices based on the needs of a specific project.

A studio rack is built within Reason on-screen using the "Create" menu, from which you choose the devices you need. When you choose to create a device, the unit appears in your rack, patched into the signal chain and ready for use (every rack includes a transport control unit at the bottom). CPU processing power may be utilized however you see fit. If a particular session calls for three analog synths and two samplers, you simply create your rack with those units. Each unit has front-panel controls much like you'd find on a hardware unit (adjustable via mouse). All parameters of a studio rack system may be saved, enabling true "total recall" of session data. Reason marks the debut of the ReFill file format, which can contain all the data (synth patches, samples, sampler setups, drum kits, etc.) needed to set up the program for a specific purpose or music style.

Perhaps the coolest feature of Reason is that, if you don't like the audio routing of the system (or if you just want to experiment), a press of the Tab key will flip the rack over, providing you with a view of the back of the rack. Inputs and outputs are displayed, as well as CV and gate connections for each device. Complicated signal routing and cross-patching is accomplished by simply drawing a patch between the output of one machine and the input of another.

One possible application for Reason would be to turn your computer into a self-contained, synth-based studio. The Reason sequencer features graphic, event-level editing tracks specifically designed for each Reason device. For instance, the drum machine editor has a separate edit area for pattern changes, while an audio file player has a waveform edit display. If you don't particularly like the default setup, you can customize an editing window.

Reason may be used together with ReWire, allowing it to be patched and synced with a digital audio sequencing program (Steinberg Cubase VST, Emagic Logic Audio, and Opcode StudioVision all support ReWire; MOTU's Digital Performer will have support for ReWire in the soon-to-be-released version 2.7). As an example, when used with Cubase, instruments from Reason are automatically routed to the Cubase VST mixer and may be processed using VST plug-ins. The output of Reason instruments may also be mixed with other Cubase tracks and recorded to hard disk.

Since the Subtractor synthesizer is fully MIDI-compatible, Reason can also be used as a virtual synth rack in conjunction with your MIDI sequencing software. Subtractor is a 32-voice-polyphonic, analog synthesizer that performs subtractive synthesis, and includes the features you'd expect to find on an analog hardware synth such as multi-mode and low-pass filtering, amplitude and filter envelopes, pitch bend and modulation wheels, portamento, and variable-waveform oscillators. All knobs, buttons, and sliders in Subtractor can be automated, and any motion may be recorded into the sequencer.

No additional hardware is required to run Reason because the computer's CPU generates all of the audio produced by the program. But since Reason includes ASIO, MME, DirectX, and SoundManager support, a wide variety of sound cards may also be used with it. Reason is available for both the Mac and PC platforms, and has the ability to export AIFF and WAV format audio files.

SYSTEM REQUIREMENTS

Mac: 604/166 MHz or faster, 64 MB RAM, CD-ROM drive, Mac OS 8.6 or later, a MIDI interface and keyboard, and OMS 2.2 or later (included).

PC: Pentium II/233 MHz or faster, 64 MB RAM, CD-ROM drive, Windows 98, NT 4.0 or 2000, 16-bit Windows-compatible sound card (preferably with DirectX or ASIO driver), DirectX (included), and a MIDI interface and keyboard.

Reason is expected to ship this summer at an MSRP of $399. For more details, contact Steinberg North America at (tel) 818-678-5100, (fax) 818-678-5199, or www.us.steinberg.net. You can also visit the Propellerheads Web site at www.propellerheads.se.

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CIRCLE 9 ON FREE INFO CARD

City Slicker

Analog and digital live harmoniously in this NYC project studio

STUDIO NAME: City Sound Productions
STUDIO LOCATION: New York, NY
KEY CREW: Bob Kirschner (owner, producer, and engineer) is a keyboard player, a composer, and a graduate of the Recording Technology program at New York University. He has worked with Yoko Ono, Bill Laswell, Jamaldeen Tacuma, Vernon Reid, Garland Jeffreys, and Maxi Priest, as well as other independent artists and producers. City Sound staff: Keith Humiston, studio manager; Corey Folta, engineer; William Bowen, second engineer
CREDITS: Deborah Harry (vocals for a soundtrack song for *Three Businessmen*), Def Squad sessions for producers Erick Sermon and Rockwaller (tracking for artists Angie Stone and Phendi Moore, respectively), Sony PlayStation game *Um Jammer Lammy*, LouKass (mixing live album), Kid Creole and The Coconuts (tracking and mixing for a recent release), production for local pop acts Blush66 and Ultralust, tracking and mixing for composer Mark Bennetts's pieces at Lincoln Center and Public Theater, mix-to-picture for independent filmmakers Reality Productions, Al Santana Productions, and Aubin Pictures
MIXING CONSOLE: Soundtracs Megas Studio (70 inputs with MIDI mute automation)
MONITORS: Tannoy DMT 15, Genelec 1031A, Yamaha NS-10M, Auratone PSC
AMPLIFIERS: Hot House M500HV [2], Yamaha P2250, Ashly FET1000M
RECORDERS: Otari MTR90 II, Pro Tools Mix Plus (18 GB of storage, 64 tracks, 32 channels of Digidesign 888 I/O), Alesis ADAT [4] plus BRC controller and AI-1 interface, TASCAM: DA-88 with SY-88 sync card, 122 Mk III cassette deck [2]; Studer Revox PR99 (1/4-inch 2-track), Yamaha CDE100 CD-R
DAT RECORDERS: Panasonic SV3700 [2], Fostex D20 (timecode DAT)
OUTBOARD: TubeTech LCA-2B (stereo tube compressor), Summit TLA-100A (tube limiter), UREI: 1176LN (blackface, two), 565 filter set; Aphex Expressor [2], BSS DPR-402, dbx: 160 VU [2], 160XT; Summit PEQ-200 (stereo tube EQ), API: 550b [2], 550A [2], 560b [2]; Rane SP-15 EQ [2], Aphex 612 gate, Gatex 4-channel gate, SPL Vitalizer, Orban 536A de-esser
EFFECTS: Lexicon: PCM70 [2], 200, PCM42; Eventide: H3000SE, H910, Instant Flanger, Instant Phaser; TC Electronic: TC2290, 1210 Spatial Expander; Ibanez SDR1000+, Roland RE-201 Space Echo, Alesis Quadraverb, Korg SDD-2000
MICROPHONES: Neumann: U67 [2], U87 [2]; AKG: C414, D112; B&K 4007 [2], Sennheiser: MKH405 [2], MD421 [4], MD441; Shure: SM57, SM58; Electro-Voice RE20, RE50; Crown SASS-P Stereo PZM
MIC PREAMPS: API: 512b [2], 312 [2]; Hardy M-1 [2], Daking 52270 Trident A-Series mic pre/EQ modules, Bryston BMP2 [2], Gates tube preamp
SAMPLERS/KEYBOARDS/MIDI MODULES: SampleCell II (32 MB RAM), Kurzweil: K2000RS (32 MB RAM), PC88MX; Roland: JV-1080 with '60s/'70s card, Orchestral card, Bass 'n' Drum card, MKS-80 with programmer, MKS-70, SH101, JD-990, TD-7K; Studio Electronics SE-1, Sequential Circuits Prophet 5, Sherman Filterbank, Alesis D4, Wurlitzer 200 electric piano, Hammond XB-2, Voce Electric Piano
COMPUTERS: Apple Power Mac G3/300 MHz, Opcode Studio 5 MIDI interface [2]
SOFTWARE: Digidesign Pro Tools with the following plug-ins: Line 6 Amp Farm, Auto Tune, DINR, Dynamics II, EQII, Focusrite d2, JVP, D-Fi, Maxim, Mod Delay, TC Megaverb; Digidesign Masterlist CD, Opcode: StudioVision Pro, Galaxy Plus; Alchemy, Adaptec: Toast, Jam
VIDEO EQUIPMENT: TimeLine Microlynx, Sony UVW-1800 Beta SP, Sony BVU-800 U-Matic, Mitsubishi 1/2-inch HiFi VTR, Horita BSG-50 blackburst generator
GUITAR AMPLIFIERS: Fender: 1964 Bandmaster, Deluxe 85; Dean Markley RM-40-DR, Airline tube amp, Ampeg B15, Sovtek 50-watt head
STUDIO NOTES: City Sound Productions was designed by Francis Manzella Design in 1994. It was built as a dual-purpose project studio/commercial facility and features separate floating control room and live room.
EQUIPMENT NOTES: According to Bob: "There is nothing like the sound of an analog synth. I was brought up on analog synths starting with an Octave-Plateau CAT, and now the Prophet 5 and MKS-80 are my all-time favorites. For electronic music production, the Sherman Filterbank (dual analog filter) is amazing for processing anything. I've taken a live drum track, looped and distorted the drums, and sent a copy of the loop through the Sherman unit. After combining the two signals, the effect is like a unique comb filter. StudioVision has a great formant-shift plug-in that changes the timbre of audio without shifting pitch. Using different amounts of it over the course of a 16-bar phase creates a great wave-like effect. The Ampeg B15 amp is awesome for sending bass through and bringing it back to the board during a mix. We also track keyboards and organs through the B15 so that they don't sound so

PHOTO BY DANIEL SMITH

sampled. The Roland Space Echo is used a lot. One cool setup is to distort an organ through an overdrive pedal, then send it through the Space Echo. The result is an eerie Pink Floyd-like sound; changing the drawbars in real-time takes it to an entirely new level. My Daking mic pre/EQ modules are fantastic. Modeled after the Trident 'A-Range' console, they have an amazing-sounding mic pre, and the EQ is my 'magic box' for mixing bass. The Amp-Farm plug-in for Pro Tools is quite unbelievable. Not only does it sound realistic, but when I first realized the flexibility of being able to change amp settings after the guitar was tracked, I was totally sold!"

PRODUCTION NOTES: Bob continues: "My addition of an analog 24-track recorder is unusual these days, when many studios are getting rid of analog machines. My philosophy of combining analog and digital to get the best of both worlds was a major factor in purchasing the Otari MTR90 II. Live drums are now tracked to 2-inch tape for its beautiful analog sound, and then transferred to Pro Tools for processing and mixing.

"My expanded Pro Tools 5.0 system is the heart of the studio. The speed and power of Pro Tools has opened up so many new possibilities in different areas: for music production, the powerful DSP lets me manipulate sounds in fantastic ways and automate the changes in real time. When tracking vocals or instruments, editing together comps is fast and accurate. Even after the musicians leave, extensive fixes or changes are no problem. When mixing, I'll use the 888 inputs and outputs to interface my nice analog outboard gear, and automate these sends and returns. For postproduction, Pro Tools is fantastic for working with timecode. We can handle mixing, ADR, or sound effects sessions easily. The OMF tool lets Avid users import their Media Composer files directly into Pro Tools.

"Last, but most important: Anyone can own racks of equipment, but the key to success is a highly skilled staff. I take great pride in our experienced engineers who can fly on Pro Tools and get a great drum sound."

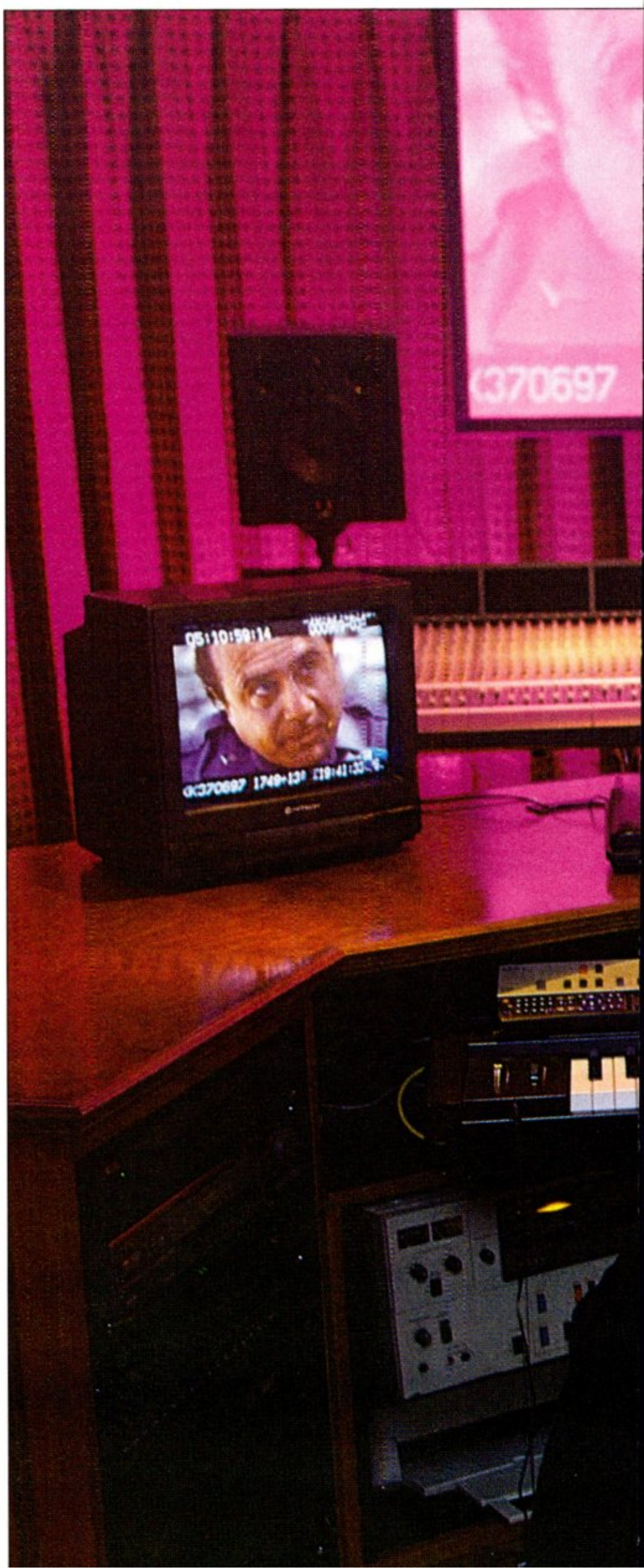
Visit the City Sound Productions Web site at www.citysound.com. EQ

Minding the Score

MIDI plays a large part in Michael Tavera's compositions for film and television

STUDIO NAME: Tavera Music, Inc.
LOCATION: Encino, CA
KEY CREW: Michael Tavera, composer; Scott Cochran, engineer and scoring mixer
CREDITS: Films: *Drowning Mona, Mr. Magoo, The Land Before Time (sequels), Rocketman, Honey We Shrunk Ourselves, Girl, Culture, Special Delivery*; Television series: *Hyperion Bay, Melrose Place*; Animated Series: *Toonsylvania, Casper*; Television movies: *Holiday In Your Heart, Two Came Back, Silent Predators, Forever Love*
MIXING CONSOLE: TASCAM M-2600 Mk II [2] with external patchbay, connected with Mackie Mixer Mixers [2] for 128 inputs
MONITORS: Quested VS2108 (active, self-powered monitors), Bose home theater reinforcement system, Fostex 6301B self-powered monitor for video dialog
RECORDERS: TASCAM DA-98, MOTU: 1224 [2], 2408; Kenwood cassette player
DAT RECORDERS: Panasonic SV-3800 [2]
OUTBOARD GEAR: SSL stereo compressor, Avalon AD 2055 stereo EQ, JVC CD player, TASCAM MM-1 mixer (for CD and cassette), Midiman 6-channel mixer (for video dialog), two 32-channel Signal Transport line drivers (–10 to +4), 48-channel Dark Star Electronics custom-made line driver (–10 to +4)
EFFECTS: Lexicon: PCM90, PCM91, MPX1
MICROPHONES: Neumann TLM170 condenser mics [2]
MIC PREAMPS: Martech MSS-10 mic preamps [2]
SAMPLERS/KEYBOARDS/MIDI MODULES: Yamaha KX-88 MIDI controller keyboard, SampleCell II PCI cards [7] mounted in a SBS external PCI Expansion Chassis, Roland: S-760 samplers [4], JV-1080 synth modules [4] (each filled with four expansion cards); Alesis Nanopiano, Oberheim OB3 organ module, Etherwave Theremin (non-MIDI)
COMPUTERS: Apple Power Mac G3/233 with 224 MB RAM, 4 GB internal hard drive, 20 GB Lacie external hard drive, 2 GB external Jaz drive, Adaptec Power Domain UW-2940 high-speed Ultra-SCSI connection, GEM 19-inch monitor on swivel (for engineer use), two computer keyboards w/mice (one set for composer, the other for the engineer), US Robotics 56k modem, MOTU PCI-324 card, USB PCI connector for use with PCI expansion module, Epson SC-740 printer
SOFTWARE: Digital Performer 2.6, Sibelius 1.2, Digidesign SampleCell 2.1.1 and Sound Designer II, MOTU AudioDesk, N2MP3 audio converter, Sound App PPC audio converter/player, many Waves plug-ins for Digital Performer, Opcode OMS 2.3.6 and MOTU FreeMIDI 1.4, Port Expander software (for switching between modem and printer)
VIDEO EQUIPMENT: Hitachi video monitor, Polaroid 211E LCD video projector, Lumalectric video projection screen with remote control, VAC RS170A black burst generator, Sony VO-5600 3/4-inch video machine, Replay TV2000 digital audio/video recorder/player, JVC 1/2-inch video machine
PRODUCTION NOTES: Michael Tavera explains, "Whether I'm writing for a purely orchestral end product or a purely synthetic end product, I always realize my musical ideas on a MIDI basis first. While I write, I monitor each module in stereo only. With this process, all of the parts are MIDI-mixed. When Scott comes over to work with me, we can always start our mix with the faders in a straight line.

"If the end product will be orchestral, my clients listen to orchestral mock-ups. This allows them to have a clear idea of what the final live version will be. We work together and make changes at this stage. I then e-mail a MIDI file of each cue to my MIDI file



clean-up assistant. My sequences are laid out in a score form so that he can easily clean up note values.

"The files are then sent back to me as a score without any articulations, phrasings, or dynamics, as a Sibelius music publishing file. I then make any necessary note value changes and add articulations, phrasing, and dynamics to the score. This final score is then e-mailed to my copyist, who will extract parts. This process gives me a high degree of efficiency and control. It also allows my clients the freedom to make changes prior to the recording date."

Regarding the use of synthesizers, Michael continues, "If there are certain parts that will not translate well via synthesizer, I record live musicians into Digital Performer prior to my client meeting. If the client feels comfortable communicating with musicians, I will have them participate in the recording by giving feedback to musicians during the session." EQ

Electro-Voice 668

A blast from broadcast and film production's past

MICROPHONE NAME: Electro-Voice 668
FROM THE COLLECTION OF: Bill Meredith, Cinesound Company, NYC
PRICE WHEN NEW: $495
YEAR OF MANUFACTURE: Circa 1965
TYPE OF MIC: Moving coil dynamic
DIAPHRAGM: Electro-Voice Acoustalloy®
FREQUENCY RESPONSE: 40 Hz to 10,000 Hz
POLAR PATTERN: Cardioid
OUTPUT LEVEL: –51 dB, ref. to 1 milliwatt/10 dynes/square centimeter
EIA SENSITIVITY RATING: –145 dB, ref. 0 dB = 1 milliwatt/10 dynes/square centimeter
HUM PICKUP LEVEL: –121 dBm, ref. 0.001 Gauss field
RATED SOURCE IMPEDANCE: 50, 150, or 250 ohms (see notes)
CONNECTOR: Cannon UA-3-11
MIC NOTES: Designed for boom mounting in broadcast and film applications, the 668 incorporates a single diaphragm made from Electro-Voice's Acoustalloy material — a stable alloy relatively immune to the effects of temperature, humidity, and shock. The 668's shock mount provides isolation from mechanical noise, while its low sensitivity to induced hum enables the mic to be used relatively close to AC lines or lighting fixtures without picking up noise.
USER TIPS: An unusual feature of the Electro-Voice 668 is a small, built-in circuit board allowing modification of the microphone's response curve. Access to this circuit board is obtained by removing the cap at the rear of the mic body. Small pins on the circuit board may be moved to various positions that change impedance, high-frequency response, and low-frequency response. Moving one pin among three positions allows the output impedance to be varied from 50 to 150 to 250 ohms (the mic shipped from the factory is set to 150 ohms). Another set of pins act as on/off switches for two filters in a passive equalization network: 80 Hz low-frequency cut and an 8 kHz high-frequency cut. Two more adjustment pins tailor the high-frequency and low-frequency response curves. EQ

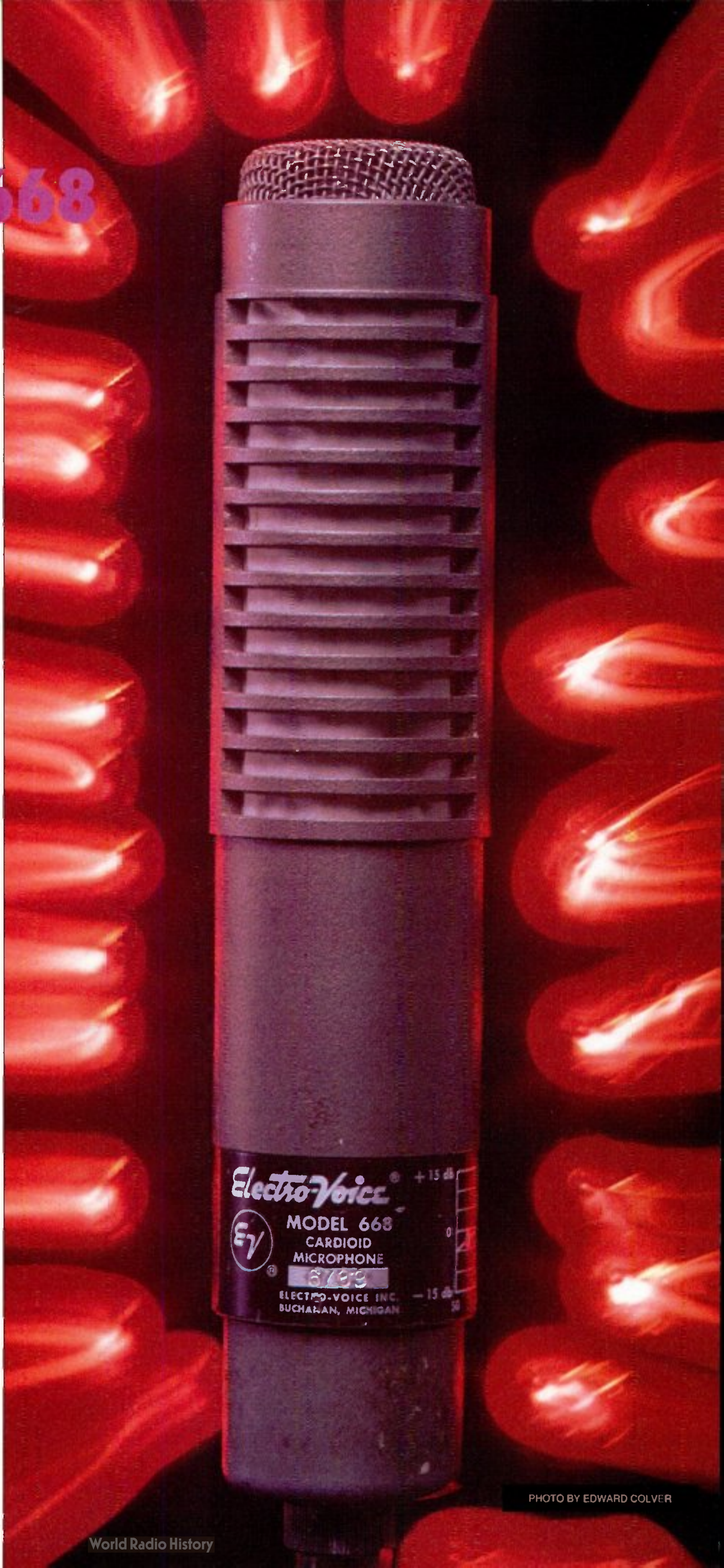


PHOTO BY EDWARD COLVER

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CIRCLE 16 ON FREE INFO CARD

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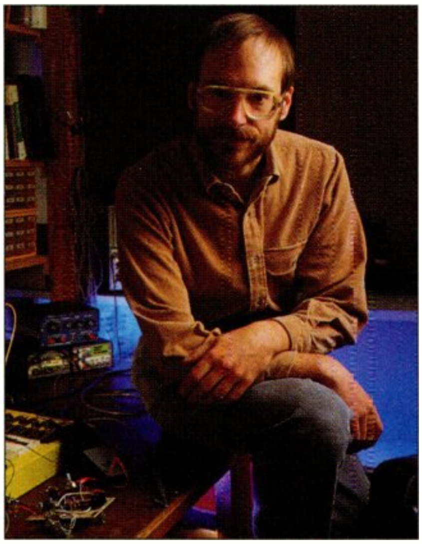
How a program named vRamDir can change your production life



BY CRAIG ANDERTON

This may tend more toward a review than an article, but the topic of kissing off your hard disk in favor of RAM deserves a little more background than would be appropriate for a review.

I spend a lot of time doing digital audio editing, which means I also spend a lot of time looking at the little progress bar graph as it moves lazily from left to right, chronicling how long it takes to rewrite any edits back to hard disk. The faster the hard drive, the better; but the process can still try your patience as you do edit after edit during a session.

However, if you have enough RAM, you can set aside a portion as a "RAM disk" for storing temp files. The advantage is that disk-intensive operations happen almost immediately because the operations occur in RAM rather than in a slow, mechanical device. The main drawback is that you need enough RAM to accommodate your temp files as well as whatever memory-draining foolishness you already have going on; for mastering work on individual tunes, 256 MB seems about right, as you can store 10–15 minutes of audio and still have room left over.

Mac PowerBook fans have known about RAM disks for years, as Apple makes it easy to set aside a portion of RAM, then load a minimal system and a few apps. This not only speeds up operation, but extends battery life by reducing hard disk access. Similar programs have existed for the PC, but none that I've seen are as elegant, simple, or inexpensive as vRamDir — a shareware program that only sets you back $10.

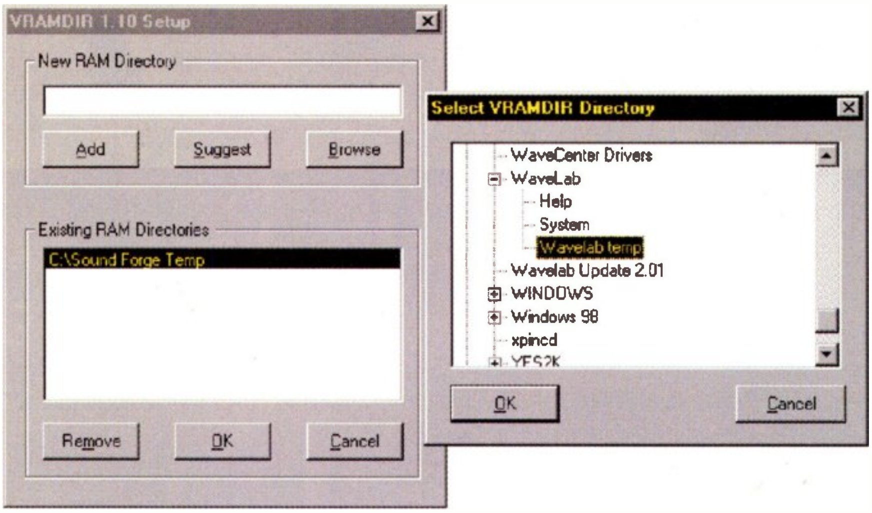
HOW VRAMDIR WORKS

vRamDir is for Windows 95/98 (not NT, sorry). In addition to the speed benefits obtained by storing files in RAM, there's another, less obvious speed benefit: reduced hard disk fragmentation. This is particularly true if you assign a temp file directory to vRamDir, because, in normal operation, you generate a lot of temp files that end up getting sprayed around your hard drive.

Unlike the Mac OS RAM disk option or the PC's RAMDRIVE program, vRamDir does not pre-allocate a specific amount of RAM; it uses RAM as needed, so if vRamDir isn't using the RAM, it's available for other applications. If your files exceed the amount of available RAM, then you're back to using the hard drive as virtual RAM, but at least the system doesn't blow up. Unlike RAM disks that utilize 16-bit real-mode DOS device drivers and therefore have to switch back and forth between 16-bit real-mode and 32-bit protected-mode, vRamDir is a 32-bit protected-mode native virtual device driver, which also gives a slight speed benefit.

Upon installation, vRamDir installs a control panel that lets you specify what folders will operate in RAM instead of on disk. You can change these at any time (although you have to reboot for the changes to take effect), and specify multiple folders to be active simultaneously if desired. For Wavelab and Sound Forge, I created two folders for temp files; one for each program. I then set the default in each program to store its temp files in these folders. (For example, in Sound Forge you go Options > Preferences > Perform Tab, and browse for the folder you want to use for temporary storage.) I've also set vRamDir to treat both folders as RAM folders, but, even with 256 MB of RAM, having both programs open and processing multiple files can put a crunch on available memory. However, I do usually keep the Windows Temp folder as a vRamDir file (you should see what this does to the performance of Microsoft Office).

vRamDir doesn't really emulate disk operation — it's more of a virtual file system. As a result, it doesn't have to create disk images and work under the FAT file system; it maps directories directly into RAM, eliminating the need for drive letter



The vRamDir control panel specifies which folders operate solely in RAM. Here, the Wavelab temp file is being assigned to RAM; the Sound Forge temp file is already assigned.

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HOW MUCH FASTER IS VRAMDIR?

Operation	vRamDir (in seconds)	Hard Disk (in seconds)	Speedup
Cut one second from file start	1.465	9.183	6.26X
Undo above operation	1.313	1.744	1.32X
Resample to 48 kHz	110.436	113.215	1.03X
Undo above operation	2.179	10.197	4.67X
Pitch shift +2 semitones	30.492	31.365	1.02X
Undo above operation	2.075	2.497	1.20X

assignments and such. Of course, there's no free lunch: because files are in RAM, you need to save often to hard disk — one power outage, and your files are history. The "recover" features of some editing programs work by digging into the hard drive and finding what was written there; with RAM files, there's nowhere to dig.

Another subtlety involves deleting files in the RAM directories via Explorer. If you send the file to the Recycle Bin, the computer has to write the file to disk in order to delete it. The advantage of this is that, should you decide you want to recover the file, you can poke around in the Recycle Bin and find it. If you're sure you don't want the file and want to save time, there is an option to immediately delete the file without saving it to disk first; this frees up memory, but eliminates your safety net.

REAL-WORLD NUMBERS

So exactly how much time do you save by taking the RAM route? Anything that accesses the hard disk benefits from using RAM as a disk. However, heavy-duty DSP operations don't really benefit if the file fits in RAM anyway. And if the file doesn't fit in RAM, then vRamDir doesn't really help all that much either, because it still has to go back to the hard disk and use it for virtual memory.

Let's take a look at some numbers. All these operations were carried out with a 40 MB file on a 450 MHz, Pentium II-class machine running Sound Forge 4.5a. The hard drives are fast EIDE suckers, so the differences would be even more dramatic with slower hard disks.

Clearly, editing operations that need to rewrite to disk benefit the most — check out the 6X speedup when cutting from the file beginning. DSP-intensive functions show little tangible benefit, but undoing those operations provides a significant speedup, presumably because the file is restored from RAM, not from a version saved to disk.

You can also use RAM as a place to record tracks in multitrack recording programs. This often allows for recording more tracks simultaneously, but, as I generally record only one or two tracks at a time, this hasn't been important for me personally. Besides, if your hard disk can handle it, it's nice to know that your playing is actually being preserved in a real, physical location that won't go away if there's a glitch that wipes out RAM before you have a chance to save the part.

By the way, more and more programs are starting to take advantage of RAM for recording processes. For example, Samplitude by SEK'D lets you create virtual projects in RAM, and MOTU's Digital Performer uses RAM for loop recording. Now that you can stuff more than a gig of RAM in today's computers, the days of being limited by hard disk throughput and mechanics may soon be behind us.

CONTACT INFO

If you want to check out vRamDir, you can download a trial version from the Web site, but it's limited to working for 30 minutes at a time. However, the program does write any files in RAM at the end of the 30 minutes to disk, which seems pretty considerate. Cost of the full version is $10 for an electronic download, although they'll send you a diskette for $15 ($20 overseas). For more information, contact Virtual Software Corporation, 15016 SE 63 St., Bellevue, WA 98006-4634. Fax: 425-649-8277. Web: www.virtusoft.com/vramdir.htm. E-mail: support@virtusoft.com.

Craig Anderton just finished mastering the debut CD from the German group Rei$$dorf Force (EMI/Harvest records). He has given seminars on technology and the arts in 10 countries and 37 states, and is the author of the classic text Home Recording for Musicians *(AMSCO, available from www.musicbooksplus.com and www.amazon.com).*

Cyber Al, At Your Service


PHOTO BY JONATHAN POSTAL

Log on and get all your pertinent industry questions answered

BY AL KOOPER

Well miracle of miracles — *EQ* has now diversified in such a way that you and I can talk one-on-one 24 hours a day! They've created forums on their Web site (www.eqmag.com) hosted by certain columnists from this here magazine: Roger Nichols and yours truly, along with George Massenburg and Ed Cherney, to answer your questions in a timely and authoritative fashion. I am humbled by my fellow forum-floggers. You can talk to the people behind Steely Dan, Earth Wind & Fire, Bonnie Raitt, Little Feat, Bob Dylan, The Stones, and Lynyrd Skynyrd anytime you want with any kind of intelligent question pertaining to the music business.

Now let's dissect how this might help you: You're working on a project that can lead to you being selected to produce an act for a major label. If there's *any* question that troubles you such as mic selection, which piece of equipment is best for solving a problem, exactly what your lawyer is entitled to commission-wise, etc., just jump on the site.

You might wanna ask George or Roger about mic selection, Ed Cherney or Roger about equipment selection, Ed Cherney or myself about production problems, or lucky me about external parts of the music business: is this person cheating you, is that person acting in your best interests, etc. Get the idea? Is this a beautiful country or what? And Roger, Ed, George, and myself welcome your participation because we can learn from your questions just what's going on out there in your worlds.

You might wanna ask George or Roger about mic selection, Ed Cherney or Roger about equipment selection, Ed Cherney or myself about production problems, or lucky me about external parts of the music business...

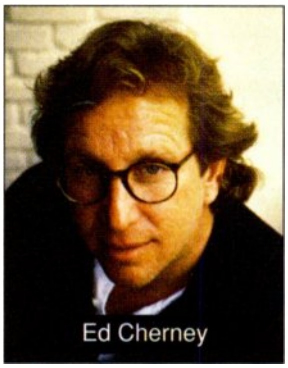
Needless to say, questions like, "What is Bob Dylan really like?" will be answered in such a way that hopefully, they won't be asked again. You guys and gals are bright, talented, and full of ambition. Hopefully, that's why you buy *EQ* in the first place. The powers-that-be have just added another dimension to it with this new free service.

When I was coming up in the late '60s, if I could have been in touch with George Massenburg at any given moment, I'm sure my technical skills would be sharper than they are today. If I could have consulted with Roger Nichols? "Hey Rog — how do you talk those companies into givin' ya all that free stuff?" — well, that would have been *my* first question. Hell, I *still* wanna know that! And Ed Cherney — "How did ya get that cool guitar sound on Bonnie Raitt's *Fundamental Things* album?" See wot I mean, mate?

So let's turn this into the wonderful learning experience it can and will be by using this gift properly. Don't waste your time and our time with silly stuff for the sake of just conversing one-on-one with these people. Ask those things that you're dying to know that are holding you back from doing your best work. These people have sold zillions of records between them, and it wasn't all luck or coincidence. They've got gold records and awards up the yang, and now you can ask them to extricate you from whatever audio mess you've gotten yourself into gratis, 24/7. You wanna rant about how corrupt the record companies are? C'mon in! I'll match ya rant for rant!

As the desk sergeant in *Hill Street Blues* used to say at the end of each morning's staff briefing: "Be careful out there, gang..." but if ya get in a jam, feel free to log on and see if we can help.

It's our pleasure! See ya all online.... EQ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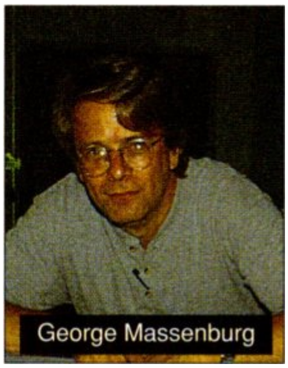
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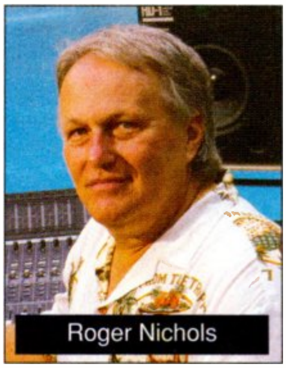
Roger Nichols

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Internet Recording Studios

Recording in the **Crows' Nest**

The Counting Crows' latest release called for some unique recording solutions

BY TONY DI LORENZO

As recording engineers, we spend a lot of time learning the "correct" way to do things; placing mics, dealing with leakage, recording drums, tracking vocals. But sometimes you can get better results by stepping outside of the box, by thinking unconventionally and being open to fortunate accidents.

In the past, Dennis Herring has worked with artists such as Concrete Blonde, Timbuk 3, and Jars Of Clay. On a recent outing with Counting Crows, his approach wasn't what you'd call "conventional." One mic on the drums? A singer lying down while recording? Sure — if the track works without following the rules, then it's a keeper.

The Crows' latest release, *This Desert Life*, was recorded in a house, *not* your average commercial studio. As Herring states, "That's actually how Counting Crows have done all their records, and I'm certainly no stranger to recording like that. I've never liked recording in formal recording studios, so I've always tried to do it as little as possible. I was certainly more than into doing it that way."

Does the band own this house or do they rent? Are they always recording in the same house?
No. It's been a different house every time.

They just grab a house and throw a bunch of gear in it?
The rule seems to be that the house has to be given back in the same condition that it was in when it was leased. So anything could happen as long as it was all fixed when it was over. Counting Crows own a lot of recording gear to begin with, and since this is the third record they've done this way, they've gotten really good at converting houses into temporary recording studios. Putting up Plexiglas over windows, etc., just doing the minimum so the neighbors don't call the cops.

They would even call the cops on the Crows?
Well, it's still noise. One man's music is another man's noise and vice-versa.

What board did you use for tracking?
Mostly a little monitor console. Toward the end of tracking, right before we started mixing, I bought an old Neve 8038. We moved that up to the house and finished tracking on it. The song "Colorblind" was recorded and mixed completely on the Neve 8038. Counting Crows have a lot of old cool gear like Neve 1073 modules and a couple of Neve side cars, so we recorded through that gear even before I bought the Neve console.

Was the house in California?
The house is in the Hollywood hills and is kind of a cross between a *Boogie Nights* and *Brady Bunch* kind of house; '60s-era architecture, pools, view of Hollywood. It was great.

Was that conducive to creating?
Oh yeah, it absolutely was. There is just nothing like being in that kind of environment. It helps them not hear the tick of the clock, which — for the most part — is a good thing for the Crows. When you're recording like this, you don't think about the artist who was in there the day before or who might be recording in the rooms around you. It's nice when you're not comparing what you're doing to other people. That's what I like about recording this way.

When I first started recording and producing, I set up my studio in my basement. This was before the age of the home studio. I did that because I always hated going to "real" recording studios. I had been a studio musician for about five years, so I was certainly used to being in recording studios. They have their place, but, at the same time, I felt, as a producer, that it was hard to get what I was trying to get at. Since I've set up my own studio, it's been easier.

Sure, because when you record at a commercial studio you block out time and you're forced to be creative on that time table. If you have the rig in your house and an idea hits you, then you can run with it.
Absolutely. I remember having this experience at Ocean Way, [which is] a great studio in Los Angeles. We just did this drum thing that we were so proud of, and we were all in the control room

DIFFERENT DRUM: Dennis Herring is known for his unusual recording techniques.

PHOTO BY CLAY JONES

CROW'S FEAT: Engineer Rich Hasal in the house's control room.

kind of high five-ing each other. We were so proud of this thing we had just done, and right at that moment I saw Sting down the hall past our open control room door. And I thought, "Sting would think this is a joke." Recording in your own place lets you wake up every day feeling that what you're doing is special and you can stay in that frame of mind. This has a way of reinforcing what you're doing and helps records turn out to be their own kind of thing.

Do you think that any future projects you work on will be recorded in a house rather than in a commercial studio?

I've had a studio for more than ten years, but it's not like a real recording studio. It's a completely homey, funky place. Sure it's got a great big Neve console and an Otari RADAR — but it's more like your living room than a studio.

Did you cut the tracks to analog tape or did you use digital?

I record exclusively to an Otari RADAR. I have my own RADAR II, and that's what we used for the Crows record. I like it.

What vocal mics were used?

My engineer has a pair of [Neumann] U 47's, and we mostly used one of those on Adam's voice [Duritz, lead singer for Counting Crows]. There was one song where Adam did his scratch vocal while laying on his back, and we liked the sound we got with the Shure SM57. The vocal got a little cloudy because of him laying on his back, and the '57 helped with this tone.

Which track was this?

"Highlife."

So the take with Adam laying on his back made it to the record?

We did some takes and comped between them — we even used some of the scratch vocal, so any of the other vocal takes that were used were done this way.

Tell me about the signal path for the vocals.

There's a guy named Frank Lacy who makes tube gear for me and [producer] Terry Maning. He calls his line of gear "Lucas." When we recorded the organ, we tried out these solid-state preamps he made, and we liked them. They beat everything else, so we decided to try them on the vocals. They had this funny quality; they made things sort of move toward you a few inches in the speakers. This worked well for Adam's voice. I still have them, but it doesn't mean that I just use those on vocals. There's no telling what I'll use on any given day. For Adam's voice, we got into that chain: the '47 through the Lucas preamps. We liked it, so we did it a lot.

Did you set up some sort of vocal isolation booth?

No. [*Laughs.*] I don't know anything about that kind of stuff. I just don't record like that at all. What I try to do is get a singer to sing in the room with me. I like to sit at the console with the singer standing behind the console singing into a mic, and listening to the monitors just like I am.

Why do you like to record that way?

I like the communication. Since it is an environment where you're going to be interacting with the singer, it's nice to get comfortable. Then I'll run everyone else off so it's just me and the singer, unless they want to do it differently. I also like sitting forward with the singer behind me, so you're not staring at them. I've always hated that about studios; where you're looking through the glass, staring at the singer.

Because you're pointing out mistakes and tuning problems.

Exactly. You've been put in this role of judge, and the last guy you want sitting staring at you is the judge. I love having the singer in the same room and being able to stop the tape, turn around, and say, "How's it feeling to you to sing this right now?" or "What if we did this another way?" This way it feels like you're doing something together.

Some of the guitar parts on "Four Days" have such a '60s feel, kind of reminiscent of the Byrds. How did you get those sounds?

It sounds like you're talking about the 12-string part. Dave Bryson had come up with that part, and it was the last thing we did to that song. I think he formulated it as a 12-string part. Then I used this old Korg delay and I kind of performed that song along with him, riding the delay as he was performing the part. That's how we got those big sweeping sounds — it's me and him working together.

How did you isolate the drums? Recording live drums in a house, there had to be some problems there.

I've outgrown the rule that leakage is bad. The engineer I work with is a man named Rich Hasal, and he's extremely sympathetic to my cause. It was very easy to get Rich into feeling that leakage is not a bad thing.

On the song "All My Friends," you got such a tight, natural drum sound. How did you record the drums for this song?

For the basic track we cut just drums and piano. I had been trying to get them to cut that song all together, but, at some point, Adam said the feel wasn't happening. He suggested we just cut piano and drums. That got us into this Memphis drum sound.

What mics did you use for the drums and what about placement?

We were into this Glyn Johns setup for a while. We'd try to use a bare minimum of mics on the drums. A common mic setup for Rich and I would be a U 47 on the kick, a '47 off the second rack tom, and another '47 behind the drummer, slightly to his left. Then maybe a hihat mic or a snare mic, but maybe not. We did a lot of the drums on this record with one or two mics. The song "Mrs. Potter's Lullaby" was done with one mic. The track was recorded live. Every element, including the vocals, was all done at the same time. If the bass sounds too woomy, then you just turn down the bass amp a little bit so that it's not going into the drums so much. If you want the bass to have a nice woomy quality in the track, then get the amp closer to the drums so you'll get more bleed into the drum mics. You'll then get some sort of stereo element to the drums and the bass gets spread out for the final mix.

It's funny, but all through the '80s, as technology became better and cheaper, leakage was outlawed.

I know.

Everyone was thinking, "We'll use the drum machine because we can get a more controlled sound," and now you've gone back to everyone playing live in the same room, like people recorded in the '50s.

I grew up in that '80s school. The first records I produced were from Timbuk Three. I was fascinated with them because it was just two people and a drum box. Everything was drum machine and I did all the engineering because, hey — I can record drum machine. The fact is that all my favorite records have leakage. I never want my real drums to sound like a drum machine. I think loops are great. I remember when all the sampled loop drum parts started to kick in around the mid-'80s with groups like Run-D.M.C. and Public Enemy. What a great way to hear drums!

The loop idea was great because you were still getting the live feel of someone's performance, but you were placing it where you needed it.

Exactly. We were talking about drum mics earlier and I remembered the way we recorded "Four Days." That song went down in a funny way. Adam had just written the song, and he was really excited to record it. It must have been about 2:30 in the morning and basically the band on that song was whoever was there playing poker. Ben [Mize, drummer] is on it, but he was drunk. I told Rich to just put up one mic. Ben was trying to get his bearings on the drum throne, and I told him we were going to cut this song. Ben said, "I haven't heard it." I said, "Yeah, yeah, you're gonna play great. Here's the thing: you get one mic. Where do you want us to aim it?" And I think he said, "Kick drum." So "Four Days" was a one-mic drum sound.

Jack Joseph Puig mixed that song for the album, but I was not in attendance for the mix. He called me and said, "Hey Dennis, I'm mixing this song 'Four Days,' where are the real drum tracks?" So I said, "What do you mean?" He said, "I have this tape and it has one drum mic, but I really want to get a bigger sound, so where is the tape with all the individual drums?" I told him, "That's it." Puig said, "So basically what you're saying is anybody who gets this tape has to mix your drum sound, and if they don't like it, then f*ck 'em." I said, "Yeah, that's pretty much it. If you don't like the drum sounds you get with that mic, then f*ck you. That's what I'm saying." There's nothing better than getting what you're really trying to do on tape right off the bat.

"Hanginaround" has an interesting overall feel to it. How was that recorded?

[*Laughs.*] Yeah that one's pretty neat. David Lowery [co-producer] had been working with the band on the demos, and the song started as a demo that the band recorded. Adam had his original demo and David was trying to record a new demo with the band. In the end, the only way he could get anything down was to take Adam's demo, put it on tape, and then have the band play to that — sort of like a click track. Just because Adam's thing had such a feel. The whole building block of the song is something that's incredibly out of time. As opposed to a click track that's in time, that song has a guide track that was fantastically out of time, and it's still in there in the final mix.

And everybody was trying to play to that?

Yeah. There was a certain amount of out-of-time stuff we wanted to keep, but the problem was that if you tried to use the drums really loud, it became so flammy and sloppy that it didn't feel good at all. It just wouldn't work. David and I came up with a new way of looking at the song: If the bass was really kickin' ass and the bass player was just so "James Jamerson" about his job that he was grooving hard, you wouldn't even notice the drummer. Do you know what I mean?

You were relying on the bass for the "center"?

And handclaps. Adam wanted these big group handclaps. I spent a whole day sliding those handclaps around to make them super in-time with the bass.

So the drums could be as loose as you wanted, as long as you had the bass and claps anchoring the beat.

Yeah, we worked a long time on that bass track to make the part and the performance so rhythmic and driving. That's what gave the song its groove. We then made the handclaps and vocal work with it. This meant that, when you mixed it, you always had to be thinking in non-traditional terms. When I mixed the song, that's the way I did it. Jim Scott did a mix and I sat there and reminded him that we had to focus on the bass, the handclaps, and the singer. If those three elements feel good, then we'll have this great record.

The strings add such a sweet touch to "All My Friends." Did you record the strings in the same house or did you go to a separate studio?

I've recorded strings before, but I was hoping I could get a sound that was more unique for this record. Rich had said, "Let's just record the strings up here in the house. There's a spot I think would work really great." So that's what we did. We had string players come up to the house and we made an event out of it. We got them up to the house early and had a chef make dinner for everyone. It felt like they just came to someone's house, you know, just hanging out.

I was getting a great string sound, but everybody kept telling me, "The strings don't sound good. We gotta move them to another part of the house." I said, "Let's do it, if that's what you're thinking — but make a note of where they are now because I like the way the strings sound." So we took an hour, moved all the chairs and miked all the strings, but to me it just sounded generic. We wound up moving them back to the first spot and they sounded great. You can do stuff like this when the budget is not an issue.

Tony Di Lorenzo has been involved with synthesizers and keyboards for the last 20 years. He has produced his own CD-ROM for the Kurzweil K2000 and K2500 called "Producer Series Vol.1." He runs his own company called Front Room Productions. You can visit Front Room's Web page at www.interport.net/~thefront/index.html. You can also e-mail Tony at thefront@interport.net.


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Putting Song **to Picture**

Greg DeBelles describes the trials and tribulations of composing an original song for film

BY STEVE HARVEY

Years after making the transition from Broadway to Hollywood, composer Greg DeBelles has found a niche writing and producing original songs for motion pictures. His involvement with the new 20th Century Fox release, Where the Heart Is, *provides some insight into the process, from conception to final mix.*

Time was, every film seemed to feature an original song: remember "Moon River" from *Breakfast at Tiffany's*, or "The Windmills of Your Mind" from *The Thomas Crown Affair*? The practice became somewhat moribund, except in Disney's animated features, to be replaced by eminently marketable collections of hits from current artists. But Jennifer Warren's "I Don't Want to Miss a Thing," from *Armageddon*, demonstrates that a well-crafted original composition still has a place in the movies — particularly when a character performs a song within the context of the story, of course. Which is where Greg DeBelles comes in.

DeBelles begins with the call for him to produce. "I've developed enough of a reputation that, when the requirement came up at Fox to have this material pre-recorded, the call came to me. I was very happy to do it because I love all aspects of making music equally — recording, writing, every bit of it."

In the screenplay for *Where the Heart Is*, one of the characters composes a song while in jail. Upon release, he puts a band together, going on to fame and fortune. Once the film's executives chose the song, it needed to be pre-recorded in several different versions for playback on the movie set, where the actors would mime their parts.

DeBelles's background and previous production work with numerous recording artists made him an ideal choice for the project. It also helped that he has a studio behind his Hollywood residence. Half control room, half live room, the studio incorporates TASCAM DA-88 recorders, Digidesign Pro Tools, and MIDI sequencers and keyboards, all hooked into a D&R Cinemix console. "I'm very happy with the sound of that board," he says. "It's very punchy sounding and transparent."

"I was asked to submit a song, "explains DeBelles. "There were people all over everywhere writing songs to be *the* song. We were given certain guidelines — they wanted to have 'beat of the heart' in the lyric. All the writers had different concepts for the song. It was interesting, because, being hired to produce it, I was privy to all the song submissions."

Happily, DeBelles's song was chosen; now they needed a singer. "There was some discussion of the actor, who of course wanted to sing it," says DeBelles. "It was decided we needed a real singer. I was given 48 hours to find one. The third day I was on a plane to Austin [where the film was being shot] with the song, recorded with seven or eight singers."

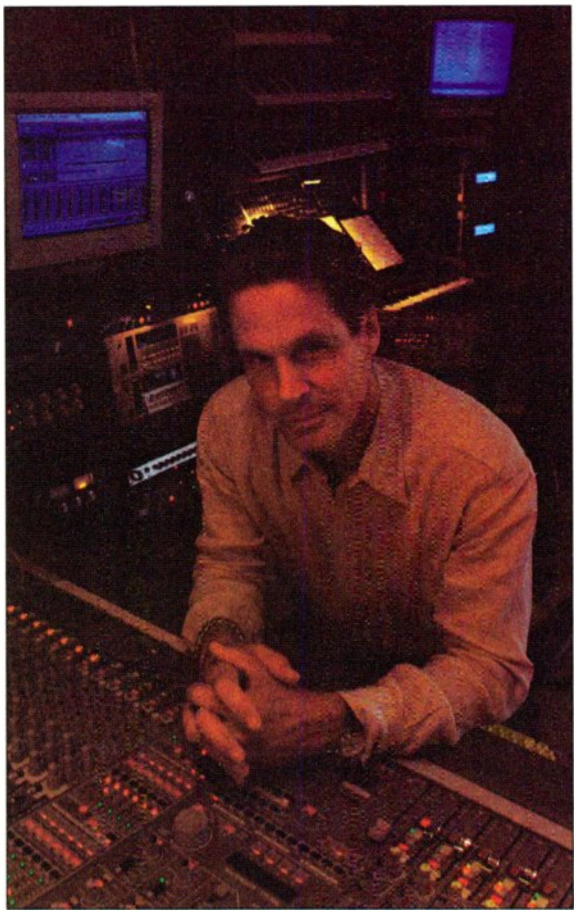
The director and producers chose the version sung by Michael McCarthy, a Los Angeles local DeBelles had discovered at an open mic session. DeBelles called McCarthy with the news and had him flown to Austin.

PHOTOS BY STEVE HARVEY

"The next day we were in a recording studio, with musicians I had never met before," continues DeBelles. "We started doing the pre-records. We had to do them over two days, with just one day of recording." Several versions were required, including the jail cell genesis, a club band version, the audition for the agent who ultimately makes the character a star, and a full recording studio version.

The sequence of events posed some challenges for DeBelles. The initial hurdle: How do you pre-record a song that convincingly conveys the act of spontaneous composition, yet can still be mimed by the actor?

DeBelles discussed the jail scene with the director, to understand "the cadence upon which he wanted the song to unfold. In other words, what were the character's emotions as he was singing it? It was a really heartfelt song, erupting in the guy's cell, from a borrowed guitar. I said to the director, 'Direct the actor and I'll conduct the musicians to follow suit.' "



"Once we had the start, stop, and stumbles on the guitar — that were enacted by a very good guitar player — I digitized the performance of the guitar and made a little digital object of 4 or 8 clicks," he elaborates. "On the stage, I could just move the clicks right up to the start of the thing, which could then cue the actor. Because there was no way of anticipating; it's completely random. He's composing this song — it comes when it comes."

There were more challenges ahead. In one scene the character is performing in a cowboy bar and a fight erupts. He gets hit on the head and the bass player gets knocked over. DeBelles asks, "How do you anticipate, when they are shooting it weeks later, when the bass player gets hit and when the melee breaks out? You end up multitracking the mix, splitting virtually everything off, so then it's just a matter of omission. The guy gets hit — just duck out the bass."

"It was delicious being back in Austin," confides DeBelles. "I actually worked analog for the first time in a long time, on an Otari machine." DeBelles's first choice for the project, Willie Nelson's studio, proved to be too much of a commute for the Hollywood executives. DeBelles settled on Mark Hallman's Congress Hall, chosen for its tracking room, monitoring, vintage equipment, and Hallman's professionalism.

At the end of the session, DeBelles left with a Pro Tools AIFF file on a Jaz disk. "I flew home with 24 tracks of all the different recordings I'd done — all the vocals, guitar, and rhythm tracks — and then I loaded them into Emagic's Logic Audio and worked with them from there."

That's when disaster struck. DeBelles discovered that, somehow, the engineer had set up the click track incorrectly, and, instead of chasing timecode, it had triggered randomly. "Which was a nightmare," he explains. "My plan was to replace a lot of the performance with some of my own musicians here in L.A., so I needed to have a click generated on the computer to lock up sequences.

"Over the course of the song's three minutes, there were thousands of timing changes. I spent a full day reclocking the songs with a function in Logic Audio called Reclock Song. It's a very powerful function that allows you to play and record anything without a click, and then it will create a click to the performance. It ultimately saved my life."

DeBelles replaced the drum tracks and the bass when he discovered that, in the heat of the moment, he had missed the bass player playing the wrong changes, added more vocals, and some orchestration.

For the final dub stage, DeBelles and his mixer, Rick Norman, prepared an eight-wide mix. "I really like stereo," DeBelles states. "If it's a melody, or a vocal, or a mono instrument, I like to have a stereo ambience. The way I split will differ from project to project, and, in this film, since it was just basically a band and a pop song, I had vocals, drums and rhythm, all guitars, and the rest of the band all split in stereo pairs."

"Because of the film format, I mix things really wide," continues DeBelles. "Then they can turn it up and it doesn't fight with the dialog and the effects. It helps you get your music louder! I try to mix it so that when I deliver it onto a stage, just have the faders at unity straight across and everything should just fly right in perfectly."

The Cinemix, which has upper and lower moving fader automation plus dynamics, made DeBelles's task simpler. "It used to be a battle, especially on a project where you're mixing for surrounds; half the console is just monitoring, with separate reverbs for every stem. When you're doing split mixes, everything needs its own ambience, there's all this extra processing. This console has 10 auxiliaries, but you can route the auxes to the busses, so it gives you 34."

The console's Advanced Routing Matrix (ARM) puts auxiliary, multitrack, and main bus routing under automated control. "Normally, since it's a true dual input console, I have all my synths coming in on the uppers and all my tape returns and Pro Tools coming back on the lowers," notes DeBelles. "My engineer has a pretty daunting assignment of splitting off and bussing things to where they need to go. Having the digital routing, the ARM, has been a real blessing."

An upgrade to Mackie HR824 speakers emphasized the importance of good monitors. "With the NS-10's, there was always the lurking *Red October* low end. You'd show up at a dub stage and, 'Oh my God! Where did that come from?' The Mackies are great sounding," enthuses DeBelles.

DeBelles is satisfied that, despite the hiccups, the project was a success. "The remix was done at Todd-AO in Hollywood. They got their splits and everything went down like a charm." EQ

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Inside View: Waves Supertap

Some tips for getting the most out of the delay plug-in

BY RICH TOZZOLI

Think you know everything there is to know about delays? That you've done it all delay-wise? If so, Waves SuperTap may have a few surprises in store for you. This delay plug-in offers a variety of capabilities that go beyond what's normally found in a standard delay, and allows you to take the concept of "delay" to a new level. Let's take a look at how to get the most from this powerful processor.

First, some background: SuperTap is a six-tap delay plug-in that can produce analog and tape delay emulations, chorusing, rhythm-based looping, and multi-tap delay effects. It provides a maximum delay time of six seconds (less with modulation), and all of the taps are "feed-forward," meaning they're all mixed together at the output, rather than feeding into each other. SuperTap is more than just a simple delay line, though, as it also features sophisticated filtering and feedback capabilities. A low-frequency modulator (20 Hz and below) can create small changes in the delay time of the taps, producing slight changes in the pitch of each tap output.

SuperTap is available for both Mac and Windows NT platforms. Mac support includes Pro Tools Mix TDM systems, as well as RTAS, VST, MAS, and Premiere formats. In addition, DirectX applications for Windows are supported.

The SuperTap graphic screen is neatly laid out with a Pan Graph, Tempo Section, Modulation area, and Gain/Output section on top. Below these are controls for the direct (dry) sound, delay lines (taps), feedback, mode selection, and an EQ (filter) section. I'll be covering most of the important controls in each area, but I won't be able to cover them all in this article — refer to the SuperTap manual for more information. There are basically two types of SuperTap processing: with long delay taps and with modulation. You can also choose between two or six taps (two taps take less power and memory), and mono or stereo operation.

Waves has included some great presets for you, such as 15 ips delay, Spring reverb, and Ping Pong, but SuperTap is such a powerful program, my recommendation is to use these as a starting point for creating your own unique effects.

GETTING STARTED

To get started working with SuperTap, simply call up the plug-in on a track, or bus a track to it on an aux input, then choose one of the various versions, such as the six-tap, six-second stereo version, and away you go!

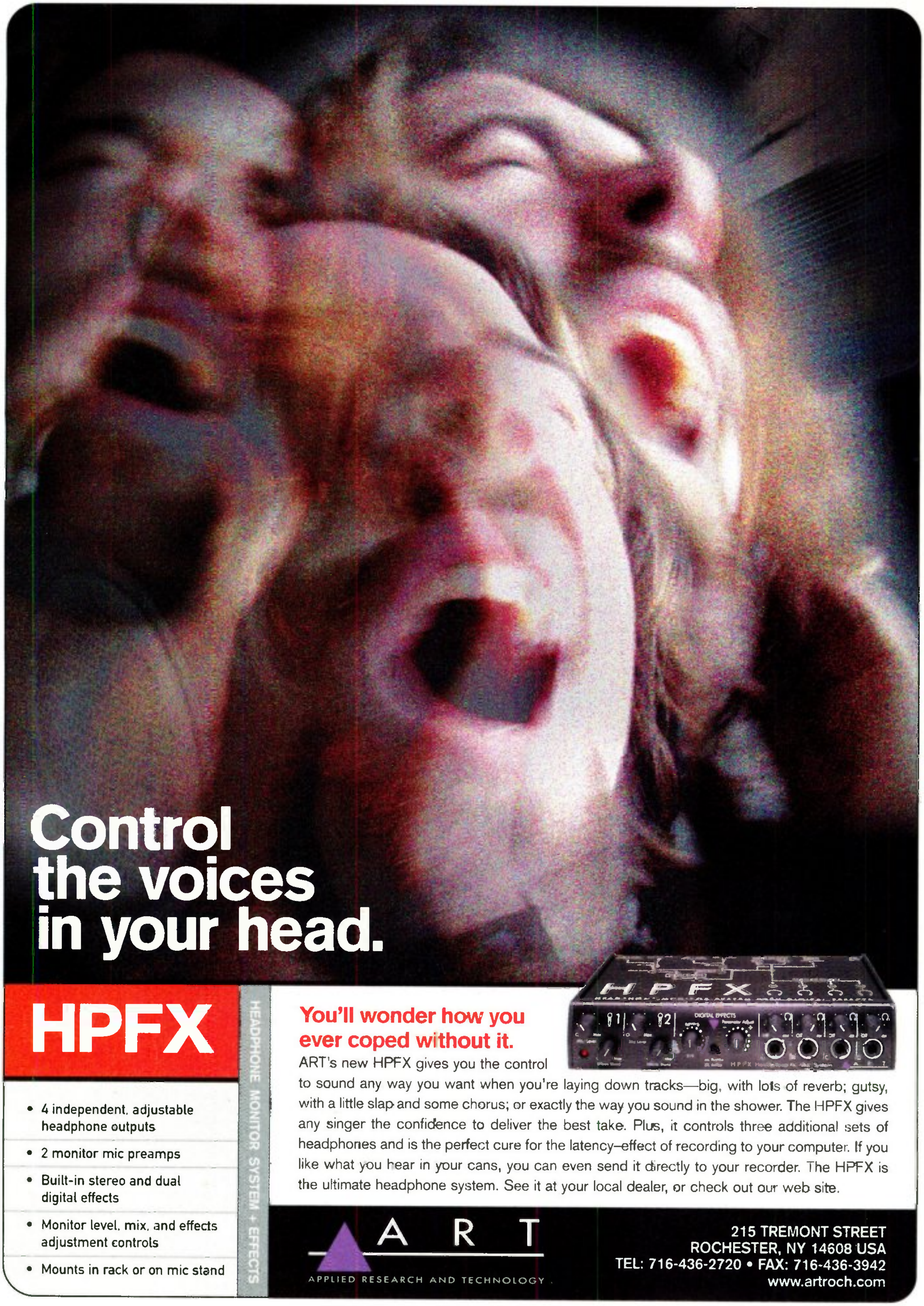
First, you have to turn the individual taps on, using their On/Off buttons. When you turn a tap on, it illuminates three things: the On/Off button itself, the delay slider, and the tap's marker within Pan Graph (stereo only). You can now drag a tap delay slider to set the delay times. Displayed inside the slider will be the delay time in either milliseconds or note divisions (quarter-notes, eighth-notes, and so on), depending on the Grid Mode setting, located at the bottom of the plug-in window.

Moving to the Pan Graph, you can use the mouse to drag the illuminated tap marker to make changes in gain and stereo position (rotation) at the same time. (If you prefer, you can enter each parameter directly using the Gain and Rotate controls.) Since SuperTap works with mono and stereo inputs, the term "rotation" is used (rather than pan). Waves uses rotation to apply to both mono and stereo input positioning in the stereo field. As you move a tap marker around the Pan Graph, the corresponding values in the delay section will change accordingly. You'll quickly adjust to working this way, and it becomes quite easy to manipulate your delays.

In the Tempo section, two modes are available: Pattern and Tempo. Pattern mode allows you to tap in (click with the mouse) a rhythmic sequence, which is automatically translated into corresponding delay times for the taps. In Tempo mode, you manually tap in the tempo of the song, then manually (using the sliders or Pan Graph) set the delay times for each tap.

As you would expect with delays, there's a feedback section, which you must also switch on. There are two feedback modes: Tap Feedback and Norm (normal). Normal works as you'd expect, simply sending each tap's output back into its input. Tap Feedback is a different thing entirely. It has its own delay time, and repeats the overall output of the tap grid, allowing you to "echo" entire rhythmic tap setups. You can apply EQ/filtering to the feedback, allowing for some very creative effects. A Rotate control for the feedback lets you play with the panning of the repeats.

The EQ section provides filters whose design is based on Waves' Q10 equalizer. Each tap has its own single-band filter, which



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has a frequency range from 100 Hz to 20 kHz. There are six different filter curve types available (various shelves and bells). The Frequency button allows you to adjust the filter's center frequency, and the Gain button boosts or lowers the gain of the filter. Using this section creatively, you can assign six different filtered taps to a single hit to create a unique multi-tap rhythm. The filters allow you to change the tone of each tap, taking things far beyond what can be accomplished with normal delay repeats or delays that simply get progressively darker with each repeat.

STEREO GUITAR DELAYS

To get some great stereo delay effects on guitars, set two of the six taps panned hard left and right on the Pan Graph. Next, pan the dry guitar as desired on your DAW mixer and set the delay levels quite a bit lower than the dry source, with the feedback turned off. To increase the stereo effect, play with modulating the delay times slightly or filter the taps to change their EQ. This can increase the impression that there's more than one guitar playing together in very tight synchronization or simply make the stereo spread seem wider.

As I'm working on a track in this fashion, I'll highlight a small selection of the audio and quickly hit the spacebar to start and stop playback. This allows me to check the delay times, panning, and filtering of the taps, and quickly adjust them to fit the rest of the mix. In general, anything below 150 ms of delay will create a nice ambient space without a perceptible delay, which usually fits into the mix better. I often set my DAW to continually cycle playback for the audio selection, and move the delay taps in real time to get the desired effect set up.

5.1 VOCALS

I've gotten great results mixing vocals in 5.1 surround with SuperTap. By taking the same approach described above, but assigning the plug-in to the rear channels, a vocal can take on an incredibly ambient sound, but, again, without the delays becoming blatantly noticeable. You notice the delay effect is gone when you take it out, but when it's in, it just blends into the song's texture. By automating various parameters, you can create some amazing effects such as tracks and delays panning behind your head, having the delays swell in volume on certain lines, and having delay times change during the song. Once again, experiment with slight modulation and filtering to broaden the effect.

DRUM LOOPS

With a drum loop and SuperTap, you can easily set the BPMs (referenced to quarter-notes) by tapping (clicking) on the TapPad in the Tempo section with the mouse. The program will automatically calculate your tempo based on the taps, and you can tap for up to two minutes to get a precise value. Of course, if you already know the tempo, you can enter it directly in the parameter value box. For some effective and wild loop sounds, I activate all six tap delays, time them rhythmically, and then enable the EQs and set them to radically different settings. In this application, SuperTap becomes more than just a simple multi-tap delay, it's more of a true audio processor and sound–design tool.

Let your imagination run wild with this plug-in — you'll find yourself taking the concept of "delay" places its never gone before. It takes time to learn how to get the most out of any plug-in, but, believe me, SuperTap is worth the investment. Get to work! EQ

"Local" Hero

Producer/engineer John Leckie gives his opinionated views on recording, as well as some of his time-honored techniques

BY HOWARD MASSEY

Every Englishman worthy of the name has a "local" — a neighborhood pub that they patronize regularly, sort of like the bar in Cheers. *John Leckie's local is the canteen at London's famed Abbey Road studios.*

Hardly surprising, actually, when you consider that Leckie's career began at Abbey Road in the early 1970s, assisting on John Lennon's Plastic Ono Band *and George Harrison's* All Things Must Pass *before going on to engineer Paul McCartney's* Red Rose Speedway *and Pink Floyd's pre-*Dark Side *album* Meddle. *In the late 1970s, he made the transition to producer, working with some of the premier post-punk bands of the time, including Be-Bop Deluxe, Magazine, Human League, Simple Minds, and XTC. By the late '80s, Leckie was working with "second wave" bands such as the Stone Roses, Gene Loves Jezebel, and The Verve. In 1994, he hooked up with Radiohead, going on to engineer and produce their critically acclaimed album* The Bends. *Leckie was kind enough to invite me to his local one evening not long ago, where he expounded on his philosophy and techniques over a pint of Guinness or two.*

When you record backing tracks, are you just going for a good drum track, with the idea that all the other parts will be replaced later?

No, I like to get as many people as possible playing together, with the attitude that what they're playing is for keeps. You don't want to set up a situation where a bass player, for example, is thinking, "It really doesn't matter if I make mistakes or if I lay back, as long as the drum part is okay."

If I'm not confident that the band can do that, then they need further rehearsal to get things tightened up so that they can play, at least, bass, drums, rhythm guitar, and a guide vocal together. If they can't do that reasonably well, ready to be recorded, they should still be in the rehearsal room.

It's very important, when you're doing a guide vocal, to have the attitude of the vocal. You should almost be going for a keeper on the guide vocal. If you're going to do a guide vocal that doesn't have attitude, that's going to be lazy and not relate to the song, you might as well just count bars. I know it can be difficult after four, five, ten takes, and I know we're really just trying to get the drum track, but the drummer is influenced by everything the singer is doing. If the singer sounds bored in the drummer's headphones, the drummer's not going to play so well.

So you set up the guide vocal as if it's going to be a keeper?

Oh, yeah, of course. I'd say in 50 percent of the tracks I record, I use the guide vocal.

Do you use leakage as a creative tool?

No, I try to avoid leakage. I might create a leakage track, route an ambient mic somewhere. It's better if you have two tracks available so you can record it in stereo. It's interesting to put microphones in different places, like on the floor or half an inch from the wall, pointed at the wall. Crazy things like that — microphones in pipes or in tubes while the band are playing. Another thing I've done is to put an acoustic guitar in an open tuning in the key of the part and lean it up against the bass amp, then mic it.

So, presumably, you typically record a bass amp track.

Always both, along with a DI. In 90 percent of the tracks I do, I use both. The amp is usually an Ampeg SVT with a

IN THE BEGINNING: John Leckie got his start assisting on albums such as George Harrison's *All Things Must Pass.*

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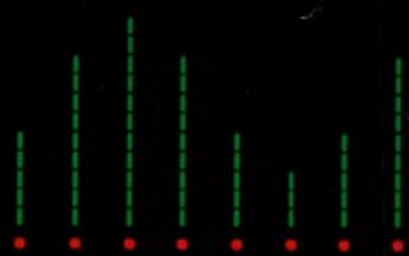

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slightly thinner, harder sound, while the '67 gives you the warmth and a broader sound. If you brighten up the '67, it's totally different to brightening up the '58, so sometimes I'll add a little brightness to the '67 and a little compression. But, between that combination, I find that I can get pretty much everything I need. Again, they rarely are used at equal level; sometimes I'll favor the '58 with the '67 at 10 or 15 dB down. Even 20 or 30 dB down, just bringing it in, it's amazing the different color you get — how much the tone of the guitar changes. Again, it's down to decisions; even if you decide to record the two mics on separate tracks, you've got to decide how you're going to monitor the signal, where to position those two faders.

Making records is about making decisions. All the time, you're making decisions. If you delay those decisions, you pay the price of having to sort them out later. And they mount up, so the sooner you make them, the better. So generally I'll record both mics onto one track and decide there and then how much the '67 is going to add to the '58; make the balance. I'll devote a lot of care and attention to doing that, even if it's the first rhythm guitar. Because it's very important — the first backing track has to have the attitude of being a keeper.

What order do you usually bring faders up when you start a mix?

I usually bring them all up. I work with as many faders up as possible, because what you're doing is making a balance of instruments. It's easy to spend six hours fiddling with the drums, but they may not work with the vocals and instruments. You've got to spend six, eight hours fiddling with things individually, then bring them in together, add in the vocal, decide it all sounds terrible, and then start all over again! [*Laughs.*]

One common mixing problem is getting a vocal to sit right with a backing track. How do you deal with that?

It's a funny thing, because there's this thing about having loud vocals, especially in the last few years here in Britain; everyone likes having the vocal shouted in your face. But I think everyone's starting to get a bit tired of that — I know I am. So I think we're going to see vocals mixed down a bit further. But there's an art to balancing the vocal within the music. The main thing is to limit or compress the voice, but without squashing the life out of it. Be careful of sibilance, but, if you have to brighten the voice up, do it without making it sound thin — keep the voice sounding warm. With a good singer, a vocal can usually go from recording to mix without any EQ.

Sometimes, with a band, if you put the vocal way up high, it doesn't sound like a band anymore — it sounds like a solo singer with backing musicians. At the end of it, you really want the four instruments — vocal, guitar, bass, and drums — at equal level, so you can hear everything. That's why, very often, rough mixes are used on an album. The idea of a rough mix is simply to hear what you've got, so you make your balance so that you can hear everything. And, really, that's all you want out of a finished record — to hear everything. The worst thing is things getting obscured. People will turn the music off when they can't hear everything; when it's a strain to hear what's going on. It's like watching a movie that's filmed in darkness so you can't see what the actors are doing.

> *"Making records is about making decisions. If you delay those decisions, you pay the price of having to sort them out later."*

In terms of vocal compression, sometimes I use the UREI 1176 full on, with the needle pinning all the time, but always at low ratios. Always use the most expensive compressor you can get, because there's a reason why it's expensive. If you only have access to a cheap compressor, you're better off not using it at all on the vocal. It's worth hiring in a good compressor for vocals.

But the most useful tool for taming vocals is automated mixing. I can spend up to three, four hours manually riding each vocal line, making it sit, getting all the quiet bits louder so they don't disappear — but, again, without changing the performance, because once you move that fader, the position of the singer changes, obviously, which means his attitude changes. The big question is where you put the vocal up against the snare drum, because the snare drum and vocal, in rock music, have to fit together. It's where each syllable hit goes with the snare drum — I guess that's what I do when I mix vocals. It's about making it sound like it's never been touched, even though the fader's going up and down. It's a great technique because all the little secrets get revealed. At the ends of lines, a lot of singers will trail off, and if you lift the fader 10 dB right at the end of the line, there's lots of things you haven't heard before, and sometimes there's a lot of character in the breath when the singer finishes his line; suddenly there are new things happening in the song.

Do you tend to set up two or three generic reverbs and route multiple instruments into them, or do you use discrete reverbs for each instrument?

I tend to have two reverbs, one delay, and one special effect — flange or chorus. That's the basic setup I would use, and I'd send multiple instruments in. I never really use discrete effects; I don't know where that all came from, actually — perhaps it's something to do with the '80s when there was lots of equipment available for the first time.

In terms of effects, you're always searching for something that catches your ear, something that's going to enhance the song. It's that little bit of magic: Sometimes you get a reverb or a setting that seems to be made for a voice, and suddenly it all fits. That's really what you're going for — one whole sound.

Sometimes you can use effects to give the impression that there's even less reverb than no reverb! You can actually make something drier by adding something. For example, using the Lexicon 480 Small Room algorithm, or something with an early reflection, 40 ms or so. Anything that's short and a little bit dark kind of makes the sound a little bigger and a little drier as well. One thing I like to do — which is made much easier by automation — is to use a small room on something, and then, once the listener is aware of the sound, just cut it out, then bring it back in again. Whenever there's a little sibilance

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or a little something that ends in a sharp attack that sets the reverb off, just trim the effect send during the mix to where you can't hear it. You've still got the bigness, but it never sounds "reverby."

You can't just say that the vocal either has reverb or no reverb, because there are all these things in-between, like the short room and the little delay. I'm a big fan of using a tiny bit of rock 'n' roll delay — 250 ms, 400 ms. That delay came about from tape echo, and you either ran the machine at 15 ips or at 7 1/2 ips and you made the delays dull so the sibilance didn't repeat. And just use that delay maybe once during the whole song; that gives it the mystery — that little bit of magic. Because when you hear the human voice, the mind instantly thinks of it singing in its place, and, suddenly, halfway through the song, there's this little other place going on. Suddenly you're not so certain, suddenly you're thinking, "Maybe he's not there at all." Maybe that's what makes people want to listen to records again — "Hang on, let me hear that again. Where was that person? Where was he standing?" All those things — like in *Dark Side Of The Moon*, where you don't quite know where the vocal is, whether it's close to you or far away from you.

The whole idea of putting reverbs on records is interesting, because it depends on where you play it and also where you record it. Different control rooms in different studios will make the reverb sound different. For instance, if you listen in a really small, dead control room, you tend to add more reverb than you need. When you go to a more live mastering room, there may suddenly be too much reverb. Through experience, you learn those rooms. But the trend now seems to be towards bigger, more live control rooms, so we've got deader records. People aren't putting as much reverb on records simply because they're hearing the reverb in the room, so they don't think they need it. But when they take it away, it sounds dry.

That's a problem that can be even worse in project studios, where often the control rooms have minimal acoustic treatment.

That's right. But all acoustic treatment's rubbish, I think. I know it's difficult to build studios, but I don't think you *can* control an acoustic environment. You listen to records in a normal room, with carpeting and a sofa and curtains. You don't listen in a room with a hardwood floor, bass traps, and a funny-shaped ceiling. As soon as I walk into a typical live-end/dead-end control room with bare floors, I ask for some carpets on the floor. The studio manager inevitably asks me why, and my answer is simply that I listen to records in rooms with carpet on the floor. Usually, it sounds great, because the room gets deader, so you create a brighter mix. And when you take the mix away, it sounds better because it's brighter, more radio-friendly. So, with project studios, don't be frightened to deaden it down. All that thing in the '80s with creating live-end/dead-end control rooms — it's all bollocks, really. All you need is heavy velvet curtains, or hang carpets on the wall, and put eggshell crates on the ceiling. It's much better than spending a fortune on bass traps and fancy acoustic treatment.

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

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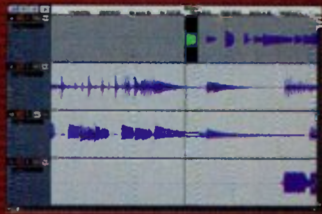

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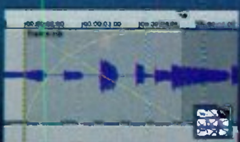

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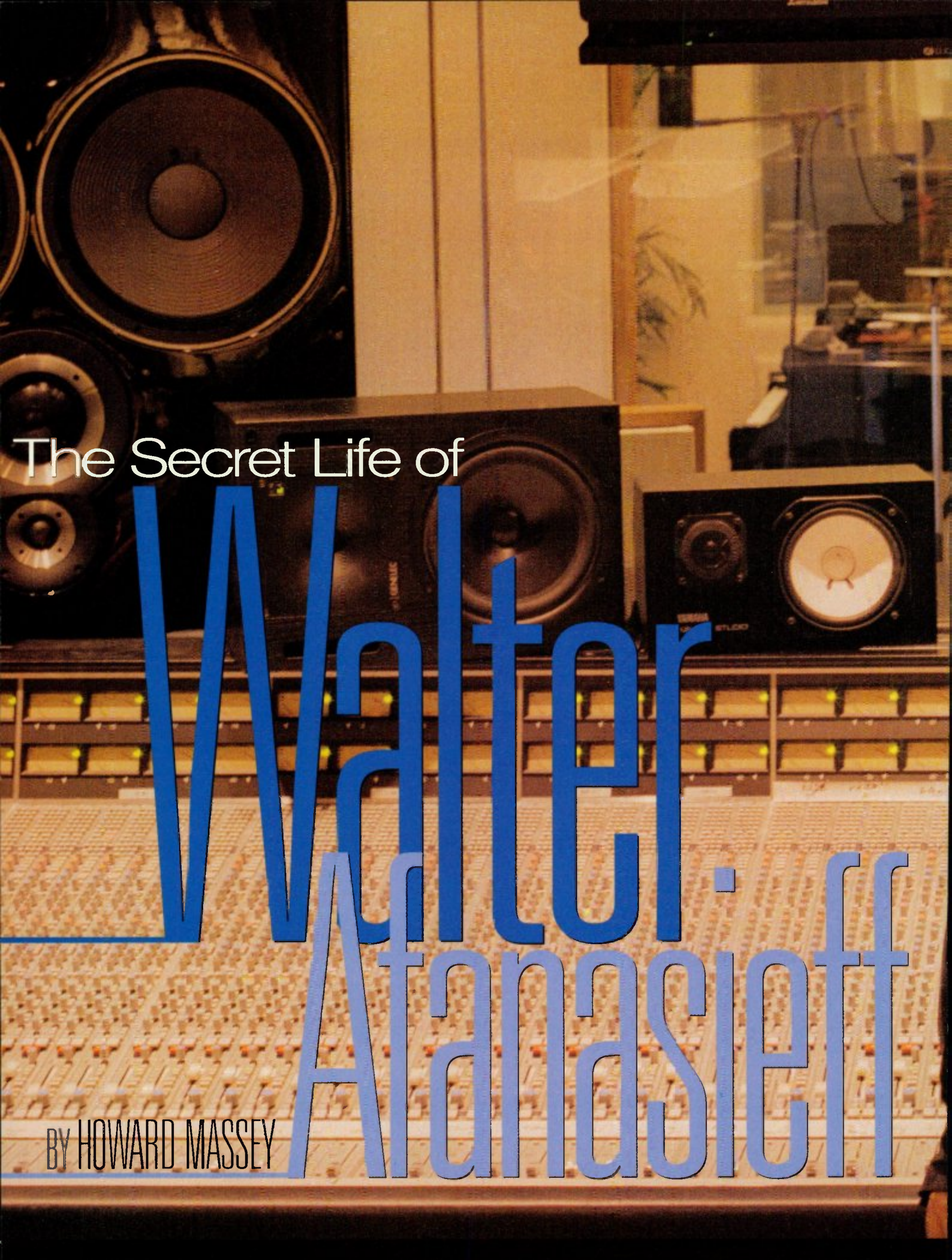

The Secret Life of Walter Afanasieff

BY HOWARD MASSEY

The man responsible for some of the biggest hits of the last decade is given his due with

PHOTO BY ROSS PELTON

a Producer of the Year Grammy win. Now find out his hands-on techniques for success.

Walter Afanasieff

Walter Afanasieff is, by his own admission, a perfectionist. He's also one of the top record producers on the planet. Coincidence? I think not. For more than a decade, he has almost single-handedly crafted a rich pop sound that has resonated with millions of listeners around the world.

A talented multi-instrumentalist (keyboards, bass, drums) and self-confessed fan of technology, Afanasieff takes a complete hands-on approach to his productions, often writing (or, with the artist, co-writing) the songs, devising the arrangements, and even playing most, if not all, the parts himself — presenting the artist with a complete backing track to add their voice to. After working with producer/drummer Narada Michael Walden for several years, Afanasieff set out on his own in 1989 and almost immediately struck gold, working with a then-unknown new artist by the name of Mariah Carey. As her career exploded, he soon found himself working with an incredible roster of internationally renowned artists, including Whitney Houston, Celine Dion, Barbra Streisand, Michael Bolton, Kenny G, Peabo Bryson, Luther Vandross, Ricky Martin, and Marc Anthony, as well as producing the band Savage Garden (*Affirmation*) and creating music for the soundtracks of major motion pictures such as *Beauty And The Beast, The Bodyguard, Hurricane,* and *License To Kill.*

I spoke with Afanasieff just days after he won the ultimate acknowledgment from his peers — the 1999 Grammy for Producer of the Year. Soft-spoken and somewhat shy, he shared his views about making records and also gave us an in-depth analysis of the famed Afanasieff sound. If you could bottle it and sell it, you'd make a fortune!

EQ: Congratulations on winning the Grammy — it had to have been an incredible moment in your life.

Walter Afanasieff: It was one of the greatest feelings I've ever had. I've only been producing for a little over ten years, so this was very special.

I seriously thought that Matt Serletic was going to get it this year because he produced the song "Smooth." It just got so many Grammys that night — the whole Santana thing — so why wouldn't it cross over into the Producer of the Year category?

But finally, for some reason, someone said, "You know what? Let's just give it to Afanasieff." [*Laughs.*]

In some ways, you're a throwback to the old school of production, where you do all the arranging and a lot of the writing, often creating entire backing tracks by yourself even before the artist sets foot in the studio. Obviously, not every producer can provide that kind of "full service" approach; especially with the Pro Tools style of recording — a lot of producers today are more technicians than musicians.

But no matter what day or year you're in, music is music. We're talking about songs, and a record is only as good as the song. Sure, you can go and Pro Tool anything to death, but, if you don't have a good song and a good performance from the beginning, you're going to get into trouble. You may get away with it once or twice, but you can't always get away with it. So my philosophy has always been, I'm a musician, this is a really good song, and let's just take it from there. We can either go into a studio with a live band or we can go into a studio and sit at Pro Tools all day long. Without question, you can do *anything* in the studio today. But I guarantee you that, without a good song, you're not going to have a good production.

I gather that arranging is something you take great pride in doing.

There are different ways to produce records. The approach depends on the song and the circumstances. Producers nowadays can simply be overseers in the studio of how a song is going to be finished, without taking any responsibility for the creation of the music and the arrangement. I, on the other hand, am involved in every single nuance of the musical performance, from every single guitar lick that's going to be played to every single vocal lick that's going to be put on. I'm sitting there and I'm actually either singing the part or I'm making sure that the part the musician created is the right part. That's just the way I do it. Other producers may not go that far, that deep, or do that much.

So you're really hands-on in every aspect.

Not just hands-on; I pretty much play every part, especially if it's a song done on a sequencer. If I have to go into the studio and do an orchestral overdub, I'm responsible for sitting with the arranger and creating the arrangement with them. And then, if there are guitar players or background vocals or anything else to do in the live domain, I'm pretty much there, creating those parts. I don't see how I can be any less hands-on than I am. Though, sometimes, it's down to the guys you have around you; if you have great programmers and great musicians and great everything, you can actually just hang out in the back and talk on the phone! [*Laughs.*]

But I imagine the total hands-on approach can limit the scope of artists that you might work with, simply because there are some artists that wouldn't be willing to give up that degree of control.

That's true, but other artists actually would flock to something like that. A lot of artists like the way I do things because they trust me. They just give it all up to me and they say, "You know what? When you do your music, I'll come in and I'll sing my song, and then I'm going to leave a happy person." You have to differentiate between the two types of artist. There's the Mariah Carey, and then there's the Celine Dion, and those two are very, very different artists. Mariah writes, co-produces, is there from every step; she's there, no question. Celine Dion — she doesn't write, she doesn't produce, she simply picks the songs that she likes to sing and trusts her producers to come up with the right track and she walks into the studio, does her vocals, and then leaves. Which doesn't make her any less of a talent, of course.

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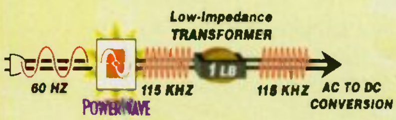

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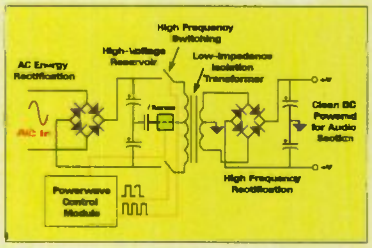

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Walter Afanasieff

Do you think there's a "Walter Afanasieff sound?"
Yeah! I know there's a sound, because I repetitively use certain tastes that I have. I have a sound because I like certain textures to go with other textures: I like big drums; I like really big background vocals; I like huge rock guitars — even if it's a really light R&B song, I'll put really big rock guitars on it. And I like big orchestras. I like to go really big where it needs to be big and really small where it needs to be small. And I think that, in a lot of the last ten years, specifically in pop ballads, I sort of came up with the tympani roll that goes into the big bridge, power chords coming in, cymbal swells — little things that make it my niche.
And, very often, it's the little things that make a record — it's all in the details.
I totally agree. When you listen to a Babyface record, you know it's a Babyface record. He has an electric Rhodes piano sound that he loves to use, he has an identifiable drum groove and drum sound, and he has that unmistakable wall of Babyface vocals that he puts down in every one of his songs. So he definitely has his sound. When you listen to a David Foster production, you have this signature big-time Chicago-sounding thing. And I think that, similarly, when you listen to my productions — when you listen to "My All" or you listen to "Hero," or you listen to "My Heart Will Go On" — you kind of know that, well, yeah, there's that tympani roll, there's that cymbal swell, there's that big power chord, and there's that big wall of backgrounds. I'm flattered that other guys out there are now trying to sound like me, whereas, for a few years, I was maybe trying to sound like someone else! So we sort of pass the baton back and forth. It's kind of good but it's kind of bad, and I know that I've been guilty of doing it. I've had records where I've tried to sound like Babyface and I've had records where I've tried to sound like David Foster, and so forth.


PHOTO: JIM MCHUGH AND THE NATIONAL ACADEMY OF RECORDING ARTS AND SCIENCES, INC. FROM THE 42ND ANNUAL GRAMMY AWARDS

Just as, when you started in the '80s, everybody was chasing that classic big stadium sound.
Absolutely. We all have our bible, of sorts. In my bible are my favorite recordings. My ultimate bible, and my ultimate guru of music, period, is George Martin. I can't tell you how much influence he had on me. Anybody who needs to get anything out of music, go listen to all The Beatles records. It's all there for me. For some people, their bible piece of music is the Beach Boys' *Pet Sounds*. To me, it's not; I don't go there when I want to refer or refresh or inspire my soul to create music. I go to other places — mostly, I go to classical music. I can put on a Chopin piano and I get inspired to play the piano parts on my next project. Or I listen to Rachmaninoff for some beautiful orchestral ideas that I want to put down. It's things like that, very personal things.
What are the most inspirational Beatles tracks for you?
When I moved to America, I was four or five years old and I didn't even speak English. I'll never forget this AM radio my

parents bought — it had a white dial and it was really old. I was thumbing through the stations, and The Beatles were playing "She Loves You." And that was it — that was exactly the moment in time that I knew that I was going to be in music for the rest of my life, all because of this song I was hearing. And when *Sgt. Pepper* came out — to me, that was the most significant recording of all time. For George Martin to create that piece of work on four tracks — that's history. In my opinion, nothing else has been done that's greater than that. I think nowadays, when we look at our studios and our Pro Tools and our 96 tracks over here and hundreds of tracks over there flying around — I mean, come on! So for this guy to do a *Sgt. Pepper* in four tracks?

Both George Martin and Geoff Emerick have said that if *Sgt. Pepper* had been recorded on 24-track, it wouldn't have turned out as well because the limitations of 4-track forced them to make decisions and not put them off. Do you find that artistic restrictions of choice can help the creative process?

I'm sure they're correct — we have too much choice today. We have so much freedom, so much room for error, that, at the end of the day, it's not about aesthetics, it's that we didn't do it right — we went too far, we did too much. There are times when I wish I had the same circumstances George had during *Sgt. Pepper*. I wish that the singers didn't come into the room and say, "I want to do twelve more tracks, I'm not happy with this." Okay, you can do twelve more tracks and I'll spend six hours comping the twelve more tracks you just did, but I know that we already have what we need. "No, just let me do twelve more." Then after that, they'll decide to do twelve *more*. I've had that with artists, and I've ended up with literally 70 tracks of vocals! That's absurd — to have a singer do 70 tracks of vocals because they're insecure in the way they sing and they're not letting you, the producer, take it and go with it in the way you believe it should be done. Then it becomes pandemonium, and, at the end of the day, I wish there were just a 4-track machine sitting in the corner and you just needed to do one track, and I would have punched in what I didn't like, and you would have gone home.

When an artist is doing 70 takes, is it because they're trying to achieve technical perfection? Surely it can't be to capture an emotion, because with so much repetition, the feel's got to start evaporating.

It can happen when the producer is seeking to get something that he just cannot get out of the singer. Personally, the only time I've had to do that is because a singer really just wasn't giving me what I felt was good enough. We're talking about a situation where the inability of the singer comes through and it's up to me, as producer, to say, "You're just not giving it to me yet." That, in combination with their own insecurity: "I can do it better, please let me do it better." But you're so free to make decisions now because you have the technology to support you — whatever you want to do, it's there; it's never a situation where you don't have any more tracks available. That doesn't exist anymore.

When you're doing an arrangement on a song, is it typically done with input from the singer, or do you do it by yourself beforehand and then just present it as a done deal?

There's two ways. If I'm co-writing with the singer, they know what

Walter Afanasieff

the arrangement is because that's how we wrote it. Then it's up to me to give them tracks, and it's up to them to say, "This is great," or "No, it's not good enough," or whatever. But, usually, I like to do everything all by myself and give it to the singer, and usually I believe in what I do enough to know that it's going to work at the end of the day — that they're going to love it. Sometimes there's an element of surprise that doesn't work in my favor — sometimes the artist will say, "I like it so much that I want it twice as long," or "Do you think we can change keys at the end?" Then it's a matter of going back to the computers and rearranging things. But, for the most part, I like to have everything pretty much done before the singer sets foot in the studio.

What sort of tricks have you come up with eliciting the best performance from a singer?

It's quite a bit to do with their space. They're very insecure, very gentle creatures, these singers. [*Laughs.*] You pretty much have to be able to be their doctor, their spiritual advisor, their psychologist, their bartender. You have to be all these personalities, and you have to stroke their egos just enough for their security, and you have to be able to solve problems that they're coming up with — even problems that may not truly exist. I've been in situations before where there really was no problem; where the singer was creating a problem out of their own pandemonium or insecurity. Then it's simply the way that you solve it — by saying, for instance, "Well, let's try a different pair of headphones," or "Let's try a different microphone; maybe this one will be better for you," even though there really isn't anything wrong. So you do this little thing where you just kind of keep stroking them; you've just got to keep taking care of them and pampering them.

"You have to be able to solve problems that [the singers are] coming up with — even problems that may not truly exist."

Not that all singers are these types of people; most singers are completely professional, dedicated hard-working people that walk into a studio and say, "Tell me what to do; I'm here for you, man." Other times, well, it just isn't coming out right. And when they hear that it's not coming out right, there's nothing bigger or deeper than that singer getting into a funk. [*Laughs.*] But when you're dealing with musicians, you're dealing with gentle souls. We're all from the same tribe; we're all trying to do something that's art. We're doing it to please someone else; we're all here for that one reason. Maybe your form of pleasing someone is to get a number one record, or maybe you just want to get that pat on the back from your fellow musicians who say, "Man, that's some great sh*t." This is what I live for. I don't live for my position on the chart or how many copies have sold, how much money I've made. My reason for doing it is to get that musician guy to say, "That sounds really good, man" — I love that. But we're all here to do that, so if it's not coming around, we feel like sh*t, we want to go run back home. So you've got to be able to anticipate it and know how to handle it.

What do you do when you and the artist are seeing something in completely opposite ways?

If I feel that what we have is there and we don't need to do another take, but the singer feels that they can do it much better, my answer is, "Well, please, go out and try it." In my mind, I'm thinking, let them do it and then I'll A/B it for them and pretty much the argument is won over what's right. And sometimes they go out there and nail it better than I would have liked!

But if the singer thinks they've given it their best and I still feel we need to do more, that's a different kind of problem. At the risk of sounding a bit graphic, if someone thinks they've shot their wad, you can't just go out there a minute later and say, "Can you shoot it again?" [*Laughs.*] So that becomes a problem — how can you ask this giant superstar person to do it all over again? That can be a bit difficult, but usually it's me knowing what they've done is so phenomenal [and them wanting to try another take].

Do you engineer most of your own productions?

Only when it comes to the mixes. It's not engineering in the sense that I don't know anything about the mathematics and science of limiters and compressors and gates and miking techniques and EQ and all that. I'm not that kind of person, but I do know that certain parts of mixes require a real hands-on approach just to ride those keyboards or bring up that orchestra in that perfect way or to make those drum fills a little bit more the way I hear them. Other than that, I completely trust the engineers that I work with. My main engineer is Humberto Gatica. Over his career, he's developed such a profound, interesting sound, and, for live recording, there's no one better than Humberto. He's just the purest, most knowledgeable gentleman in the studio, and he's full-on. There are no pretenses; he really knows what he's doing [see the story on Humberto Gatica in the January 2000 issue of *EQ*].

The second person I've been working very closely with, who is a masterful, masterful engineer, is Dave Reitzas [see

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the July 1999 issue of *EQ*]. Dave has gone inside of that computer, he's gone inside of that Sony 3348 digital multitrack machine, he's gone inside of that SSL console, and he knows every single thing about these pieces of equipment, to the point where there's not a second wasted of him trying to figure out how to make something work. And he's a wonderful musician. He knows music to the point where he anticipates what you're going to say — he's already there, unlike any other engineer I've worked with. He knows how to read your mind, because he's a super musician and a super technician and a super engineer.

And Mick Guzauski is pretty much the only guy that I would ever let mix a record for me with me not in the studio with him.

That's quite a compliment.

I'm so into mixers right now. To me, Chris Lord-Alge is an incredible mixer; Dave Way, who mixed the last Savage Garden album for me — he's a wonderful mixer. One of the more up-and-coming guys who works in my studio is Dave Gleeson, who's just wonderful. And then there are other people I adore who I just haven't gotten a chance to work with yet — for example, I really, really want to work with Al Schmitt.

One of the hardest things to accomplish in a mix — especially when there's a dense backing track — is getting the lead vocal to sit right. A number of engineers have given me technical solutions to that problem, but have you come up with any arrangement ideas that can accomplish the same thing?

There used to be a lot of truth in the statement that you could "fix it in the mix," but that's obsolete; you really can't say that anymore. It's got to pretty much be there at the time we're doing a vocal. If it's the right microphone, the right EQ, the right reverb, then you're safe. Then when the mix comes, it's just a matter of riding the vocal. To me, it's not a matter of changing it afterwards in the mix; if it's not done right during the recording, then I don't know what you can do.

Are there ever times when you'll change the arrangement afterwards because the vocal didn't seem to be sitting well in the framework that you originally created?

Not too many times. I think that the only problem we've ever had is if there was a bareness to the track when the vocal was actually done. Where it sounded pretty good at the time, but then, when you started filling up the whole thing with backgrounds and orchestras and guitars, we'd get into a situation where the vocal needed to sound bigger and stronger. Then you have to start making the engineer do a little more work. Sometimes it becomes a little dangerous there because you start having to take away certain musical things that you've done since the vocal was done. But pretty much the rule of thumb is, "It's got to sound like it's supposed to sound when you're actually cutting the vocals."

PHOTO BY ROSS PELTON

At what point do you lay the vocal down? What instrumentation will have been recorded beforehand and what goes on afterwards?

Sometimes it's actually the whole track; sometimes it's just the basics; sometimes it's just the basics with a good background section there to support the singer so the singer can go off and improvise, doing licks during the out choruses. I record background vocals before the lead vocal because I'm confident that the singer is going to love them, but it's kind of a hit-or-miss thing. Background vocal sessions are really, really expensive. I don't know why that is, but it is. I've yet to determine why a background singer can come in and sing for five minutes, and just because the song is a certain length and because you used a certain amount of tracks on it — damn, these people make a lot of money! [*Laughs.*]

But we pretty much commit — and it's a very expensive commitment — to putting the backgrounds down before the singer comes in to do their vocals. That's kind of a dangerous move, because if they don't like the backgrounds, or if they want to do the backgrounds themselves, or if the backgrounds aren't right, you've got to go in and do them again, and it's usually a big deal because it's a lot of money. But, if it works, it works great, because now the singer doesn't have to sing where he thinks he has to sing; now the singer can do a lick up here or a harmony over there. It becomes more fun and more creative.

Do you tend to use the same backing singers for every project?

I do. I have two groups of singers that I use — the San Francisco group and the Los Angeles group — though it usually is the San Francisco group that I use. I have created what I consider to be a masterpiece sound and technique with my background vocals. I have two guys and two girls, and we usually have the four of them sing a particular part in unison, and then I usually double, and then I go and I add a harmony that they sing in unison and then double. Sometimes I just have the guys sing, and sometimes just the girls. This technique and this sound — in my room with my microphones — is really pleasing to me. Listen to the end title song in the movie *Hurricane:* it sounds like a 100-piece gospel choir, but it's just the three or four people that I use. So I'm kind of married to that; I always want to do it, and I always depend on it.

"My reason for doing it is to get that musician guy to say, 'That sounds really good, man.' "

But I've [also] been realizing that there's so much in the artists themselves singing their own backgrounds. On the Savage Garden album, Darren [Hayes] sang every single part — it was like walls of background.

A lot has to do with how the backing singers blend with the lead singer's voice.

The blend, absolutely. Though not every lead vocalist can do their own backgrounds. There are some people out there who sound horrible singing their own backgrounds; I don't think Michael Bolton is known for singing any of his backgrounds, because he just doesn't sound good doing it. Luther Vandross never sings any kind of a background. They'll do harmonies with their own leads, but that's just a perspective that lead singers do. With Darren in Savage Garden, it just sounded so great. Doubling, tripling, backgrounds, harmonies, everything — it's just all Darren. Other singers need to have that really good background action, those background vocals.

So, on each take, you have the group of four backing singers sing a unison note?

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Pretty much, unless they physically can't go that low or that high. When you hear the total, though, there's a lot of harmonies going on. In the old school, you have your baritone, your tenor, your alto, your soprano voices in there — be it three or four voices — and they're all singing their parts. And you would never ask a baritone to sing a soprano part, and you would never ask a soprano to sing an alto or baritone part — it's just never done. But why not? I'll ask that baritone guy to sing in falsetto [in order] to sing the soprano part. So the guys are all singing falsetto, and the girls are all singing down low, and it's just a really good way to get different textures and tones in your background vocals. Plus, the singers really have fun doing it!

So the way you build up your chords is that each track has a different note.

I've done songs where all four singers do one unison part, then we double that, we triple it, and we quadruple it. So one line, one part, all unison, now becomes sixteen voices singing the one note. Same thing with the next harmony up: All the same four voices sing the same harmony note, four tracks of that. So that's another sixteen voices doing that harmony. And then we do another harmony and then we do a lower harmony, and then just the guys — because the girls can't sing that low — give me some really low notes.

Part of the reason for this, simply put, is that my level of perfection is unlike anyone else's. I just don't like the human error; I don't like the human part of it as much as I like the real, nailed, in-tune, perfectly executed part. I would love sometimes to put *everything* through a computer and Autotune and clean it up and quantize it and fly it around left and right, because I really like that very clean, very precise style of recording. So to have three or four singers out there with each one singing different parts, they'd have to be the greatest singers in the world to me because, sometimes, somebody's going to be a little flat while the other three are perfect — and then you've got to do it again. And I just don't have the patience for that. [*Laughs.*]

Based on your records, I would have guessed that you were a perfectionist in the studio. There's a real polish, and there's no question that what you're hearing is a production, as opposed to a capturing of a live performance.

Well, when you start doing things on your own, you get into thinking of ways of doing things perfectly. You're in that school, and that's where I found my training. My school days were when I was in the studio with Narada Michael Walden, sitting with my computer or my [Akai] MPC60 or my Fairlight or Synclavier. We were already being taught to be perfect, because that computer was making it perfect; that keyboard is already in tune perfectly, and we were all doing things pretty much with perfection in mind. Then, when you finally get out there in the main professional workplace, you don't really like to hear something that's out of tune; you don't really like to hear something that's out of time, because you've been taught not to do it that way.

On the other hand, a lot of people have been brought up in the live band domain. Recently, I've been working with this group Train. There are five guys in the group, and all five of them are live musicians, and there's forgiveness, there's compromise, there's all these human allowances. They're not all playing in tune and they're not all playing in time — they're not all doing it completely like my computers have always been doing it. So now I need to go, "Wait a minute — are they doing it wrong, or am I hearing it wrong?" [*Laughs.*] Do I make them do it again and make them do it so good and so clean and so polished that it takes away from their live thing, or do I have to do it so I allow for their own human mistakes to sort of be what their sound is? If you listen to Crosby, Stills, Nash and Young, they're not singing completely in tune, and that's the reason they sound the way they do. But sometimes you do hear live recordings and it's like, man, they're singing perfectly and playing perfectly. Is it because Mutt Lange sat there for months and months and months making AC/DC or Def Leppard do it again and again and again until it's completely right and in tune and in time?

I don't know. My training, my school, has come from listening to perfect recordings and then trying to create them on computers. Technology sort of prevails in my life. I appreciate the philosophy that, if you're doing your job right, then you're going to do it perfectly. If you've learned to sing correctly, you're going to sing in tune; if you've learned to be a masterful musician, you're going to play in time.

If the performance is there — if it's evoking the correct emotional response, making you smile or cry — but it has technical imperfections, will you have the artist do it again?

No, no, no, no. On Savage Garden's record, there's a song called "Two Beds And A Coffee Machine." The song is a profound ballad — it's a really moving piece of music, and it's a very emotional subject. It's very poignant and very dramatic — it's about spousal abuse. And none of us were actually ready for Darren to sing the song — we were just doing the music recording; I was in the middle of doing piano. He said, "Let me just put down this guide vocal so everyone can build on top of it," and I said, okay. So we just ran out there into the room and put up any old microphone, and there were no baffles and it was really not done in the way you would normally do a vocal in any sense of the word. In fact, I can't even remember if there was an engineer in the room! Anyway, he went out there and he sang the song and everyone in the studio was just moved to tears — we were just sobbing. At the end of the day, that was it — that was the vocal that was on the record because he could never ever do that again. It was the first time out of his body, the first time into our ears, the first emotion, the virgin part. If a plane crashed in the room, I would have kept it! [*Laughs.*]

So you strive for technical perfection, but if you get that magical performance, you go with it anyway, even if it's not perfect.

I think what I'm saying is that everyone should really learn their craft so that when you do need that magic performance, it would be done masterfully.

But there's a very fine line between chasing perfection and polishing the life out of a track.

Yeah, there's no question that you can make an orchestra play so many

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Beyond Utopia

story & photos by Mr. Bonzai

Legendary performer/producer Todd Rundgren muses on recording technology and the impact of the Internet on the industry

Over the past three decades, Todd Rundgren has carved out a unique niche in modern popology as a gifted artist, inventive engineer, provocative producer, and pioneer in rock video, interactive music, and the Internet. Today, he spends much of his time at his command center in Hawaii working on his own projects and, recently, producing such acts as Bad Religion.

Born in Philadelphia, Rundgren formed the progressive band the Nazz in 1967, scoring regionally with an early version of the later hit "Hello It's Me." The band broke up in '69 and Rundgren put together the studio band Runt, which hit the top 20 with 1971's "We Gotta Get You A Woman." He then began an association with manager Albert Grossman and the new Bearsville Records, which led to producing/engineering Badfinger, James Cotton, Paul Butterfield, the New York Dolls, and Grand Funk Railroad, among many others.

In 1972, Rundgren produced and engineered his solo album, *Something/Anything*, on which he played most of the instruments, stacked up his vocals, and mixed a modern masterpiece which yielded the hit singles "I Saw The Light" and "Hello It's Me." His following grew. In 1974, he launched his band Utopia and the fan base expanded even further. Along the way, he produced Meat Loaf's *Bat Out Of Hell* (1977), one of the best-selling albums of all time.

In the '80s, Rundgren continued with his own recordings, as well as producing such groups as Cheap Trick, the Tubes, XTC, and The Psychedelic Furs. He explored new media with video and computers, creating a backdrop for his one-man shows, which were augmented by pre-recorded audio. In 1993, he released *No World Order*, reputed to be the world's first interactive music-only CD, and in '94 played ten shows in a special pod pavilion at Woodstock II.

We met at his home in Hawaii and discussed his past, the present, and his plans for this freakin' new millennium.

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Mr. Bonzai: Could we call this your "studio"?

Todd Rundgren: Well, I don't really have a studio, *per se*. Ideally, I have to find a space that I don't have to share with the rest of the family. This setup is mostly for mixing. I am doing a little bit of recording here for myself, but the CDs I've done recently have been tracked and overdubbed in other locations in Hawaii. I usually just rent a house, and in one case we rented a project studio that someone had built in their basement. For the Bad Religion album I produced last summer, we rented a converted barn nearby that is mainly used by photographers — high ceilings and cement floors.

What records have you made during the past year?

I produced Bad Religion and the second album from a band called 12 Rods on B-12 Records, and, at the same time, I have been working on a long-term dance-techno instrumental project with composer Michael Gallagher. It's a mixer's record.

In your entire career, which are the albums you are most proud of?

Well, I consider pride a sin, so I don't catalog the albums like that. There are albums for which I set certain goals, and got very close to the goals — in some cases, certain aspects of the projects exceeded the goals. Albums by XTC come to mind, and any of the albums where my influence had more than a custodial effect on making the record.

In the case of some records, the artist has a very strong idea of what is to be done, and that's fine. In other cases, the artist has very little idea of what's to be done, and that's fine, too. But those aren't the kinds of records that I feel are the best records for me, from the standpoint of either production or what I would do as an artist myself. This is because there are compromises in there — I have to contribute more than I normally might as an artist, and at the same time, I don't have the last word in terms of what's there in an artistic sense.

It's hard to avoid mentioning an album like *Bat Out of Hell*, not for the fact that it was so hysterically successful, but because I did it for completely other reasons and it seemed to have achieved those goals in the process. I did it because it didn't seem like any other producer would do it with any enthusiasm. I approached the whole album as being a spoof on Bruce Springsteen. That's what it seemed like to me, and that was the principle influence. Bruce Springsteen was on the cover of *Time* magazine and had all this other visibility, and it was all being taken so seriously. No reflection on the quality of his music, but I thought it was a great opportunity to do something in the pop cultural realm, rather than simply making a record. I had no expectations that it would be so commercially successful. The songs were so freakin' long, but that was what sold it in the end — those shaggy dog jokes. Long songs with goofy punch lines at the end.

Are you still making big bucks off that album?

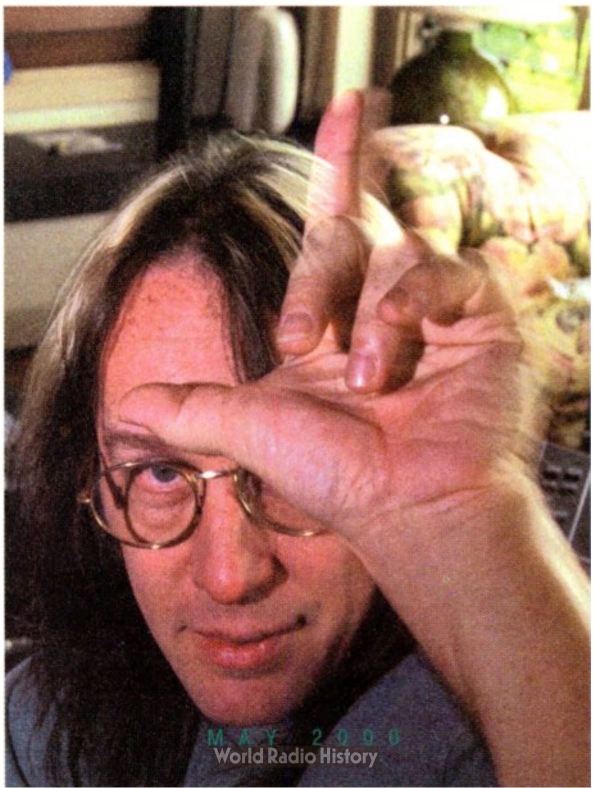
No, as a matter of fact I am not making any bucks off that album. There was a legal confrontation with Sony — who had inherited the CBS catalog — over discrepancies in how much royalties had been paid. Essentially, I sold my participation in the record back to the label because I didn't want to spend years in court and having to pay for it. I essentially cashed out and used it to buy my stake in Hawaii.

Do you do most of the engineering as well?

Yes, it keeps me awake. [*Laughs*.]

How did you learn about engineering?

I learned about engineering around the same time as I learned about producing. When I was in the Nazz, I payed close attention to what was going on, but was very satisfied with the kind of engineering that we got, so I wasn't aching to get my hands on the board as long as the engineer did the right thing. But after we did our first real album, we had a producer come in and just kind of whip through the mixes — which is not what we were looking for. I wound up going through the mixing process, which means you are on the other side of the board. It was the first time that I focused in on what was involved, and started putting my hands in there. By the time we got to the second album, I pretty much wound up mixing the whole thing, as well as producing it in conjunction with the rest of the band.



Chris Stone, founder of the Record Plant, once told me that you used to come in every day and blow up the speakers.

Well, I was one of the first clients in there, but with Hendrix working there as well, the speakers were probably gone on a nightly basis. Prior to that, most of the studios didn't have anything approaching what most bands were producing on stage. When you saw those

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about the artist's state of mind and getting a performance out of them, I don't want to deal with equipment problems. The new technology makes the process go a lot faster, and, personally, I have always campaigned against that inertia that makes albums take longer and longer to produce. In the end, it is questionable whether the time was well spent.

Either it doesn't seem to be economically in balance, or, in many cases, people move away from their ability to actually perform a song. I think there is a narcissism that artists can develop in the studio from listening to themselves too intimately, and not realizing why other people listen to music, what other people appreciate in music.

stacks of amps, you had to have them and then suddenly you go into the studio and everything sounds puny because they were used to doing R&B records on little Altec 12-inch speakers with little tweeters.

In most cases, the technology was skewed toward no distortion — keeping everything well below the level of distortion. In performance, it was all about loudness and distortion. Then there was an evolution in the studios, and I got in there fairly early — 1968 was when they opened the doors at Record Plant — for my very first production where I picked up on engineering. They had a new custom-built console in the front room at the time, and the engineer had never used it before. I decided to do it myself, and my first production was also my first album where I engineered from beginning to end. It was called *American Dream*, after the Nazz broke up and I went to work with Albert Grossman. This was one of the first releases on Bearsville Records — it was Ampex Records at the time.

Leaping across 30 years of engineering, do you find that the technology serves you better or worse today?

I am very happy with the technology today; the flexibility and the ability to approach the sound that you want. Everything is better than ever — particularly with the digital world. I did a project in 1998, and after that I swore I would never use tape again. I haven't used tape in any substantial way since. If I have a problem with equipment and it's my project, I can live with it. But if I am in there and it's all

> "Everything is better than ever – particularly with the digital world. I did a project in 1998, and after that I swore I would never use tape again."

I have a very strict formula, an upside-down pyramid in terms of what's important. The most important thing is the song. If you have a great song, there is a whole lot of slack you get everywhere else. Then it has to be performed as if you are interested, with some sympathy to the song. The performance doesn't have to be perfect — it just has to be sympathetic to the song. The very last thing that people care about is the sound. Sound is the most subjective thing there is about a recording, and, if you compare two hit records, you find that they may not sound anything alike. How can they co-exist as hits? It's because the sound is completely subjective.

The worst thing in the world is getting into the studio with someone who is just anal about the sound. The best thing about this digital methodology is that you never get locked into anything. You start with the raw materials and everything beyond that is completely re-configurable. As long as you get the performance, you can defer that sound stuff — it's completely flexible. This technology makes the entire environment so easy to package and transport. "You don't like the way I mixed it? Take it and put it on a system similar to this one and just mix it to death until you are satisfied."

What are you mixing on here today?

[Digidesign] Pro Tools' latest setup — not the biggest, but the latest. Pro Tools|24 MIX, 64 channels simultaneous recording. Virtually unlimited virtual channels, running on a 350 MHz G3. DSPs are on cards. Pro Controller unit, which is essentially eight virtual faders and a bunch of stuff to control the software. Multisync flat screen monitor. Five Miller & Kreisel speakers for 5.1. These M&Ks are THX-certified, the first portable system certified, and it makes it very easy for us to go to other environments. Take two of these speakers and the subwoofer, and there you are.

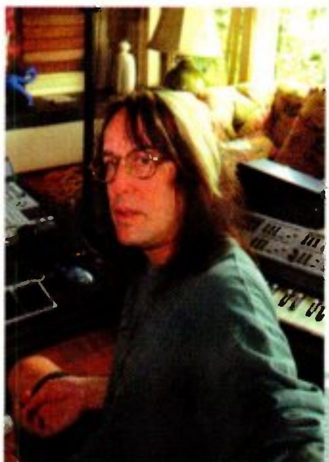
What about this keyboard here?

That's just my old JD-800 that I use as a keyboard controller. I don't often use the sounds out of it much anymore. I'm using a software synthesizer called [Bitheadz] Unity DS-1 that interfaces with my sequencer, which is [Steinberg] Cubase, and essentially anything I can sample, I can put into the software synthesizer and use it just like a hardware sampler.

One of the things we used a lot on the Bad Religion album is this Line 6 TDM amp plug-in [Amp Farm], which is an acoustic modeling of an amp. We plug the guitars in totally dry, right into the preamp — I've got 16 preamps and 16 digital I/Os — into the Pro Tools system. We ran the guitars direct and the only things that we miked were the drums and the vocals. Everything else goes directly in and we put the guitars into these TDM plug-ins so we can dial in any one of a dozen vintage amps, and thirty amp/cabinet combinations. When we are going for a guitar sound, we aren't tediously setting up amps, and miking amps — we are dialing them in. Would you like the Soldano head? How about an old Vox AC-30? 4x12s? Maybe an old Bassman cabinet?

I guess the old gear will certainly have a place in the "Old Engineers Home."

Well, a lot of people have been driven crazy doing it the old way — having someone come in with six amps and a pile of cabinets, looking for their ideal lead sound, and the ideal rhythm sound. And days go by. I just won't stand for that anymore. To me, it's all about the performance. The final "sound" is right down there at the bottom of the pyramid. I've done enough albums that I can at least get everything onto the disk, or onto the tape. From then on, you deal with all the tortuous twists and turns of those subjective things called "sounds." Everyone hears it differently, and if you can build a consensus and get everyone happy, it's like getting the members of Mormon Tabernacle



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Choir to all agree on the same thing.

It's also hard to do anything new. There has been a lot of taxonomy in the world of sounds. We have catalogs of sounds that artists can use as reference for how they want a particular song to come out. We may not be so far from a time when software has a robotic ability to take over where the producer leaves off. Just talk to your computer and say, "Could you make it a little fatter? Make it sound funkier." They'll actually have a program that will interpret that and change the sound. Eventually you will just describe it and the computer will take over through voice recognition. You won't even need a producer at all — save yourself a bunch of money.

How has the Internet affected your career?

It doesn't affect me in the way that it affects a more contemporary artist who is cashing in on the traditional market right now. There's all these scare tactics about how the Internet is going to siphon away all your profits through bootlegging and things like that. The reality is that the relationship between an artist and a record label is an entrepreneurial investment on behalf of the label toward the artist.

The artist is fronted their own money, which is spent to make and promote records, and, over the course of a deal, the likelihood of going into the black — for most artists — is low. Most artists will not come out of their record deals in the black. They may not even get past their second album before they have to change labels, and that's usually a bad sign. Not many people make it from one record deal to the next one without having shown some success in there.

It's a very dicey, chancy, cutthroat business. No matter how good the quality of the product, the traditional marketplace has a very narrow pipeline. If someone of great stature releases an inferior record, they get all the attention anyway. It doesn't matter if your record is better than Michael Jackson's. If his comes out the same week as yours, odds are that you won't get on the radio if the records are in any way similar.

In any case, the idea that the Internet is going to significantly impact your record sales — that's not what's going to bring you down. What's going to bring you down is the fact that the old model is so undependable for most artists. One of the reasons for this is that it is inventory-based and transactional. You press up the records before you know how many will sell [*Laughs*] and then you send them out there. Next comes the "demand and inventory" juggling act, which can go awry very easily. You want to create demand, but you don't want to press up too many records and wind up holding a bunch of unsold inventory. That's what kills most artists' profits in the end. It's not CDs being bootlegged; it's unsold inventory.

> "You can live as an artist and not have recordings as a significant factor in your life – it has to be possible, because so few artists make a living off their records."

Is it true that anyone can subscribe to Todd Rundgren on the Internet and get regular shipments of new material?

Well, that was my original offer, but I discovered that you could only do that if you had nothing else to do. You can go to tr-i.com and check it out. A scalper took ToddRundgren.com, and I refused to pay him for the domain.

The idea is that there are two factors in the life of an artist who makes recordings. You can live as an artist and not have recordings as a significant factor in your life — it has to be possible, because so few artists make a living off their records. You need underwriting to make a record, and then you have to market the record. Traditionally, the record company takes care of both of those responsibilities. The idea I had was that if you have a core audience that you know will always buy your records anyway, you elevate them to the status of record label.

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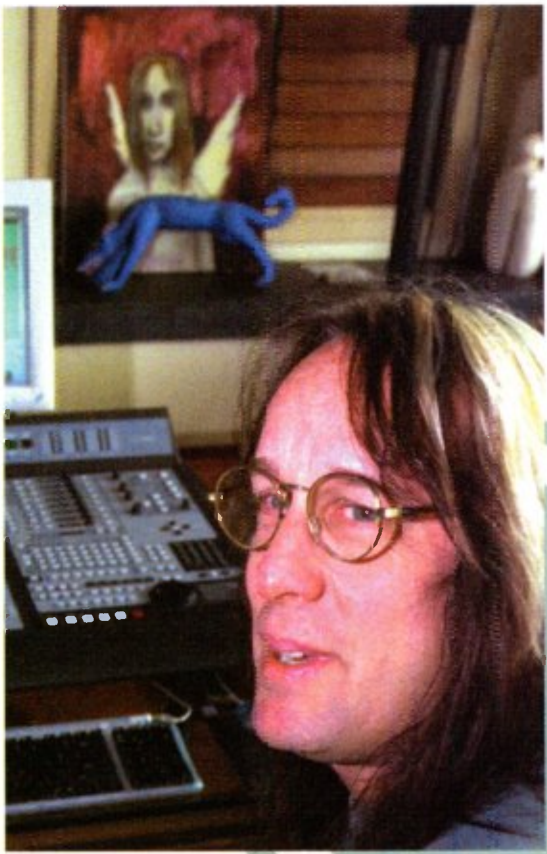

You are saying, "If you will pay me in advance, and underwrite the record I am about to make, then I will keep you up to date on it as I make it." Then I don't have to go to a record label for an advance. And when I am done with it, I can just make a straight distribution deal for those people who aren't members of the devoted core audience, and they pay regular retail. And the record will be finished when they get it. There is the core audience, the completionists, and the devotees who want to know everything that you are doing while you are doing it, and want to feel involved with you rather than the sales clerk at the record store.

I developed a concept called Patronet, which is a mechanism that allows artists to get connected to that core audience and have that audience subscribe to them. And then you give the artists the tools to deliver. In the past year and a half, I have been the only artist because it is an experiment. Everything I have made has gone to pay for the Patronet. We now have new partners and it will be an aggressively marketed service, and more artists are going to take advantage of it.

Who chooses the artists?

It won't be a question of choosing; it's more of a public utility, like the phone company. Patronet is a place where you can go and get into the business of soliciting subscriptions from devoted listeners or followers. It doesn't even have to be music. It could be a magazine format, or something like that. You don't have to go out and get your own hosting service, your commerce solutions, and hire your customer support if you need fulfillment of hard goods. All this can be provided. If you need a design partner to help put the space together, that could be provided. Essentially, it's a single source to get all the things you need to get into the business of offering a subscription service over the Internet.

The goal is that artistic survival is enhanced?

The idea is that there is a certain type of artist who can benefit from this immediately, while the concept is in its primeval stages. For instance, it is for artists like myself, who have a devoted core following, who are interested in knowing what I am up to now that I'm not in the mainstream anymore — as so many are after a certain number of years. It's for artists who are interested in having more of a relationship with their audience, rather than the old transactional model, so they can send me e-mail and get e-mail from me when certain things happen. Possibly they could be in a live chat with me. There are other benefits associated with a more privileged relationship such as this.

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CIRCLE 37 ON FREE INFO CARD

By Howard Massey

The Voice of Experience

Famed engineer Mick Glossop talks about Peter Gabriel's accidental drum sound and dealing with the transition from analog to digital

For more than 30 years, Mick Glossop has been lending his considerable talents to top-flight recordings, including a long-term collaboration with Van Morrison and work with Queen, Mott The Hoople, Frank Zappa, Renaissance, Tangerine Dream, Waterboys, and John Lee Hooker. Glossop is unique in that he's served as staff engineer at some of London's finest studios, including Wessex, Townhouse, The Manor, and Nova. This long apprenticeship gave him the opportunity to study and absorb numerous different production techniques — invaluable experience that has undoubtedly aided him in his own illustrious career.

On the rainy London afternoon we sat down to talk (what other kind is there?), we were plagued with a power outage, resulting in the interview being conducted by candlelight while frantic staffers in his office ran around trying to conduct business by battery power. None of this seemed to have any perceptible impact on Glossop's powers of concentration, however; for more than two hours, he remained focused on the task at hand, giving thoughtful, insightful answers that reflected his intensive training and singular dedication to the craft of making records.

EQ: What's the most important piece of gear in the project studio?

Mick Glossop: It depends on what you want to achieve with your studio. I suppose you could divide equipment up into the stuff that you are creative with and the medium upon which you record — the tape machine or hard disk recorder. I think it's worth getting quality for the medium, because that directly affects the reproduction quality of what it is you are creating. You can buy a computer-based system with a PCI card, but use the best converters you can, so at least you can do transfers with the best quality; it's quite important and you can't really bypass that. I suppose the next thing on from that would be to make sure that your listening environment — by that I include the speakers and the amplifier — is as good as it can be. That doesn't mean you have to have a big massive system with 18-inch bass drivers. You can get very good results using a pair of Yamaha NS10's. A tremendous number of producers and engineers work for maybe three-quarters of their major projects on NS10's.

Is the quality of the mic preamps more important than the quality of the mics?

There are companies like Mackie that are making desks that have mic preamps

PHOTOS BY SARAHPHOTOGIRL@HOTMAIL.COM

that are great. This is very subjective, of course, but, as far as I am concerned, they are better in some respects than SSL E-series mic amps, and we made a lot of records in the '80s on SSL E-series consoles with E-series mic amps. So in terms of mic amps, if the budget's restrictive, then get a Mackie. Not just Mackie — there are other cheap consoles with reasonable mic amps, like the Soundcraft ones. And rather than just one mic, it's good to have a choice of mics — at least one dynamic and at least one condenser mic for different things, because they sound different.

Do you feel there are too many options available to today's engineer, as compared with when you started out in the '70s?

I'm in favor of restriction of choice from a creative point of view. One case in point is the third Peter Gabriel album, the one that had no cymbals and the big Townhouse drum sound. The room wasn't designed in any sense whatsoever — it was more or less accidental. It's live, with a stone slabbed floor, which was done at least 50 percent for aesthetic reasons. Given the context of the day, it was very, *very* live. If you hit a snare drum in the room, it was so loud, it was incredible.

We were contemplating changing the acoustics in there because we felt it was a bit uncontrollable, but it went on to become the vehicle for a classic drum sound!

How did that happen?

Peter had decided before he even went in the studio that he wanted to restrict the choice of percussion sounds inasmuch as he didn't want any cymbals on the album — he didn't want any crashes, rides, or hihats. So the rhythmic components that were traditionally played on hihats and rides had to be played on floor toms and that kind of thing.

On the Townhouse SSL, there was a talkback system that used a mic rigged up in the ceiling, connected to a really vicious compressor. So vicious that, if somebody hit a drum, the volume of that drum was the same as the level of their voice, which is exactly what you want [in a talkback system]. It was probably Phil Collins who was playing the drums at the time; they were talking about something and he started playing, and in the control room they suddenly heard this amazingly compressed drum sound. Peter Gabriel heard it and said, "Wow, that's fantastic! Put it on the record!" and I don't think that they were working towards that. So [engineer] Hugh [Padgham] then set about using the SSL channel compressors to reproduce the sound of the talkback compressor. In the course of that, he was playing around with the gate, and, again, Peter responded because Phil hit the snare and the gate chopped off the reverb, which is the classic sound that has become legendary. It was really Peter who insisted that the sound be used, though it came about by accident.

But the point is, if they had been using cymbals with that kit, they wouldn't have been able to use that sound, because the gating would have chopped off all the cymbal decays in a very haphazard way, which would have messed up the sound. The sound is basically the sound of a drum being hit and the ambience compressed and then, as the ambience tails off, it chops off sharply because the gate's got a short release. It just doesn't work if you play cymbals as part of the rhythm. So it was only possible to get that drum sound — which then went on to be the drum sound of Phil Collins's "In the Air Tonight" — without using cymbals. That's an example of a massive sonic creative effect being made possible only by a restriction of choice.

And a series of accidents.

And a series of accidents, yeah, especially if you think about the room design not being planned. So good things can come about if you say, "Right, we're not going to use that piece of equipment" — whatever that equipment is, a musical instrument or whatever.

> "Sound is not a separate thing from the music; the two are totally interlinked."

So you're suggesting giving yourself some restrictions beforehand in order to find a way to accomplish what you want with limited options.

Absolutely. Instead of yearning after all the gear, choose wisely and think what the essential things are — just make sure that they are the right individual bits of equipment. Given the kind of recording technology we have now, with hard disk recording, etc., you can create fantastic results with very little musical equipment, with one keyboard that has a selection of tones, one guitar; you don't even need a bass guitar, really. If you're working on your own in a project studio, you can create bass parts with either the keyboard or the guitar, using octave dividers. You can speed the tape up and play a bass part on the bottom strings of the guitar, use that sort of thinking. If you get into that kind of boy scout mentality of making it work with a piece of string and a pen knife — seeing what you can do with not much equipment — that will bring an originality to your work, and it's fun as well! You just have to work a little harder mentally, rather than just calling up preset 25.

The thing about sound is that it's not a separate thing from the music; the two are totally inter-linked. When you're creating effects on records — not in an effects box sense, but in a musical, sonic sense — the two are the same. So choose two or three bits of equipment with care and then set about doing what you've got to do with those.

What are the common mistakes you are hearing in tapes that are coming out of project studios?

It's difficult to answer that, really, because they're not really mistakes. It's just to do with skill, and, to a large extent, it comes out of experience, for which there's no substitute. Things like not being able to use a compressor in the right way; not knowing whether something is too compressed or not compressed enough; over-equalization.

I've listened to a few records recently — finished CDs that have been recorded digitally — and the reason I haven't liked the sound of these records is because there's something about the transients that are excessive, particularly in cymbals and drums. There's an aspect of digital recording that results in transients being too present, for me, anyway. The hit of the stick on the cymbal is too clear, too apparent. It's something that needs controlling.

It's very difficult to describe, but one of the benefits of analog is that those transients are controlled and they enable a blend of the instruments that is very appropriate and very pleasing to me. A lot of digitally recorded releases are made on low budgets, and that's usually the reason for it: lack of expertise and cheaper digital formats where those transients aren't being controlled. As a result, the instruments aren't blending properly and a cymbal crash, for example, is sticking out too much and dominating the listening experience. It's inappropriate because it distracts your attention from the main event, which is the vocalist. That sense of how to deal with what's happening in a digital recording is something that you need to have an awareness of. So I think a pitfall that's being fallen into is the allure of those transients.

When you put a microphone up and you start getting a drum sound, the fact that digital recording preserves and in some ways actually accentuates those transients is something that initially can be appealing. You think, "Oh wow, that sounds like quality." It's like sometimes when you add high frequencies to a sound, it immediately makes it shine more and you think, "Oh, that's better." But it isn't necessarily better. It's just that you're being kind of drawn into that; it's seductive, I suppose.

What techniques can you use to tame those transients?

One of the things that I've really only learned in the past three or four years is that, if you're going to record on a digital medium, you can't expect to use the same mics you've used for analog. There's an upper midrange quality that can be accentuated by digital, [a quality] that was always smoothed off by analog. So you could use a mic that had a spike — an upper-mid boost to it — on an electric guitar or a cymbal, something that maybe had a tendency to be a bit harsh. Because the signal was going to analog tape, it would be dealt with. When you record on digital, it's going to be enhanced, so you need to choose a mic that has got less of that upper mid. Choose an Audio-Technica or a B&K or an Earthworks — there are several mics that have been developed in the past five or ten years that have a smooth top end and nice, smooth low-frequency response, but not much midrange enhancement. Those kinds of mics are a bit more appropriate for recording spiky sounds or transient sounds. So that's one thing: Use a different mic.

Do you tend to record everything on analog tape or just drum tracks?

It depends on the project. My philosophy is that, if possible, everything should go through one analog generation somewhere along the line, even if it's the half-inch mix. It's better, though, if it's before that because it helps to blend the instruments. I find it more difficult to get instruments to blend together when they've all been recorded in a digital format. If you're leaving the analog process to the end, then you're not going to hear it until you play back the analog tape, so it's a bit limiting. The analog process is perhaps not essential for vocals, but I think it helps drums. But that doesn't mean you have to record on analog first; I've done a lot of projects where I've used Pro Tools as a tape machine and then copied it to analog at some later date. A lot of people work like that with ADAT, for example. Again, you're not hearing the effect until after you've done the recording, so it involves a certain amount of experience to know to record onto the ADAT and to listen into the future. You've got to think, well, it sounds like this at the moment but I know that when it comes back off the 2-inch [analog tape], it's gonna be smoothed off a bit.

Analog tape does have a beneficial effect on a lot of instruments straight off the bat. There are different ways you can achieve that. If you're working with hard disk and you've got an analog 1/4-inch machine sitting in the corner, you can come out of the hard disk through the 1/4-inch machine back to hard disk and then slide the track back into place to compensate for the delay. It's easy to do that — you know what a tape time delay is, so you just match that. So you can use the analog tape machine as a processor, working that way.

When you're mixing, in what order do you bring faders up? What are you listening for and what are you thinking about?

I start off with an initial mental picture of how the song is supposed to sound — a general kind of personality for the mix. If it's something I've recorded, then I know what's on the tape and I would have been recording with the mix in mind anyway, so I'm halfway there. Then it's a matter of just trimming because I've done so many monitor mixes in-between time, and I know how things can blend. The panning positions are probably already worked out, and most of the compression will have been done. If it's a track that I'm not that familiar with, then I'd listen to everything up and get some kind of a rough balance.

If it's a rhythm section-oriented thing, then I'd probably work from the drums up, but I wouldn't listen in solo to the drums. I generally mix on an SSL, and I generally tend to assign each group of instruments to its own VCA master fader. So I'll have the drums on one, the bass on two, the guitars on three, acoustic guitars on four, keyboards on five, etc. I'll pull all the faders down to about –15 [dB], except for the drums, and I'll leave those up so I can work on the drum sound while listening to the other instruments at a reduced volume. That way, you're still listening in context.

Are vocals the last thing that you work on?

Well, you kind of move them around. All the midrange instruments have to be related in terms of their frequency band and the musical harmonic aspect. The vocals have to relate to the guitars and keyboards; generally, the vocals are in that midrange, so there's no point in getting a bunch of guitar sounds that leave no space for the vocals. So I'll work on the guitars for a bit while listening to the vocal. Whilst you are doing that, you are kind of with another ear listening to the sort of problems there are in the vocal sound that need to be dealt with. For example, if it's a song that has a quiet verse and a loud chorus, then it's very common for a singer's voice to develop a hard edge when he's singing loudly, which doesn't happen when he's singing quietly. So while you are doing the drums and guitar, you might be thinking, well, perhaps I'm going to have two channels for the vocals — one for the quiet parts and one for the loud parts.

An alternative to that may be using a frequency-band compressor where you control that particular hardness factor in the vocals. The problem with using static EQ to get rid of the hardness is that it might solve the problem of the vocal in the chorus, but, when the quiet verse is being performed, it will lack some presence because you've taken out some 4 or 5 k. So you could use a frequency-band compressor to control that band. I use a BSS DPR901, which is 4-band; it's a really, really, good device for controlling that kind of thing. It has a sidechain [listen function], so you can tune one band to that problem area. I use it a lot, particularly in vocals.

Another thing I do is that I have an AudioDesign Scamp rack, and I have a couple of modules that are frequency splitters designed for band compression for radio use. You put a signal in and it's got four outputs for low, low/mid, high/mid, and high, so you can split the signal up into four frequency components. But then it has four inputs that combine those four components back to the original, so if you take the four feeds out and put them straight back in again, there's no difference in sonic quality because it's designed to have no phase shift. If the gains of those bands were changing and there was a phase shift between the bands, you'd get a swooshing effect. So it means you can take just one band of the signal, compress it, and do whatever you want with it — pan it, whatever — but if you process it in a mono way, you can feed it back in with the other bands. You can compress just one band and it works really, really well — it's fantastic. I use it in conjunction with four dbx 160 compressors — one on each band. The nice thing about those compressors is that each one has an output level control, so you can do the compressing and then balance up the four bands as they are mixed back together.

So in effect you're equalizing as well.

It's dynamic equalization, but you can have a different compressor on each band. You don't have to recombine them through its combining network; you can bring those four signals up on the desk and you can use different effects on the different bands of the vocals. You can put them through four auto-panners and have them moving around in different frequency bands in the vocals, which is quite a nice effect. You can add a digital delay just to the upper mid, you can pan the high frequency just on one side and have the lower mid on the other side, things like that.

And presumably you've done all these things.

[*Laughs.*] Oh, sure. I simply realized when I started messing around with these boxes that you can split the signal up into these bands with a hi-fi respect for the quality. It's quite transparent; it doesn't compromise the sound in any way. It just splits and recombines if you want to. But once you've got those four components, you can do all kinds of things; you can use a phaser on just the high frequency.

Do you have any general advice for the reader who wants to be the next Mick Glossop?

Just don't be afraid to experiment; that's important. Most of what we do comes down to experience, and there's no substitute for that. So just spend a lot of time on it and mess around and experiment. Do crazy things, break the rules, and, if you like the sound of it, then that's great.

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

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In the Internet Age, The Future of the Studio Business Begins Here...

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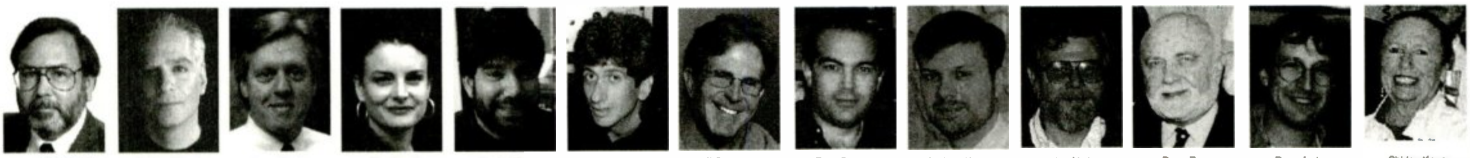

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The future of the studio business in the Internet Age Featuring the Nashville Surround Conference & Showcase sponsored by SPARS and Pro Sound News

eStudio Summit

Conference Program*

Friday May 19, 2000

9:00 a.m. - 9:30 a.m.
Keynote: *Digital Rethink: How to Succeed in the E-Recording Business*


Graham Strong

How should you rethink your business strategy for the dynamic world of e-business? And how can you best equip your facility for this changing era of e-business? Graham Strong, director of technology at PSW will guide listeners through his vision, illuminating his secrets for rethinking your studio operation for the digital (e-business age). Strong was the principal architect behind the overwhelming success of Priceline.com and other leading Internet start-ups.

Speaker: Graham Strong, Director of Technology, PSW

9:30 a.m. - 10:15 a.m.
Producer Power


George Massenburg

Some of the world's greatest producers speak out on where the industry is headed and what studios need to do to get their business.

Moderator: George Massenburg, GML
Panelists: Scott Hendricks, David Malloy, John Hampton, Dann Huff


David Malloy

10:15 a.m. -10:30 a.m.
Coffee Break

10:30 a.m. - 11:15 a.m.
Internet Opportunities


Dann Huff

The Internet revolution now permeates all aspects of the studio. In this session, company founder Willy Henshall examines how the Rocket Network envisions the interconnected world of the music creation and distribution space and what this means to the music business infrastructure.

Speaker: Willy Henshall, Rocket Network

11:15 a.m. - 12:00 p.m.
Mastering the Equation


Willy Henshall

Right now, the Internet equals money. The only question is, now can studios and mastering facilities improve the end product when it's going to be distributed on MP3 or some other Internet format? Discussions include billing for the service, understanding the formats, documentation, costs, etc.

Speaker: Scott McConville, Gateway Mastering

12:00 p.m. - 12:45 p.m.
Fun & Games


Tommy Tallarico

Award-winning musician/producer/composer Tommy Tallarico reveals the hidden money making opportunities in Video Games, multimedia and the Internet. Tommy has participated in some of the world's most famous games and interactive ventures, and will thrill the audience with a presentation that offers multiple opportunities for the modern studio to make money in this arena.

Speaker: Tommy Tallarico, Tommy Tallarico Studios

12:45 p.m. - 1:30 p.m.
Networking Luncheon

1:30 p.m. - 2:15 p.m.
Surround Sound and the Money Myth


Bobby Owsinski

The advent of surround sound seems inevitable, but how can the studio offer services in this area and get paid for it? There will be significant investment in equipment and personnel by studios to accommodate demands in this area. How will the economics work themselves out? Bobby Owsinski unveils the truth behind the myth of surround sound, discussing where the potential for increased revenue lies with this unfolding technology.

Speaker: Bobby Owsinski, Surround Professional

2:15 p.m. - 3:00 p.m.
Working with Project Rooms


Mitch Gallagher

This panel of commercial and project studio owners will look at the ins and outs of sharing projects with smaller pro project rooms. Topics will cover everything from formats to billing to cooperative marketing to use of the World Wide Web for exchanging files or promoting each other.

Moderator: Mitch Gallagher, EQ magazine
Panelist(s): Tommy Dorsey, Producer/Engineer


Tommy Dorsey

3:00 p.m. - 3:15 p.m.
Coffee Break

3:15 p.m. - 4:00 p.m.
Case File

It's all about billable hours. Rather than threatening to deteriorate sound quality, the Internet offers an opportunity for studios to extend the hours by offering clients an additional service. Matt Ward shows how he uses Liquid Audio technology to enhance revenue by offering open format encoding for clients.

Speaker: Matt Ward, Liquid Audio

4:00 p.m. - 4:45 p.m.
Living With Surround


Tomlinson Holman

Everything you need to know about building a surround sound studio operation, from the technology and techniques, to the room acoustics, equipment placement, and staff training. Surround Professional magazine's founding editor and multichannel audio pioneer Tomlinson Holman provides a detailed tutorial that covers all the bases in this eStudio exclusive presentation.

Speaker: Tomlinson Holman, TMH Corporation

Saturday May 20, 2000

9:00 a.m. - 9:30 a.m.
Keynote

Speaker: To be announced

9:30 a.m. - 10:15 a.m.
Can Surround Save the Studio Business?


Chuck Ainlay

Jake Nicely, owner of 17 Grand Recording, moderates this panel of industry luminaries as they examine all the issues facing the modern studio, and how surround sound may grow to be the biggest money maker.

Moderator: Jake Nicely, 17 Grand Recording
Panelists: Chuck Ainlay, Producer/Engineer; Kerry Moyer, CEA; Rory Kaplan, DTS; Hank Williams; Denny Purcell

10:15 a.m. - 10:30 a.m.
Coffee Break

10:30 a.m. - 11:15 a.m.
New Opportunities for Internet Audio


Matt Fine

Matt Fine, vice president of program production at Audible.com, discusses business-to-business opportunities for studios including creating audio programming for everything from spoken word projects streamed over the Internet to using the Internet to distribute corporate projects recorded in the studio.


Keith Hatschek

Speaker: Matt Fine, Audible.com

11:15 a.m. - 12:00 p.m.
Studio Revolution

What will the future of the studio look like? Will it include consoles, workstations, hard disk recorders? How will studios market their services in the brave new world of Internet distribution?


Dave Gustafson

Moderator: Keith Hatschek, Keith Hatschek & Associates
Panelists: Brad Wood, Dave Gustafson, Roger Wiersma, Ramzi Haidamus

12:00 p.m. - 12:45 p.m.
The Business of Archiving


Brad Wood

Malcolm Davidson, who heads up Sony Music's archiving operations details the massive undertaking the label has engineered in archiving its library.

Speaker: Malcolm Davidson, Sony Music


Roger Wiersma

12:45 p.m. - 1:30 p.m.
Networking Luncheon

1:30 p.m. - 2:15 p.m.
Who Cares About Quality Anymore?


Ramzi Haidamus

With pressures mounting to stream audio, to compress it into MP3 files, and with the trend toward using the Internet as a distribution medium, the concern for quality has been raised. A panel of the world's most renowned mastering engineers tackles this sticky issue.

Moderator: Frank Wells, Pro Sound News
Panelists: Glenn Meadows, Masterfonics; David Glasser, Airshow Mastering; Bob Katz, Digital Domain; Bob Olhsson, Bob Olhsson Audio


David Glasser

2:15 p.m. - 3:00 p.m.
Windows Media Perspective


Bob Katz

Microsoft's operating system has been in the studio before, but now, with the impact of the Internet, Windows Media technology is poised to influence the creative production side of music, and its distribution via the Web.

Speaker: Brad Brunell, Microsoft Windows Media

3:00 p.m. - 3:15 p.m.
Coffee Break


Glenn Meadows

3:15 p.m. - 4:00 p.m.
How to Deliver Surround?


Paul West

Frustration has built up in the business as we await the resolution of new formats that will deliver on the promise of surround sound. It seems the record labels are preparing for a soft launch of titles on DVD-Audio in June. This timely panel will explore the potential of DVD-Video and DVD-Audio formats and look at some projects that have been completed in these formats.

Speaker: Paul West, Vice President, Studio Operations, Universal Music Group; Ken Caillat, president of audio and production, 5.1 Entertainment Group

*** Schedule Subject to Change***

For more information, call 212-378-0400. For sponsorship information, call Margaret Sekelsky 212-378-0491

VISIOSONIC DIGITAL 1200SL *MP3 PLAYER*

BY DJ RADAR
(A.K.A. PROFESSOR JAM)

Leave your discs at home with this computer-based professional DJ system

Being raised on *Lost in Space, Star Wars,* and *The Jetsons,* I've longed for the day when I could beam my entire DJ setup over to an event — who wants to carry all those records, turntables, and mixers? Although that's not in the cards just yet, MP3 data compression (see sidebar) brings us much closer to that ideal.

Sure, some DJs are already forsaking tapes, CDs, vinyl, and videos as ways to promote themselves for gigs. Instead, they create high-tech multimedia demonstrations on notebook computers (backed up by a live mix on the system during the presentation). But now the computer can actually become part of the DJing process with VisioSonic's Digital 1200SL MP3 player ("PCDJ" for short), a computer-based professional DJ system. The software "virtualizes" a pair of turntables, a mixer, and a record case on your computer screen. After encoding your music collection to MP3 files and storing them on a big hard drive, you can use PCDJ to set up your playlist, cross-mix between tracks, change volume levels, beat-match one song with another, and so on. The interface is so friendly it's like working a real-life system, but with finer control. Interestingly, this doesn't make for a less creative experience; in fact, having everything in one place lets you really focus on the musical flow.

Installation and setup is easy. The software auto-tests the PC's sound cards and generates a list of PC settings that tech support can use if there are installation problems. Copy protection consists of e-mailing or calling the company to "unlock" the software.

VisioSonic recommends two sound cards to allow main and cue stereo outputs, but the system works with one sound card if you can forego cueing. PCDJ supports WAV files, MP3, the new Xing VBR MP3, and Encrypted MP3. Commands can be triggered with MIDI note-ons (*e.g.,* a keyboard controller patched to the sound card's MIDI in) for exceptionally fast response. Currently, note assignments are fixed, although you can use any MIDI channel. Audio processors (such as an EQ) patch between the mixer and amp, as usual.

The system is password-protected to guard against unauthorized changes while the program is in operation. Also, encryption of the encoded MP3 files makes them safer from theft and piracy.

CUE ME UP, SCOTTY

Unlike mechanical devices, the PCDJ offers real-time instant start, cueing to within 0.01 ms, pause with "audio looping," and up to 20 cue points for each track. (You can see, as well as hear, the cue points while editing cues.) The "Auto Cue on Stop" feature skips the silent part at each track's beginning and end. For precise beat-matching, there's a "match pitch to BPM

(Beats Per Minute)" conversion button; the program's metronome-assisted BPM counter allows you to determine a song's tempo (if the tempo changes, you naturally need to re-adjust). Pitch controls for each player range from +4 percent to –52 percent, and a mini player allows auditioning a track directly from the record case without having to load it first. And where do you get those MP3 files? The PCDJ does audio extraction from CDs, using a fast encoder.

However, there is one major limitation of this technology: You can't use dynamic scratch techniques unless you get really facile with the pitch control or create an MP3 scratch file as filler at a particular cue point. Even then, it doesn't come close to the real theatrics of scratching. Also, the system is picky about what CD drives and sound cards you can use (see *EQ* Lab Report). While other devices can work, VisioSonic guarantees optimum performance only with the recommended components.

ON THE FLY

The PCDJ's labeling, grouping, and find capabilities simplify mixing on the fly. There is a search engine for the track database, and you can create an unlimited number of groups and subgroups in the record case to store links to tracks on the hard drive. Removable media subgroups are sortable by BPM, title, artist, version, or comment, and DJs can easily tag edit with the new ID3V2 Version 3.0 tag standard. (This simply means that each track can be tagged or labeled with information about the artist, song, etc., which will be displayed in the "record case.") You can name each cue point to make it easier to find.

To get an idea of how the process works, from a category of twenty to thirty songs incorporated into a specific playlist, you can drag and drop titles in a pre-selected order and load them into both players (A & B), similar to what happens with CDs and vinyl. From there, mixing and matching the beat from one player to the other simply requires pressing a button to set your next track to the same BPM, adjusting the pitch control to your desired key, bringing up the volume, and tagging the beat. It's almost an automatic mix. Of course, slam mixing and dropping works just fine as well.

IN THE Groove

MANUFACTURER: VisioSonic, 21939 US 19 North, Clearwater, FL 33765. Tel: 727-799-3828. Web: www.visiosonic.com.

APPLICATION: Professional DJ mixing where portability and control are paramount.

SUMMARY: PCDJ brings the professional DJ to a higher level of technology, offering exceptional capabilities and ease of use.

MINIMUM SYSTEM REQUIREMENTS: Pentium II or AMD 300+ MHz (no Cyrix chips), Windows 95/98, 32 MB RAM (64 MB RAM for SoundBlaster Live sound card), 10 MB hard disk space (more required for MP3 songs), 20X CD-ROM or above (must support digital encoding), one SoundBlaster Live or two Turtle Beach Montego A3D Xtreme sound card(s), 800x600 SVGA/256 color display.

STRENGTHS: Simple to use. Beat match capabilities with the press of a button. Thorough, readable documentation. Easy setup with auto-testing. Instant start. Integral search engine. High-resolution cueing. Supports dual sound cards for cueing. Includes CDDB (Compact Disc DataBase) functionality.

WEAKNESSES: Requires particular CD-ROM drives and sound cards. No scratching.

PRICE: Software PCDJ PHAT Digital 1200SL $199; rackmount system with computer $1,699; mini-tower version $1,099.

EQ FREE LIT. #: 101

TRAVEL LIGHT

You can put your entire music collection in a laptop — with a 10 GB hard drive, the PCDJ holds *well* over 3,000 tracks at 192 Kb VBR. There's even an Auto Pilot function that automatically plays tracks from the record case should the DJ not make it back in time for the next mix.

Overall, the PCDJ has been a great tool. Many large radio broadcasting companies have used MP3 technology for years; now this reliable compression format has made its way into the mobile and club disc jockey market. MP3 technology gives me the ability to mix easier, but the PCDJ allows me to "mix smarter."

[*Note:* A "junior" demo version, PCDJ Phat, is downloadable from VisioSonic's Web site.]

From a meager start as a vinyl disc jockey at a local roller rink in 1973, Professor Jam has become a full-time entertainer, and owns the Florida-based company, "A Spinnin' Crazy Production." He is often found speaking and demonstrating at DJ and music conventions, and his level of community involvement has made him one of the few mobile disc jockeys entered into the Congressional Record.

MP3 BASICS

MPEG (pronounced "em-peg") stands for *Moving Picture Experts Group,* an industry standards group that defines universally compatible ways to code and transmit audio and visual information. MPEG is also a sophisticated technology that compresses and transmits music and video over networks (satellite, the Internet, and telephone) with minimal loss of sound or picture quality.

MP3, an extension of MPEG technology, specifically covers audio file compression and storage; Layer 3 is a third-generation coding standard that can produce near-CD-quality sound, while requiring only a fraction of the amount of disk space used by uncompressed hi-fi audio. Quality varies with the compression ratio (typically 5:1 at the high end to 22:1 for low-fi applications).

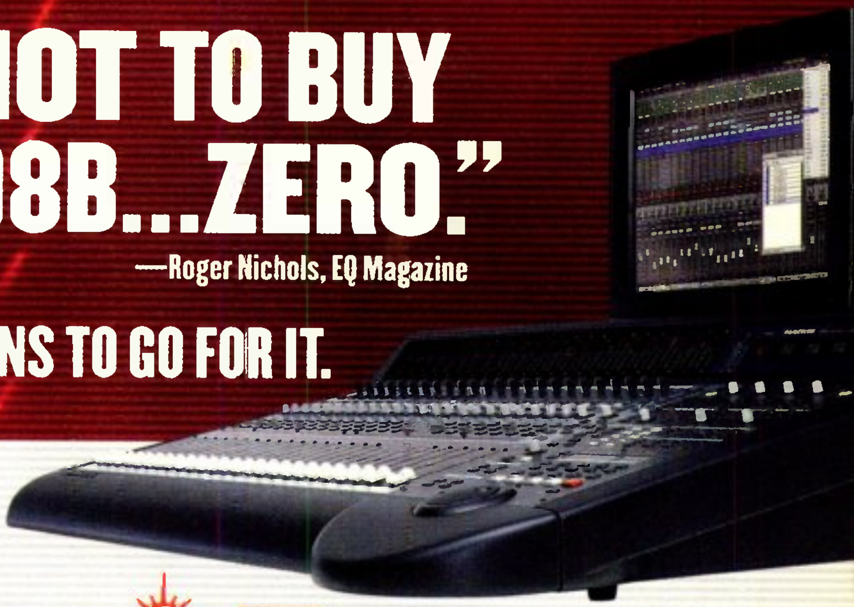
MP3 compresses data by analyzing a file's audio and not coding sounds that it deems imperceptible (*e.g.*, sounds masked by other sounds). Discarding unnecessary audio information saves tremendous amounts of disk space, and the fidelity remains acceptable for all but the most audiophile-oriented applications.

There are two main encoding processes, VBR (Variable Bit Rate) and CBR (Constant Bit Rate). With CBR, the bit rate stays constant. With VBR, the computer analyzes the song being encoded and changes the bit rate on an "as needed" basis, attaining the maximum bit rate only when requiring maximum fidelity.

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

—Roger Nichols, EQ Magazine

PLUS 3 MORE REASONS TO GO FOR IT.



1 FREE UPGRADE! NEW OS 3.0 ADDS OVER 30 NEW FEATURES!

Our Programming Department has been chugging the double lattés to create Mackie Realtime OS™ Version 3.0, packed with more new features and enhancements than you can shake a mouse at. Here's just part of what 3.0 adds to the already amazing D8B.

- **New key (sidechain) inputs** for all 48 onboard dynamic processors featuring soft knee architecture and single band 20-20k parametric EQ for frequency dependent processing such as de-essing
- **3rd-party plug-ins** via our new UFX card. Up to 16 simultaneous plug-ins on the first 48 channels, pre or post DSP, pre-fader via up to 4 UFX cards. Each plug-in is available twice — once when tracking, and again at mixdown!
- **Multiple Undo List** — 999 levels!
- New **Snapshot libraries.**
- Externally or internally accessible **inserts across Mains and Buses** plus **channel inserts pre and post DSP.**
- **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.

DIGITAL MACKOIDS AT OUR FACTORY IN WOODINVILLE WA, USA

The list of top engineers and producers who use the award-winning Mackie Digital 8 • Bus is growing daily. For info on the D8B, new UFX and Optical • 8 cards, 3rd-party plug-ins and how D8B owners can get their free OS upgrade, visit www.mackie.com or call your local D8B dealer.

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CIRCLE 30 ON FREE INFO CARD

ELECTRIX MO-FX PROCESSOR

Electrix scores with a multieffects processor that will please DJs and engineers alike

BY CRAIG ANDERTON

Although the Electrix Mo-Fx is a multi-effects (distortion, flange, tremolo, and delay) designed to satisfy the DJ's/performers appetite for effected sound, it's great for studio engineers who like to *play* with a mix. Parameters aren't hidden behind an LCD menu; every function is brought to a large, playable button or switch so you can bash around in real time. Like other Electrix units, the Mo-Fx works in a rack or on a tabletop (my preferred option), and is solidly constructed.

You can't program or recall settings, although each control produces MIDI controller data that can be recorded into a sequencer or fed back to re-create a performance. In the studio, I usually just ignore MIDI and send the track to be processed through the Mo-Fx to an empty recorder track.

There are no input or output level trims for this unity-gain device, although there are in/out level LEDs. If you need level control in the studio, patch the Mo-Fx into an aux bus, and use the bus sends and master to adjust levels.

THE MODULES

All effects have a Bypass/Active button and a large, lit Momentary button. Hitting this inserts the effect if bypassed, and bypasses it if inserted — great for inserting effects when desired.

Regarding individual modules, distortion offers level and drive controls; it's more of an overdrive effect than a "stack o' Marshalls" buzz. Unlike the other effects blocks, it can go in parallel, with distorted signal hitting the out — but you can also feed distorted or undistorted signal to the remaining effects chain. The inserts go pre-distortion, but minimizing the distortion drive setting causes the Bypass and Momentary buttons to control the external effect without introducing distortion.

The flanger, delay, and tremolo work as expected, with a few twists. Each has a "Band" button that selects among seven combinations of high, mid, and low bands, so you can process the entire signal, individual bands, or various combinations — for example, delay the hihat, but not the kick. However, the Sync button is

IN THE Groove

MANUFACTURER: Electrix, 6710 Bertram Place, Victoria, BC., Canada V8M 1Z6. Tel: 250-544-4091. Web: www.electrixpro.com.

APPLICATION: Performance-oriented multieffects (distortion, flange, tremolo, delay) for processing recorded tracks, live instruments, loops, or turntables.

SUMMARY: This multieffects is designed for real-time playing, not programming — it's fun, flexible, and unique.

STRENGTHS: Solid tempo syncing options. Big knobs, lit switches, and bold typefaces for player-hostile stage environments. Insert jacks. Well constructed. Rack or tabletop operation. Clear documentation. Accepts turntable inputs.

WEAKNESSES: No input or output level adjustments. Flanger not as flexible as some other devices. Delay tails continue to decay even after you release the "momentary effect" button.

PRICE: $549.99 **EQ FREE LIT. #:** 104

these modules' most important feature, as it allows syncing sweeps to tap tempo or MIDI clocks. When synced, the speed controls become "division controls" for different rhythmic values (from 1/8th speed to 8X speed). Furthermore, each block's control can have its own division value.

Other goodies include multiple tremolo modulation waveforms along with a stereo ping-pong option, delay times up to 2,600 ms (max delay regeneration produces a "freeze" or "hold" effect), and a tap tempo feature that averages multiple taps. My only complaints: I'd prefer a wider-range flange sweep, positive and negative flanging could be better differentiated (I don't think the regeneration phase changes when you change the mix phase), and the delay tails don't cut off when you release the Momentary button. I don't mind that they continue after being bypassed, but I'd like some way to stop them with a switch. As things stand, you have to turn a control.

I seldom used all of the effects blocks at once, because each block offers so many playable options. I tended to concentrate on playing with and effecting a particular sound. For some trance music, you could probably just feed a couple low-frequency sawtooth waves into the Mo-Fx and play with the controls all night.

BLAME CANADA

With its first three products (Filter Factory, Warp Factory, and Mo-Fx), Electrix has scored a signal processor hat trick. More importantly, this small Victoria company has taken a new slant on existing tools, and given hands-on engineers/DJs/musicians extremely cool sound mutation machines. If you thought you'd seen it all when it comes to multieffects, Mo-Fx will convince you otherwise. It's an outstanding and creative product. **EQ**

REAR PANEL CONNECTIONS

- Left and right 1/4-inch line-level (+4) in/out jacks
- Left and right RCA phono jacks, switchable between +4 or phono (turntable) level
- 1/4-inch footswitch jack for bypass on/off
- Left and right TRS 1/4-inch jacks for inserting a pre-distortion effect
- MIDI in/out/thru, with a 16-position, rotary channel-selector switch
- IEC AC cord receptacle (settable to 110/120/234/240 volts)

Bomb Factory Classic Compressors

Who needs hardware — get classic compression with your Pro Tools system

BY MITCH GALLAGHER

Vintage mania continues to run rampant in many parts of the audio universe. How many of us haven't lusted after a time-washed U 47 mic, a Neve 1073 EQ, or an original lava lamp? But unfortunately, owning vintage gear comes with a price (that is, aside from its *purchase* price): You have to maintain it, and, in this squeaky-clean 24-bit digital age, you also have to deal with '60s- and '70s-level noise floors. And, of course, all that is assuming you can even *find* one, let alone find a *good* one. What's a poor slob consumed with vintage gear envy to do?

Buy a TDM/AudioSuite/RTAS-compatible system and call the Bomb Factory, that's what. Admitted vintage freaks themselves (check out www.bombfactory.com for a lust-inspiring look at their collection), the Bomb Factory has collaborated with respected gear designers Dave Amels (Voce and others), B. Andrew Barta (the man behind Tech 21), and Bob Moog (the man himself), to re-package a number of classic processors into a group of very cool software plug-ins.

Common fixtures on many vintage wish lists are the tube and solid-state dynamics processors of days gone by. To address those wish lists, the Bomb Factory has also released the Classic Compressors package. Classic Compressors comprises plug-in versions of two of the most sought-after vintage dynamics processors, the venerable Universal Audio LN-1176 solid-state and LA-2A tube compressor/limiters. TDM, AudioSuite, and RTAS (Real-Time AudioSuite) versions are available now, an MAS (MOTU Audio System) version is on the way.

OVERVIEW

I was immediately impressed by the graphic interfaces used for these plug-ins. The Bomb Factory gang has duplicated the front-panel appearance of the originals pixel-for-pixel. They've even gone to the trouble of duplicating the mechanical VU meter ballistics of the originals, so, if you're accustomed to watching the real things, you'll feel right at home. My only complaint with the user interface is that, while the knobs may move authentically, I found that unless you clicked very, very carefully, the selected knob would move with the click, even if you didn't want it to. Once it was clicked, rotation, etc. was fine. I just found it very hard to click without moving the knob. For tweaky adjustments, holding the Command key while clicking gives finer resolution.

Bomb Factory has gone a step past the originals and added a few cool features such as sidechain inputs, storing and recalling presets, mono and stereo operation, and so on. The only other significant difference between the hardware and software units is that the LA-2A's Comp/Limit switch appears on the plug-in's front panel, instead of the rear.

With the TDM version, you can get three instances of either compressor on a Mix card DSP chip. The plug-ins will work on either type of chip on the card. The only limitation is that you can't run a LA-2A and an 1176 on the same chip simultaneously.

The entertaining manual discusses the concept of compression and the operation and specifics of each model, provides tips and tricks, and also covers topics such as de-essing, "un-pumping" drum tracks, breathing life into lame tracks, and using predelay to achieve faster attack times. The manual is a combo affair, covering the entire line of Bomb Factory plug-ins. Be prepared to be constantly tempted by those you don't own.

IN USE

The beauty of the hardware versions of both of these processors is that they're simple to use. Dial a few settings in, and they sound good — that's it. The same is true here with the modeled versions. While you can certainly over-compress the signal or cause pumping/breathing very easily, these are some of the easier-to-dial-in compressors out there that still give you a reasonable amount of control over the unit's operation. Speaking of over-compressing, there is a preset (or you can double-click or option-click a ratio button) for the 1176's infamous "all four buttons depressed" setting, which squashes the signal completely. Not a sound I generally find particularly musically useful, but, hey, the original could do it and so can this modeled version.

The question with modeled devices and plug-ins is: "Do they sound like the originals?" In this case, the answer is, yes, Classic Compressors sound close enough for most ears. But, in my opinion, that almost misses the point of these processors. Taken on their own merit, they're effective, good-sounding plug-ins. They add a nice roundness to the signal and also soften or warm the

top end a bit. I used them on drums, vocals, acoustic and electric guitars, basses, flugelhorn, percussion, and full mixes. In every case, I was pleased with the results. In fact, during the time I spent with the Classic Compressors, they became my first-call dynamics plug-ins (and my Pro Tools rig is loaded up with most of the competing products).

CONCLUSIONS

The Classic Compressors don't fall into the "so-flexible-they-can-do-it-all" category. There are some cases where you'll want more tweaky parameter control, multi-band compression, less coloration, and so on, but, for what they do, the 1176 and LA-2A top my list.

While I don't think many of us would turn down the opportunity to own a great example of one of the original hardware units, these plug-ins are more than sufficient to give you similar results under many circumstances, and they're a heck of a lot more convenient and flexible — and cheaper, especially since you can load multiple instances simultaneously. If you're a Pro Tools user, and you're looking for some vintage-style dynamics processing, check out the Bomb Factory offerings: Great performance, noise-free, no maintenance, good sound, they're the bomb! (Sorry, couldn't resist....) EQ

EQ LAB REPORT

MANUFACTURER: Bomb Factory Digital, 3917 W. Burbank Blvd., Burbank, CA 91505. Tel: 818-558-7171. Web: www.bombfactory.com.

APPLICATION: Virtual vintage compressor/limiters.

SUMMARY: Impressive re-creations of two popular vintage dynamics processors; the Classic Compressors package provides convincing tones and equally convincing operation with classy graphics, and couple of cool modern features.

STRENGTHS: Authentic operation. Stereo link. Sidechain. Fairly DSP-efficient. Cool graphic interface — love those VU meters!

WEAKNESSES: None to speak of.

SYSTEM REQUIREMENTS: TDM/AudioSuite requires Pro Tools 4.0 or higher. Real-Time AudioSuite (RTAS) requires Pro Tools 5.0 LE or higher.

PRICE: $599

EQ FREE LIT. #: 105

SANSAMP MODEL PSA-1

In addition to the Classic Compressors package, Bomb Factory is also offering a number of other modeling processors, including Voce's Spin rotating speaker simulator, Bob Moog's moogerfooger Lowpass Filter and Ring Modulator, and Tech 21's SansAmp Model PSA-1. All feature the same type of impressive photo-realistic 3-D graphics as the compressors. Just for kicks, I took the PSA-1 for a spin around my hard drive.

If you've checked out the hardware version of Tech 21's SansAmp, then you have a great idea of what the Bomb Factory plug-in can do for you. The plug-in provides a tube amplifier simulation with built-in speaker emulation and flexible sound shaping in the form of Buzz, Punch, Crunch, and Drive controls, which shape the distortion characteristics, and High and Low EQ controls for shaping the tone. A Preamp control sets the input sensitivity and first stage of distortion, and a Level control acts as a master output volume. All of the controls are automatable in Pro Tools; using automation you can also access a Master Bypass control.

Initially, I experimented using the SansAmp as an amp simulator for recording guitar tracks. Plugging a Les Paul and a Strat through a Radial Engineering DI straight into my digital mixer interfaced to Pro Tools, and later through a JoeMeek VC3Q preamp/EQ/compressor straight into Pro Tools, I was able to dial up a range of convincing tones. All 49 of the original presets from the hardware PSA-1 are included, as well as new presets, including two simulating the power amp distortion in a Leslie cabinet. (Handy for use with Bomb Factory's Voce Spin Leslie simulator plug-in, one would have to think.)

For guitar, if you liked the sound of the original SansAmp, you'll like this re-creation. (Hmm, a model of a simulator; what will they think of next?) To my ears, the plug-in sounds especially good on some of the heavier, higher-gain tones. As with most devices and processors similar to this, my ears were less deceived by the lower gain and clean guitar settings, but they're still pretty good and would likely work well in a mix.

As much fun as I had annoying the neighbors with screaming guitars, I had even more fun when I settled down to process other types of tracks with the PSA-1 plug-in. I first used it on a variety of drum and percussion parts. I found that the plug-in could add a nice quality to the sound if it wasn't overdone — although overdoing it produced some nice sounds, too! In one extreme case, a resonant djembe sample turned into a ringing, punchy, almost-pitched electric guitar-type *chunk* when processed with high gain and careful EQ. Perfect for the grinding industrial track I was working on. As effective as the distortion and tone-shaping is on guitar, it can be equally powerful on drums. The Punch and Crunch controls, in particular, are voiced well for massaging the presence of drum tracks into shape.

I next tried the PSA-1 on some Ebowed fretless bass tracks. Here, it helped to clarify and add presence to the sound without necessarily adding distortion; a nice trick.

On vocals, you can use SansAmp almost as an EQ, to add some edge, to overload the signal, or to torture the vocals into outright distortion. The combination of the flexible distortion- and tone-shaping controls gives you the ability to fine-tune the tone exactly the way you want, and, by automating changes, you can easily bring the effect in and out for certain passages or even words or syllables within a song. Very cool!

Negatives? At $499 list, it's not cheap, but keep in mind that you can run multiple instances simultaneously — as long as you have DSP chips available. Each instance scarfs up an entire chip on a PCI or Mix card. Fortunately, it will run on any of the DSP chip varieties found on the Pro Tools cards.

The verdict? A powerful distortion processor and amp simulator capable of broadly expanding your Pro Tools plug-in palette. Even if you're not into out-and-out distortion or direct-recording guitars, you'll probably still be able to find a multitude of sound-shaping uses for the PSA-1. *—Mitch Gallagher*

Furman SRM-80 Signal Router/Monitor

Get your monitor and dubbing needs under control

BY MITCH GALLAGHER

It's no secret that, over the past few years or so, much of the recording, mixing, and processing power in many studios has been migrating toward the computer. My DAW, for example, handles just about everything involved in the production of audio; what isn't accomplished in the DAW is handled by synths, samplers, and a few processors, which are directly connected to the DAW's ins and outs. Microphones are routed through stand-alone preamps straight into the DAW's converters. It's all pretty neat and simple.

I still have a mixer, but at this point its main purpose is to control the volume of my DAW's output as it feeds my powered monitors. Occasionally, it's used for monitoring a CD player, a cassette deck, and a DAT machine, and for routing analog signals between the DAW and the various decks or from deck to deck.

Kind of a waste, really. It's a great board, but about all I'm using from it is the output section. So when I saw Furman Sound's SRM-80 signal router/monitor, I was intrigued. Here was a box that was apparently designed exactly for my situation. Would I at last be able to get rid of my mixer?

OVERVIEW

Let's make one thing perfectly clear: Despite the question at the end of the last paragraph, the SRM-80 is not intended to replace a mixer. Mixers are still the most efficient way for routing and combining multitudinous signals. But for many DAW-based rigs, mixers serve little or no purpose. And, in many studios, the small, rack-mountable mixers being used aren't capable of routing signals to and from multiple stereo master recorders or switching between multiple sets of speakers. In either case, the SRM-80 can be a useful addition. Furman also offers an optional remote for the SRM-80, the SRM-RU ($79), which provides control over the speaker selection, mono, and dim switches.

Here's the short story: The single rackspace, SRM-80 comprises three sections: line level I/O routing, level control and metering, and speaker output and selection. Let's take a closer look at each section.

LINE LEVEL I/O

The SRM-80 provides inputs and outputs for four stereo sources/destinations in addition to a fifth stereo "Source" (input only), which is normally connected to your mixer, sound card, or DAW output. The Source input pair uses 1/4-inch, balanced connections. The Stereo Device A I/O connections are also 1/4-inch balanced. A switch is provided for changing both of these from +4 to –10 level operation. Too bad you can't set their levels independently. Stereo Devices B, C, and D all use unbalanced RCA connections and operate at –10 level.

I hooked up the +4 balanced outputs of an Alesis MasterLink to the Stereo Device A inputs, and the MasterLink's unbalanced RCA –10 outputs to the Stereo Device B inputs, and found that the SRM-80's balanced I/O tends to run about 3 dB hotter than the unbalanced. This is definitely an audible difference and could be a problem if you're trying to critically A-B between two sources. (Furman says that this input sensitivity problem will be fixed in new units. Owners of existing SRM-80's can contact the company for upgrade instructions.)

Whether or not the SRM-80 has the right configuration of balanced and unbalanced I/O depends on the gear you have in your studio. It's likely that there simply wasn't room on the back panel for any other combination of jacks, but, in an ideal world, I would have preferred to have seen all the line level I/O as switchable-level balanced/unbalanced 1/4-inch connectors, or, even better, switchable-level combo XLR/1/4-inch connectors.

Four rotary switches are used to route signals between the Stereo Device connections. Any source can feed any output, although Furman has wisely set things up so that a source can't feed itself. You can dub between more than one set of I/O at the same time. For example, you can dub from A to B and from C to D simultaneously while also monitoring any of the five inputs you choose. One input can also feed multiple outputs simultaneously. One application might be to feed the Source Stereo Device to all four of the other Stereo Devices for making multiple dubs or to mix to multiple machines/formats at the same time. There are no pops or clicks when switching from device to device.

MONITOR CONTROL/METERING

The monitor control and metering section features another rotary


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switch, which lets you choose any of the five inputs (Stereo Devices A through D or Source). A Listening Level knob sets the monitor volume, and a Dim switch is provided for quickly reducing the volume level by 15 dB — handy for when the producer needs to take a phone call during the session. Not that *that* ever happens....

Rounding out the monitor control section is the Mono switch, which sums the left and right signals, and an LED bargraph meter. The meter is switchable from average- to peak-reading operation, and concentrates most of its 40 steps of resolution in the range from –7 dB to +12 dB (in peak mode).

The Listening Level control exhibited the biggest flaw in the unit: As the volume was reduced to very low levels, the stereo balance skewed off to the left side. Once the volume was up to more reasonable levels, the image centered and was fine. Fade ins/outs and quiet input levels in the audio signal didn't exhibit this behavior; the problem seemed to be with the pot or circuit used to set the output volume of the SRM-80 itself.

SPEAKER OUTPUT/SELECTION

The final section of the SRM-80 deals with routing the signal to various pairs of speakers. Connections are provided for one set of powered monitors and two sets of passive (unpowered) monitors. Routing to the unpowered monitors is accomplished by sending a line-level output to the power amp, bringing the power amp's speaker-level output back into the SRM-80, then routing the SRM-80's two speaker-level outs to the monitors themselves.

You could quibble with this selection of speaker outputs. One retailer I spoke to confirmed my suspicion that powered monitors currently outsell unpowered by a large margin. However, there's no debating that there are thousands of pairs of unpowered monitors in use out there, so Furman can certainly justify their choice.

If you're like me, and mainly use powered monitors, it is possible to use the line-level feed intended to go to the power amp's input as a second powered monitor feed, although this isn't documented in the SRM-80's manual. This lets you switch between two sets of powered monitors, although the third SRM-80 speaker connection is wasted.

Three switches with accompanying indicator lights allow you to choose which set of speakers you're listening to; deselecting all the switches mutes the SRM-80's monitor output, although not the headphone output, which is handy.

Speaking of the headphone out, I found that the SRM-80 was capable of providing more than enough power to drive my AKG K240m headphones to ridiculously loud levels. I would have liked to have seen a separate volume control over the headphone output, but in practice I found that a reasonable listening level over the speakers also resulted in a reasonable level in my 240's.

Below the speaker selection switches you'll find three tiny, recessed trim pots for balancing the relative levels of your monitors. You'll need a small Phillips-head screwdriver to adjust them, but these are likely to be "set 'em and forget 'em" controls, so the hassle is minimal.

In my experience, setting the trims properly is critical to getting the most out of the SRM-80. At first, I followed the manual's advice and set the trims to around 75 percent. But I found that, with this setting, I was running the Listening Level control barely cracked open for reasonable listening levels — resulting in the stereo balance problem discussed above. Instead of following the manual, I turned the trims all the way down, which let me bring the Listening Level control up to the middle part of its range. This solved the stereo balance problem, as well as a problem I had with the volume jumping abruptly on as I cracked open the Listening Level. With the trims

continued on page 128

EQ LAB REPORT

MANUFACTURER: Furman Sound, 1997 South McDowell Blvd., Petaluma, CA 94954. Tel: 707-763-1010. Web: www.furmansound.com.

APPLICATION: Monitoring, routing, and dubbing controller.

SUMMARY: If you've been looking for a way to get your monitor and dubbing needs under control, the SRM-80 deserves serious consideration. It's particularly cool for DAW owners who don't want to fool with a mixer just for controlling monitor levels and dubbing.

STRENGTHS: Plenty of inputs. Flexible routing and dubbing capabilities.

WEAKNESSES: Stereo balance shifts at low monitoring levels. The +4 balanced signals connected to the +4 inputs tend to be 3–4 dB hotter than the same signal at –10 dB run into the –10 dB inputs.

PRICE: $499

EQ FREE LIT. #: 102

FURMAN SRM-80 SPECS

Line Level Inputs: *Source:* Stereo 1/4-inch, balanced +4 dBu or –10 dBV
Input A: Stereo 1/4-inch, balanced +4 dBu or –10 dBV
Input B, C, D: Stereo RCA, unbalanced –10 dBV
Speaker Level Inputs: *"From Power Amp":* stereo, binding post
Line Level Outputs:
Output A: Stereo 1/4-inch, balanced +4 dBu or –10 dBV
Output B, C, D: Stereo RCA, unbalanced –10 dBV
Speaker A: Stereo 1/4-inch, balanced +4 dBu
"To Power Amp": Stereo 1/4-inch, balanced +4 dBu
Speaker Level Outputs: *Speaker B, C:* Stereo, binding post
Headphone Output: Front-panel 1/4-inch TRS
Metering: 40x2 LED bargraph meter, switchable from peak to average response
Monitor Control Options: Listening level, Dim (15 dB attenuator), Mono (sum), speaker level trim (3), speaker selector switches (3)
Frequency Response: +0, –1 dB from 20 Hz to 20 kHz
Dynamic Range: Greater than 96 dB

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Event Project Studio PS6 Monitors

Event's latest nearfields offer big sound in a compact size

BY MITCH GALLAGHER

Over the past few years, Event Electronics has garnered a significant reputation for producing great-sounding affordable nearfield monitors (they refer to them as "Direct Field" monitors for technical correctness). Most of their designs are self-powered, although there's one passive (unpowered) model in the line-up.

For this review, we're checking out the middle-sized member of Event's new Project Studio series, the PS6. (See "Meet the Family" for more on the Project Studio family.) The PS6 is a self-powered biamplified design, driving a 6.5-inch low-frequency driver with 70 watts and a 1-inch silk-dome tweeter with 30 watts. The individual monitors are all identical — meaning there's no dedicated "left" and "right" speaker. They can be used in either vertical (standing up) or horizontal (lying on their side) orientation. The front-panel features include two ports for increased bass response and a green power LED in the speaker surround that doubles as a clip indicator for those of you who get a bit too aggressive with the volume control. (You can bet that clip light never flickered while *I* was using them....)

Around the back, we find a reassuringly simple layout: There's a standard IEC three-prong power cable jack and a power switch. Near the top (assuming vertical orientation) are 1/4-inch and XLR inputs, either of which can operate balanced or unbalanced at –10 or +4 levels. Apparently two jacks were used (rather than the increasingly common Neutrik combo 1/4-inch/XLR jack) to allow for parallel "pass-through" signals to be sent on to a subwoofer, if you're using one.

A screwdriver-adjustable input sensitivity control is located between the two input jacks. It offers a broad 20 dB of range, and there's a détente at –5 dB. While the sensitivity control allows you to easily match the speaker levels to just about anything else you might need to deal with, it can be difficult to match the levels of the speakers to each other — the level control is fairly touchy, and it's tough to eyeball or otherwise ensure they're truly set the same. To be fair, just about any monitor with a sensitivity control will suffer a similar problem; even "notched" level pots don't guarantee level matching. The best solution is to run the sensitivity controls all the way up.

There are no controls provided for adjusting high- or low-frequency response. Event says that the PS6's are designed to be flat, and that, if you adjust the response, you no longer have a flat reference. Further, they feel that if the speakers are set up properly, there should be no reason to have to

EVENT PS6 SPECS

LF Driver:	6.5-inch polypropylene cone
LF Amp Power:	70 watts
HF Driver:	1-inch silk dome
HF Amp Power:	30 watts
Frequency Response:	45 Hz-20kHz, ±3 dB, ref 500 Hz
Input Sensitivity:	1.1V produces full output
Crossover:	2.6 kHz, active 4th order asymmetrical
Input Connectors:	Balanced/unbalanced XLR and 1/4-inch
Input Level Control Range:	20 dB
Indicators:	Combination power on/clip LED
Magnetic Shielding:	Yes
Dimensions:	8.25-inch x 12.5-inch x 10-inch
Weight:	23 lbs. each


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tweak their response. I still prefer to have the option to tune the speakers; many project studios have less than optimal acoustic and placement conditions.

Size-wise, the PS6's are only slightly larger than Genelec 1030A's, although they weigh about six pounds more. Their size and weight is comfortable for most, if not all, normal nearfield applications and placements. With their front porting, they can be placed nearer to walls behind them than some other designs. It's still probably a good idea to keep the PS6's some distance away from large flat surfaces to prevent unwanted bass build-up. The six-page manual gives a rundown on how to place the monitors for best results, as well as a description of the various connectors and controls.

Some speaker manufacturers suggest that, for best stereo imaging, the monitors be placed "tweeter in" when in horizontal orientation (setting on their sides); the Event manual suggests the opposite for the PS6's. It's probably best to follow the specific manufacturer's recommendations on this point (and other points, for that matter), but you might experiment with the other orientation to see which you like best. I tried them both horizontally and vertically, and, being a good little reviewer, I followed the manufacturer's instructions for placement.

EQ LAB REPORT

MANUFACTURER: Event Electronics, P.O. Box 4189, Santa Barbara, CA 93140. Tel: 805-566-7777. Web: www.event1.com.

APPLICATION: Nearfield monitoring.

SUMMARY: Clean, defined biamped monitors that pack a punch into a good-sounding, compact, low-priced, high-value package.

STRENGTHS: Big sound for compact size. Excellent price/performance ratio. Articulate midrange.

WEAKNESSES: No response tailoring. Somewhat difficult to match sensitivity levels between the speakers.

PRICE: $699 per pair.

EQ FREE LIT. #: 103

THE SOUND

I set up the PS6's in my studio and settled in for extended listening to a variety of commercial releases; everything from Hans Zimmer's orchestral score for *A Thin Red Line* to the classical guitar of the Los Angeles Guitar Quartet's *Air and Ground* to Pantera's *Reinventing the Steel* to the ubiquitous Roger Nichols's work on Steely Dan's new *Two Against Nature.* I also scrutinized an array of my own mixes and works in progress. The *de facto* monitors in my studio are Genelec 1030A's, which I know more intimately than perhaps a human being should (no comments please, this is a family magazine).

Guess what: The PS6's sound different from the Genelecs! Okay, no real surprise there; all speakers sound different (despite them all claiming to be "flat" and "accurate"). To be more specific, as with all Event monitors I've listened to, they tend to be brighter, and to have more presence in the upper-midrange and lower-highs than some other speakers. (Event says their speakers seem brighter because they use a silk-dome tweeter, which has no resonant frequency that must be scooped out.) They're nice and tight on bass, but on first impression don't seem to reach down quite as far as some similar-sized monitors I've used. This can be deceiving though, since the enhanced upper-mid presence tends to emphasize the "punch" of bass sounds, rather than the "boom." By focusing in and carefully comparing the bass of the 1030A's with the PS6's, it became clear that the bass range of the two is actually quite similar; it's in the higher-bass/lower-mid frequencies that there are differences. The Genelecs sound rounder and fuller here, the Event's more present and perhaps clearer.

I did find that, to my ears, the Events are ever so slightly better suited to pop/rock/electronic music than to acoustic and classical material. With the Los Angeles Guitar Quartet's nylon-string classical guitars, for example, the PS6's got a bit "stringy" sounding, emphasizing the pluck portion of each

SURROUNDED BY EVENT PS5 MONITORS

While there are many medium-to-large nearfield monitors currently available for surround monitoring, there are a number of applications that require speakers that are smaller and more portable. Take DAWs, for instance: With many new software applications containing some sort of surround implementation, there's a coming need for a surround monitor system that fits within the limited space normally found in an editing situation. Bring on the new Event PS5's, which seem destined for this application (and others) thanks to their size and surprisingly big sound.

I used the PS5's on location during a live recording of the Ives Quintet at Colburn Hall in Los Angeles with great results. They were small enough to not only be easy to pack and carry, but also to fit nicely into the available space allotted for a makeshift control room. (In this case the "Green Room" just off the stage of the hall.) Not only were the members of the quintet surprised and pleased with the playback, but we found that the tracks we recorded (done at 24-bit/96 kHz resolution) translated well once we got back to the studio for editing and mixing. It's always interesting to measure the reactions of players who've not yet experienced surround sound, and the PS5's certainly helped to make the Quintet's first exposure most enjoyable.

As stated before, the other perfect application of the PS5's is with a DAW. Thanks to their flip-down stand, the monitors are always correctly aimed and their small size makes them relatively unobtrusive. But the thing about these monitors is that they have a very big sound for such a small box. They will fool you into thinking that they're larger than they are, as they put out much more low end and overall level than you'd expect. They also maintain their frequency response and, therefore, their "sound" at just about all levels until clipping.

If you're looking for a budget surround system or if space is a primary concern, the Event PS5's are certainly worth considering. The PS5's have a manufacturer's retail price of $299.50 each.

—Bobby Owsinski

MEET THE FAMILY

The PS6's are part of a family of three models that also includes the PS5 (5.25-inch woofer, $599 per pair) and PS8 (8-inch woofer, $849 per pair). The three models are very similar, differing only in the size of their cabinets and woofers, their respective frequency responses (the larger speakers go lower), and the design of their active crossovers. All three models share the same amplifier, which provides 70 watts for the lows and 30 watts to the highs. The PS6's and PS8's share the same 1-inch silk-dome tweeter, and the same controls and layout. The PS5's sport a slightly different tweeter, a front-panel volume control, and a built-in tilt stand. All three models are shielded for use near computer monitors and video displays.

Event says that the PS5's were based on their Tria system, while the PS8's share many similar characteristics with the popular 20/20bas monitors (200-watt, 8-inch, biamped design). —MG

note. We're talking real subtleties here, but it is something I also noted in listening to my acoustic steel-string guitar tracks, whether fingerpicked or strummed. On the other side of the coin, the Events offer a nice depth of field; you can hear "into" a mix and discern background and lower-level sounds well.

This points up one of the best things about the Events: They can be very revealing in the mids. Things like acoustic and electric guitar, vocals, and snare jump right out and are available for close scrutiny. Working on my own mixes, I found it easy to discern reverb levels on the Events. Interestingly, I also found it easy to hear the effects of compression on the Events — squashed attacks and peaks are readily apparent with that cool upper-midrange clarity.

Don't take this discussion of "enhanced midrange" to mean that the PS6's are harsh-sounding, however. The Events are easy to listen to for long sessions; ear fatigue was not an issue with them. The mixes that I did on them also transferred well to other systems, probably the most important thing a studio monitor — no matter what its size or type — can provide.

continued on page 130

Røde Classic II Multipattern Tube Mic

Event releases an improved sequel to the original Classic

BY STEVE LA CERRA

Several years ago, Røde introduced a multipattern tube microphone called the Classic. While the Classic certainly had a successful run, Røde engineers made improvements to the design and have now introduced the Classic II. There are some obvious differences between the Classic and the Classic II such as a new suspension-type shockmount, slightly different body styling, and more rugged connectors on the mic and power supply. Not-so-obvious differences include a redesigned capsule, updated power supply, and refinements in the electronic circuitry. The redesign even extended to the mic's multi-pin cable; Event claims the new cable is much more robust than the one included with the original Classic. The Classic II ships in an aluminum flight case with compartments for the microphone, power supply, shockmount, and cables. In addition to the shockmount, a swivel-type stand adapter is supplied for applications where the shockmount isn't needed.

Out of the box, there were several things we liked about the Classic II, such as the heavy-duty multi-pin plugs on the cable that connect the mic to the power supply. Any guesswork regarding pin orientation is eliminated by white dots indicating the 12 o'clock position (the mic body has a gold dot indicating its front). A ground-lift switch is provided on the rear panel of the power supply, should you find yourself in a situation that requires it (we didn't). Though we didn't like the fact that the shockmount was made of plastic, it attaches to the mic quickly and easily: you rest the mic in the cradle and then screw the cable into the rear of the mic body, sandwiching the bottom of the mount in between. Done. This is an important improvement over the original Classic's mount, which was susceptible to stand-transmitted noise.

In the studio we used the Røde Classic II on a variety of acoustic instruments as well as vocals. On a session recording steel-string acoustic guitar, the Classic II sounded great. Top end was shimmery and extended (no tube murkiness here); in spite of the fact that the guitarist thought the instrument sounded bassy on playback, we felt it was quite well balanced. Since he was singing while playing, we tried the cardioid and the next-more-directional pattern in an effort to isolate the guitar from the vocal. Our success varied. On a song where the guitar was strummed, isolation was adequate. On a song that was fingerpicked rather softly, the gain on our Yamaha 02R mic pre had to be cranked up quite a bit to get a usable signal level to tape, and so the vocal was subjectively much louder.

In any case, the Classic II picked up a sense of immediacy on the fingerpicked guitar with a crisp, percussive element when our player slapped the strings. Initially the timbre for the fingerpicked track was a touch heavy on the bass; moving the rolloff to the 12 o'clock position cured this. These sessions clearly revealed that the Classic II's noise floor is lower than that of the Classic. Since we had the Classic on hand, we were able to A/B the two mics, confirming that Røde has indeed succeeded in making the Classic II quieter than its predecessor. (Event says the difference between the self noise in the two models is 10 dB — down from 32 dB in the Classic to 22 dB with the Classic II — a significant difference.)

During the process of positioning the mic and getting levels, it was possible for us to experiment with the rolloff and pattern selector switches while not disturbing our artist.

EQ LAB REPORT

MANUFACTURER: Distributed in the U.S. by Event Electronics, P.O. Box 4189, Santa Barbara, CA 93140-4189. Tel: 805-566-7777. Fax: 805-566-7771. Web: www.event1.com.

SUMMARY: A flexible, quiet, large-diaphragm, multipattern tube microphone.

APPLICATIONS: Acoustic instruments, vocals — a great all-around mic.

STRENGTHS: Nine polar patterns, remotely switchable from power supply. Two-position bass rolloff. Two position pad. Bundled shockmount. Quiet.

WEAKNESSES: Plastic shockmount.

PRICE: $1.995

EQ FREE LIT. #: 106

Although switching wasn't totally silent — we wouldn't do this during a take — it was quiet enough not to annoy the guitarist. We quickly took to using the pattern switch as a means of tonal adjustment. Much as we expected, switching the pattern toward omni flattened out the bottom end, while moving the pattern switch to the one o'clock position added a touch of bass. We found the pad switch to be noisier when moved, and suggest muting input to the tape machine when changing it.

The transition between polar patterns is very smooth. In the first position from omni, we could barely detect a slight rejection of sound from the rear. As we switched the pattern toward cardioid, rear rejection increased and proximity effect became more evident. We found the cardioid pattern to have a wide front lobe; although it rejected sound well at 180 degrees, sound from ±45 degrees was picked up almost as strongly as sound directly from the front — and with very little off-axis coloration. In the figure-eight pattern, the 0- and 180-degree points sounded virtually identical. When used as a room mic for a live drum kit, the Classic II captured an excellent translation of the drummer's performance. The bottom end had plenty of slam without overcoming the rest of the kit. Cymbals were bright without sounding harsh, and stick hits on the toms were well defined.

For an *a cappella* lead vocal, we began by using the Classic II with the rolloff switched out. A popping "p" or two led us to switch the rolloff to the first position, curing the problem. The timbre of the vocal was very natural, and, since the singer was almost a foot away from the mic, the low-frequency response was smooth, with just the right amount of "chest" to the sound. At one point, our vocalist began whistling, and the Classic II clearly captured the whistle without picking up wind noise — which can be a problem with some microphones. When our vocalist began belting it out, the mic produced a bit of an edge in the sound, which we liked.

Between the pattern selector, rolloff switch, and positioning, this microphone can produce a huge range of sounds, making it a good all-around mic. Røde has made worthwhile improvements to the design over the Classic, without abandoning the personality of the original — which fans of the Classic will appreciate. Like the Classic, the Classic II appears to be a solid piece of equipment that ought to last for years. EQ

TC Works Spark 1.5 Digital Audio Editor

The Mac continues its comeback into the digital audio editing arena

BY CRAIG ANDERTON

In the beginning of Mac 2-track digital audio editing, there was Digidesign's Sound Designer, and it was good. Then there was Blank Software's Alchemy, and it was better. Then dark times fell upon the land: Digidesign cast away Sound Designer, Blank Software was destroyed by an earthquake, and many were exiled to the Tribe of Gates. Then a miracle occurred: Steve Berkley said, "I will render unto the Macintosh a professional 2-track editor," and he did, called Peak. The Mac had a real editor again.

TC Works further legitimizes the Mac's comeback with Spark. You probably know editing program basics: cut, paste, copy, normalize, and the like (if not, go to www.musicplayersnetwork.com, and check out the "Basic Digital Audio Editing" slide show). So, this review will concentrate on Spark's distinguishing features.

THE BROWSER VIEW

TC Electronic, the parent company of TC Works, knows how to create interfaces, and TC Works has adopted an elegant, simple interface. There are two main elements; the Browser view (fig. 1) integrates waveform editing/region creation with overview, playlist creation/editing, and file selection panes (we'll cover the Master View later). You can change the proportions of the various panes, so if you're looking for files, make the file section bigger at the expense of other panes. After selecting a waveform, make the editing pane proportionately bigger. This isn't a huge technical innovation, but not having to deal with multiple, overlapping windows works for me. The one thing I miss compared to Windows is "mouseovers" so when the mouse passes over an icon, you see the function spelled out. [Ed. Note: This has been fixed in Spark XL (see sidebar), and will also be included in the next Spark release.]

Spark lets you "build" a database of audio files by adding folders, adding files, etc. So, you can browse your hard drive(s) and create a collection of files pertinent to a particular project. If that's too much work, drag and drop files from the desktop to the waveform editor, or import directly from audio CDs (requires QuickTime 3.0 minimum).

Recording is straightforward, although a bonus is that you can apply up to two plug-ins in the recording path. Note that these plug-ins sit in the chain after the analog-to-digital stage, so a limiter here won't necessarily help prevent clipping. Still, it's cool to be able to EQ or compress with a plug-in during recording. You can also sample rate convert in real time, even on the fly while recording. Destructive editing options include the usual suspects (cut/paste/clear/etc., normalize, fade, change gain, reverse, invert, remove DC offset, convert sample rate, time stretch/pitch shift, and add plug-ins), but the resampling is particularly good, and the stretch/pitch functions are excellent with reasonable changes. The transport works as expected, save the welcome implementation of a smooth jog/shuttle-type control. The only problem is a very *s-l-o-w* rewriting processing with long files, as the program seems to rewrite the entire file and regenerate a new overview, even if you only change a small section.

The playlist is where you collect regions. You can add CD index markers (as well as sub indices), change level, etc. The best feature here is the cut editor, a window that specifies eight transition/crossfade types between playlist entries. The crossfades are all real-time, and have no length limitation other than your file's length. The inability to create your own curves is compensated for by being able to choose either Overlap or Extend Crossfade modes, which place regions on top of each other or extend regions into the crossfade area, respectively. You can also place the index marker at the crossfade's beginning, end, or middle — very handy.

MASTER VIEW/FXMACHINE

The Master View includes several elements.

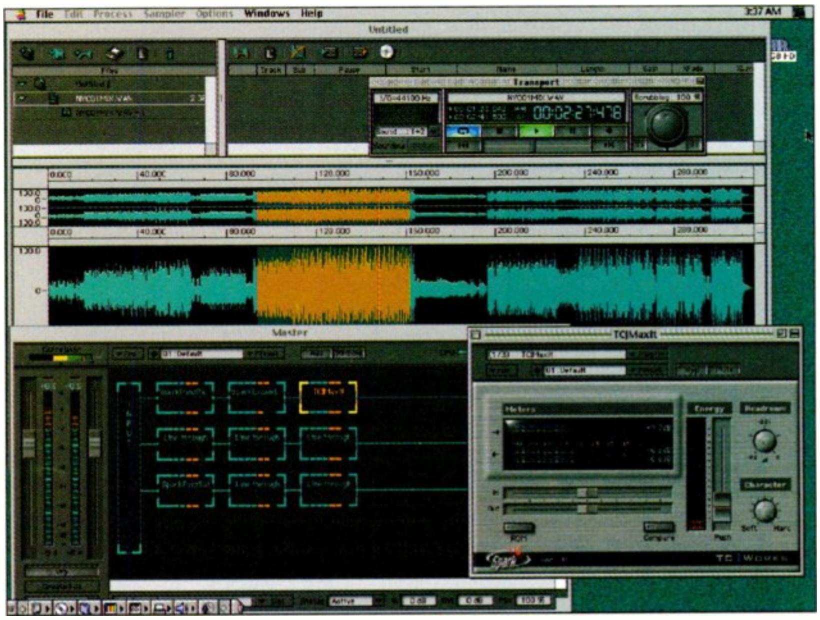


FIGURE 1: Spark adding punch to a remix. Three plug-ins in the top row isolate the bass (about a 60 Hz bandpass filter), expand it like a noise gate, then compress the sound that's left. This gives a percussive, strong pulse. The middle line carries the straight signal in parallel. The bottom row brings in a very tiny amount of distortion. The Browser View is labeled "untitled," and the Master View, "master."

Metering consists of peak level meters and a separate correlator meter that shows phase differences between the two individual channels, from which you can predict mono compatibility and the ability to successfully make vinyl masters (which still matters for DJs, especially with the new do-it-yourself acetate cutter from Vestax). There's no average level metering, though; I'd like an option to see average level, with the peak hold function tracking peaks.

Dither accommodates dithering to 8-, 16-, 20-, or 24-bit resolution.

The *FXmachine* will be familiar to anyone who has used TC Electronic processors. It's a 4 X 5 matrix where you can insert/remove stereo plug-ins. The four ins allow for parallel processing, but you can also split and mix within the matrix for parallel/series effects combinations.

And it gets better: Duplicate a plug-in, including settings, by option-dragging from one position to another. Futhermore, individual level controls and metering for each "slot" ride herd on possible overload situations. You can also mute or bypass slots, and a "live" mode allows for real-time processing through the matrix. Depending on what you're doing and your audio I/O, etc., latency can become an issue here.

Best of all, the FXmachine is itself a VST/MAS-compatible plug-in. So you can even stick an FXmachine collection of plug-ins into a slot in the FXmachine (in practice, though, loading in too many plug-ins will bring your computer to a crawl). When used as a MAS plug-in, you can finally use VST effects within Digital Performer. The manual apologizes for not letting you nest an

continued on page 130

EQ LAB REPORT

MANUFACTURER: TC Works, 742-A Hampshire Rd., Westlake Village, CA 91361. Tel: 805-373-1828. Web: www.tcworks.de.

APPLICATION: Edit and/or master two channels of digital audio, as well as edit samplers and burn CDs.

SUMMARY: An innovative digital audio editor that further revitalizes the Mac as a digital audio editing platform.

MINIMUM SYSTEM REQUIREMENTS: PowerMac with 604/166 MHz, MacOS 8.1, 64 MB RAM, ASIO compatible sound card highly recommended. (Tested with PowerComputing 604e/200 MHz processor and SeaSound Solo interface.)

STRENGTHS: FXmachine is absolutely brilliant, and also works as a MAS or VST plug-in. Elegant interface. Sampler and CD-burning support. Excellent batch processor. Supports MP3 file format. Runs on older 604 machines. Twelve VST plug-ins included. Imports Akai samples from SCSI CD-ROM. Supports Digi Direct I/O and ASIO.

WEAKNESSES: No FFT views, spectrum analysis, or test tone generation. Entire file is rewritten, even with small changes. Supports VST format plug-ins only. No average meter readings, only peak.

PRICE: $499

EQ FREE LIT. #: 107

SPARK XL HITS THE STREETS

Just as this issue was going to press, TC Works released a new version of Spark, Spark XL. *EQ* was fortunate enough to receive the very first copy in the U.S., so that we could bring you this look at the new product.

Spark XL ($699) is aimed at the higher-end of the digital audio editing world. It doesn't replace Spark 1.5, rather it's aimed at a different set of potential users. (Spark owners can upgrade for $199.) Primary among Spark XL's enhancements are support for Digidesign Pro Tools Mix systems, TDM, and two new native restoration plug-ins, DeClicker and DeNoiser. Spark could access Pro Tools systems for audio input and output through Direct I/O drivers previously, but the XL version uses Digidesign's new DirectConnect plug-in. This allows the software to take full advantage of Pro Tools' TDM capabilities — including support for TDM plug-ins. Up to five TDM plug-ins can be used simultaneously, Spark XL routes through them in a serial fashion using a new TDM Master View. As a bonus, you're still able to access Spark's FXmachine while using TDM plug-ins. Yes, that's right, you can run VST and TDM format plug-ins at the same time! In practice, this worked flawlessly for me. However, be aware that Spark XL, when using DirectConnect, requires a whopping total of five DSP chips; this only leaves you with one open chip for running plug-ins on an unexpanded Pro Tools Mix system.

A few other things are included in Spark XL that will be rolled into upcoming versions of Spark, such as support for reading Akai-format CD-ROMs from IDE drives. The lack of "mouseover" labels Craig mentions in his Spark review has also been addressed; now when you roll the mouse over the toolbar, labels pop up to tell you what each thing does.

Since I had only a short time with XL before this issue's deadline, I immediately put the program to work on the very worst-sounding audio I could come up with. I dug out a recording of my first band from back in 1980, recorded live to a low-budget boom box using the built-in mic. We're talking hiss, pops, auto-compression, distortion — yikes! With a minimal amount of effort, I was able to come up with a surprisingly clean, almost-listenable end product. I used a combination of VST and TDM plug-ins to do the work, and I was especially impressed with the new real-time DeNoiser plug-in, which was able to remove a surprising amount of the wash of noise in the recording with minimal damage to the "glorious" music living within it. The DeNoiser works in similar fashion to other noise reduction programs and plug-ins, by taking a "fingerprint" of the noise you want removed. You can dial in the amount of noise reduction you want, "bias" the fingerprint for better results, and set the release time for the processing.

If you're a Pro Tools Mix user, Spark XL is definitely a worthy upgrade to Spark. It's also a strong contender if you're doing restoration work or remastering old recordings. The DeNoiser plug-in operates very effectively, and being able to combine VST and TDM plug-ins gives you a great deal of processing power and flexibility.

As I said in my *Keyboard* magazine review of Spark (June, '00): "If your needs run toward audio file editing, mastering, audio CD burning, or sound design, you owe it to yourself to check out Spark. The program is very capable, and, perhaps more important, very easy to use — clearly the programmers behind it thought about what people would actually be *doing* with it, and how they'd want to work." Extrapolate this statement to include the new features in the XL version, and you have a solid winner. *—Mitch Gallagher*

dbx 386 Dual Vacuum Tube Preamp

Analog and digital outputs combine to make this one versatile unit

BY STEVE LA CERRA

The dbx 386 is a rackmount, two-channel microphone preamp intended for use with both analog and digital recording equipment. In addition to the typical analog audio outputs, the 386 includes both S/PDIF and AES/EBU digital output connectors fed by the unit's built-in analog-to-digital converters. This enables the device to interface directly with a variety of digital audio gear, including multitrack recorders, audio cards/computer interfaces, mixing consoles, and digital effects. Acknowledging the gradual move toward higher-resolution formats, the 386 can output audio with 96, 88.2, 48, or 44.1 kHz sample rates, at 16-, 20-, and 24-bit word lengths. A-to-D conversion is at 96 kHz, using dbx's proprietary Type IV Conversion System. Essentially, Type IV Conversion "stretches" the top few dB of the analog-to-digital converter's dynamic range, creating a region of extended headroom. This prevents high-level signals from producing distortion during the conversion process, and allows a user to safely hit the input with more signal level than would be possible without Type IV.

OVERVIEW

Along with the more familiar analog controls such as gain, mic/line switch, +48-volt phantom power on/off, 20 dB pad, phase (polarity) reverse, and 75 Hz highpass filter, the 386's front panel has push buttons for dither, noise shaping, sample rate, word length, and output format. Separate rotary pots for analog and digital output level are provided, with a corresponding 12-segment meter that can be toggled between analog (dBu) and digital (dBFS) output level scales. Vents in the front panel reveal the 12AU7 vacuum tubes employed in the input stage of each channel; 1/4-inch TRS connectors are provided for use with instrument-level signals. (Note that this instrument input has priority over the rear-panel 1/4-inch input.)

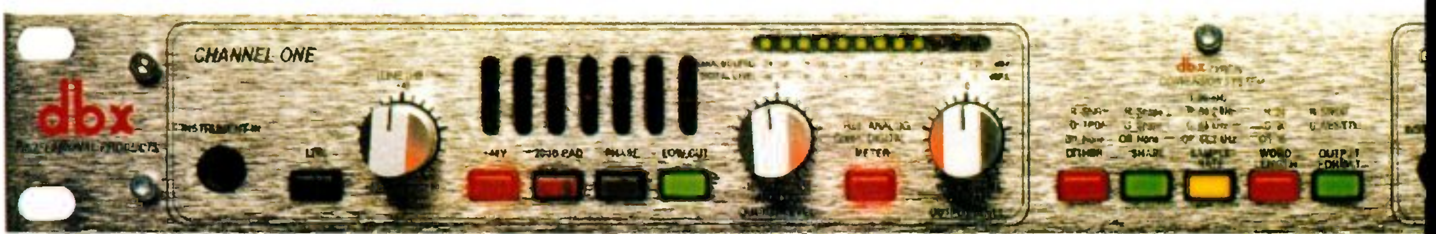
Rear-panel connections for each channel include balanced XLR mic and 1/4-inch line inputs, balanced XLR and 1/4-inch TRS line outputs, and unbalanced TRS inserts (tip = send, ring = return). Any of the balanced TRS jacks may be used for unbalanced operation simply by inserting a TS connector. Also on the rear panel of the unit are the aforementioned S/PDIF and AES/EBU connectors, BNC word clock I/O, a power switch, and an IEC power receptacle. dbx recommends leaving a centimeter of space above and below the 386 when the unit is rack-mounted. We found this to be good advice, as the unit did run rather hot.

IN USE

We took the dbx 386 for a spin on sessions, using it for miking acoustic instruments as well as DI'ing synthesizers. Recordings were made via both the analog and the digital outputs of the 386. For some of the digital recordings, the 386 slaved to the external word clock of our Yamaha 02R, while, on others, the 386 served as the clock master. Either clocking arrangement worked flawlessly. Analog outputs were used to route signal directly into the 02R or TASCAM DA-98/88/38 tape machines.

The 386 provides both dither and noise-shaping for use when reducing the bit depth (word length) of digital signals. These two circuits are active only on the digital output. Three settings are available for the Dither push button switch: off (the switch is unlit), TPDF (the switch lights green), and SNR2 (the switch turns red). These cryptic labels refer to dither algorithms, though nowhere in the manual could we find what they actually stand for. As you might expect, dither was most useful during 16-bit operation. We preferred the SNR2 algorithm for reducing background noise as compared to the off or TPDF (which we found actually increased background noise) settings, although the effect of the SNR2 setting was extremely subtle. When using the preamp with 20- or 24-bit word depth, we found dither unnecessary.

Much more effective were the 386's psychoacoustic noise shaping curves. Labeled simply Shape 1 and Shape 2, engaging these functions was like pressing a magic button — making hiss almost completely disappear and allowing audio to emerge from an extremely quiet background. It was particularly effective when using the front-panel instrument input for synths, where it seemed to mask some inherent noise from the output of older analog synths (an Oberheim Matrix 6, for example). Perhaps it's the way the 386's instrument input loads the output from a synth, but the bottom line is that it just plain sounded good.



EQ LAB REPORT

MANUFACTURER: dbx Professional Products, 8760 South Sandy Parkway, Sandy, Utah 84070. Tel: 801-568-7660. Fax: 801-568-7662. Web: www.dbxpro.com.

SUMMARY: Two-channel vacuum tube mic pre with analog and digital audio outputs.

APPLICATIONS: Analog or digital multitrack recording, front-end for computer workstations, "upgrade" analog-to-digital converter for DAT machines.

STRENGTHS: Supports 44.1, 48, 88.2, and 96 kHz sample rates, as well as 16-, 20-, and 24-bit word lengths. Balanced analog I/O. Front-panel instrument input. S/PDIF and AES/EBU digital outputs. Separate level control for digital output.

WEAKNESSES: May be too noisy for critical recording. Performance of the two channels is not consistent.

PRICE: $599.95 **EQ FREE LIT. #:** 108

Unfortunately, there's more than a hint of noise (hiss) present in the preamp's output. As gain is increased, the amount of noise being generated at the analog stage is enough to diminish the benefits of both noise-shaping and dithering.

When set to 16-bit/44.1 kHz resolution, the amount of noise present at the analog line output was only slightly higher than the noise present at the digital output, which is good, considering there's an extra amplifier stage in the analog path. With word depth set at 20- or 24-bit, the digital output won the "quiet" contest hands-down. Both the analog and digital outputs were relatively colorless, with barely a hint of that "tube sound."

The performance of the 386's two channels was a bit inconsistent. For example, we found the output of channel two to be noisier than that of channel one. Specifically, a low-level 60-Hz hum was present in both the analog and digital outputs of channel two. Under most circumstances, it wasn't an issue, but when we used a Beyer MCE82 stereo mic to record a finger-picked guitar, the hum became apparent. Also, on our unit, the results of the Shape and Dither switches were much more readily apparent on channel two than channel one.

The rear-panel inserts are an intelligent addition to the 386, being invaluable for processing signals with compression or EQ. Since the inserts are pre-A/D converter, we were easily able to insert the requisite effect while maintaining our digital connection from the 386 to the destination device.

One of the 386's great strengths is as a "front end" to a computer-based workstation. Particularly in cases where the audio interface lives in the computer, the dbx unit can make a drastic improvement over both the clarity and noise level in the system. Removing the analog-to-digital conversion process from the computer's housing makes a big difference in the noise floor, and the 386's headroom is better than anything you're likely to find in a PCI slot. We also found that the 386 makes a nice front-end for DAT machines with inferior (or older) analog-to-digital converters, allowing it to do double duty during mixdown.

CONCLUSIONS

dbx has done an intelligent job constructing a useful preamp, although the unit could perform somewhat better from a noise standpoint. It may not appeal to "high-enders" who already have $5,000 invested in a set of preamps or converters, but for the home or project studio looking to take a step up, it could prove a useful addition. EQ

Internet Update

PHOTO BY STEVE JENNINGS

Something old, something new, something borrowed, something black and blue...

BY ALLEN WHITMAN AND JON LUINI

Ceviche, gaspacho, and consommé, like revenge, are best served cold. Microsoft is now, officially, a monopoly. There's a shocker. They won't go easy, though. "We're in this legal battle for the long term," says MS CEO Steve Ballmer, assuring the continued existence of half the lawyers in the country. Anyone who heard Ballmer's defiant posturing couldn't help but notice his voice is a dead ringer for Loony Tunes' Marvin the Martian, with the animated broomstick headpiece, Roman Centurion garb, and comments such as: "I'm here to destroy the Earth with my Acme Disintegrator!" Coincidence?

But there's a silver lining to this *pro forma* posturing. The technology could see improvements as each individual spin-off unit of Microsoft must actually compete in the marketplace. Also, oddly, stockholders may possibly see gain as the spin-offs gain strength. But enough about matters over which we have no control. Let's get to matters that have some relevance.

NEW BROWSERS

Netscape 6 beta (http://home.netscape.com/browsers/6/) and Microsoft's Internet Explorer browsers (MSIE 5) for Mac (at last!) (www.microsoft.com/downloads/) are available for download. The new IE is, atypically, a decent Microsoft product for the Mac. It displays Web pages as they were designed to look (and how they do on other platforms), a nice change from the unreadable text we've had to deal in the past. It's always good to have the latest browser so sites will appear as intended. However, our first reaction to the Netscape 6 beta left us a bit flat in terms of the user interface. The FezGuys recommend waiting for the real release.

NEW IUMA ARTIST UPLINK

The Internet Underground Music Archive (IUMA) has new, redesigned features for its Artist Uplink offering. The administration area has been expanded to let users choose where to have IUMA send the checks (IUMA shares advertising revenues with bands). Other new features include quicker first-time set up and detailed reports of statistics broken out with page views, MP3 downloads, RealAudio files streamed, and ad revenue accumulated. Though the site looks good, the page layout was a bit off under IE 5 for the Mac. Functionality wasn't affected, however. (www.iuma.com)

EARJAM

This new MP3 player/encoder is approaching full release and, for a week during April, Earjam gave away free copies to users reporting verified bugs. Bug reporters were also entered into a contest for a free Rio 500 MP3 portable. The FezGuys applaud Earjam's fearless tapping of the online music community to help them develop a solid product. Perhaps the new Internet software model will be taking feature requests from users and awarding stock to those who suggest ideas they implement. Hey, weirder things have happened. (www.earjam.com)

EMUSIC

Here's the stats: Emusic has announced selling over one million MP3 songfiles since their launch in 1998. This incorporates single-track sales as well as tracks included as part of albums and special collections. The site boasts more than 100,000 MP3s for sale from over 600 independent record labels and popular artists.

Here's what it means: People are paying money for online music. This is good. Audiophiles are not happy, however, viewing MP3 songfiles as low-fidelity audio. It's true. MP3 songfiles *are* low-fidelity audio. MP3 was never intended to be used as a retail delivery format. But the ever-egalitarian FezGuys figure the more people understand the value of music separate and apart from the physical product the better. Get used to songfiles flying around! It's imperative that people understand what they're buying. It's not the plastic disc! Despite the questionable legalities of MP3.com's "MyMP3" service, this partnership with physical CD distribution is an appropriate combination of paying for music once and having it in multiple formats.

Perhaps in the near future we'll purchase an album online and instantly get streaming MP3 versions for immediate listening while overnight the full 44.1 kHz files download to our desktop for us to burn our own CDs. Maybe a band logo coffee mug and signed poster get put into the mail to arrive the next day. Wouldn't that be nice?

NAPSTER

Napster (www.napster.com), for anyone hiding under a rock for the past six months, is *the* hot ticket for MP3 use. This freely available tool for sharing MP3 music files with other online users even has a chat system. Now that's community-building on the Internet! The problem, of course, is that some people upload their entire music collection for all to have without the artists' (or the record labels') consents. The FezGuys like free music (even bad music is forgivable if it's free), but we're a touch old-fashioned on this one. We think the artist should be the one to decide. Napster is currently being sued by the whole music industry: the RIAA, the labels, the artists, even our cat. The outcome is uncertain. As if that weren't enough, Napster has also pissed off U.S.

colleges whose networks have been crippled by increased MP3 songfile traffic as a result of the popularity of their software. While legal issues continue in a spinning, sucking vortex, plucky students at Indiana University built a clever workaround. The fix, planned to be implemented at other college campuses where Napster is currently banned, reduces the impact Napster has over a network. At SXSW in March, Napster hinted at future business plans that would alter their status as the latest music industry whipping boy (having temporarily dethroned MP3.com). Napster execs (the ones over 21) said the company planned to add support for secure music files and (gasp) hope to work with the music industry to ensure artists are paid for traded music within Napster's network. For the moment, the embattled company is reaching out to independent musicians, encouraging them to download Napster, share their band's songs, and post a nice Napster logo on their Web site. Nothing like free viral marketing!

In related news (showing what can happen when a small company is eaten by a large corpo-entity), recent AOL acquisition Nullsoft (makers of the popular MP3 player Winamp) leaked out a beta version of a Napster clone called Gnutella last March. Unfortunately, that news sprinted up the chain of command to AOL execs (amidst planning logistics to acquire Time Warner) who promptly put a jackbooted foot down. But, of course, this is the Internet. It's impossible to do recalls. Currently, the software appears to have found a new home online at (www.gnutella.wego.com). One key difference between Gnutella and Napster is that Gnutella shares files of all types. This distances Gnutella from a purely musical focus. Now the issue becomes censorship instead of outright piracy. Gnutella also is a friend to UNIX folks, with new versions planned for Linux, FreeBSD, and others. Plus, somebody at the site has a sense of humor. Very refreshing.

Until Napster releases their Mac version, the common Mac Napster clone is Macster (www.macster.com). One thing we aren't too grinny with is Napster's lame slogan: "Music At Internet Speed." That doesn't say too much to people on 28.8k modems!

MORE PLACES TO UPLOAD YOUR MUSIC

Fortunecity (www.flynote.com) — Flynote.com's beta site does not, thankfully, claim to be your Internet Label or Industry Daddy Warbucks. Billed as a place to "Store Your Music Online," the site has a very easy registration process (they don't ask for your phone number!), but then throws us into the larger "community" of fortunecity.com. Fortunecity.com requires a much more lengthy registration process that effectively triples the amount of time it took to sign on to flynote.com. What a waste. There are pages and pages of "offers" to click "no" on and, like a multi-level warfare game, you dodge spam ordinance like bullets. There is some music sorted by a few genres (*i.e.*, the "Electronic" category has, as of this writing, only four songs).

Ease of Use — Simple and tedious
Design — Inoffensive
Tech Support — Undetermined
Expected User Experience — "Great! I have a 100 MB free storage space!"
Overall — "Free storage space!"

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MOTU 1296

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(128 MB preferred), Windows 95 or Windows 98, and a 9 GB hard drive. The 1296 interface is 100-percent compatible with VST, MAS, Premiere, DirectX, and all other host-based native plug-ins supported by popular audio software. MOTU is shipping the 1296 in the second quarter of this year at a suggested retail price of $2,095 for the 1296 core system (includes the PCI-324 card and AudioDesk workstation software) and $1,795 for the 1296 expansion I/O (adds 96 kHz recording to any MOTU system).

For more information, contact Mark Of The Unicorn at Tel: 617-576-2760. Fax: 617-576-3609. Visit their Web site at www.motu.com.

WALTER AFANASIEFF

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takes that the soul is completely taken out of it, because you're striving for a better intonation. They're just going to be rolling their eyes and playing you something that's very in tune — the intonation's there, but the soul has been taken out of it. That goes without saying for everything; it's just a matter of being able to rightfully compromise and juggle all the human and technical aspects and not make it a stale thing. Again, if it's a good song, it's in the song. At the end of the day, I really just believe in the song.

But not every song is a great song, and every album is going to contain some songs that are B-list and some that are A-list.

Of course, but that whole argument of what's an A-side and what's a B-side is just record company language — B-sides are just as important to me. I don't know why and how it was mandated that [to be played] on the radio, a record needs to be this type of song and only a certain length, that whole thing. I still remember a world where we were all driving around and "Stairway To Heaven" came on the radio. Or "American Pie," or "Inna Gadda Da Vida," or "Freebird." Come on, those are all seven-, ten-, twelve-minute songs. These days, if you go past three minutes, thirty seconds, forget it.

And all of those songs you mentioned have a great feel, but are technically lacking.

All of them are completely technically lacking. That's what I'm saying — it's just the song. You can't play those songs any better because it's the song.

Any advice you'd like to pass on to the reader who wants to be the next Walter Afanasieff?

Well, I'm just a musician who likes to talk a little bit about what I do and maybe set people into being a little more forgiving and believing. That's the thing: Just believe in yourself. Just really know that there's a place for everyone and everything here. EQ

TODD RUNDGREN

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The economics of it are much more favorable to the artist, as opposed to getting your underwriting from a record company at 90 percent deduction rates. You get it from the fans, and the cost of actually making disks and sending them, when you know exactly how many to make, is really low — less than a dollar a CD in any appreciable volume. The fans are paying the shipping and handling, so it works out better for both. They pay shipping and handling, plus $25 to get four CDs, so it works out to a grand total of about 40 bucks. Some of them are CD magazines, but the final one is a full CD with the artwork and everything.

Artwork. Do you miss those old record covers?

Everybody misses them, and, since the labels decided to scrunch everything down to microscopic print, I think they should include a magnifying glass with the CD. How about a jewel box with the magnifier built in? It seems like it's gotten to the point where they aren't paying attention. They lay it out large, and when it comes back from the printers it's so damn freakin' small, I don't think any normal person could read it without taking it out under the noonday sun.

What's the biggest mistake of your life?

[*Laughs.*] Well, as I consider pride a sin, self-importance is not an indulgence I care to get into. I do imagine myself in a certain way — not to say I am not vain — I imagine my best self, how I'd like to be and where I'd like to be, and what I'd like to become. But in terms of thinking that any single act of mine is that important or unredeemable, I don't have that interest. There is no moment in my past that sticks out: "Gee, I wish I had that to do all over again, with all the karmic repercussions that would entail." I'm at a point where I feel I have a pretty good handle on how life works — for me. You can only say "for me." Others might say, well, gee, how did you do that? And I say, all you have to do is think this way. That's the hardest thing for anybody else to do — to change the way they think.

The most important thing to me has always been to satisfy myself and, therefore, you can't really have a lot of regrets. I'm going to satisfy myself, and know when I'm satisfied, and I can't have any complaints later. And I really don't blame anyone else for the way my life has turned out. EQ

FURMAN SRM-80

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turned down, the volume faded in and out smoothly as expected.

A third thing is affected by the trim levels: Switching between speakers that have radically different trim levels can occasionally result in a small pop in the audio output. By balancing the input sensitivity controls of my monitors against the SRM-80's trim levels, I was able to almost completely eliminate this problem.

TURN IT UP

Once I had everything wired up properly and had figured out how to best set the speaker trim controls, the SRM-80 was a joy to use. The sound quality was high, at least as good as the mixer I was previously using (and those that I've seen many others use) to control outputs levels and routings. Its selection of line-level connections matched well with the gear I needed to connect and switch between, and its dubbing/source selection capabilities worked flawlessly. I would prefer more balanced inputs and powered speaker outputs rather than the combination of two passive and one powered out, but your needs may differ.

This points out the main factor in evaluating whether the SRM-80 meets the requirements of your studio: Does it have the I/O you need, both line-level and speaker-level? If it does, then go for it; you'll find it performs very well. If not, then you may be better off staying with your current mixer or monitor control system while you search for another solution (but good luck finding a similar box in anything resembling this price range). You could put together a system of patchbays, speaker switches, and a volume-pot-in-a-box, but it wouldn't be as foolproof or as cool to use as the SRM-80. All in all, the SRM-80 is a handy necessary tool for today's DAW-based studio. EQ

EVENT PS6 MONITORS

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WELL?

The array of powered compact nearfield monitors we have to choose from these days is approaching the ridiculous. Event Electronics alone offers six models to choose from; combine this with the number of manufacturers flooding the shelves with speakers and you've got a devastating case of Option Anxiety Syndrome just waiting to happen (and there's no vaccine).

Having said that, the PS6's distinguish themselves with a great combination of power, sound, size, and price point. For stereo nearfield monitoring they excel, comparing favorably to monitors costing significantly more. Looking for an affordable surround mixing monitor solution? You would do well to consider five PS6's and a subwoofer; for $2,000 to $2,500 (or so) you're set.

In short, Event has another winner here. The PS6's easily live up to their predecessors, and also set the bar up another notch for affordable powered nearfield monitoring solutions. EQ

FEZGUYS

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Kanoodle (www.kanoodle.com/mp3/) — Kanoodle's chirpy tone fits well with their message. They want to host your music and they want to sell and promote it, too. "It's Your Music. Shouldn't You Make All The Money?" boasts the tagline at the top of the front page. There is a regular upload section (create your own genre), a special Lend A Hand Program (LAHP) for the technically challenged ($15), and a CD production and sales area called Music Production System (MPS). MPS creates individual, one-off CDs of your MP3 files to sell to prospective listeners. You set the price and you get all the money. That's what we might call "raising the bar" for other Web sites! Innovative Marketing Solutions, Inc. of New York is the parent company, and they apologize for the legal contract and blame the lawyers. The contract is thankfully short, but, when viewed in Netscape, is set in a frame requiring scrolling up, down, left, and right to read the damn thing. IE5 does not do this. We copied and pasted it into our own document to read it. Overall, the site is simple and straightforward.
Ease of Use — Simple
Design — The monthly bill from the power company
Tech Support — Undetermined
Expected User Experience — Typical of the genre
Overall — Recommended

ASCAP AND BMI GROK THE WEB

The two behemoth performing rights collection agencies, though still plodding along in the traditional world of inaccurate and inefficient payouts, are moving forward in the online world. By taking the first baby steps toward automated reporting of Webcasted music, it's more likely that smaller bands who typically don't show up in a traditional radio station's top-ten list may some day actually see some revenue from online plays. Here's hoping....

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TC WORKS SPARK

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FXmachine as a plug-in within an FXmachine that's also a plug-in, but c'mon — how many plug-ins do you really need?

Twelve VST plug-ins come with Spark, including a compressor/limiter, level "maximizer," downward expander, reverb (better than you'd expect given its "freebie" status — TC tells us that the same algorithm is used as in their Native Reverb plug-in), stereo delay, hi/low cut filter, bandpass filter, single- and 3-band parametric EQ (with hi/low shelving and notch), high-pass/low-pass synth filter with resonance, distortion, and Grainalyzer (a sort of "lo-fi" downsampler). The plug-ins mercifully avoid baroque interfaces, giving you fast access to parameters, while not taking up too much screen space. TC Works has even come up with a way to allow direct numerical input of VST parameter values — a real time-saver in certain situations. The plug-ins range from basic to creative. (The tape delay, resonant filter, distortion, and Grainalyzer units are particularly useful for sound design.) For mastering, the TC Native CL plug-in provides a compressor/limiter with a Level Histogram feature, which gives you an overview of the dynamics processing before/after.

Also remember the power of series/parallel processing. For example, if you want a multi-band compressor, you can load four bandpass filters into the four audio streams, then compress each band individually.

Spark burns CDs by communicating its playlist to Adaptec's Toast (included) and creating a CD image that contains all the crossfades, level changes, and so on, defined in the playlist. Creating a CD image is a more bulletproof way to create CDs than doing them on the fly, but requires enough disk space to hold everything that will be on the CD *and* a copy of same. The bundled version of Toast is full-featured, so it's good for more than just burning audio CDs (*e.g.*, data backup).

There's also a batch converter, which is great for, say, taking all those cool field samples recorded on DAT at 48 kHz and converting them to 44.1 kHz while you normalize them, remove DC offset, and apply some compression from a VST plug-in. It does format translation among WAV, AIFF, MP3 (using the Fraunhofer codec), and Sound Designer II formats — although when I tried to import a WAV file created on a Windows machine, I had to rename the file type with Disktop. Once the computer could recognize it, the file loaded without a hitch.

The best batch converter feature is the ability to audition each converted file, A/B it with the original file, and reset the converted file back to its unconverted form if there's some problem — very cool.

To edit the audio from QuickTime movies, just import them; Spark splits off the audio and shows the video in a separate window. Edit the audio, then export to meld the audio and video back together again into a new movie. There's sampler support, too — Akai, Emu, Kurzweil, Roland, Yamaha, and SDS/SMDI.

Unfortunately, my only remaining samplers (I do most of that stuff in a computer now) aren't supported by the program, so I can't comment on the effectiveness. You can also import Akai CD-ROM samples from SCSI CD drives. Of course, the imported file doesn't retain any tuning offsets that might be part of a patch within an Akai sampler.

So how does Spark compare to Peak? You'll just have to wait for my upcoming Peak review, but suffice it to say that Spark definitely delivers the goods. And honestly, the comparison may not be as cut and dried as you might think. There are some big differences in the way the two products are aimed — Spark definitely goes for the project-level mastering approach with lots of real-time processing, and it also has a very different approach to the user interface issue.

Speaking of user interfaces, Spark's single-window approach is quite effective. The overall feel of the controls is very responsive, which makes it all the more unfortunate that matters bog down whenever you have to deal with saving or modifying long files. On the other hand, you have moments of flat-out brilliance, such as the FXmachine.

In the long run, I suspect that Mac owners can finally look forward to enjoying the fruits of the same kind of healthy competition that has led to major improvements in Windows editing programs. EQ

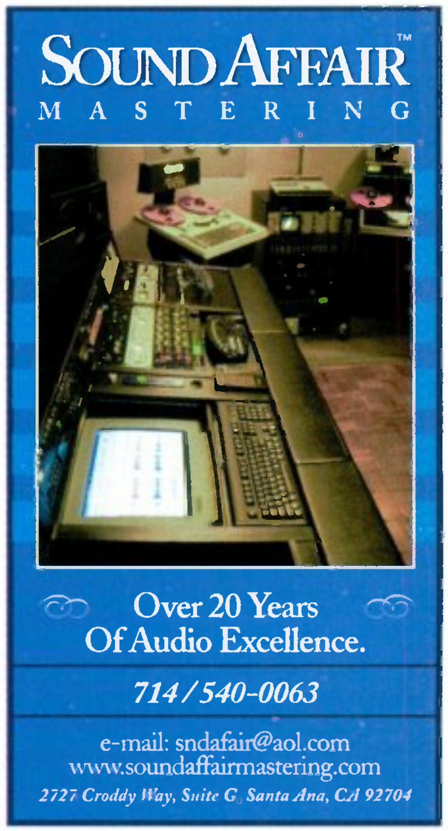
CIRCLE 53 ON FREE INFO CARD

CIRCLE 51 ON FREE INFO CARD

CIRCLE 50 ON FREE INFO CARD

CIRCLE 55 ON FREE INFO CARD

CIRCLE 56 ON FREE INFO CARD



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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

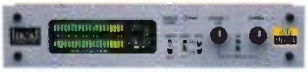
The high-end quality analog to digital solution for the project studio. With support for both professional and consumer digital formats you can now record your audio at a higher resolution and with greater detail than standard converters found on MDM's, DAT's and DAW's. Ideal for mastering or tracking.

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

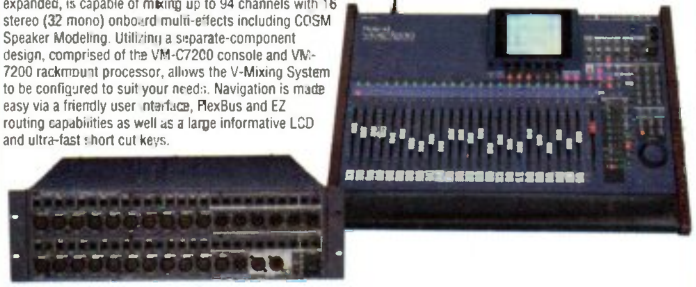


Transparent analog to digital conversion designed to bring your music to the next level. XLR balanced inputs feed true 24-bit converters for revealing all the detail of the analog source. 16-bit masters can take advantage of the AD9624's noise shaping function which enhances clarity of low level signals.

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 COSM 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 processes using optional VS8F-2 Effects Expansion Boards (2 stereo effects processors standard)
- COSM 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 COSM 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
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- 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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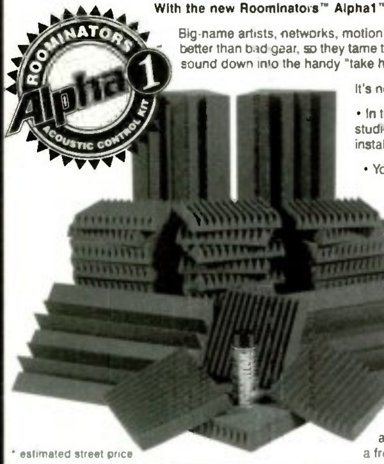

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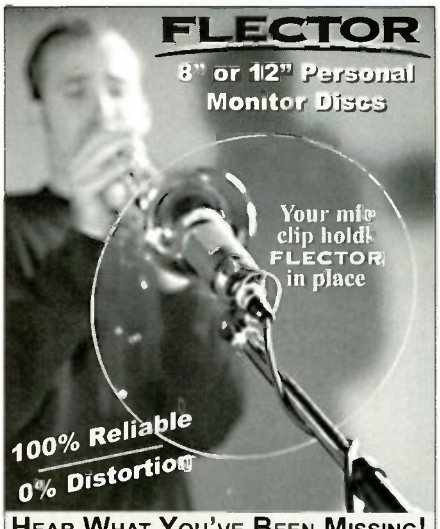

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ACROSS THE BOARD

continued from page 146

movements in this area take advantage of the smaller 0.1 dB steps, and make riding the vocal much more accurate.

Small format digital consoles use larger steps for automation data. The typical value is 512 steps. This means that, at the most sensitive fader position, the smallest step will be about 0.2 dB. As you move the fader down, the steps quickly change to 0.4 dB then 0.8 dB and 1.2 dB. You can quickly realize the benefits of 1,024-step faders.

Pro Tools uses 0.1 dB steps for automation, and so do most of the DAW-type systems. It is actually easier in a DAW because you are just dealing with computer information and can choose any step value you want. The value just becomes a multiply value stuffed into a DSP to get the levels you want. The Pro Control and the HUI provide the same resolution of control.

If you must mix by using the mouse, here is a tip: Most people set the mouse sensitivity so that the mouse pointer moves a lot with a small physical movement of the mouse. If you try mousing a fader, you will see big jumps in the level corresponding to the coarseness of the mouse setting. If you set the mouse to the slowest response, you can more easily control the fine movements of the fader in 0.1 dB steps.

FADER QUALITY

The high-quality automated Penny & Giles faders used in large-format consoles cost about $1,000 each. They have very smooth and silent audio qualities and are touch-sensitive. I mean *really* touch-sensitive. The conductive fader knobs are electrically connected to a circuit that detects the change in capacitance caused by the touch of a finger. Without moving the fader, an LED will light when the fader is touched.

Part of the cost of the fader is the accuracy of the resistive element that passes audio in an analog console. Digital consoles do not pass audio through the fader, so this element can be removed. No signal actually passes through the fader in a digital console. There is only a motor to move the fader and a resistive ladder that sends the current position of the fader to the automation. The level of the audio is changed in the DSP. It doesn't really matter whether it is a physical fader or a picture of a fader on a screen.

Small-format digital consoles use a different method for touch sensitivity, which allows for very inexpensive faders. The automation continuously looks at the position sent by the fader. If the position of the fader is different than where the automation set the fader, then it must have been moved. Notice I said "moved." The fader must be moved. The automation does not actually sense that the fader was touched. The fader movement sensing is usually set to trigger if the fader was moved two fader increments to avoid false movement indications. This means that, if you are going to make a small ride in a vocal track, you must move the fader by two steps (0.4 dB in the sweet spot) before the automation knows about the move. This means that the initial movement will be late (it won't be detected until you have already been moving the fader for awhile) and the change in level may be larger than you want it to be. I always try to move the fader in a hole between lines of the vocal to get the automation's attention so that automation data is being written before the spot that I actually want to ride.

Now you know everything about console faders. Next month we will cover the neutron capture cross sections of transuranium elements and why the increments are measured in Barns. EQ

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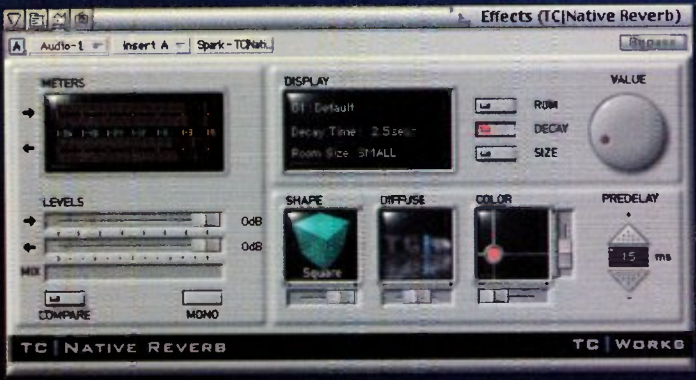

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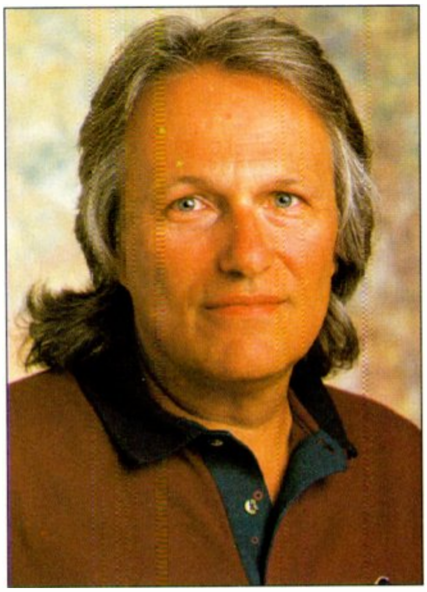

Everything you wanted to know about console faders — both real and virtual

BY ROGER NICHOLS

For some reason it is hard to control delicate computer functions with a mouse. The mouse is fine for pointing and clicking, but when the task at hand requires fine continuous movements, the mouse just rolls over and plays dead.

Have you ever tried drawing a picture on your computer screen with a mouse? The pointer jumps because of the inaccurate roller ball interface, and the scale of what you see on the screen does not easily translate to the amount you are moving the mouse, especially when you run out of mouse pad and have to pick up the mouse and move it in the middle of a stroke. This is why artists and designers use tablets to input drawings into the computer. The movement of the hand and stylus are more natural and are easier to translate into computer vectors.

The same thing happens when trying to mix audio with a mouse. You click on a screen fader and move the mouse. If you move the mouse one inch forward, the fader moves some amount on the screen. Because of mouse stutter, if you move the mouse back the exact same inch, the fader on the screen will not be back exactly where it was when you started. A physical fader that you can move up and down is much easier to use for riding vocals and other gain-varying instruments. When I want to ride something up and then bring it exactly back to where it was, I place my finger below the knob as a stop. I can ride the fader up, and then snap it back to my finger and the level drops back to exactly where I want it to be. You just can't do that with a mouse.

Because of mouse stutter, if you move the mouse back the exact same inch, the fader on the screen will not be back exactly where it was when you started.

Now that we have determined that nothing is better than a real fader, we should talk about the resolution of the real fader. It has already been determined that 100 mm is a good length for a straight-line fader. If you lean your forearm on the armrest at the front of the console, the range of movement in your wrist is comfortably 100 mm. You can move the fader from the full off position to the loudest position without lifting your arm from the armrest. The highest resolution available in the fader movement is at the point where the wrist is relaxed. At this point, you also have the most amount of control over small movements in your wrist.

When console automation came on the scene in the early '70s with the Allison 56k, the range of fader movement had to be quantified into discrete steps that could be detected and reproduced by the automation computer. After subsequent generations of console automation, and input from scores of automation users, a value was reached of 1,024 steps to represent the position of an analog fader. At the "sweet spot" (about 25 percent down from full fader level) one step in the fader movement corresponded to 0.1 dB of gain change.

One dB is defined as the smallest amount of change in level detectable by the human ear. This is based on the level change of the entire sound that is being heard. If you change the level of the mix, or the level of a soloed vocal, this definition holds true. If, however, you have a reference to compare the level with, then 0.1 dB change in level is readily apparent. If you are comparing the input to the output of a DAT machine, you can easily tell if the levels are 0.1 dB different. If you are listening to relative balances in a horn section, it is easy to hear the difference when one horn is turned up 0.1 dB.

One more factor comes into play with audio faders: the logarithmic taper of the audio fader. As the fader is pulled down toward the off position, the amount of attenuation increases rapidly for the same amount of movement of the fader. A half-inch movement at the top of the fader is approximately 5 dB. At the bottom third of the fader, the attenuation is 20 dB for that half-inch. The next half-inch is 50 dB.

If you recorded the vocal much louder than all of the other instruments, then the fader position will be much lower in the final mix situation. Because of the taper of the fader, small physical moves will translate into larger changes in level. If the moves are automated, the steps will be 0.2 dB or greater on our 1,024-step faders. If this happens, use the line trim to lower the level of the vocal so that the fader can be moved up into the "sweet spot" of the fader, or up near the top 25 percent of the travel. The sweet spot is usually marked as zero on the fader. Levels above this reference will be marked as +5 and +10, while markings below this level will be –5, –10, –20, and –40, down to infinity. Any fader

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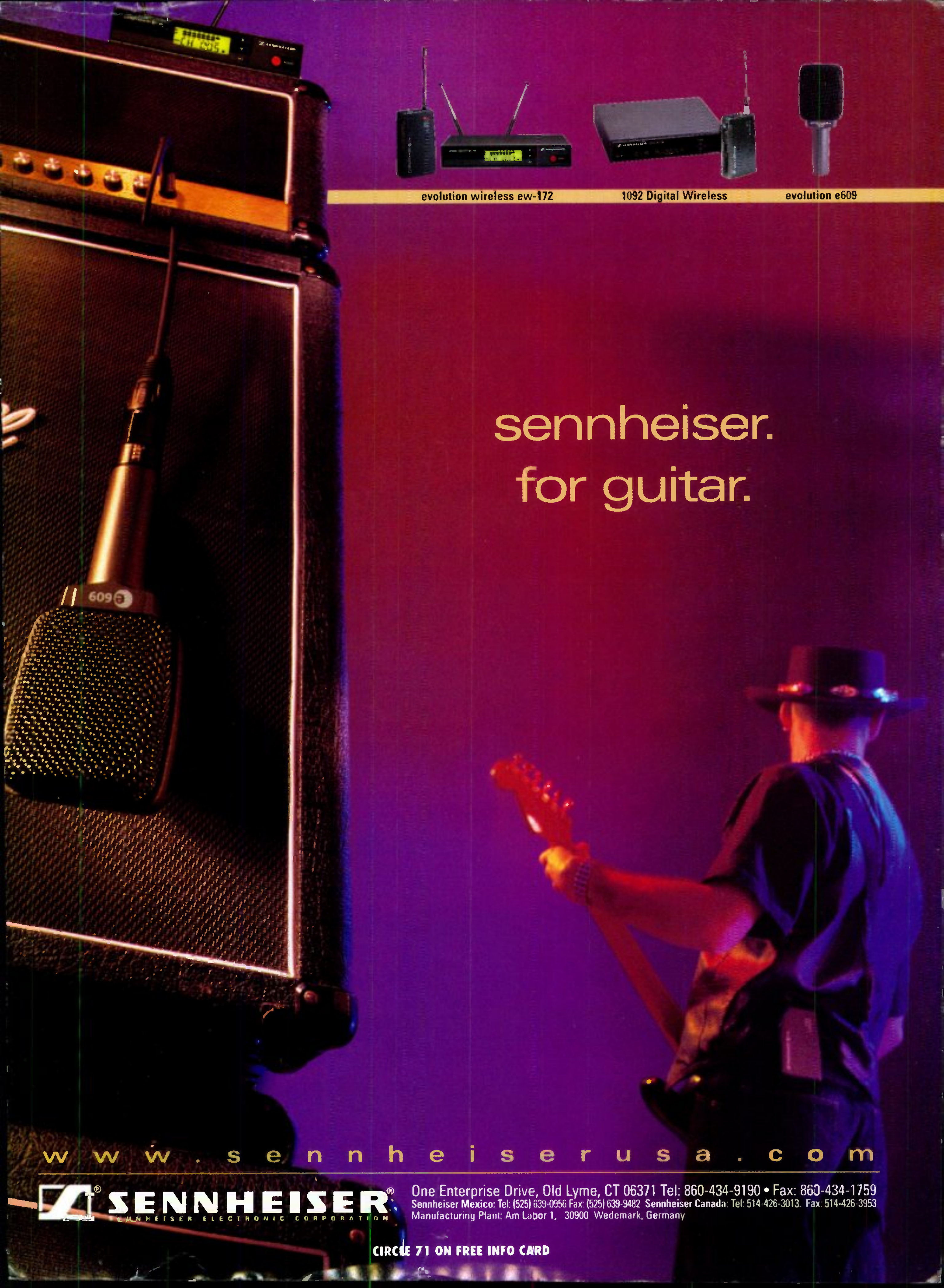
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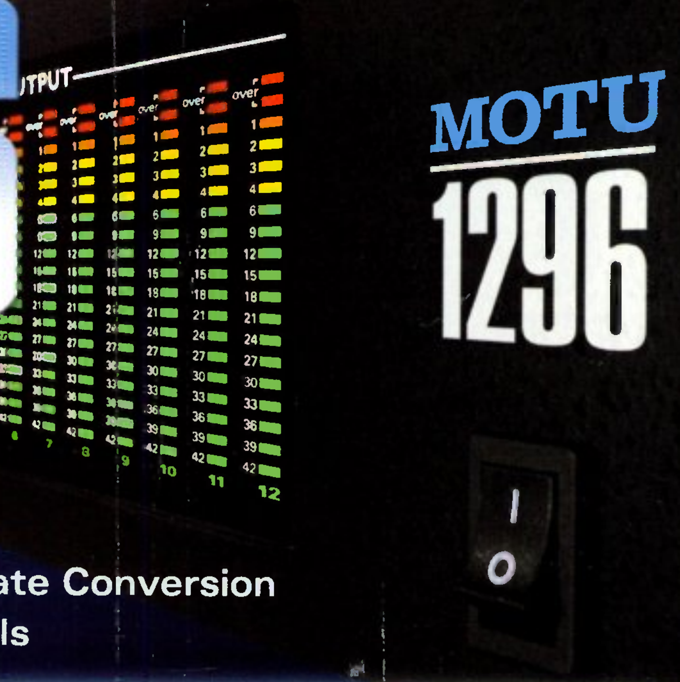

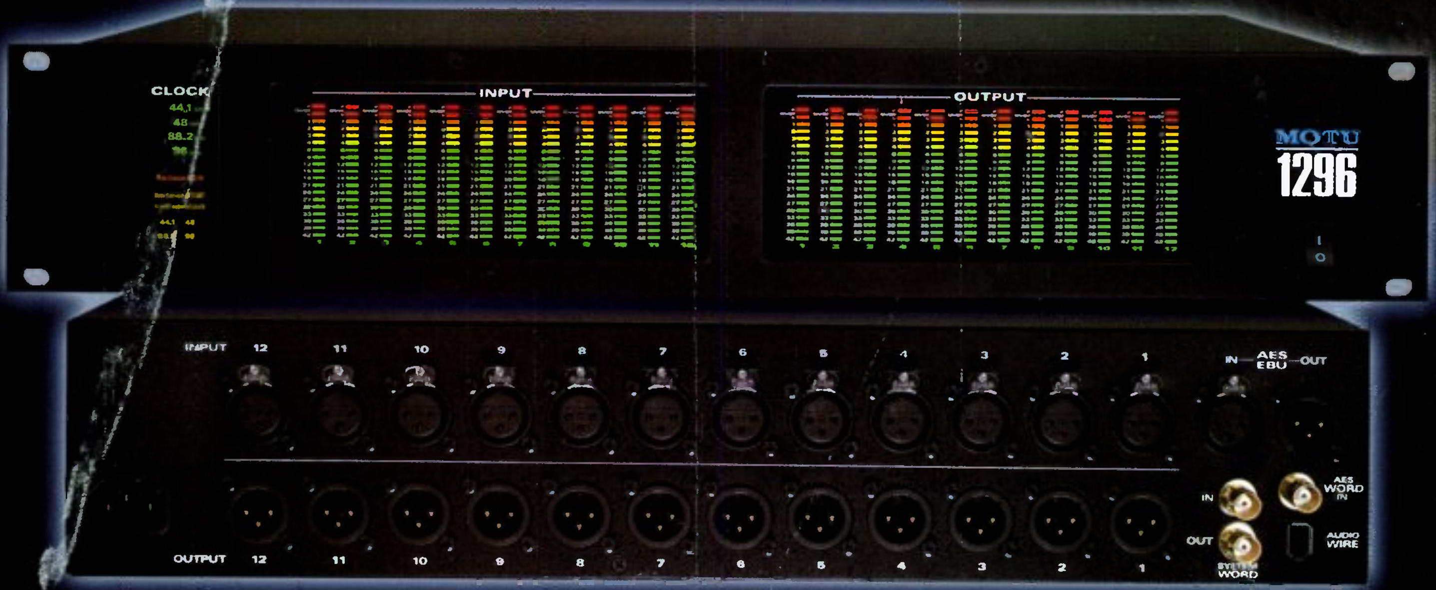

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