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AUGUST • 1999

EQ

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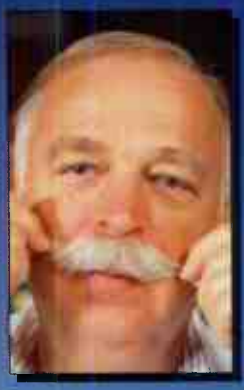


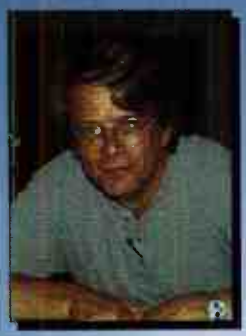




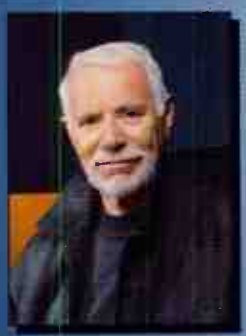
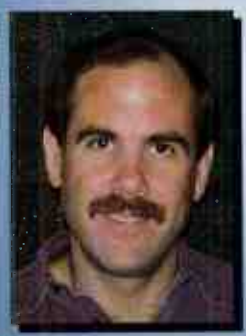




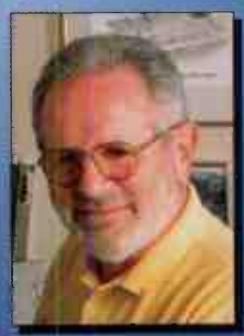

Recording in the New... millennium

Project pros look toward the future of recording

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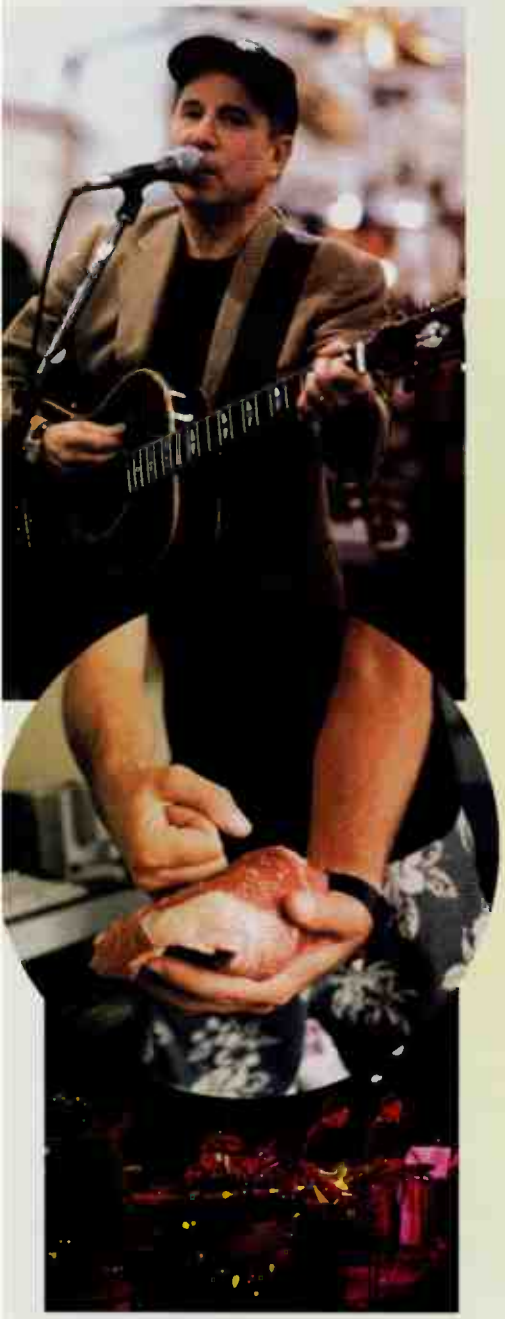
& SOUND TECHNIQUES
VOLUME 10, ISSUE 8
AUGUST 1999



ON THE COVER:
(Left to right, top to bottom) Bruce Swedien, Tommy Tallarico, Al Schmitt, George Massenburg, Leslie Ann Jones, Ed Cherney, Arif Mardin, Phil Ramone, Jac Holzman, Dave Smith, Jack Douglas, Dust Brothers, Tom Holman, Bob Moog, Chris Stone, Frank Fillipetti, and Alan Sides predict the future of recording.

FEATURES

TECHNIQUES/WORKSHOPS

EQ LIVE

DRIVE!

GROOVE

MULTIMEDIA

COLUMNS/DEPARTMENTS

EQ (ISSN 1050-7868) is published monthly plus Buyer's Guide in December by Miller Freeman PSN Inc., 460 Park Ave. south, 9th fl., New York, NY 10016-7315. Periodicals postage paid at New York, NY and additional mailing offices. POSTMASTER: Send address changes to EQ, P.O. Box 0532, Baldwin, NY 11510-0532. SUBSCRIPTIONS: U.S. $29.95 for 1 yr. (13 issues); CANADA add $10.00 per year for surface; other countries add $15.00 per yr. for surface; All add $30.00 per yr. for Airmail. All subscriptions outside the U.S. must be pre-paid in U.S. funds by International Money Order, checks draw from a bank located in the USA Visa, Master Card or American Express. Back-issues $5. Printed in the U.S.A.

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SPECIFICATIONS: ***Amplifier power:*** 250W rms/ch, 400W (100ms peak). ***Peak acoustic output:*** 117dB SPL (100 ms pink noise @ 1M). ***Residual hum/noise:*** ≤10 dB SPL (A-weighted @ 1M). ***THD @ 90 dB SPL:*** <0.4% (100Hz - 10kHz @ 1M). ***Response ±2dB (1/3 oct. swept noise):*** 48Hz - 20kHz @ 1M, 45Hz - 20kHz @ 2M. ***-6dB LF cutoff:*** 40Hz. ***Control Amp Dimensions/Wgt:*** 3.5"h x 19"w x 10.75"d (2u), 42lbs.

SYSTEM: ***Type:*** Modular, active near/mid/far-field monitor. ***Configuration:*** 2-way acoustic suspension. ***Woofer:*** 6.5" treated paper. ***Tweeter:*** 1" metal dome. ***Magnetic Shielding:*** Partial. ***Monitor Dimensions/Wgt:*** 14"h x 7.5"w x 11.9"d, 17 lbs. ***Monitor Enclosure Materials:*** 3/4" mdf w/HP laminate. ***FEATURES/CONTROLS: Connectors: Input:*** XLR, TRS. ***Output:*** XLR. ***Controls: Input sensitivity:*** -10, -3, +4, +11, ∞ dBu. ***Listening proximity:*** 5 position (near/mid/far). ***Boundary proximity:*** 5 position (0,1,2)

A-20 Everything, *as in every single thing*, about the A-20 points to the concept of unmitigated clarity and razor sharp reference – revealing every nuance in detail, in balance and in sonic image. The amplifier is a horse (check out those specs), and due to its outboard nature, there is more efficient heat dissipation and head room than when crammed inside a more conventional wood-based monitor enclosure. Moreover, this puts acoustic controls and diagnostics within your fingers' easy reach. Incorporate some of the finest drivers made and the result is a monitor that not only helps make each session as predictable and repeatable as humanly possible, it makes for a recording that is cut with considerably more precision than any previously known.

NHT PRO

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EQ

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An Educated **Consumer**

Or "The Geek Who Shagged Me"

My goal, to date, in reviews and my monthly column, has been to share information. In essence, I've wanted to assist readers (and Web site visitors) in making choices according to their needs. It's my belief that — to paraphrase the motto of a NY clothier — "An Educated Consumer Is the Best Customer."

I have been in St. Paul for two months (as of July), and I'm finally ready to address what has been, until now, an avoidable concept: that some consumers just want to be told what to buy. Well, which is it: education or dictation? I wanna know!

Your goal may be to save time and/or money, so being told what to buy might justify your (temporary) submissiveness. If, though, your goal is to buy "the right tool for the job" and make no mistakes, it might be easier to be a monk. Relying on information from experts is one thing, but keep in mind that an expert is someone who has made enough mistakes — and observed others doing the same —to become knowledgeable. The process of educating yourself involves more than just reading *EQ* (sorry) or copying someone's notes. You need to experiment after you read. Then document your successes and failures. Finally, give detailed feedback to manufacturers. *Carpe Diem*, baby.

Once you've got the gear, the next issue is product support. When a problem arises — usually at that rare moment of creative inspiration — it is frustrating to call product support and get a busy signal. Few people realize how small our industry is relative, for example, to the consumer VCR market. When you encounter a logjam on the way to product support, it's easy to speculate that a design flaw is the cause. However, it might also be that the profit margin of the product is so low that it does not allow for enough support personnel.

Though it doesn't always deliver instant gratification, most manufacturers also have a Web presence. For every site that is more concerned with marketing than support, there is probably a user-created site or newsgroup dedicated to the product and the problems you have. With all due respect to fans of Tesla, one of my favorite quotes is from Edison, who defined genius as "one percent inspiration and ninety-nine percent perspiration." If you don't see what you need on the Internet, get motivated and roll yer own "help" site. I did!

Taking the time to *document* a problem, in a real or virtual letter, forces you to get all the details straight while they're fresh in your mind. It provides valuable *feedback* to the manufacturer, and could upgrade your status from a user to a "beta tester."

One friend nearly sent a nasty e-mail to a European company complaining about the poor English translation of their user's manual. Rather than provide crustacean-style feedback, I suggested he offer to rewrite the manual in exchange for "perks." After submitting the rewrite, my friend now gets free software updates and a discount on hardware. So, the next time you're ready to blow a mental fuse, consider the proactive alternatives to anger.

Here's another alternative — although you may think that I am crazy for making this suggestion. Based on my own informal survey, it appears that many products meet 80 percent of the user's expectations. Yet, some of those users "flame" the manufacturer as if they were only 20 percent satisfied. This despite the fact that, in reality, none would give up the toys that seem to be destroying their lives. ("Flaming" is the art of harassing via e-mail and bulletin boards.)

If you are one of those "80-percent satisfied" types, ask yourself if you would pay for those extra features or fixes in order to achieve that last 20 percent of perfection. If you answered "yes," how *much* would you pay? For software, I would arbitrarily set the price of a "fix" or a "custom" feature as being what you'd already paid. (Luckily, a software tweak can improve some hardware; so, for that, let's set the price at 25 percent of the purchase price.) The next time you need a bug-fix or a custom feature added, make a dollar offer to the manufacturer for that specific feature, and then let me know the price of your success or failure. Who knows, you could even drum up investors on the Internet!

—Eddie Ciletti

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EX'PRESS YOURSELF

I am a new sound arts student at the Ex'pression Center for New Media in Emeryville, CA. The library here had a copy of the July issue, so I plopped down to try and get a little current. The article on Dave Reitzas was dynamite, and I just thought you should know. It really got me fired up to start the school, but I must admit that it made me a bit nervous as well.

I am not a technology junky, but I am looking to become one. I imagine that, as the courses go on, I will understand more about the jargon, but right now I am feeling a bit inadequate. The rest of the students here have a strong background in playing music, technology, and computers. These days it seems harder to get deep in the industry like Reitzas.

I having been playing bits of guitar since 1994, but I do not carry the stature compared to those around me. I have been told here a few times that knowing an instrument is not as important as what my ears can do. But don't those two go hand in hand? If I could train my ears better, my guitar playing would probably climb to another level.

I guess what I am trying to ask is, have you seen many top-quality engineers that did not have experience recording their own music or playing in bands? Also, should I rededicate myself to my guitar, really make an effort to understanding the finer points? Will studios and bands hire/trust me if I cannot tune a guitar?

Ned Creed
via Internet

[*Engineers have diverse backgrounds, and there are no hard and fast "rules" that make a person qualified or unqualified to be an engineer. You should check out some of our back issues, paying particular attention to the cover stories and engineer/producer technique profiles by Bobby Owsinski and Howard Massey. These show how different engineers can be from one another — both in background and in style — but still be successful. —Ed.*]

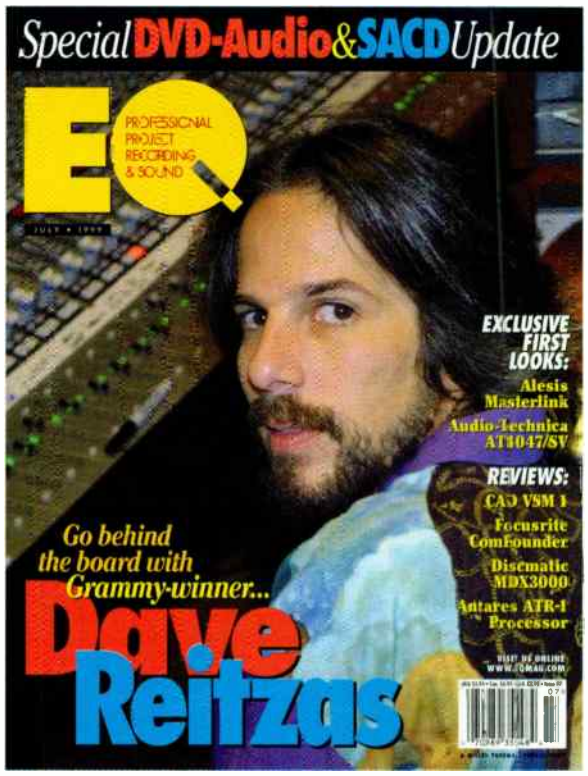
POWER PROTECTOR

Awesome, awesome magazine!

Regarding the issue of which way the XLR pins should go, although Mike Sokol (May '99 issue) and Tony Giverin (letters, July '99 issue) have interesting mental schemes for remembering, let's go to the engineering reason the pins are the way they are: consoles supply phantom power, and if they had pins sticking out, somebody could come along with a screwdriver, short the power, and ruin the show. Power in a system always presents itself at a female connector. Example: your house 110 AC outlets are, thank god, female! You can use this rule for any and all cabling and connectors where power is supplied.

Keep up the great publishing!

George Biner
Field Applications Engineer SW
Cirrus Logic/Crystal Semiconductor
via Internet



WATCH WHAT YOU EAT

I think there is another explanation beyond the one given by Mike Sokol for the vocalist who wants a lot of monitor gain, but refuses to eat the mic [EQ&A, July '99].

Most of the popular dynamic vocal mics for live work have a strong proximity effect, which means that, without EQ, they will greatly increase in bass output the closer a vocalist gets. If 200 Hz or below is rolled off, at either the system EQ or the channel strip EQ, the mud or boom produced by eating the mic goes away, the vocal sound is much clearer, and there is more gain before feedback in the bottom end for that mic. Of course, the channel strip EQ is the better place to do this, since you do not have to sacrifice the sound of the bass guitar and kick drum to clean up the vocals by using the main system EQ.

The problem? Far too many live mixing consoles, for reasons I do not understand after 30 years of doing live sound, take the monitor/foldback send before the channel strip EQ, rather than after. So the house soundguy can fine-tune the sound of the vocal mic out front, while having no effect at all on the monitor feed from that mic. Thus, if the vocalist eats the mic, it is going to boom, and if he/she backs off, it will clear up tonally, even though the gain is weak.

The solution is to change consoles, or modify the existing one, to provide a post-EQ, prefade monitor send. This is not a real tough mod in most cases; I first did it in 1975. I come from the days (early 1970s) when live mixing boards had no channel strip EQ at all, and a pre-EQ monitor send is really no better than that situation. Some manufacturers addressed this as long as 25 years ago, and I wish the rest would get on board. It shouldn't even cost any more to do it right. I am amazed that, after all these years, we cannot, in most cases, have two separate channel strip EQs for main output and for monitor send; that would cost a bit more, but it would be worth it for bands with only one soundman, or none at all.

Tom Shevlin
via Internet

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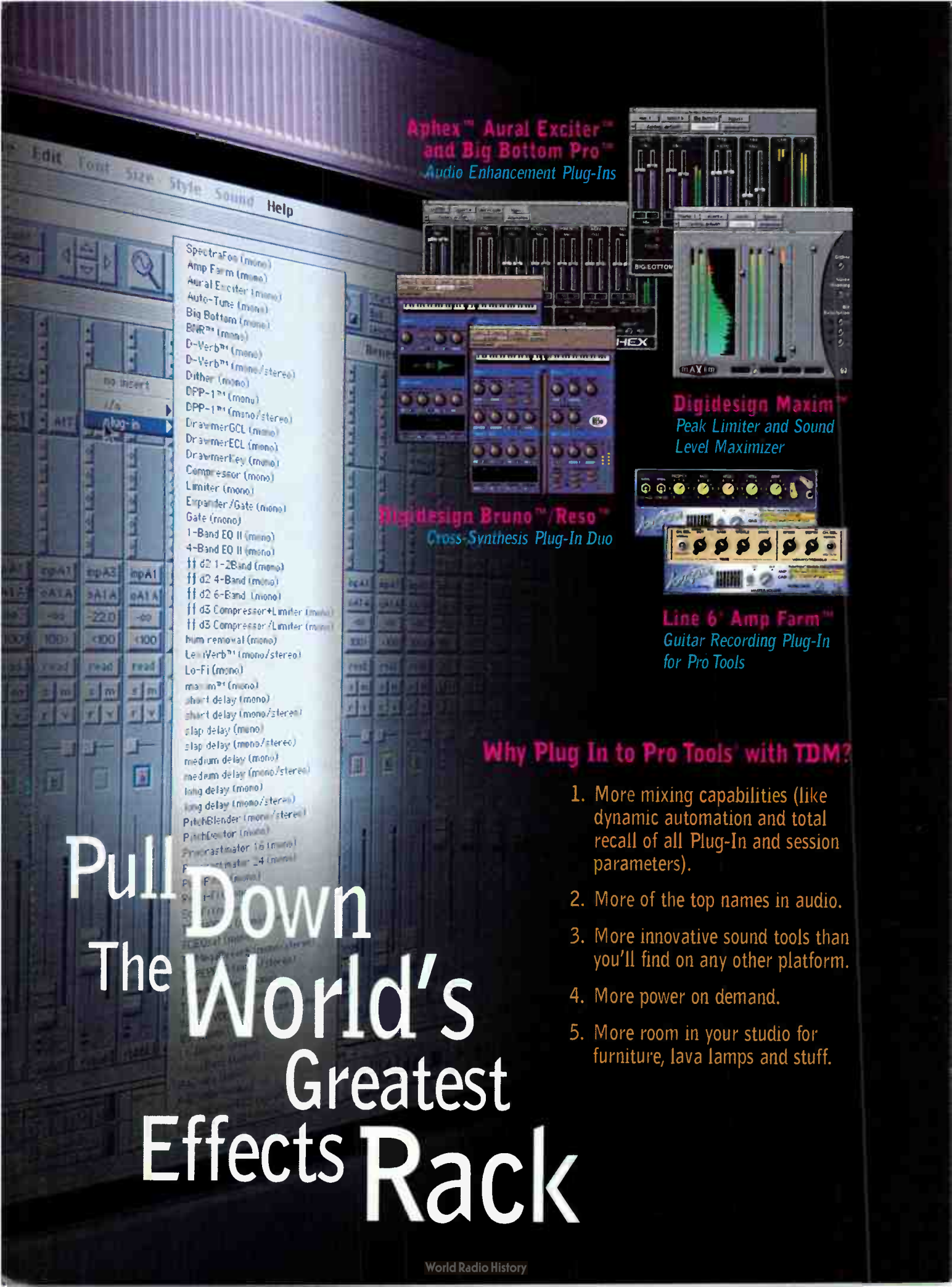

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DVD RE-EDITS

Q *I read with great interest Mike Sokol's article on the way AC-3 sounds can be burned to a CD and played back on a DVD player. [See* EQ, February 1999.] Other than Sonic Foundry, is there any other company that sells soft encode for AC-3?

Another question: Is it possible to edit an encoded AC-3 data stream? The reason I ask is that we are doing editing for movie titles, and some clients who pressed DVD for a specific region want to release them in another, but do not want to spend money encoding the elements again.

Knowing the limitations, we are looking for cuts only. Is there any software that you know that will handle cuts for encoded AC-3?

Lim Tai Wah
via Internet

A I believe that Minnetonka Software (612-449-6481; www.minnetonkaaudio.com) is selling a Dolby Digital (AC-3) encoder plug-in. Plus, I just got a press release that Digigram has licensed the Dolby Digital Professional Encoder. However, I'm not sure this is the solution for you.

The only possible editing with an already-encoded AC-3 file would be cuts only, and then there's the possibility of big ticks and pops at the edit points. I've played with it, and there's no good way to know where the safe cut points are. I don't know of any program that will let you hear the edit points directly, but it's possible to output the S/PDIF output to a standard Dolby Digital decoder for monitoring. Then you have to be careful to cut "in the gap" without corrupting the whole data block. If you look at an AC-3 file that's been saved as a WAV file, there are bursts of data separated by quiet. Even if you cut and splice in this quiet gap, however, bad things can happen when the decoder gets a corrupted data packet. I don't know of any automatic way to do it in AC-3, only Dolby E.

Crossfades or volume changes are not possible at all on the raw data without decoding and re-encoding the AC-3 file. If you want to change elements (for instance, put in a new center narration voice for a different language or change the level of a track), the Dolby Digital (AC-3) file has to be decoded, edited, and re-encoded. Still, in all, AC-3 is a lossy encode/decode process that was not designed to be taken apart and put back together. The noise floor and encoding artifacts quickly become audible. For that process, Dolby has designed something called Dolby E, which functions much like AC-3, but which can be decoded and encoded up to 10 times or more without losing fidelity.

In the distribution food chain, you would use Dolby E for network and satellite feeds where crossfades and dialog replacements are needed. Furthermore, Dolby E's block size can be matched to video frame rates, which means it can be used in a standard audio-follow-video editing system to cut together a video piece with the proper audio for each picture element. But Dolby E is a broadcast/distribution system only. Before final consumer distribution, the 5.1 (or other flavor) audio still needs to be re-encoded as Dolby Digital (AC-3) so it can be used for digital television broadcast or put on a DVD. In addition, because "E" is designed specifically to work in this fashion, it will pass on all the Metadata to the AC-3 encoder without operator intervention, which includes all the dialnorm and compression data.

Mike Sokol
Contributing Editor
EQ magazine

[*Tim Carroll, Dolby Laboratories adds:* Unfortunately, even if you edit "in the gap," you will produce a stream that has discontinuous blocks of data. The decoder will determine that there is an error and will most likely mute. I would also be very concerned about A/V sync, as there will definitely be a slip at the edit points. Fifty edits means fifty slips. Not a great idea, and a risky bet. I for one would quickly return the poor-quality DVD that resulted from editing raw AC-3 streams.]

SURROUND BASICS

Q *I've been trying to find out more about surround sound. I've been reading your magazine, and just have a basic question. Will all the recording formats up to this point (ADATs; Yamaha's D24 MO recorder) be able to accommodate surround? Yamaha tells me that you need a decoder, which they will not develop.*

I would like to know before I sell the cattle ranch to finance a studio that will be obsolete in a few years!

Tony Loeber
via Internet

A Thanks for reading *EQ*. Since you are a loyal reader, you're with us on the cutting edge of the industry (sometimes the bleeding edge). So here's the basic media requirement for mixing surround sound.

As you probably know, we need to consider several different parts of the mixing chain. The first part is the source tracks, which realistically can exist in almost any multitrack format. You can use any analog — or digital — recording deck from 8 to 24 channels.

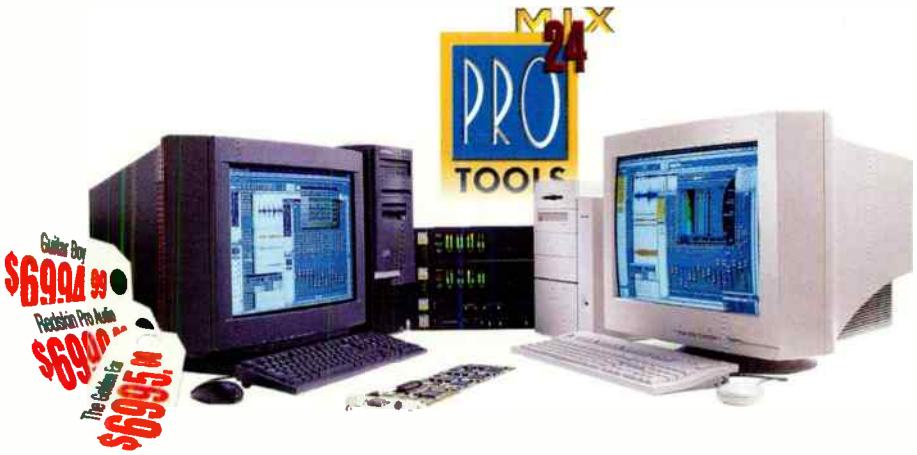
I've mixed using tracks from a dedicated 48-track deck, as well as one, two, or three affordable digital multitrack recorders (e.g., ADAT, TASCAM). I'm presently working on a field-recording project using a Zaxcom Diva hard-disk recorder, which is really cool. Of course, you'll want to use as high a resolution as possible, so while 16-bit/48 kHz is acceptable, 20 and 24 bits are even better. Higher bit rates, while still a big buzz in the industry, don't mean that much to me, but if you can afford the overhead of 96 kHz, then go for it.

Next up is the console: good examples of available multilayer consoles include Panasonic, Spirit, TASCAM, Mackie, and Yamaha. Larger-format boards include Euphonix, SSL, Amek, AMS/Neve, and a few others. They have various bit rates and depths, but 48 kHz by 24 bits seems standard. Of course, you don't really need a console and can do a surround mix internally in your workstation, with Minnetonka's MX51 being an excellent choice. There are also a few Pro Tools plug-ins for surround mixing, but I don't have any experience with them.

I won't go into speakers and monitor controllers right now, since you can just check the last few issues of *EQ*. Suffice to say, the subject is almost worth a book on its own.

Finally, what you asked for: final

Looking for the lowest price?



Fact is, if you purchase your gear from one of the major retailers, you're going to get a great, low price. The big stores all carry the top brands, receive volume discounts from the manufacturers, and then "price-shop" each other to make sure they're not undersold.

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recording formats. You really don't need to worry about encoders and decoders — be they DTS or Dolby Digital. That's only for the mastering house that's assembling the whole mess to a DVD or DTS disc. You just need to get six discrete tracks down on media of some sort. All you really need is a good 8-track deck to record on, and it doesn't have to match the media (or sample rate) of your source tracks. Up to this point, the standard has been DA-88, but that's due mostly to inertia from the movie and video industry.

Since the newest DVD-A spec allows up to six tracks of 24/96 audio without lossy compression, you can do a little better than 16/48. There are a few interesting contenders, but I don't think a clear leader has risen from the primordial stew. Of course, the DA-98 is a heavy-duty, DTRS-format machine, while the Alesis M20 looks like a very serious ADAT entry. In addition, the Yamaha D24 could be a real good choice as well, since optical disc is a great format (much more robust than those Jaz® things). One of my engineering friends uses six tracks of a spare 24-track analog deck with Dolby SR to mix surround tracks. He says it's the best-sounding deck he has, and clients can sample it at anything they like and it will still sound great. This may be the time to rebuild that monster 8-track, 1-inch machine and put modern electronics in it. (I think Eddie Ciletti should offer a rebuild and upgrade for those great old analog decks.)

You can see that all mastering houses are going to have to deal with a variety of surround formats in the future. I think you just need to pick the best-sounding 8-track format you can afford and start mixing. Someone else will sort out the DVD or DTS encoding question down the line. I'm using an Alesis M20 digital recorder for some mixes I'm working on right now, and it's doing a great job.

That said, before you mortgage the farm, you need to get to one of my surround mixing seminars, where I go into these questions plus a whole lot more. I'll be doing a bunch more this fall at various recording schools, and maybe a few music retailers, around the country (check with Hector La Torre at *EQ*). Stay tuned to this station for updates of when the 5.1 Surround tour takes to the road again.

BTW: I'm thinking about writing a book on the subject of mixing for 5.1 surround. Any takers?

Mike Sokol
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SCOPING IT OUT

Q *I really like Eddie Ciletti's very informative articles. I am planning to purchase an oscilloscope and was wondering what kHz/MHz would be okay for audio equipment (head alignment, etc.).*

Tom Coker
Coker Audio
Woodland, CA

A You don't need much for aligning a tape recorder: 100 kHz would be enough. However, they are so "plain vanilla" in features that you may want to shoot just a little higher. Oscilloscopes for audio are very affordable. A new 20-MHz dual-trace (2-channel) 'scope is around $400. On the used market, a 60-MHz–100-MHz 'scope can be had for a similar price.

Considering that "stereo" is normal, a dual-trace 'scope is great for watching both channels. A single-trace 'scope can display the "lissajous" pattern used to observe phase — as would be the case for analog head alignment. A dual-trace 'scope, though, is far more versatile.

The more bandwidth available, the more detail you can observe. This is especially true if you are looking for oscillations and RF. Incidentally, if you ever want to peek at a digital tape deck's RF envelope — if only to confirm basic alignment (for compatibility) or head cleanliness — 20 MHz is a good place to start.

Eddie Ciletti
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ASK US

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EQ NEWS

AUGUST • 1999

NAMM Techno Update

Your neighborhood pro audio dealer took a break from the store last month to get a taste of the Nashville music scene at the annual, mid-year NAMM bash. This event is rarely a watershed of new technologies, however, it did reveal several impending movements that will impact the way you demo and buy your products.

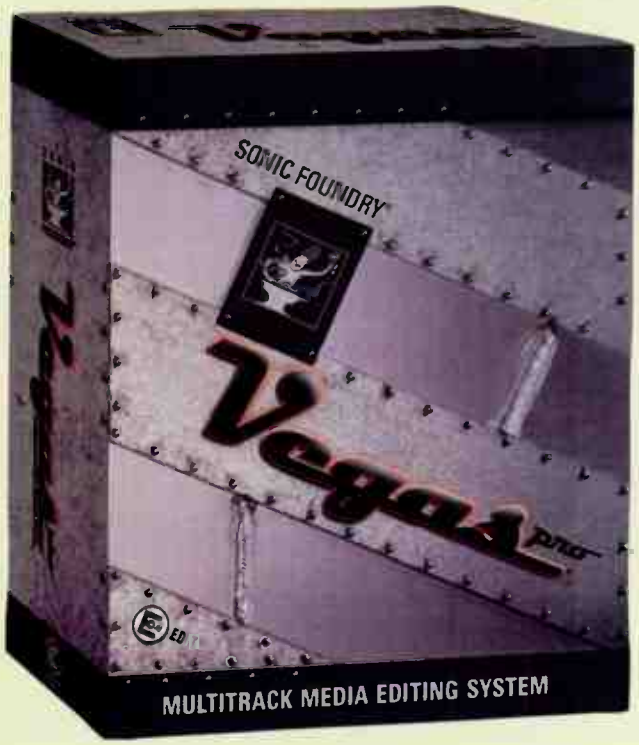
The big news was the fact that the Sam Ash chain had purchased the assets of the venerable Manny's store on 47th Street in New York City. For many industry veterans, this signaled the sad demise of yet another independent retailer, especially one that has played such an integral role in music history.

Consolidation is clearly the trend, with Ash, Guitar Center, and MARS playing an increasingly powerful role throughout the entire pro audio scene. Meanwhile, Sweetwater Sound was passing around CD-ROMs and brochures to hire knowledgeable music professionals who can help them handle the increase in sales calls they're receiving on a daily basis.

Otherwise what was new? Here's a quick lowdown:

• The hot NAMM party award went to Audio-Technica, which hosted a rare duet performance by country heavyweights Vince Gill and Rodney Crowell in the "big room" at Alan Sides's Ocean Way Recording on Music Row. The event kicked off the release of A-T's AT4047/SV large-diaphragm cardioid capacitor microphone, which is reviewed (exclusively) in this issue.

• Sonic Foundry made the world debut of its new Vegas Pro nonlinear multitrack multimedia editing system for Windows-based PCs. Vegas Pro combines audio media production with Internet authoring in a single software environment. Vegas Pro provides 24-bit/96-kHz class performance, accommodates unlimited tracks, and includes DirectX plug-in support. It supports MP3 and the creation of streaming media content, including the ability to incorporate timeline metadata for both Windows Media Technologies 4.0 and RealNetworks RealSystem G2 file formats.



• On the software front, Emagic announced that Version 4.0.1 is now available for Mac users of Logic Audio Gold and Platinum. The new rev adds direct support for Emagic's new AMT8 and Unitor8MkII USB MIDI interfaces. In addition, it offers sidechain support for TDM plug-ins and prefader mode for audio track level meters.

• Liquid Audio and Roland have inked a deal whereby Roland customers will be able to utilize Liquid's Web music tools to upload and download their tunes off the Roland site.

• Keyboard players were going gah-gah over the new S80 synthesizer keyboard from Yamaha. Featuring an 88-note, weighted piano keyboard action combined with an expandable synthesis engine, the keyboard is designed as a controller/synthesizer for the heart of a complex MIDI system. It's especially impressive since it's priced at under $2000.

• Cakewalk got in the groove at NAMM, announcing that it has signed a licensing agreement with Italy's IK Multimedia Production to exclusively distribute GrooveMaker and GrooveMaker MAX in North America. GrooveMaker, which runs on both Mac and Windows PC, is a loop-based mixing software for creating electronic dance music. It can be used as a standalone application or with other Cakewalk software products.

• The list of plug-ins for Digidesign's Pro Tools has grown even longer with the introduction of ChannelStrip from Metric Halo Laboratories. It's literally a channel-strip audio-processing section of a digital console with a TDM plug-in interface, providing PT users the ability to work with the program the way they would with a dedicated mixing console. Its "all-controls-available-at-once" user interface was designed to help the user arrive at a particular sound faster and more intuitively than using plug-in combinations.

• On the PA front, JBL showcased the SR-X Series portable live sound reinforcement loudspeakers at the show, bringing touring sound performance to the average performing band and DJ. Meanwhile, EV introduced the newest member of its Eliminator line — a 15-inch, two-way monitor — at the show, featuring the exclusive Ring-Mode Decoupling found in the acclaimed X-Array Touring System.

• Other news: TC Electronic was at the show, exhibiting its new Intonator intonation processor, which is now shipping...TASCAM introduced the new CD-A700 CD player and auto-reverse cassette deck at the show...Mackie introduced its first completely new compact mixer in over four years, the 1642 VLZ PRO.

KRK Systems Hosts TOTO Listening/Production Session

KRK Systems, Inc. recently hosted a special seminar featuring producer/engineer Elliott Scheiner and members of the band TOTO. The purpose of the event was to explore the multitude of factors involved in music production, recording techniques, distribution, and numerous additional issues that confront both artists and other members of the creative community. Open to a select group of invited guests, the seminar was held at Royaltone Studios in the heart of the North Hollywood/Burbank production community and included a facility tour and an opportunity to hear the just completed TOTO CD, *Mindfields*.

In addition to Scheiner, TOTO bandmembers Steve Lukather (guitar), Mike Porcaro (bass), and Simon Phillips (drums/percussion) were on hand to discuss their individual and collective contributions to the project. Elliott Scheiner and TOTO employed KRK's E-8 studio reference monitors throughout much of the CD's production, and, as a result, all listening segments of the event were played through a pair of E-8 speakers. Miller Freeman PSN west coast editor Roger Maycock moderated the discussion.

As much of the TOTO project was recorded at Royaltone Studios, the seminar was held in Studio A's control room area. After listening to the first of six chosen selections, many in the audience were immediately drawn to the fact that the music had a very wide, big sound. This prompted a discussion of analog versus digital technologies. Scheiner talked about his decision to record the instrumental tracks initially to Studer A-800 MKII 2-inch analog machines prior to being transferred to Sony 3348 DASH recorders. "There continue to be a number of compelling reasons for choosing analog recording technology — among them a certain fullness that still eludes digital under various conditions," says Scheiner.

While instrumental tracks were recorded to analog, it was revealed that vocals were recorded directly to Digidesign Pro Tools. Steve Lukather, who, along with Bobby Kimball, handled many of the lead vocal performances, stressed the importance of being able to conveniently comp several takes and then compile them into a "best of" performance. "For each song on the CD, I usually sang roughly eight takes, which Steve MacMillan then used to create the finished tracks," notes Lukather. "Aside from the fact that it's much easier to do that sort of work with a random access recorder, it's also much faster. We frequently recorded our parts and then split — leaving Steve to work his magic."

IN TOTO: (Foreground) Simon Phillips, Steve Lukather, and Mike Porcaro of TOTO. (Standing) Keith Klawitter, KRK; Elliot Scheiner; Karen Brinton, KRK: Jane Scobie, Royaltone Studios; and Steve MacMillan.

Another production issue evoking considerable discussion was the recording of Simon Phillips's drum tracks and the issue of microphone selection, placement, and room conditions. Scheiner stressed the importance of carefully selecting a room that enhances the quality of the drum sound and paying particular attention to the positioning of the kit in the room itself. Simon Phillips, who is an accomplished engineer in his own right, stressed his preference for top miking and shared his views on recording drums. "A big part of the sound is getting to know the room and paying attention to the tuning of the drums," says Phillips.

TOTO, who, both domestically and internationally, had a huge impact on rock music throughout the '80s and early '90s, has in recent years suffered in the U.S. due to a lack of aggressive promotion. The band continues to be a major attraction in Europe — selling out concert venues and doing exceptionally well with their record releases. When asked about their current distribution arrangements, Steve Lukather expressed his disappointment with the existing situation and noted that the band is currently in the midst of negotiations to rectify the situation.

The TOTO seminar was KRK's first attempt to organize an educational event of this nature and, judging by the turnout and enthusiastic response of the audience, it was an absolute win. According to KRK marketing director Karen Brinton, "This seminar enabled those in attendance to get first-hand advice from one of the top engineer/producers in our industry and discuss specific production techniques and performance issues with one of the most respected bands in the business. The information conveyed was invaluable, and we hope that this will be the first of many similar events for the future."

Serletic Starts Record Co. in Joint Venture with Arista

Multiplatinum producer Matt Serletic has entered into an exclusive joint venture agreement with Arista Records in which his newly established Melisma Records will join the Arista family of labels. The partnership was announced by Clive Davis, president of Arista Records.

"Matt Serletic is one of America's great producers," says Davis. "I'm personally thrilled that he chose Arista as his home for Melisma. I know we will celebrate many successes together."

Serletic added: "We have learned through hard work, studio experience, and extensive time on the road what successful artistic development encompasses. The energy and expertise of the Arista group rounds out a perfect marriage. Melisma's focus is to discover and develop the very best in contemporary music, and with the resources of Arista and BMG Distribution, we now have the leverage to launch a highly-competitive music label."

Serletic is best known as the producer of Matchbox 20, whom he discovered and signed out of Orlando in 1995. Serletic also produced the #1 Aerosmith hit, "I Don't Want To Miss a Thing," which garnered back-to-back Grammy and Oscar nominations and is featured on the multi-platinum soundtrack album of the Touchstone film, *Armageddon*. Serletic has also worked with Collective Soul, Edwin McCain, and Santana.

For more information, visit www.arista.com

Studer Announces Final Run of A827

Studer of North America has announced that the final run of the its flagship analog 24-track recorder, the A827, is underway at the company's manufacturing plant in Switzerland. The news is causing a stir among many top recording studios and producers who have been competing to purchase a used Studer multitrack machine.

According to John Carey, president of Studer of North America, "The continued demand from our customers for the 827 surprised us, but it resulted in planning this final production run and the development of a special collector's edition of the 827. We believe the A827 Gold Edition will be valued by our customers as a top-of-the-line, handmade machine."

According to Studer, leading American studios have already placed orders for multiple machines, with the first deliveries slated for later this summer.

"Each [A827] will include a commemorative numbered ID plate," says Carey, "as well as a test generator, an RS 232 card, an audio channel remote, and an auto locator with stand. Additionally, head production techniques employed on the A827 Gold Edition will further reduce gap scatter."

For more information, call Studer of North America at 510-297-2711 or check out www.studer.ch.

CM Automation Announces Motor Mix

CM Automation has announced that its newest product, Motor Mix, is now fully operational with Digidesign's Pro Tools. Motor Mix is a fully automated, yet compact, console control surface.

Motor Mix is user programmable and expands the Pro Tools software with 100-mm motorized faders and a wealth of dedicated and programmable controls. It also features a wide-screen "scribble strip" read-out that displays ongoing operations. Beyond Pro Tools, Motor Mix is capable of operating any DAW software supporting system.

ADK Responds to Neumann

Audio DeutchKraft, Inc. (ADK) was recently served a cease and desist request by G. Neumann over the company's "SD" series microphone. The reason for the request involved Neumann's concern over the grille design employed by ADK for the mic. ADK has responded to the request.

"Out of deep respect for the Neumann Company, we have withdrawn our A0-51sd both from the shelves of the few dealerships who have advance prototypes, as well as reviewer's demo mics," says Larry Villella, president of ADK. "ADK accepts that Neumann's rights in Germany are clearly defined. We, however, do not waive our rights and remedies internationally. Whether the time-honored principle of using isosceles shapes to attenuate standing waves belongs to one company [or not] remains unanswered....At least three major companies sell a microphone with a virtually identical grille. It is at least a 40- or 50-year-old design, and has been widely imitated. We were led to believe that this 'industry-standard' grille was in the public domain."

Pulling the new mic from retailers' shelves may have an effect on ADK's new product development and roll-out timetables. But, according to Villella, the company is confident it will be short term. ADK's flagship A51 and A-51s, however, will remain unaffected by the Neumann dispute.

AES 1999 Update

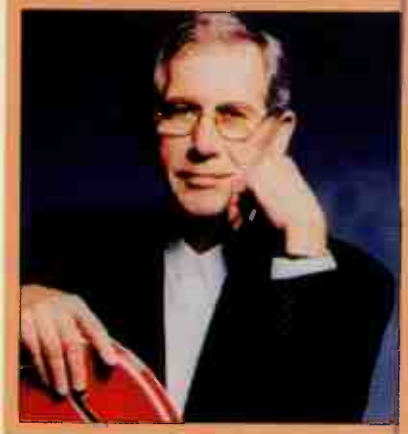
AES Convention co-chairs Kathleen Mackay and Doug Cook, along with AES executive director Roger Furness, have announced that country music legend and audio pioneer Chet Atkins will present the keynote address at the 107th AES Convention.

A 14-time Grammy winner and 9-time CMA (Country Music Awards) winner, Atkins is one of the world's most universally recognized guitarists. As a producer from 1957 through the '70s, he pioneered a pop-flavored country music style that became known as the Nashville Sound. He has produced and developed such legendary artists as The Everly Bros., Roy Orbison, Waylon Jennings, Willie Nelson, Dolly Parton, Floyd Cramer, Don Gibson, and Perry Como.

In other AES news, an honorary AES Membership will be Awarded to Sony Corporation Chairman of the Board Norio Ohga, one of the industry's most accomplished pioneers.

Ohga's career has been characterized by technical, artistic, and business achievements. In 1959, he was named manager of Sony's Tape Recorder Division. He later served as head of Sony's Product Planning Division, where he created the Industrial and Advertising Division. In 1968, Ohga was appointed senior manager of CBS/Sony Records, Inc., and helped nurture the new company. He played an integral role in ushering in the digital audio era with the development of the CD in 1982. Ohga is also celebrated for his distinguished work as a conductor of symphony orchestras around the world.

The 107th AES Convention will be held Sept. 24–27 at the Jacob K. Javits Convention Center in New York City. For more information, call AES at 800-541-7299 or visit www.aes.org.



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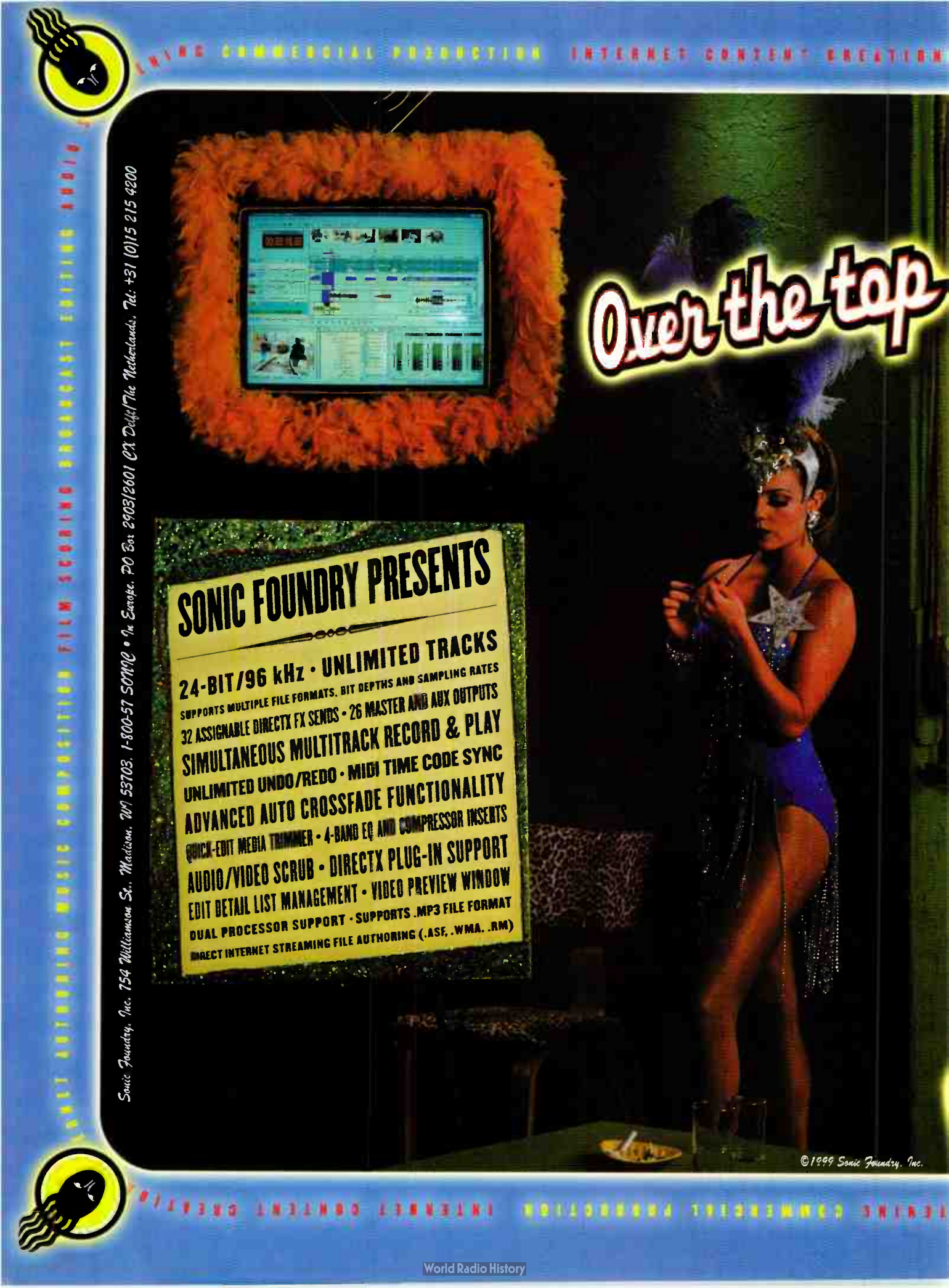


EQ PRODUCT VIEWS

IT'S IN THE TAPE

Quantegy has introduced an enhanced version of its 467 digital open-reel DASH audio format tape. The new formulation offers low error rate specifications combined with mechanically stable DASH format tape media. The 467 tape uses a durable oxide and binder system that is coated and calendered through a clean-room manufacturing environment. This controlled atmosphere, says Quantegy, ensures a smooth and blemish-free surface for very low error-correction activity. Quantegy 467 is available in 1/4-inch, 1/2-inch, and 1-inch precision reel configurations to meet all DASH and ProDigi format needs. For more information, call Quantegy at 334-745-7643 or visit www.quantegy.com. Circle EQ free lit. #101.

BOARD MEMBER

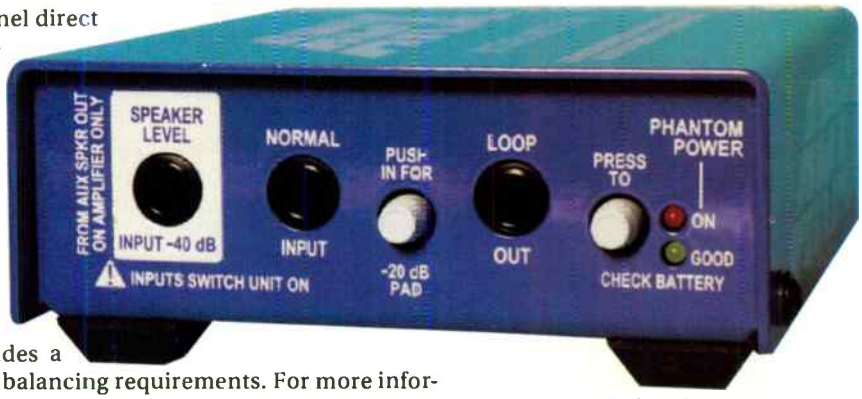
MTA has launched the Series 924, a 24-bus small-format analog console built in the Trident heritage. The Series 924 is designed as an inline console featuring: balanced mic/line inputs with switchable phantom power; phase reverse; 4-band sweep EQ plus high-pass filter on inputs; eight auxiliary sends; stereo AFL; channel pan and mute; Alps K series 100-mm fader; signal present and peak LEDs; and 24 track-assignment switches. Four dedicated effects return channels are provided with high/low shelving EQ, eight auxiliary sends, stereo AFL, mute, and more. An integral meter bridge for monitoring the group outputs or tape returns is provided as standard. The suggested prices are $13,999 for the 24-input console, $16,899 for the 32-input console, and $19,799 for the 40-input version. For more information, call MTA America at 716-589-2100, fax them at 716-589-7498, or visit www.toft.co.uk. Circle EQ free lit. #102.

3-POINT CONVERSION

The z-3src sample-rate converter from Z-Systems is the successor to the company's industry-standard z-2src. It retains all of the older unit's features, while adding support for higher sample rates and longer word widths. The z-3src supports input and output sample rates up to 96 kHz at word widths up to 24 bits, and can perform sample-rate conversions over a 1:3 to 3:1 range. This means that 96 kHz to 44.1 kHz and 44.1 kHz to 96 kHz are well within its range. In addition, the unit is equipped with an output word width control that allows the output to be set to 16, 20, or 24 bits. The z-3src boasts better than –120 dB THD+N and a dynamic range better than 120 dB, which is usable for demanding high-quality audio applications, such as mastering. The unit lists for $1500. For more information, call Z-Systems Audio Engineering at 352-371-0990, fax them at 352-371-0093, or visit www.z-sys.com. Circle EQ free lit. #103.

DIRECT ISSUE

ARX has released the DI-Plus single-channel direct box. Inputs are via a normal gain, high-impedance, unbalanced input, and a separate –40 dB speaker level input for signal derived from the external speaker output of an instrument amplifier. Both inputs can be attenuated further by a global 20 dB Input Pad switch. The DI-Plus will automatically switch itself to the mixing console's own phantom power, if available. If not available, the DI-Plus reverts to the internal battery. Complete with input loop through socket, balanced XLR output, and ground lift switch, the DI-Plus provides a rugged solution to all direct injection and signal balancing requirements. For more information, call ARX North America at 818-225-1809, fax them at 818-225-1309, or visit www.arx.com.au. Circle EQ free lit. #104.

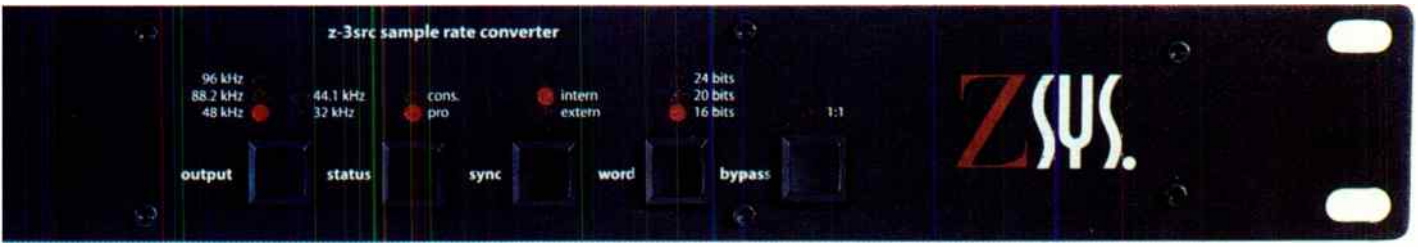


XL-ENT

Dbx is offering its dbx 223 stereo two-way/mono three-way crossover and dbx 234 stereo three-way/mono four-way crossover in "XL" versions, featuring TRS differentially balanced XLR ins and outs in place of their previous 1/4-inch connections. Like their 1/4-inch-ready siblings (which are still available from dbx), both units feature switches on the rear panel for stereo or mono operation, LF mono sum (for a subwoofer feed), and a range of individual channel crossover frequencies. LEDs on the faceplate indicate when "x10" switches on the back panel are activated and whether the unit is in stereo or mono mode. Recessed 40 Hz low-cut switches on each channel are available to remove unwanted low frequencies. Both units feature Linkwitz-Riley 24/octave filters. The suggested retail price is $299.95 for the 223XL and $349.95 for the 234XL. For more details, call dbx Professional Products at 801-568-7660, fax them at 801-568-7662, or go to www.dbxpro.com. Circle EQ free lit. #105.

DARLING DARLA

Event Electronics has introduce the Darla 24 multitrack digital audio recorder. The new Darla 24 boasts a number of improvements, including balanced inputs and outputs, clock sync among multiple Darla 24's, and a desktop audio interface box loaded with 1/4-inch TRS connectors. The system's 24-bit 128x oversampling A/D and D/A converters provide over 108 dB of dynamic range for quiet operation and high headroom. And with support for sample rates up to 96 kHz, Darla 24 is ideally suited for DVD-Audio and surround sound production. Darla 24 ships cross-platform (PC and Macintosh), and the package includes multitrack recording/editing software for the PC. The system is fully compatible with WAV device driver-based and ASIO-based host application on the PC, and ASIO-based applications on the Macintosh. For more information, call Event Electronics at 805-566-7777 or fax them at 805-566-7771. Circle EQ free lit. #106.



EQ PRODUCT VIEWS

STORM OF THE CENTURY

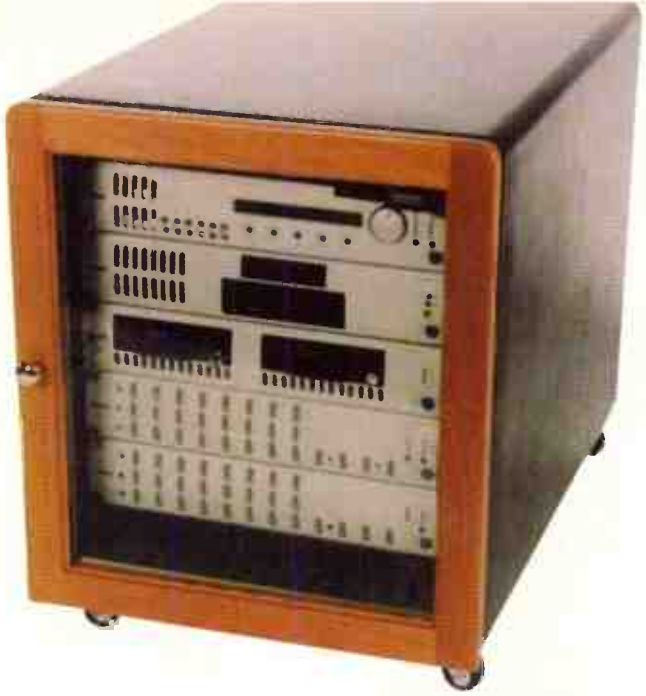
Digital Audio Research (DAR) has debuted its new range of SoundStation audio editing systems — the SoundStation STORM. The SoundStation STORM is initially available as a 32-track system, scalable to up to 128 internal tracks. DAR has designed the STORM as a bankless system, removing the need for a disk-to-disk copy when moving a segment from track 1 to track 128, and therefore greatly improving the system speed. The STORM supports sampling rates up to 192 kHz, plus 16- and 24-bit operation. For more information, call Digital Audio Research at +44 (0) 1372 742848 or visit www.dar.uk.com. Circle EQ free lit. #107.

FIGHTING THE SURGE

Housed in a 2U metal chassis, the SX-2120 from Surge-X provides surge protection and power conditioning for up to 15 pieces of equipment. The SX-2120's rear panel features 14 standard grounded AC receptacles. These are divided into two separate banks of six outlets that may be switched on or off via front-panel rocker switches, plus two outlets that are permanently on. Price is $599. For more details, call New Frontier Electronics at 215-862-9344, fax them at 215-862-0270, or visit www.frontierelec.com. Circle EQ free lit. #108.

QUIETING NOISY GEAR

Noise created by a hard drive or other sources can be fatal to productivity when working in the digital domain. The ISO Box from Sound Construction and Supply of Nashville is designed to eliminate hard drive and other noise, enabling important digital equipment to remain in the center of the work environment. The ISO Box is fabricated from medium-density fiberboard to contain seek noise (hard disk read and write crunch). It is lined with 1-inch Auralex acoustical wedge foam to help it minimize spindle (audio drive) and white noise. The unit's exclusive HVLP fan cooling system brings fresh air to the contents of the ISO Box every 20-30 seconds. The unit's 1/4-inch laminated glass front door adds a greater degree of noise reduction. The ISO Box is available in 12-, 16-, and 20-space configurations. Prices start at $750. For more information, call Sound Construction and Supply of Nashville at 615-313-7164, fax them at 615-313-7799, or visit www.custom-consoles.com. Circle EQ free lit. #109.



CUSTOM POWER

Juice Goose has introduced a new concept in power distribution, the PD-*ONE*. This made-to-order, multiple-circuit power strip is designed for serious amplifier racks. The PD-*ONE* features a choice of one to six individual circuits with 12 duplex outlets (24 total outlets), providing up to 120 AMPs of power. No longer is it necessary to spend valuable time assembling expensive power distribution pieces to get multiple circuits in a single chassis. Juice Goose provides an easy-to-use menu of PD-*ONE* options and costs. Options include number of circuits, length of the wire harness, and where the wire exits the chassis. The company also offers an isolated ground version of the PD-*ONE*. For more information, call Juice Goose at 713-772-1404 or fax them at 713-772-7360. Circle EQ free lit. #110.

TASCAM TM-D4000 Digital Mixer

TASCAM takes the middle ground in completing their digital console triumvirate

BY ROGER MAYCOCK

TASCAM became one of the first manufacturers to bring a digital mixer to market with their introduction of the TM-D8000. This was followed later by their first small-scale entry, the TM-D1000. Now their third offering in this category, the new TM-D4000, is being introduced to the market. This new console takes the middle ground of the "affordable digital mixer" market from a price perspective, but leans toward the upper echelon in terms of features and functionality.

TASCAM's TM-D4000 brings a number of important features to the party, including flying faders, on-board snapshot automation, dynamic automation via an external PC, and dedicated machine control for accessing your recorders. These and other functions make the mixer well-suited for music production at both the project and commercial studio level, as well as equally comfortable in postproduction, particularly in the offline editing suite.

The TM-D4000 provides 36 channels configured as 32 mono and 2 stereo — each with 4-band fully parametric EQ, six aux sends, and dynamics processing. The mixer's 16 long-throw motorized 100-mm faders are layered to facilitate control of the 32 mono channels, which may be ganged to create additional stereo pairs. The console's eight busses and six aux sends derive their fader control from a third layer. The two stereo channels and the stereo master have dedicated faders. For additional flexibility during mixdown, the console provides eight fader groups and eight cut groups.

The TM-D4000 provides an on-board stereo digital effects processor that includes reverb, gated reverb, delay, chorus, flange, pitch shift, phase, exciter, and de-esser. These signals can be assigned to either Aux 1–2 or Aux 5–6, and are returned at the stereo Digital In 2.

Depending upon the type of equipment you intend to use with the TM-D4000, you should plan to add upwards of three IF-Series expansion cards. Without these, you'll be limited to eight analog mono inputs, two analog stereo inputs, and the analog, stereo master bus output. The IF-Series cards provide 8-channel digital I/O in three flavors: IF-TD4000 (TASCAM's TDIF); IF-AE4000 (AES/EBU on a 25-pin D-sub); and IF-LP4000 (ADAT optical). A fourth card, the IF-AD4000, provides eight channels of +4 dBu balanced analog I/O on a 25-pin D-sub connector.

The TM-D4000 is best thought of as a console for 24-channel digital I/O operation. The IF-Series cards 1, 2, and 3 (in any combination) are defined for channels 9–16, 17–24, and 25–32, respectively. The only exception to this is that channels 5–8 and 13–16 can be assigned to the two digital inputs. Mixer channels 1–8 are the integral analog channels and primarily serve for inputting signals or perhaps additional returns at mixdown. These eight channels provide balanced XLR mic inputs with 48-volt phantom power, 1/4-inch balanced TRS jacks, and 1/4-inch TRS inserts. All A/D and D/A converters, including the final stereo digital mix, operate at up to 24-bit resolution. Additionally, the mixer supports surround sound mixing via busses 1–6.

For automation control, the TM-D4000 provides 80 user-definable snapshots, which can be recalled by way of MIDI Program Change (PG) commands or from the control surface. Dynamic automation is handled via an external Windows PC, providing offline editing control in addition to a vastly superior visual in-

IF YOU COULD ONLY HAVE ONE MICROPHONE...

The popularity of the C414 B/ULS remains an enigma. Historians boast that in the last decade the C414 B/ULS has contributed to more #1 selling records than any other microphone in existence. From Barcelona to Rome. From Paris to New York, the C414 B/ULS can be found in almost every professional recording studio. The majority of world-class studios like *The Hit Factory* in New York, have at least 10. Purists have aptly nicknamed the C414 B/ULS "The Chameleon." A master- piece of Austrian technological know-how and precise crafts- manship that knows no boundaries or outer limits. Perfectly suited to any genre of music or instrumentation, the C 414 B/ULS delivers what the world hears over and over and over...

C414 B/ULS

THE MOST POPULAR MICROPHONE IN THE FREE WORLD

AKG

terface than could be achieved from the internal LCD.

The TM-D4000 provides MIDI In, Out, and Thru connectors. In addition to receiving PG commands for recalling mixer snapshots, the console provides support for MIDI Machine Control. The TM-D4000 supports jog/shuttle and includes Record function switches for arming tracks on the connected recorders. Software enhancements to the mixer are installed via MIDI system exclusive messages, which can be played into the console from any MIDI sequencer.

Central to the TM-D4000's operation is the 320- x 240-dot, variable contrast, backlit LCD and its array of rotary encoders and switches. The graphical user interface facilitates inspection of EQ curves, pan positions, and numerous additional per channel or global parameters, and is also the focal point for accessing the on-board EQ, dynamics, effects, and snapshot libraries.

The mixer operates at 44.1 or 48 kHz (with ±6 percent varispeed) and provides the ability to define various clock configurations. Word Clock In and Out terminate on BNC connectors. Additional control/sync connectors include: RS-422 (9-pin D-subminiature); TO HOST (for connection to a PC for dynamic automation, also on 9-pin D-subminiature); Timecode In (RCA unbalanced); Meter In (37-pin D-subminiature for the optional model MU-4000 meterbridge); and Cascade In and Out for interconnecting upwards of three additional TM-D4000's.

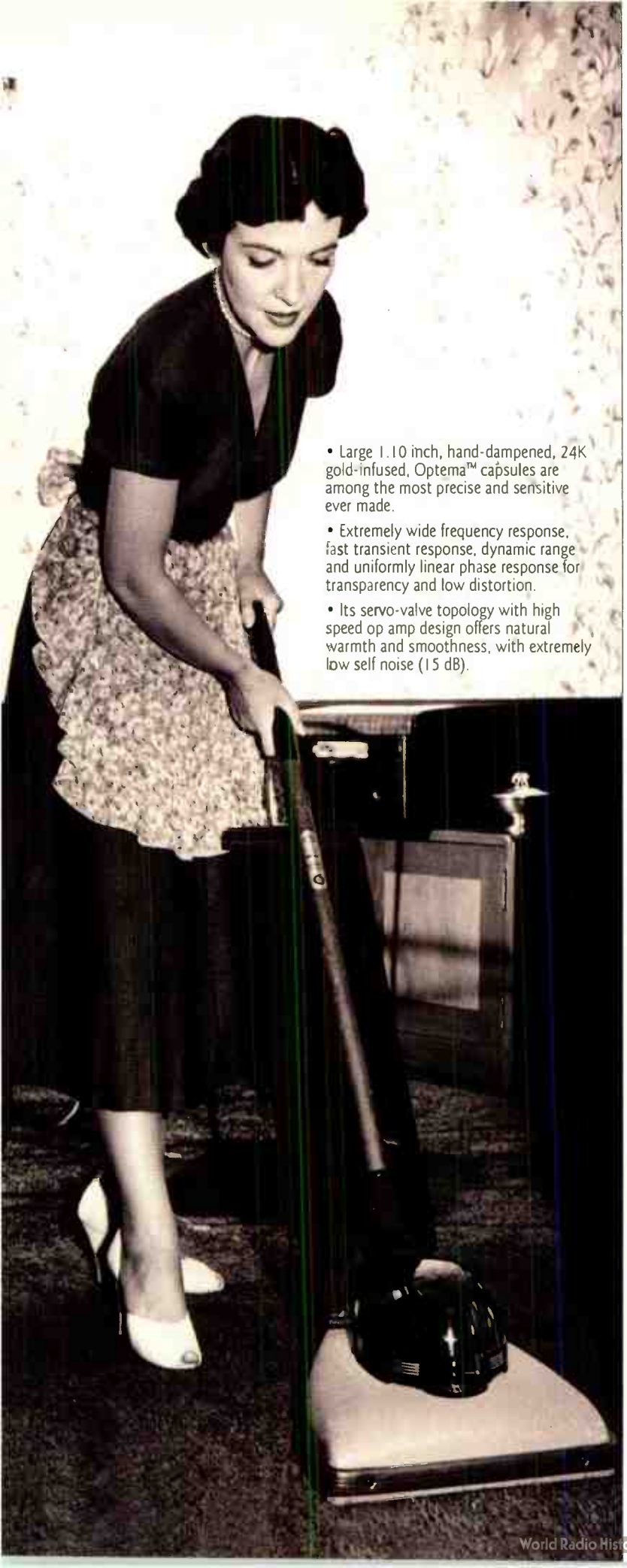
The TM-D4000 is best thought of as a console for 24-channel digital I/O operation.

In addition to a relatively small footprint, the TM-D4000 benefits from an ergonomic design that makes every system parameter, whether hardware or software based, easy to access and adjust. Both the LCD and the timecode display areas are easy to read — keeping you in touch with your project every step of the way. There are full control room and monitoring facilities, including an integrated talkback microphone. Other than the four IF-Series expansion cards and the meterbridge, the only remaining option is the model PW-4000CS cascade cable for interconnecting multiple TM-D4000's.

TASCAM's TM-D4000 digital mixer provides a wealth of I/O options, a solid feel, facilities for expansion and system enhancement, and a user interface that won't have you scratching your head in bewilderment.

Price is $4299. Accessories: IF-TD4000, $199; IF-AE4000, $250; IF-AD4000: $499; IF-LP4000, $199. Model PW-4000CS cascade cable, $99. Model MU-4000 meterbridge, $550. For more information, contact TASCAM (TEAC Professional Division), 7733 Telegraph Road, Montebello, CA 90640. Tel: 323-726-0303. Fax: 323-727-7635. Web: www.tascam.com. Circle EQ free lit. #111.

Picks up what other mics leave behind.



- Large 1.10 inch, hand-dampened, 24K gold-infused, Optema™ capsules are among the most precise and sensitive ever made.
- Extremely wide frequency response, fast transient response, dynamic range and uniformly linear phase response for transparency and low distortion.
- Its servo-valve topology with high speed op amp design offers natural warmth and smoothness, with extremely low self noise (15 dB).

The VSM1 offers the warmth and smoothness you expect from a large diaphragm tube condenser. But what makes it different is its enhanced sensitivity—to capture and articulate voices, instruments and rooms with their distinctive character intact.

Its uncanny accuracy and ability to convey a faithful sonic image can evoke chills. Its ultra-low distortion and virtually non-existent self noise make it among the cleanest, quietest mics available, certainly in its class.

Unlike many of the new "vintage" replicas, the VSM1 is an original. It's the result of decades of microphone wisdom combined with our own innovations, including the inspired circuit topography developed for our acclaimed VX2 twin tube condenser. Unlike many of today's so called "classics," the VSM1 is built in the US, in our state-of-the-art facility. With meticulousness found in mics costing several times as much—and it sounds like it!

Visit your CAD microphone dealer for a listen, compare it with anything on the shelf. Hear what you've been missing. In fact, with VSM1 you'll hear so much more you may want to clean your studio.

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CIRCLE 11 ON FREE INFO CARD

Power Technology DSP FX Virtual Pack

Put the equivalent of 12 signal processors into your PC

BY ROGER MAYCOCK

Over the past several years, Power Technology has gained a solid reputation as a developer of high-quality effects processing software. Known as the DSP FX Virtual Pack, the company's collection of software-based DirectX plug-ins for the PC/Windows environment is currently released as Version 6.2. The DSP Virtual Pack includes a full array of processors, including Parametric EQ, Auto-Pan, Tremolo, Multi-Element Chorus, Analog Tape Flanger, Stereo Pitch Shifter, Acoustic Verb, and Multi-Tap Delay. Recent additions to the software suite include the Optimizer, Studio Verb, Aural Activator, and support for Nullsoft's Winamp MP3 player. As an added bonus, Version 6.2 adds a new Stereo Widener that enables you to enhance and manipulate the stereo content of both full mixes and individual tracks that have been recorded with stereo miking techniques.

As DirectX-compatible plug-ins, this collection of signal processors will be accessible from within any program that supports Microsoft DirectX. Thus, if you use (for example) Sound Forge and Cubase VST as your 2-track and multitrack digital audio programs, the DSP FX Virtual Pack will be available from within both applications. Other compatible programs include Cakewalk Audio, Steinberg WaveLab, Emagic Logic Audio, Cool Edit Pro, Sonic Foundry ACID, SEK'D Red Roaster, and all 32-bit versions of Innovative Quality Software's SAW (Software Audio Workshop). The software uses true 32-bit, floating point processing, and will operate as a stand-alone collection of processors, thus enabling you to process your sound card's "live" stereo inputs and send the processed signal to your sound card's stereo outputs.

System requirements are quite modest. You'll need a Pentium-class PC, 10 free megabytes of disk space, a mouse or other pointing device, a VGA display with at least 640 x 480 resolution, and a sound card.

As this is a comprehensive suite of software processors, we'll look at some of the new and unique aspects of the DSP FX Virtual Pack. Trust me when I say, "There's a hell of a lot of software here."

One particularly interesting aspect of the DSP FX Virtual Pack is its ability to be controlled remotely through common MIDI Controller messages. This capability enables you to have tactile control over the various parameters from an external MIDI controller. With little effort, I was able to use my trusty Yamaha MCS2 MIDI control station (anybody remember that?) and control pan position on the Multi-Tap Delay. As most recording professionals hate using a mouse for this type of common function, the MIDI support is really a welcome addition.

The DSP FX Virtual Pack provides two reverbs: the Acoustic Verb and the new Studio Verb. The Acoustic Verb is particularly useful for adding ambience to a recording, while the Studio Verb is a more dense, warm reverb effect reminiscent of high-end systems such as those from Lexicon. With each reverb, as I rotated the on-screen room size knob, a box representing the room would change in size — enabling me to visually get a sense of what I was doing in addition to hearing the changes. This type of visual feedback is indicative of virtually every processor in the collection, and can be a valuable aid in assisting you with the many intricate as-

pects of shaping your sound.

The Aural Activator and the Optimizer are new to this version of the DSP FX Virtual Pack. The Optimizer, intended for use as a mastering tool, combines a look-ahead peak limiter with high-quality dithering algorithms. It is best used as the final processing step on a stereo mix or individual track. This processor can significantly enhance the impact and perceived sound level of all PC-based audio production. The Aural Activator adds carefully generated high-frequency harmonics to the output signal. You'll want to use this tool to brighten up material that has experienced high-frequency loss, such as older recordings or those from cassette tape.

INDIVIDUAL AND MULTITRACK PROCESSING

With an application like Sound Forge, using the DSP FX Virtual Pack is about as easy as it gets. The processor of your choice is accessible via the drop down DirectX menu and impacts the active window on screen — regardless of whether this is a mono or stereo file. With multitrack applications, you'll need to learn how to use the track insert capability of your program if you wish to process an individual track. Think of this as though you were patching an insert cable into the insert jack on a conventional mixer. If you wish to process multiple tracks such as a group of percussion instruments, you'll access the DSP Virtual Pack via one of the aux sends/returns of your program.

What's so cool about this is the fact that all "patching" takes place without you ever once crawling around on the floor with a flashlight or getting frustrated by a dirty patchbay. No added noise or ground hums to contend with; just clean, easy to use, sophisticated, virtual signal processing.

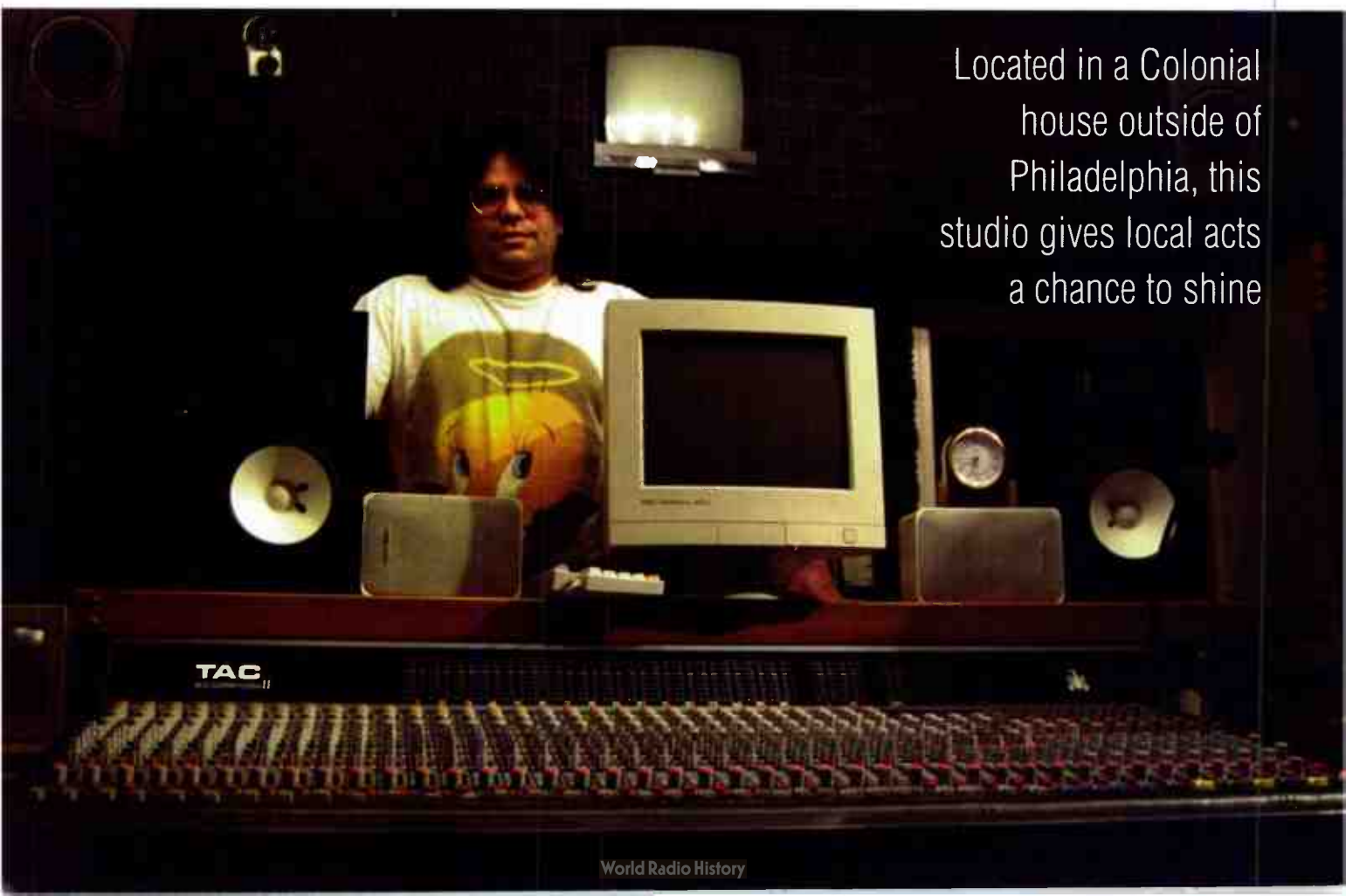
Power Technology's DSP FX Virtual Pack is like having a rack of 12 dedicated signal processors at your disposal. For anyone using a Windows PC for audio work, it's difficult to imagine not taking a keen interest in this product.

Price is $299. For more information contact Power Technology, 100 Northill Drive, Bldg. #24, Brisbane, CA 94005. Tel: 415-467-7886. Fax: 415-467-7386. Web: www.dspfx.com. Circle EQ free lit. #112.

Local Hero

STUDIO NAME: Gold Key Recording
LOCATION: Suburban Philadelphia
KEY CREW: Bob Grosso, Staveley Andrews, Mike Harmon, Mike Klein
PROJECTS RECORDED: Numerous local bands including Papa's Gun, Jades Parlor, Wykked Wytch, mastering for the Philadelphia Music Conference compilation CD
CREDITS: Engineering and recording of several albums on DCC Records/Solutions Entertainment, including Dixie Hummingbirds, Paris, and the Bruce Wacker Band
CONSOLE: TAC Scorpion II (30 x 8; 50 inputs available during mixdown)
KEYBOARDS AND MIDI GEAR: Yamaha SY77, RY30, and DX7 II; Roland D50 and R8M; EMU Proteus; Oberheim Matrix 1000; Alesis D4; DrumKat 3.0 triggering system; MOTU MIDI Timepiece, MX-8 MIDI patchbay
SAMPLERS: Prophet 2002+
MONITORS: KRK 9000B; Yamaha NS10M; Minimus 77; Rane HC6 headphone amp; Symetrix SX204 headphone amp
AMPLIFIERS: Hafler Pro 2400
COMPUTERS AND SOFTWARE: Apple PowerMac 8100 with Pro Tools and TDM (this 8100 also runs Digital Performer and Cubase VST); PowerMac 8600 with CD burner and the Media 100 NLE; Amiga running Bars and Pipes Pro 2.5
RECORDERS: Studer A80 Mk IV; Alesis ADAT [3] with BRC and AI-1; Ampex 351 1/4-inch 2-track; TASCAM 112 cassette deck; JVC TD-W777 cassette deck; TimeLine Lynx for 48-track lockup
DAT MACHINES: Panasonic SV-3700 and SV-3200
OUTBOARD GEAR: Eventide H3000SE Ultraharmonizer™; Lexicon PCM70; TC Electronic 2290; Yamaha REV7, REV5, and SPX900; Drawmer DL241; UREI LA2A; RSP Tube Saturator; Symetrix 564e; Gatex Quad Gate; DigiTech DSP128; Alesis 3630 and Midiverb IV; Behringer Composer; Aphex Aural Exciter Type C2; Aphex 107; BBE 862; Summit EQP200 tube EQ; API 550 EQ [2]; API 550A1 EQ [3]; Rane ME30 1/3-octave EQ, and FME15 EQ [2]; dbx 1531x EQ and 160x compressor; Orban 672 parametric EQ; Altec 436A; Ensoniq DP/4+
MICROPHONES: AKG C414, C451, D12E, and D112 [2]; Neumann KM 84 [2] and KM 184 [2]; Sennheiser MD421 [3]; Electro-Voice PL20; Microtech Gefell UM70; RCA 77DX and BK5; CAD Equitek II; beyerdynamic M400 and M88TG; Shure SM57 [2], SM58, and VP88 stereo mic; Soundeluxe U195
INSTRUMENTS: Five-piece Yamaha Club Custom drum kit with GMS snare drum; 1976 Gibson Les Paul and EB3 bass; Fender American Stratocaster and Sidekick Reverb 25 amp; Madeira 12-string acoustic guitar; Martin D28 and Washburn acoustic guitars; Peavey Mace guitar amp with 4x12 cab; Roland JC120 amp; Intersound guitar preamp
STUDIO NOTES: According to Bob Grosso:

continued on page 137



Located in a Colonial house outside of Philadelphia, this studio gives local acts a chance to shine

The Complete Digital Studio

Truly a complete digital studio, the DPS12V2 combines a powerful 12-track random-access digital disk recorder and 250 virtual tracks, with a digital mixer, and a 4-bus effects processor. This compact package fits easily into any personal studio space, and it even includes the ability to master direct to CD. Audio is recorded in uncompressed format in order to maintain the complete integrity of the source, and the graphic waveform display provides a detailed view of the recorded material. Recordings are stored to SCSI hard disks, allowing the flexibility to create recordings of any length, limited only by the size of the storage medium. Akai's M.E.S.A. PC mixer software allows total control of the transport and 20 channel mixer via computer. When you are finished with your project, it is a breeze to archive all tracks, virtual tracks, effects and mix settings to CD for permanent storage.

Contact your local Akai dealer today to learn more about the DPS12V2 and how to turn your home into a digital studio.

- Master Direct to CD
- CD Archiving
- Timestretch
- M.E.S.A. Mixer Software
- Graphic Waveform Display
- Uncompressed Audio

AKAI professional

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Akai Musical Instrument Corp. 4710 Mercantile Drive, Ft. Worth, TX 76137, 817.831.9203, fax 817.222.1490
In Canada contact Power Marketing • 372 Richmond Street W. #112 • Toronto, Ontario, Canada M5V 1X6 • 416.593.8863

Shure 540S

The mic formerly known as Sonodyne II

MICROPHONE NAME: Shure 540S
YEAR OF MANUFACTURE: 1961 through 1979
PRICE: $49.95, MSRP in 1961
TYPE OF MIC: Dynamic
POLAR PATTERN: Omnidirectional
FREQUENCY RESPONSE: 50 Hz to 13,000 Hz (see User Tips)
RATED SOURCE IMPEDANCE: 250 ohms
OPEN CIRCUIT VOLTAGE: –76.5 dB or 0.149 millivolts (0 dB=1 volt/microbar)
OUTPUT LEVEL: –57 dB for low impedance operation, 0 dB=1 milliwatt per 10 microbars; –52.5 dB for high impedance operation, 0 dB=1 volt per microbar
SENSITIVITY: –148 dB, 0 dB referenced to EIA standard SE-105, August 1949
CASE MATERIAL: Die-cast zinc and Armo-Dur
DIMENSIONS: 1-31/32-inch diameter x 3-5/8-inch length
WEIGHT 18 ounces
MIC NOTES: Also known as the Sonodyne II, the Shure 540S microphone was the first in a small family of 540-series microphones manufactured by Shure from 1961 through 1983. The "S" designation indicates that this microphone has an on/off switch; the 540 — manufactured from 1962 through 1979 — lacked the switch (and the "S" designation). From 1979 to 1983, Shure produced the 540SH, which had the on/off switch and a black base as opposed to the silver-colored base of the 540 and 540S.
USER TIPS: One of the more unusual design aspects of the Shure 540S is that its frequency response can be user-modified by making a few internal changes with a resonator plate and a set screw included with the microphone. Unscrewing the grille cap reveals this plate, which can be removed and replaced with an 18-hole resonator plate. At the same time, a spacer (also included) is inserted between the grille cap and the mic element, resulting in a sharp high-frequency rolloff above 6000 Hz. The low-frequency response can be changed by removing a small (4-40) set screw located at the rear of the microphone. The absence of this screw creates an acoustic port that produces a rapid rolloff of frequencies below 400 Hz. Making both frequency response modifications will leave the mic with a midrange emphasis that is useful for increased voice intelligibility. EQ

PHOTO BY EDWARD COLVER

D-108

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CIRCLE 19 ON FREE INFO CARD

Mackie D8B Console

Part one of a two-part series on using the popular digital board

BY RANDY NEIMAN

As a product specialist for Mackie Digital Systems, I am questioned most often about two topics: [1] system integration needs; and [2] "insider" tips and tricks for the Digital 8 Bus console. That's why I've divided the following Tech Tips into two parts. Part one discusses integration issues; Part two deals with helpful hints for quick mixing.

EVERYTHING BUT THE KITCHEN SYNC

Moving from either an all-analog recording scenario or a hybrid system (analog console/digital recorder) to a fully digital system can deliver many advantages in terms of control, DSP, and conversion processes. One of the most important, and often misunderstood, issues to consider in making the switch to a fully digital system is synchronization, or sync (not sink).

The two most commonly used (and confused) types of sync are word clock (or sample-sync) and LTC (Longitudinal Timecode) like SMPTE or MTC (MIDI Timecode). Longitudinal Timecode references linear time, showing either hours, minutes, seconds, and frames (HR:MN:SC:FR) or bars, beats, and ticks (BR.BT.TC) to indicate a point in time. This lets you know, for example, that you are 00:03:12:10 into your audio (or video) program. This method of sync is also how you can lock different devices together in some sort of coherent way (e.g., your sequencer and DAW locks to tape, so they're all at the same place in time).

Word clock often seems a bit more mysterious to the digital user. There's a plethora of specifications and technobabble we can throw around here, but instead, let's take a look at why this may be one of the most important parts of your digital audio chain.

ROCK AROUND THE WORD CLOCK

Most *EQ* readers know that, in an analog audio system, the frequencies and amplitudes of the sound are represented by voltages that are analogous. Hence the term "analog" — to the original waveform as it would appear in the conductor (in most cases, air). In order to get this information into a digital domain, where these values are now represented by a series of 0's and 1's, it is sampled at a rate determined by the frequencies you want. Time to put on the old propeller beanie.

According to the Nyquist theorem (named after a Swedish dude — not the nighttime cold medicine), in order to sample a frequency without errors, it must be sampled at a rate twice as high as the desired frequency. For example, to sample a sound at 20 kHz, you must have a sampling rate of at least 40 k. Since the upper reaches of human hearing is generally calculated to be around 22 kHz, the sampling rate of 44.1 kHz was adopted (the extra hundred samples allow for error-correction) for most applications, including commercial CDs. This means you need a clock that can accurately present timing pulses 44,100 times a second or higher. Word clock stability is critical to high-fidelity audio.

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as well: all of your digital audio devices need to be on (or resolve to) one single master clock. This clock tells each device when to move on to the next sample phase; it is critical that all pieces move on simultaneously. Failure to lock your clocks will result in clicking and pops at best, full-on digital noise or no audio whatsoever at worst. Two important considerations when putting your digital studio together: [1] Each product must be able to slave to a master clock; and [2] choose the most stable clocking source you can afford. This is definitely not the place to skimp.

One of the most important issues to consider in switching to a fully digital system is sync.

GETTING IT TOGETHER

In most cases, there's no problem getting your clocks together. Many digital protocols, such as ADAT Lightpipe or AES/EBU, carry the clocking information as part of the signal. It is, however, important to let your devices know (via interfaces or software) to which clock to lock. Standard word clock or super clock (256x sampling rate), as well as black burst (used in video as the clock source to establish the black phase of CRT scanning rates) use a BNC coaxial cable to connect devices' clocks. The preferred cables are high-quality, 75-ohm cables. In a pinch, 50-ohm cables can be used; in these cases, using a terminator at the receiving end of the connection can prevent standing waves in cabling. Otherwise, the standing waves can introduce jitter and unwanted artifacts in your audio.

Finally, make sure all your digital audio devices are set to the same sampling rates. Most products now have at least two available options — 44.1 k and 48 k. Some have many more, including lower sample rates, pull up/down for video drop frame rates, and higher rates, like 96 k. Once you have selected the desired sample rate (most people choose 44.1 k to avoid sample-rate conversion, or SRC, when the project is being mastered for CD), and have made sure that all your connections are in place and everything is locked to the master clock, the rest should be fairly easy. At least until Y2K.

Just kidding.

Randy Neiman is a product specialist for Mackie Digital Systems in Woodinville, Washington. Known as the "Audio Guru to the Stars," he spends a great deal of time helping folks understand the wiles of digital audio. He is also a complete beer snob, so don't offer him any of that typical bar swill. E-mail him at Randy@Mackie.com.

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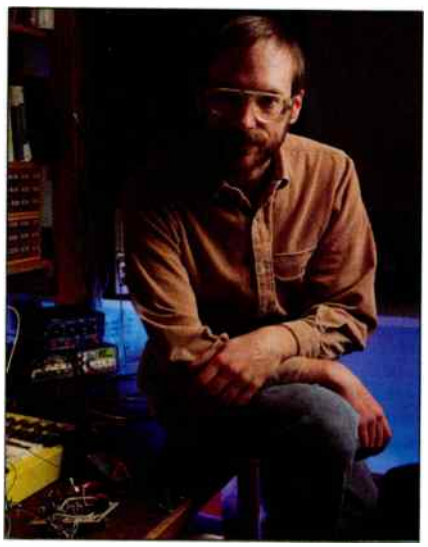
CIRCLE 49 ON FREE INFO CARD

What's the Deal with Digital Mixers?

A look at some of the objections to digital boards, and whether or not they're valid



BY CRAIG ANDERTON

I think it's fair to say that manufacturers have, largely, been disappointed with the level of interest in digital mixers. Frankly, I wasn't interested either until I used one for several weeks in the studio while writing a review, and found that it absolutely smoked the analog console I had been using. I became a convert, and now, I could never go back to analog.

Part of the problem has been manufacturers' inability to elucidate a compelling reason to switch to digital. It seems that almost all companies are guilty of stressing that their digital console is "as easy to use as an analog console." If you already have an analog console, that's not exactly an earth-shattering sales pitch. In addition, most specs simply relate to the number of busses, inputs, stages of EQ, etc. These specs aren't radically different from analog consoles; in fact, judged by these specs alone, digital consoles aren't cost-effective because they typically cost more than analog models with equivalent capabilities.

Nor are most musicians, engineers, or music stores looking beyond the hype to see what digital mixers are all about. In the course of talking with people about the topic, I run across the same general reasons for the lack of interest in digital mixing. So, as a public service to *EQ* readers and digital console manufacturers, let's look at the main objections to digital mixers, and whether they're valid concerns or not.

"I don't like the sound of digital mixers." At one time, that was a valid comment. However, times have changed in digital audio-land. Better converter design, improved bit resolution (and even higher bit resolution for internal processing), oversampling, better EQ algorithms, control smoothing, and many other factors have made all digital audio devices (not just mixers) more transparent and accurate. Compare contemporary digital tape recorders, hard-disk machines, sound cards, CD players, and DATs to offerings from the early '90s; the difference is profound. Those same advances have been incorporated into today's digital mixers.

"I tried earlier ones, and they sucked." The answer to this is partly the same as the previous paragraph, but, in many cases, the complaint relates to user interface, not just sound. Moreover, early devices, admittedly, could be confusing to use. But, again, times are changing. Manufacturers are heeding the call for better interfaces, better support, and better documentation — because, if they don't, customers will seek out manufacturers that do. This doesn't negate the fact that the interface is certainly *different* from an analog console, and, because there are more functions, is bound to be more complex. But there's a flip side to that complexity, because you don't have to learn everything all at once. Initially, you can use just the faders and apply some EQ, like an analog console. Over time, you can get into whatever "power user" functions the mixer offers.

In many ways, you have to realign your thinking somewhat. That's because today's digital mixers are more like musical instruments than simple traffic routers, as you can improve your technique and discover new sounds as you delve further into what these babies can do. Although that's more demanding than just twisting knobs and pushing switches, in my opinion, the rewards are well worth it.

"I like the sound of analog mixers." This is certainly reasonable; you use gear whose sound you like. But it's important to remember that like analog gear, not all digital gear sounds the same. It's important to audition mixers with program material you know well so that you can make an informed decision about which sounds best to you. Personally, I prefer the sound of a good digital console over analog because I can always "accessorize" a neutral-sounding mixer to get the warmth I want (tube preamps are a good start). It's much harder to take a mixer that already has a sound and try to make it neutral.

"I want to be able to see all the controls at one time." For many live applications, this is a totally valid reason *not* to go digital. Digital mixers, because they use relatively costly motorized faders, restrict the number of physical channels to anywhere from (typically) 8 to 24, which isn't enough if you're used to running live mixers with several dozen inputs.

In the project studio, though, it's a different story. Many mixers have some sort of "layering"-based design, where you deal with particular groups of faders (e.g., 1-8, 9-16, 17-24, etc.) at a time. Thanks to automation, you can automate the "set-and-forget" channels, then stick them on a layer and let them hang around in the background. Meanwhile, if you want to actually play some faders, you can assign those to the active, physically present layer.

One big advantage of this scenario is that you don't end up being distracted by a zillion controls you're not using (admit it, how many times have you adjusted a control one channel over from the one you really wanted to adjust?). Also, this approach allows for a more compact footprint, leaving more space for your monitors, MIDI controllers, signal processors, etc.

"I use Cubase/Performer/Logic/Vision/ACID, etc., and having a virtual mixer means I don't need one in hardware." Technically speaking, this is true: these types of programs do indeed allow

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you to do all your mixing on a virtual, on-screen console. However, just because you can do it doesn't mean this is the most efficient way to do a mix.

There are two main problems, the first being mixing with a mouse. Mixing one channel at a time is *way* tedious — an analogy would be trying to record a rhythm guitar part by recording one string at a time on each of six tracks. It's possible, but who wants to work that way? Feeding a sound card's multichannel digital output (typically an ADAT Lightpipe or TDIF connection) into a mixer's digital input lets you use real hardware faders, often with automation (which can typically sync to the MIDI or MTC signal generated by the sequencer). A side benefit of using external faders is eliminating latency; with slower computers, there may be an audible delay in level changes when you use an external hardware box to control the on-screen faders. Even though these changes play back without delay, it can be disconcerting to experience delay while recording.

The second problem with the software-only approach is processor loading. Once you start laying on the tracks and plug-ins, the computer often slows to a crawl. Furthermore, plug-ins have to tread a fine line between sound quality and processor loading. By feeding a digital mixer, you can use the mixer's on-board EQ, dynamics, reverb, etc., thus freeing up the computer's CPU for maximum track count.

Combining a digital mixer as a "front end" with MIDI + digital audio programs gives the best of all worlds: quality audio, a human interface with real faders, minimal processor loading, and the ability to use the software for what it does best — recording, editing, cutting, pasting, and copying.

"I can't afford a digital mixer." While the price tag may initially seem higher than analog models, in many cases you're getting on-board effects, an automation package, a set of D/A and D/A converters you can interface with other digital gear (DATs, sound cards, CD-Rs, digital tape machines, etc.), programmability, and a slew of other features. Add these all up, and even the more costly digital mixers can end up looking pretty cost-effective.

For me, the decision to buy a digital mixer was predicated on the assumption that it would save me enough time compared to analog mixing that it would pay for itself within a year. So far, that assumption is right on track. Just being able to recall presets and automation has made a huge difference in working more efficiently.

As I said at the beginning, once you've actually experienced what it's like to work with a digital mixer, you'll have a hard time going back to analog. Yes, analog still has its place; my trusty Mackie LM-3204 continues to do a great job as a line mixer for my synths and drum machines. Nevertheless, digital mixing has its place, too, and its usefulness will continue to grow in the years ahead.

*Craig Anderton's books (*Home Recording for Musicians, Multieffects for Musicians, Digital Projects for Musicians, *etc.) are available through Amazon.com. To hear sound bites from one of his latest music production projects, surf to www.kac-cohen.com.*

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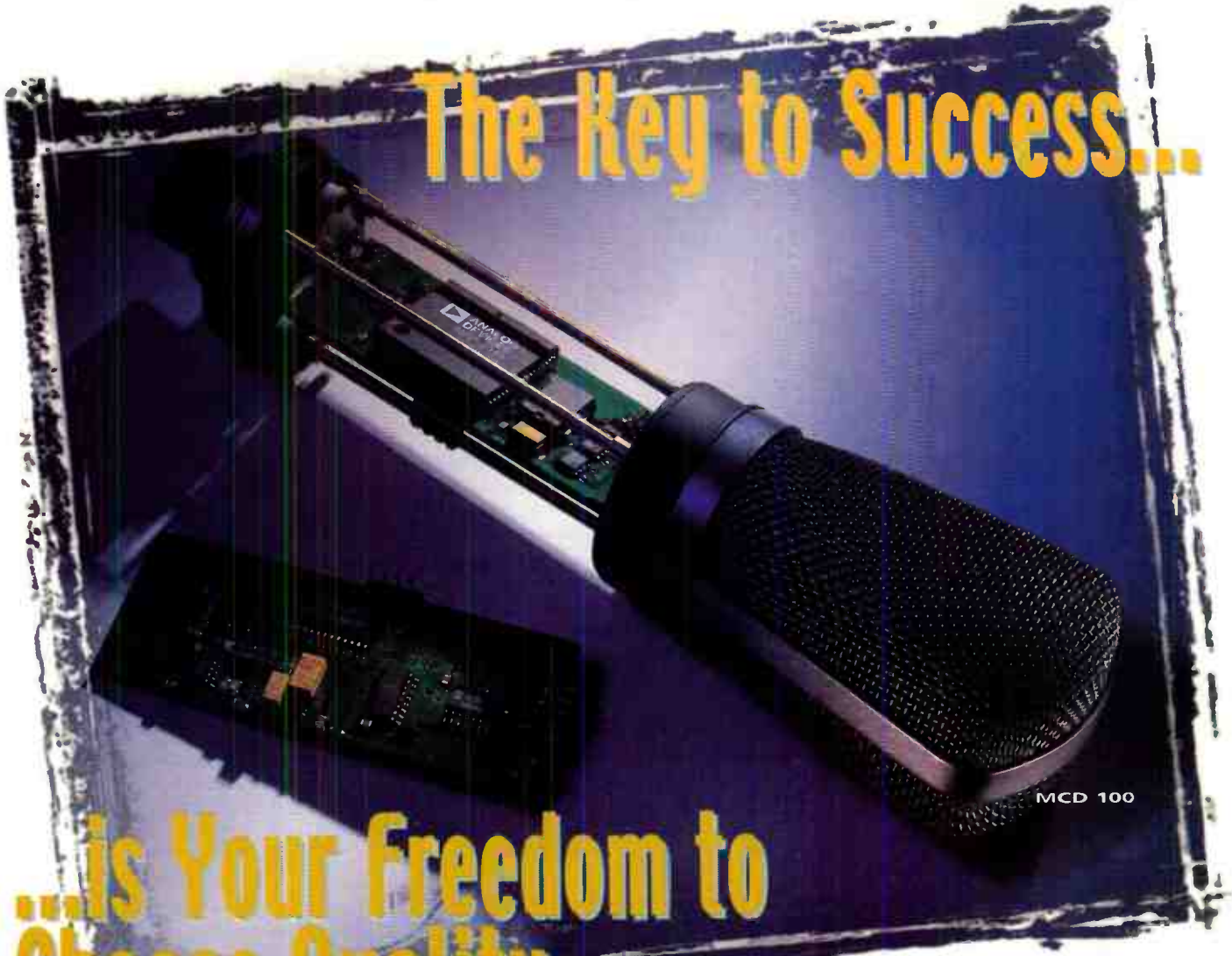

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CIRCLE 10 ON FREE INFO CARD

There's No Denyin' Brian Survivin'

PHOTO BY JONATHAN POSTAL

Finally performing under his own name, Brian Wilson brings the house down

BY AL KOOPER

It was with great trepidation that I earmarked June 21st to see the Boston concert debut of fellow-former-*EQ*-coverboy Brian Wilson in that classical bastion, Symphony Hall. Now, more than a few of you might denigrate the value of the Beach Boys legacy in popular music, but you'd have to answer to me. I was never suckered into the Pendleton shirt wardrobe, the general appearance of the mayonnaise-headed performers themselves, or some of the lyrics, but I was extremely moved by the musicality of it all. The composer cleverly interspersed jazz chord changes over doo-wop structures and thick, never-heard-before voicings in the background vocals. I sat up and took notice. Slowly, the instrumentation caught up to the rest of these divergences, and soon cellos, bass harmonicas, harps, and flutes adorned these tales of teen angst. Then the lyrical content began its admittedly slow ascent to the top, and the package became complete.

This was no ordinary surf group — believe me. I knew my slavish devotion had finally paid off when the almost-classically constructed instrumental introduction remained unedited on the hit "California Girls," and I was completely vindicated with the release of Brian's masterpiece, *Pet Sounds*, in 1966. As a matter of fact, I was invited into Brian's home two weeks before the release of said album for an advance solo listening party. It was the era of the piano-in-the-sandbox and the soda fountain-in-the-living room. This New Yorker was bowled over by the music, and felt like he was in the company of a latter-day Mozart or Beethoven (the Beethoven comparison is closer, as Wilson is stone deaf in his right ear). *Pet Sounds* scared the Beatles into creating *Sgt. Pepper*, but their creation was a pale second to their original muse. Today, 33 years after *Pet Sounds*' release, I still can't figure out the opening chord changes to "Don't Talk, Put Your Head On My Shoulder."

Today, 33 years after Pet Sounds' release, I still can't figure out the opening chord changes to "Don't Talk, Put Your Head On My Shoulder."

But Brian had his share of setbacks. Like any great genius, he was belittled and misunderstood, and, with aggravated drug use, it all came crashing down around him. Misplaced in the custody of some Dr. Nick-like shaman, he slipped further and further away from the creation of his former glories. His long-suffering wife left with the kids, and it seemed to the faithful fan like he/she would never hear the likes of that musical Camelot again.

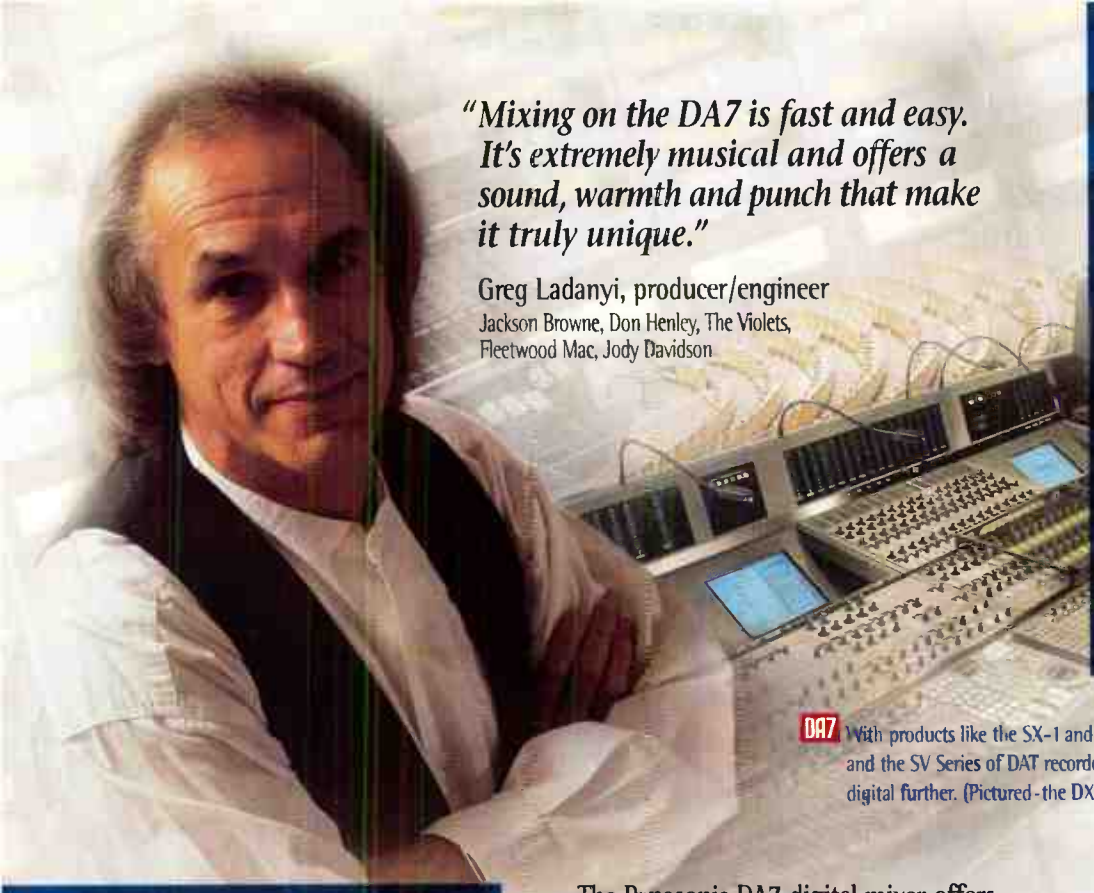
And that is why, as I walked into Symphony Hall last month, I silently prayed that everything would be OK. That Brian would surmount the recent deaths of his brother Carl and his mother Audree, leaving him the sole- (soul) surviving Wilson.

As the house lights dimmed, a makeshift screen on the side of the stage unspooled a 20-minute cinema-verite video account of what I just wrote in the preceding paragraphs. The image transformation of the bright-eyed strapping youth on Dick Clark's '60s TV dance party, *American Bandstand*, into the foreboding Broderick Crawford look-alike of today was instantly sobering. But Brian was now situated in a new, sane marriage replete with two infant children and a new-millennium grin that seemed to never quit. The lure of audience-interaction was finally too much for the Master. The screen was taken away and the audience rose for its first of many standing ovations, as the man himself strode to center stage.

To open, he chose the 1964 oldie, "She's Not The Little Girl That I Once Knew." In retrospect, gender aside, it seemed like the perfect introduction lyrically, in a vaguely analogous way. His vocal range has been decimated by the years, and he chooses to shy away from his trademark falsetto, leaving it for the other nine singers on stage to sort out. The entire show was set up as a fans' fantasy set list. "I Get Around," "California Girls," "Caroline No," "All Summer Long," "God Only Knows," "In My Room," and the other 24-or-so chestnuts were precision-like replications of the original recordings, obviously lovingly assimilated by the orchestra and choir. Even the two instrumentals on *Pet Sounds* were trotted out successfully.

Not only did this evening not suck, but it was actually a watershed moment in time that I shall never forget. In the appropriately selected Symphony Hall, I sat with tears in my eyes and triumph in my heart. It seemed, by the response after each song, that each and every audience member in the sold-out concert hall shared the same feelings, adoration, and respect, and that we were all in the same secret handshake Brian Wilson Society. At $65 a ticket, we got more than our money's worth. In our senior years, we will surely take our great-grandchildren on our laps and tell them how we saw the great composer Brian Wilson perform all his masterpieces for the first time under his own name.

That is, unless Brian literally outlives us all the way he's survived his entire nuclear family. And I'm certainly not betting against that eventuality... EQ



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James West

PHOTO BY ED FREEMAN

"Weird Al's" guitarist scores with his project studio on the beach

BY MR. BONZAI

Mr. Bonzai: How did you happen to meet up and work with Al?
James West: There was this contest for the world's stupidest guitar solo. The winner got the gig with Al. As fate would have it, I won with a twisted rendition of "Mary Had a Little Lamb."

No, it wasn't like that at all. A former bandmate of mine, bassist Steve Jay, hooked up with Al on his first album. When Al was putting together a live band, I got a chance to audition. I've always considered it an extremely fortunate and bizarre twist of fate.

What is the most important thing you've learned from Mr. Yankovic?
Never to take myself too seriously. And how to avoid the dangerous objects that people throw at us when we perform live. It certainly doesn't help things that he's named his new record *Running With Scissors.*

What are your main recording tools here in your project studio?
The main brain is a G3 Mac with StudioVision, Pro Tools hardware, three SampleCells, and various plug-ins. Most of my stuff starts out MIDI, even if it's just a click, so I tend to record direct to disk because it is so convenient. I do have DA-88's as well. I have the obligatory rack of synths and samplers — Roland 760's, 1080's, 990, EMU stuff and others, and way too many CD-ROMs of sounds. I love my Neve and API preamps and EQs, and my ADL and UREI compressors. As far as mics, I've been using a Lawson M-47 on my acoustic guitar — it's very cool. For reverbs I've got Lexicon PCM91, PCM70, and 200. For effects, an Eventide H3000/DSE, a Boss SE-70, an RSP unit, and a Zoom thingamajig.

How did you learn about engineering?
I started playing with tape recorders when I was about ten years old. My older brother, Allan, had a Webcor reel-to-reel that I would fool around on — you know, record the toilet flushing and play it back over the phone sort of prank stuff. Once I got to college, my friend John Esak and I would sneak into the electronic music studio at two in the morning to do our band demos. He sort of knew what he was doing, and I picked up a bit from him. Later on I got a TEAC 4-track and learned how to get

Suspect: James "Jim" West

Ancestry: English, Canadian

Occupation: Guitarist, composer, producer, arranger, engineer

Birthplace: Toronto, Canada

Residence: Hermosa Beach, CA

Vehicle: Plymouth Minivan. "Utterly theft proof!"

Diet: Omnivore

Identifying Marks: Guitar pick surgically implanted on first finger

Pet Peeves: "This is more than a peeve: big media companies that are taking writer's royalties for themselves. Oh, and you can imagine the endless comments about the *Wild, Wild, West* character, like 'Where's Artemus?' "

Credits: West has recorded ten albums as guitarist with "Weird Al" Yankovic — most recently the new release, *Running With Scissors.* Al's previous album, *Bad Hair Day,* was certified Platinum. For TV, he has scored *The Weird Al Show, Iron Man,* a Cousteau documentary, and the films *Lady Dragon, Blood Warriors,* and *Land of Dreams.* West's commercial clients include Honda, Nike, ESPN, Nestle's, Yamaha Motorcycles, NASCAR, and Straw Hat Pizza.

Notes: West and his wife, Jan, have a nifty little studio in a 100-year old redwood beachhouse, and a second home on Maui. An accomplished slack key guitarist, his new solo album of Hawaiian-inspired tunes is *Coconut Hat* on Westernmost Records.

JAMES WEST PHOTOS BY MR. BONZAI

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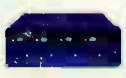

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a good sound by bouncing between two machines. Since then, I've had many opportunities to watch real engineers in action, and I always learn something valuable from them. I also learn tons from reading recording magazines.

Who was your most influential teacher of music?

Well, I was a visual arts student and never formerly studied music or guitar. My older brother showed me a few guitar chords, but most of what I know I learned by ear and through music books and watching guitarists play on TV. I definitely was exposed to a lot of interesting music through some of my bandmates who were music majors.

Most of what I've learned has been by ear, starting with cover bands and eventually Weird Al, learning tons of guitar parts for his parodies. Once, I had a job re-scoring about a dozen classic Fox films (*Diary of Anne Frank, Journey to the Center of the Earth*, etc.) for foreign re-dubbing. I re-created all the music under dialog, much of which was full orchestra, by ear with my sampler. It was a great study of composing and orchestration — the scores being by people like Alfred Newman and others.

What are your favorite guitars?

As far as electrics, I use a couple of Tom Anderson guitars. His are by far the best I've ever played. I also have a small vintage collection, including a nice old Silvertone [manufactured by Sears], a

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Danelectro, a '50s Martin Tenor Guitar, 1938 Rickenbacher lap steel, a great '60s Guild acoustic, and a couple of Martin ukuleles. The other day, Martin Simpson showed me his Stefan Sobell acoustic and I *have* to have one! It's the most beautiful-sounding guitar I've ever heard.

What do you listen to while you're driving?

L.A. public radio stations like KCRW and KPCC, with a generous dose of freeway mayhem as a background.

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

The first human singing would be cool — I could sample it and use it without any hassles. I'd also like to hear the sound that gigantic meteor made when it hit the earth and caused the Ice Age.

Is there anyone in the world you would like to record with?

That's tough — there are so many. Right now I'd love to play with any of the great slack key artists: Ledward Ka'apana, Cyril Pahinui, Ozzie Kotani, to name a few.

What is the secret of Hawaiian slack key guitar music?

It's harmonically simple, but the great players play from the heart. It's a deeply felt music.

What's wrong with the music industry?

They need to wake up to the fact that the important artist they just rejected is now selling his music via MP3, and they are out of the loop. See ya.

What is your strangest characteristic as a human being?

I get very uncomfortable and sometimes twitch uncontrollably when there is no guitar around to play.

How would you like to be remembered in history?

To be remembered by a melody or two would make me happy. Or as that Canadian guy who played with Weird Al and composed a bunch of music and played slack key guitar a lot .

What music would you like played at your funeral?

"It's My Party" by Leslie Gore. Anything but "Stairway To Heaven."

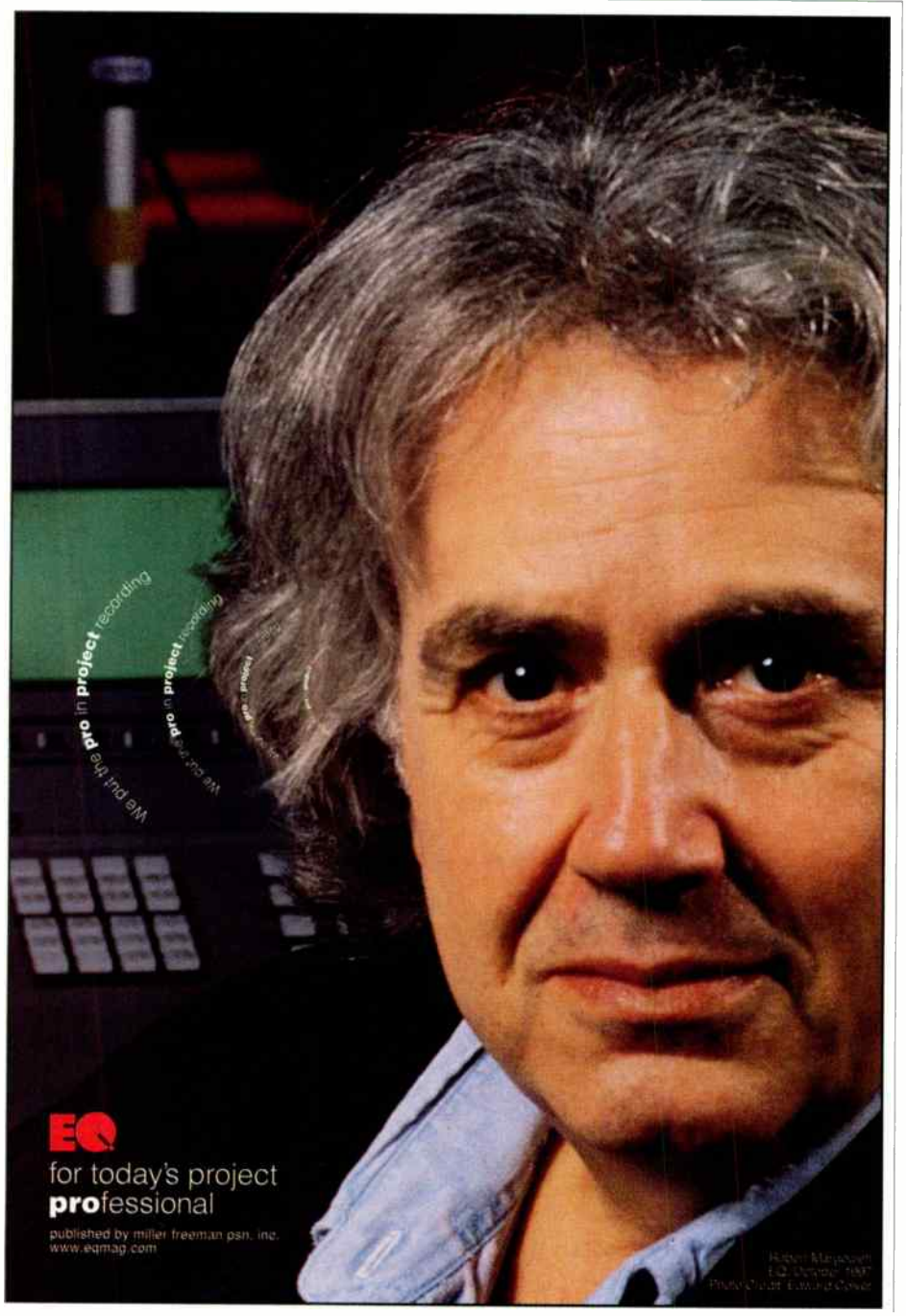
Do you know any interesting business tricks?

Use one of those voice-changing boxes when you answer the phone to make clients think you've got all kinds of receptionists and secretaries and such.

Can you recall your most ridiculous experiences in a recording studio?

Al had his dentist show up with a huge drill and a human tooth so that he could record the ultimate tooth-drilling torture for one of his parodies: "Cavity Search." For another Al session, the band was called in to do some backing

continued on page 137



What To Do? Build Bus'tr

Here's a quick and easy way to sum two audio signals into one

BY MIKE SOKOL

We've all been in a fix from time to time when we needed to sum two audio signals into one. Most of the time, it involves taking a stereo signal and turning it into a mono feed. The other situation is when you need to add the outputs of two mixing consoles together, summing the FX, left, right, and aux levels so that the two boards act as one large mixer. The quick (and incorrect) fix is simple: use "Y" cords, jamming two outputs into one with variable results. If the output drivers of the gear to be summed have build-out resistors, sometimes this will work. However, most professional and pro-sumer gear is designed around a 600-ohm output stage. If you attempt to sum together most pieces of gear this way, bad things can happen, the least harmful being a level imbalance between the two pieces of equipment. In some cases, severe oscillations and distortion can result, with possible circuit failure.

Mackie manufactures a very nice active bus mixer called Mixer Mixer, which, for a few hundred bucks, does individual buffering and allows master volume controls to be added where desired. Bus'tr isn't quite so fancy: I designed it as an easy-to-build passive mixer in an aluminum project box with 1/4-inch TRS circuit jacks and 680-ohm build-out resistors.

To use Bus'tr, you just hook two outputs of anything you want to mix together into two of the jacks and the signal comes out the third jack, simple as that. Don't try to mix mic and line-level signals together, since there's no way to match the signal levels. The pair that's to be summed needs to have similar levels.

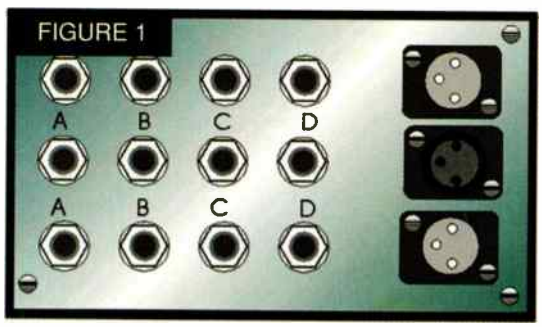
You can modify Bus'tr to sum as many separate channels as you like; from two to infinite sets are possible. Don't try to make it something it's not by modifying it to allow more than two signals to be summed into an output—the losses become too great very quickly. In that case, buy the Mackie unit. I like to make my Bus'trs with eight sets of inputs, allowing me to bus all of the functions of two mixing boards. Since this is a dirt-simple passive mixer, there will be a 3-dB insertion loss because of the passive summing, but you've normally got lots of extra signal to play with, so losing a little output won't hurt a bit.

Notice that all three jacks (see diagram) have build-out resistors, allowing any two jacks to act as an input. In addition, since this is a purely passive circuit, you can use this box to route a single channel into two outputs, while adding 680 ohms in line. This won't affect the signal at all, except under some conditions of very long line runs.

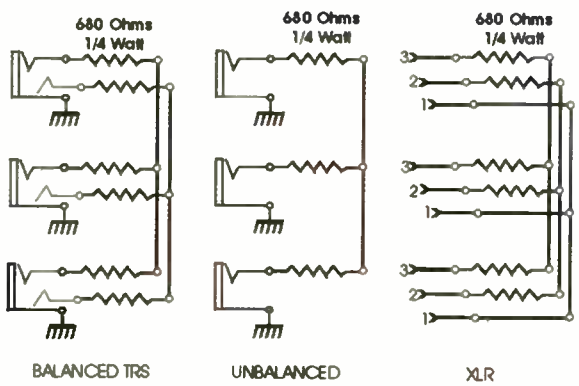
You can substitute RCA jacks for the 1/4-inch jacks if desired, just be sure that the equipment you're hooking together is rated at 600 ohm or less output impedance. You can make higher impedances work by substituting a large resistor value, but I've never encountered a modern piece of musical gear where this is necessary. I've also included a diagram to allow XLR jacks to be summed, and, yes, this will also work with microphones in a pinch.

I keep Bus'tr handy in my gig box for those times when I've got to jack in a second board in an emergency, feed a press mult, or make stereo into mono, and it has never let me down. There's no batteries to wear out, no possibility of overloading active circuits with excessive signal levels and no induced noise. Plus, it's cheap to build. What's not to like? EQ

FIGURE 1

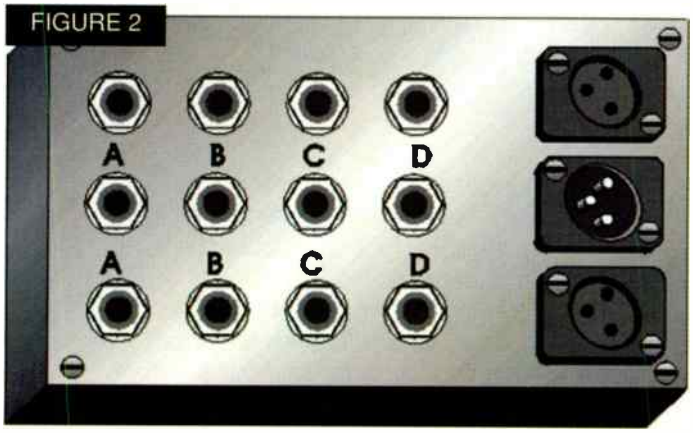


BUS'TR
TOP VIEW



BUS'TR
SCHEMATIC DIAGRAM

FIGURE 2



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Froom with a View

A walk through the creative mind of producer Mitchell Froom

BY HOWARD MASSEY

Mitchell Froom is one record producer who marches to the beat of a different drummer. Working with engineer Tchad Blake and a diverse crop of artists that includes Sheryl Crow, The Pretenders, Los Lobos, Del Fuegos and wife Suzanne Vega, Froom is the man behind some of the most individualistic, off-kilter recordings of the '80s and '90s — recordings that seamlessly merge the musical traditions of the past with modern technology. We recently visited with him in his project studio in the middle of the meatpacking district (really!) on NYC's West Side. Surrounded by his impressive collection of vintage keyboard and percussion instruments, Froom discussed his philosophy and approach toward making records.

These days, so many demos sound like masters and vice-versa. Where do you see the demarcation between the two?

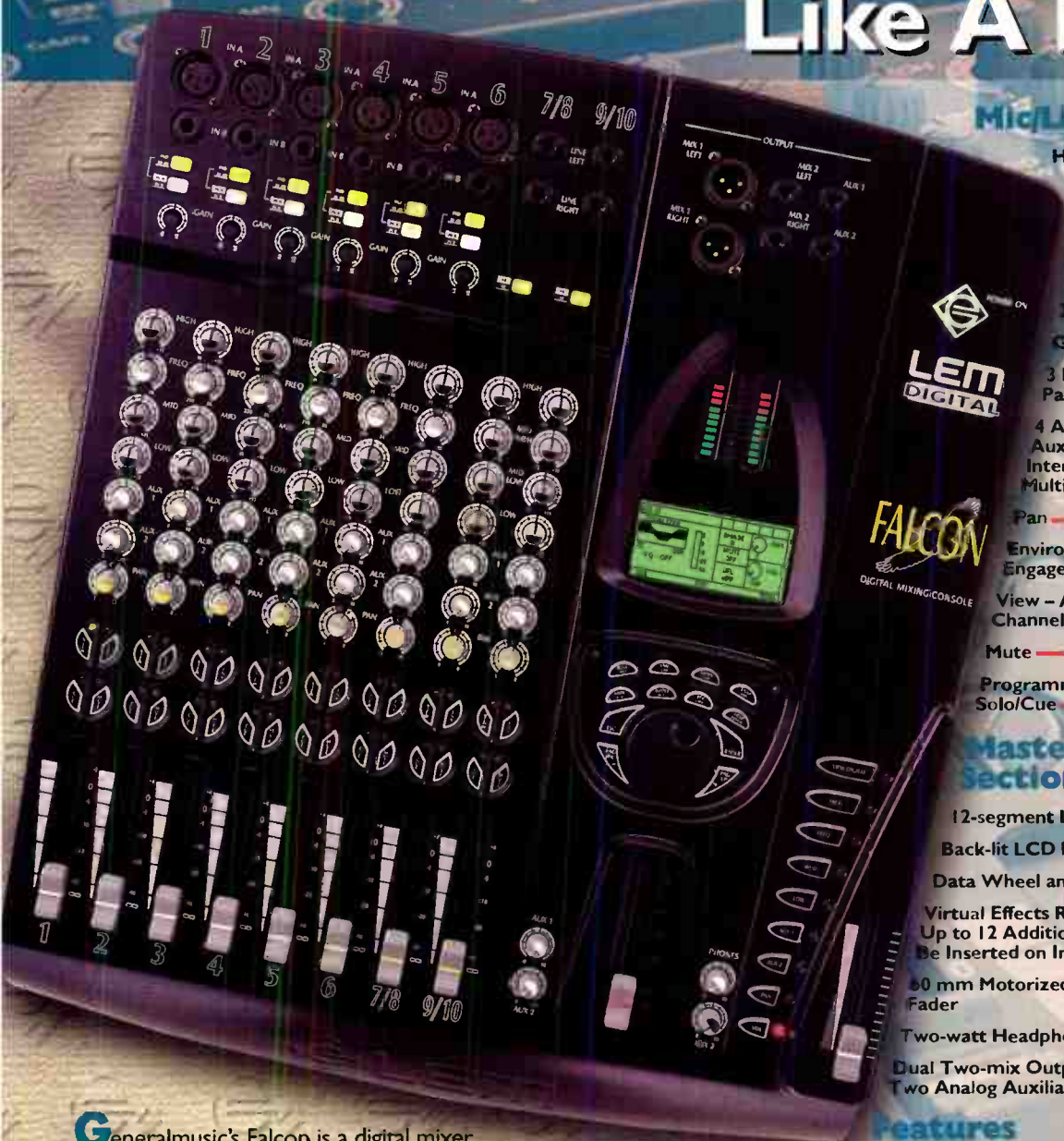
Mitchell Froom: One of the most popular records I ever worked on was a demo recorded on a badly aligned 4-track cassette. However, [the music] was so beautiful, there was no way I could suggest that it be re-recorded. This was the first Latin Playboys CD (*Latin Playboys*, Slash/Warner Bros., 1994), with David Hidalgo recording a bunch of music at home in his kitchen, at night, just using guitars, spoons, drum machines, stomp boxes, whatever. The result was probably the most interesting tape I've ever heard. It didn't have any singing on it, so we decided to bounce the 4-track to 24-track and overdub on it. The thing about 4-track cassettes that's wonderful is that they have a very cool sound. They have a fantastic low end and a lot of character — the tape hiss, the slight warbly quality. It's instant atmosphere, built into the machine. Even if you were thinking in terms of releasing your own record, I'd still use a 4-track cassette and something else. I happen to like hard-disk products for their editing capabilities. You can record onto the 4-track and bounce that to the hard disk to edit, maybe record clearer vocals on the hard-disk system. You can make a fantastic record for almost no money if you have the time, the knowledge, and the desire.

And these days, low-fi is hip again.

I think for awhile it became *too* hip. I find it disappointing when people say, "That's slick, so it stinks; this is cool 'cause it's so rough." You know, the "I'm so cool because I use cheap equipment" school of thought. In many cases, things sound rough because the artist is unaccomplished — because the person's not ready to be making records. It makes me long for the jazz days, where musicians would jam in clubs after hours. If you didn't cut it, you'd get thrown off the stage. You'd have to go home, figure out what you did wrong, practice, then come back. So before you ever got to record, you were at a certain level. You had to develop yourself; you couldn't just strum a chord, sing badly, and think it's cool because it's rough. That doesn't mean anything in the real world.

PHOTO BY TCHAD BLAKE

A Digital Mixer That Works Like A Mixer



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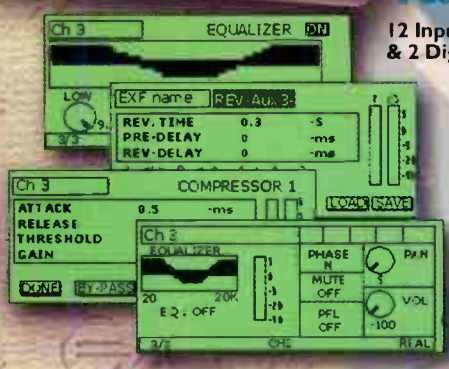

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What *is* the point at which someone is ready to make a record?

It's the point at which you have something with a distinct personality to offer. For the singer/songwriter, it's when he starts to come into his own. You know when you're hearing a true individual. For a band, it's the way that you play together. There has to be something unique about it, something that makes you stand out from the crowd.

One of the great myths of the music business is that if you have a great song and a great performance, it doesn't matter how well it's recorded.

It's a huge myth. As a producer, I want to be involved with the definitive recording of any song on which I work. I want to make the hit record of the song. And making a hit record is very deceptive; it's not an easy thing. It's not just getting good sounds and getting a strong performance. A hit record is the mystery of it, discovering something, and having something that's surprising that cuts through the airwaves and makes people want to hear it again.

So where does one start in pursuit of this Holy Grail?

A lot depends on the kind of artist you are. What I look for [in an artist] is his or her eccentricities and strengths, and sometimes that's difficult for someone to see in himself. If what you're about is words — say you're not a great singer or a great guitar player — you want to make sure that the lyrics come off in the strongest possible way. So perhaps you'd do things as sparse as possible. Vocally, you want to make sure that the song is written within a range where you can be convincing. Learn about mic techniques; find out how you can record your voice in a way that's attractive. When Tchad [Blake] and I work in the studio, 99 times out of a 100 there's only one effect that we ever use on a vocal, and that's a little analog slap — just a little stomp box pedal analog delay, to different degrees, different widths, or whatever. That's all you need to fit a vocal into an environment. If it sounds great dry, leave it dry. The key is to search for what makes you different and accentuate it.

One thread that runs through most of your productions is an intimacy in the sound. Your records tend not to be bigger than life. It always feels like there's a bunch of musicians hanging out in the room with the listener. How do you achieve this?

Well, you start by putting everybody in a room together! [*Laughs.*] Popular music at its best, if it's going to be artistic, is a form of a folk art. Bombastic pop music is about the most ridiculous-sounding thing to my ears that I can imagine; you have people singing love songs and they're singing from the top of a mountain, which is the least convincing way of conveying an emotion. Somebody screaming at you that they love you...you're gonna want to run away from them! So we don't place music on a big stage. It's not Beethoven; it's not Tchaikovsky. To me, the word "big" conveys big tones that are almost confrontational.

Do you use a lot of close miking?

I use some. The drums are usually in a very small space, and Tchad will use some fairly wild compressors to get a sense of ambience and power, though he doesn't use very many microphones. Often we'll have the drummer not play too loud. It's important that the drummer puts together a kit that's right for the song. But we don't go for the classic, modern, punchy drum sound with everything perfectly separated. We look at drums more like the way they used to be viewed — that they are part of the group. We try to get at least two or three people playing at the same time, if not in the same room together, at least able to see each other. Sometimes it will just be a drummer and a guitar player, sometimes with bass, live singing, and keyboards.

Bombastic pop music is about the most ridiculous-sounding thing to my ears that I can imagine; you have people singing love songs and they're singing from the top of a mountain.

Even your final mixes sound very much like it must have sounded during live tracking.

But the whole thing is still like entering a kind of fantasy world. We're trying to express emotion. We're not trying to make it sound authentic like you're in a club and you're watching a band. There's no way you can approach the experience of watching people play live in a club. There's not the dynamic range, you don't have the music bouncing off the walls, there's nothing to look at. You have to embrace the studio to create an emotion from sound.

As a keyboardist, you have an obvious predilection for older, analog instruments. Do you use a lot of older analog recording gear as well?

I use anything that's full of character. We don't necessarily use the latest microphones. I'm very happy to use a piece of modern gear, though. Sometimes we'll do a rough mix to DAT and it will sound better than an analog mix. I don't necessarily love the fact that something's analog. In fact, I sometimes curse the fact because analog gear takes up so much room. I wish that the 800 sounds in today's keyboards were all great, 'cause that would make my life much easier!

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

The most important thing is to have good instruments. If you've been playing one guitar for your whole life, just keep playing that guitar. You shouldn't think you have to change to a different one just because it's more fashionable. That guitar you have probably sounds better in your hands than a brand-new model will. I'd also spend money on little stomp boxes, things that might do radical things. You might want to get a 4-track cassette and some kind of digital recorder, so you can move between lower and higher fidelity in an easy way. Get a few mics that you like the sound of, that function well in a number of ways.

Any mics that are personal favorites?

Well, I like really rotten mics...like the little square things that used to come with old reel-to-reel machines. They distort and have a limited dynamic range, but they have a really cool sound. I would always pick up a cheap mic and see if it sounds good on

something by chance. On the other extreme, the only microphone I personally own is a Telefunken 251. It's an incredible microphone, but I don't use it much anymore. I used to use it almost exclusively on vocals and acoustic guitars. It's beautiful, very big, with a very natural high end.

Any favorite signal processors?

One piece of gear I'd definitely recommend is a compressor called the [Empirical Labs] Distressor. It's really great for just about everything, and, if you had to use just one compressor, you could probably get away with just using that.

Any tips for recording keyboards?

I don't usually use amps for recording keyboards. I usually go direct. If, however, I want an amp sound, I use the [Tech 21] SansAmp Classic. It provides a lot of variables so you can get an amp-like sound without the problems of having an extra amp in the room. Sometimes I'll run stuff through a Leslie. I do that quite a bit. And I use a lot of stomp boxes — anything I can find. I just plug it in and see if something happens. Sometimes the buzz [from the stomp boxes] is a big feature! It's the glue that holds it together, something that we usually find really attractive. I don't worry about buzz and hiss. You have to trust the moment when something sounds great to you.

How do you record bass?

We always record DI and usually also a SansAmp track. So we don't use an amp, but we get that growl, that midrange kind of thing, sometimes with just enough distortion to make it pop out better.

Do you record with effects?

I often do, if it's possible to keep them on separate tracks. But things like the reverb coming from a guitar amplifier — you always want to record with that, because it sounds completely different than the reverb an outboard processor would add later. The processor's reverb will have all these high frequencies that don't exist in the guitar amp reverb. The reverb sound from the amp is much more attractive simply because it's coming through a guitar speaker. It's a better sound for a guitar.

Any mixing tips?

If you view a recording as living in a kind of fantasy world, you can make your music more emotional if you are prepared to pan things in a radical way. Don't put one guitar just a little bit to the left and another one just a little bit to the right. Go all the way! Use far left, far right, center, and don't mess very much with those areas in the middle. Panning something hard left or right can also allow you to hear the character of the playing more. It's good to try to be bold with panning.

Is there an order in which you bring instruments into the mix?

No, I don't fuss and fret over a mix. If we're doing an album, I'll usually do two or three mixes a day. There's almost a degree of improvisation to what Tchad and I do; we don't feel bound by any rules. We try to get into where the groove is and how to project the song in a straightahead way. Also, we're very attracted to things being slightly out of balance. You want to push that threshold, because, if something's "too loud," it often sets the groove off in a whole different way. A lot of Sly Stone records are like that. There's just this great feeling going on although it's completely askew. Oftentimes it ends up that the rough mixes are the final mixes. There's no big problem in that. It's a tough thing to explain to artists, because they want the most for the music and they feel that they have to go to the absolute limit, but that's not true. You just have to get it really good.

Many people go past the point of diminishing returns and end up sabotaging themselves...

Yeah, it starts getting smoother around the edges and losing everything that's vital. If you can't get it going really fast, go to a different song, do something else. I made this mistake a lot when I first started — just tucking it in, getting it more and more precise, and the results were absolutely terrible. I would work that hard if the result was better, but I've never heard a three- or four-day mix that sounded any good.

How do you feel about the trend toward people recording themselves in their own studios?

It's a very good way to start and to learn. But because so much is written about it, there's all this common information out there, so you don't get as many distinctly characterful recordings. You used to have to go to Memphis to get one kind of sound, and to a different place to get something else. Now you get the same sound everywhere [because] everybody's got the same equipment. These days, I couldn't tell you where anything is recorded. My deepest wish is that when somebody hears one of my records, they say, "Where was *that* done? Siberia?" EQ

Champagne Sound on a Beer Budget

Pete Woodroffe creates aural euphoria with Def Leppard

BY ALAN DI PERNA

"Right at the beginning, we decided to do a real Def Leppard album," says Pete Woodroffe, the band's engineer and co-producer. *Euphoria*, the new Def Leppard album, is something of a calculated risk for the band. It's a return to the big '80s production values of the Leps' multiplatinum discs *Pyromania* and *Hysteria*. It's a sound they'd abandoned on their prior album, 1996's *Slang*, opting for a more '90s-friendly grunge style of rock. But in the wake of indifferent sales on *Slang*, not to mention the death of grunge, the band decided to go back to what it does best — high-gloss hard rock with mile-high guitar sounds and lavish layers of creamy vocals. They're betting the farm that the public has regained its appetite for such rich fare.

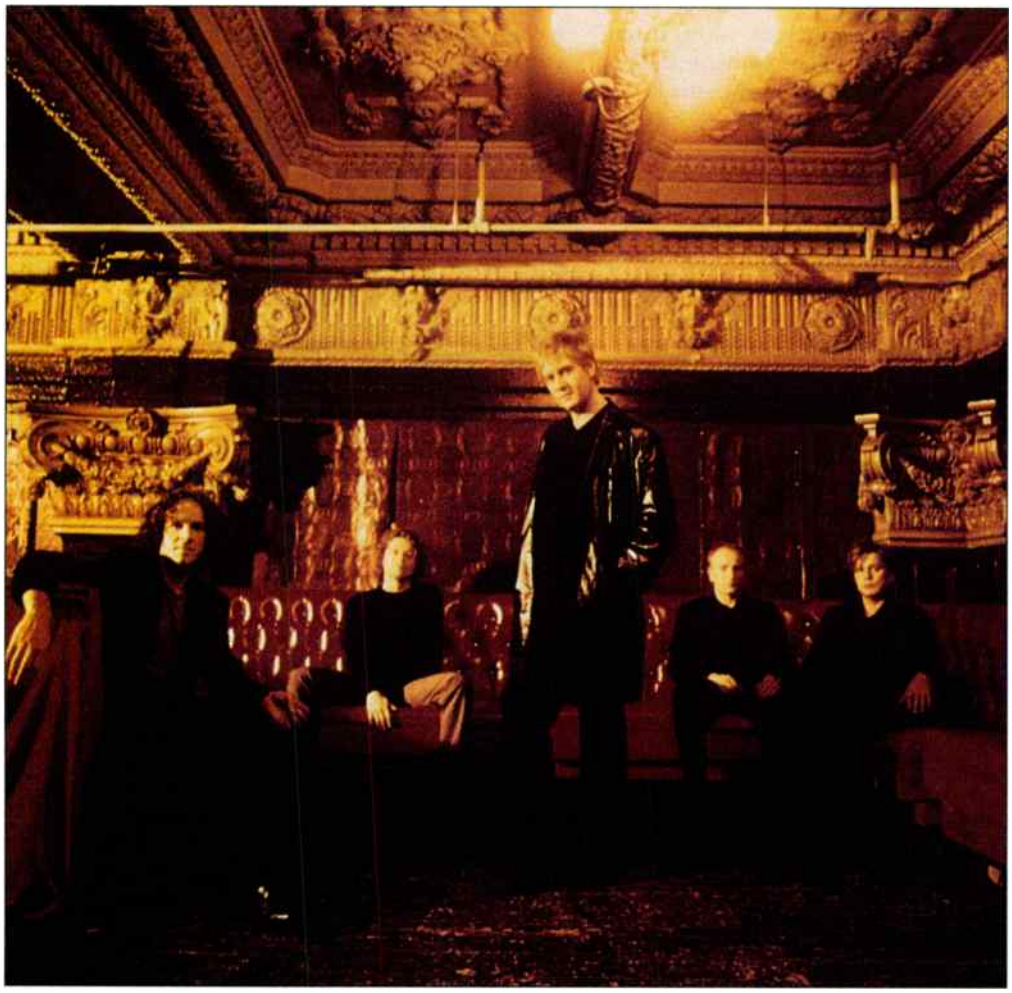
The challenge for Woodroffe was to re-create those big-budget sounds in lead singer Joe Elliott's modest project studio in Dublin. "The main room downstairs has an Amek desk and Yamaha NS10M's," says Woodroffe. "No big monitors. Mid-level gear. The multitrack we used was a Mitsubishi XZ80, which is a slightly old-fashioned, 16-bit machine that isn't being made anymore. If we'd spent $35,000 at the beginning on new tape machines, it would just have seemed too expensive. We were trying to make the album cheap: Do it at home, put the hours in, but don't spend a lot of money. Just in case the album sold nothing."

Woodroffe first began working with Def Leppard on 1992's *Adrenalize* album, engineering for producer Mike Shipley. "Mike promised me the album would take six months to make," Woodroffe recalls. "It took three and a half years."

Creating that big Def Lep sound is a highly labor-intensive process. Each vocal harmony and guitar part is double tracked anywhere from one to 16 (!) times. But for the *Euphoria* project, Woodroffe and the band devised a few strategies for economizing their time in the studio and completing the album more quickly than usual. One was to set up a two-studio system at Elliott's house. In addition to the main studio downstairs, an upstairs bedroom was converted into a second studio.

"It's just a little room with a Mackie 8•Bus desk and [Alesis] ADATs," Woodroffe explains. "We'd do backing vocals, lead vocals, and the occasional guitar track upstairs, while downstairs we were mainly doing guitars and some vocals. There was kind of a friendly competition between the two rooms: who could be more productive."

The other great time- and labor-saving device on the project was Woodroffe's 64-track Digidesign Pro Tools Mix Plus system, which was running on a Mac 9600 that was set up in the downstairs studio. "I have two 8-GB drives," says Woodroffe. "So it was 16 GB, and on that I had the entire album — every overdub, every vocal, every lead vocal outtake, every bass outtake. Pro Tools made it possible to pick between takes really quickly and easily. With lead vocals, especially, you could bounce things down very quickly. You're not stuck with six



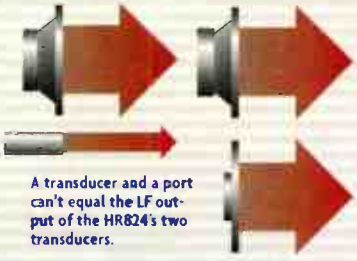
PHOTOS BY DANNY CLINCH

BLAST FROM THE PAST: For their new album, *Euphoria*, Def Leppard went for the sound that made them famous in their '80s heyday.

NO WONDER THE HR824 HAS THE BEST BASS RESPONSE OF ANY 8" MONITOR. IT'S REALLY A 12" MONITOR IN DISGUISE.

TIGHT, RESPONSIVE BASS FLAT DOWN TO 39HZ. Reviewers and owner's warranty card responses are unanimous: The HR824 has the most accurate bass they've ever heard from an 8-inch monitor. And bass **quality** is as astonishing as the quantity. Fast low frequency transients like kick drum slaps and electric bass notes have a crisp, articulation that makes other monitors sound like mush.

ANOTHER TRANSDUCER INSTEAD OF A PORT. The more LF transducer cone area a speaker has, the more bass it can produce. But a huge low frequency transducer isn't an option on a compact near field monitor.



A transducer and a port can't equal the LF output of the HR824's two transducers.

To augment primary bass output, other monitors resort to using ducted ports that can convert cone movement into extra low frequency air movement. But for optimal output, a ducted port needs to have the **same area** as the low frequency transducer — an 8-inch near field monitor would need an 8-inch vent. Needless to say, you haven't seen any vents this big on other near field monitors. When vent size is reduced, bass output is compromised. And, forcing a lot of energy out of small ports can create audible wheezing and whooshing.

Instead, the HR824 adds a large passive transducer with the cone area of another 8-inch woofer. This ultra-rigid, honeycomb laminate piston tightly couples with the HR824's active bass transducer. With a combined cone area greater than a single 12-inch woofer, you get exceptionally extended bass without port noise complaint.

MASSIVE POWER THAT WOULD PROBABLY POP A PASSIVE MONITOR. Punching out crisp bass requires a lotta watts. The FR Series™ high-current bass amplifier module inside the HR824 delivers a solid 150 watts of power with peak output in excess of 250 watts (plus another 100 watts for mid and treble). That's significantly more than any other 8-inch active monitor. Moreover, the HR824's servo coupling and ultra-short signal path put that power to work far more effectively than a passive monitor and a 250-watt stereo amp could.

PART OF A TIGHTLY-INTEGRATED SYSTEM. Our servo bass system is only one contributing factor to the HR824's amazing accuracy.

Internal power amplifiers are "fed" by phase-accurate, low distortion electronic circuitry instead of a crude coil-and-capacitor passive crossover. The HR824's proprietary logarithmic wave guide not only widens treble dispersion but also smooths the midrange transition between high and low-frequency transducers. Thanks to the wave guide's flaring design, the HF transducer's output is **acoustically the same diameter** as the LF transducer's at the critical 3500Hz crossover point.

Rear view: The HR824's electronics conceal an ultra-rigid, honeycomb composite passive transducer.

The HR824's LF transducer even contributes to midrange accuracy. In many monitors, woofer cone harmonic vibrations bounce around inside the enclosure and then exit through the thin woofer cone. The result: smeared imaging and muddled details. Instead of a chintzy chunk of fluff, the HR824's enclosure is utterly packed with high- density absorbent foam. Cone vibrations go in, but they don't come back out.

DON'T SKIMP. It's amazing how many studio owners will mortgage the farm for money-is-no-object, esoteric microphones... and then monitor on cheap, passive loudspeakers. If you aren't using *some* brand of ACTIVE near field monitors you're seriously compromising your creative product

HEARING IS BELIEVING. We urge you to visit your nearest Mackie Designs Dealer and carefully audition all of their active monitors with some demanding, bass-rich program material. Judge our claims (and those of our competitors) for yourself. We think you'll agree that the HR824 is truly the best of the best.

Last fall we won the pro audio industry's coveted TEC Award for best near field monitor. Modesty prevents us from listing the impressive field of competitors but you'll probably encounter their ads in this very magazine.

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tracks of tape that you have to bounce onto another track. It's all in Pro Tools. At this point, it would be very hard for me to go back and make an album without Pro Tools. This album only took a year to make; *Adrenalize* took three and a half. And we weren't sitting looking out the window all that time. It was work the whole time."

Def Leppard has recorded all their guitars direct ever since 1987's *Hysteria*. Drums are programmed, and vocals are done one track at a time. So Woodroffe and the band were able to record all of *Euphoria* with just two Neumann U 87 microphones: one upstairs and one downstairs. The signal path was very simple: just an 1176 into either the Amek or Mackie desk, depending on the room in which the vocal was recorded. The baroque elaborateness of Def Leppard's vocal recording procedure lies not in the equipment, but in the sheer number of tracks that go down to tape.

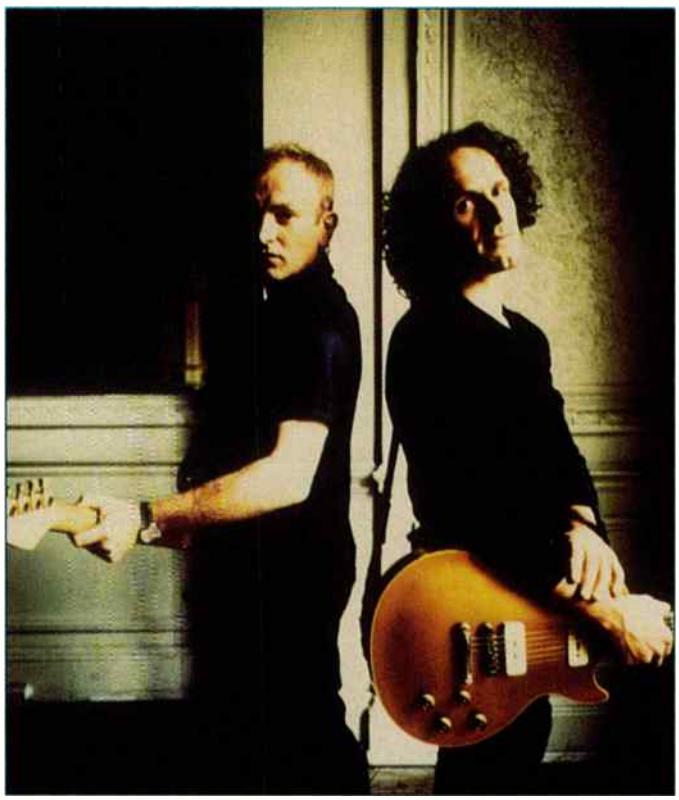
"On the song, 'Paper Sun,' for example, it's a four part harmony," Woodroffe elaborates. "There were 16 tracks of [Lep bassist] Rick Savage singing the low part. Viv Campbell, Phil Collen [the band's two guitarists], and Sav all sang the middle part. So that was 16 tracks for each of them. And then Viv did the high part: 16 more tracks. So you're talking many, many vocal tracks, which Pro Tools really helped me manage. The whole way through the sessions, I ran just the one 32-track tape machine with a stereo pair of backing vocals on two tracks, balanced and EQ'd. All the individual backing vocal tracks were in Pro Tools. And if we were working and someone said, 'Oh, I can't hear the high harmony part,' I would then just adjust it in Pro Tools, put it back on tape as a stereo pair, and off we'd go again. It was only in the mix where I put all the backing vocals down on a reel of tape separately, so I could have access to each individual track and rebalance the backing vocals."

The band's approach to layering guitars is somewhat different from their layered vocal technique. "People often think that with Def Leppard you'll have eight tracks of exactly the same rhythm guitar part," says Woodroffe. "But that's a misconception. Generally, each part will be played by a double-tracked stereo pair of guitars. But there are lots of different parts, and that's what makes it sound bigger. On the chorus of 'Guilty,' I think there's 20 tracks of guitars, all doing slightly different things. One will do an eighth note part, another will do a jangle. So it's more complicated than just piling up tracks."

The band's array of Marshalls, Gallien-Kruegers, Randalls, and other amps went through one of two Palmer speaker simulators: again, one upstairs and one down. Rockman effects preamps were routed through a Marshall head before going to the Palmer. "The outputs of the speaker simulator then went to a pair of SSL EQs," says Woodroffe, "and over the SSLs I'd have a pair of Amek EQs."

Subtractive EQ — particularly in the 1 to 3 kHz range — is the key to Woodroffe's approach to recording Def Leppard's guitars. "When you've got 20 tracks of guitars all playing at the same time, it sounds like a really noisy racket. All these hard frequencies build up. So once I've got the whole balance, I'll solo each pair and take out the really nasty, horrible frequencies. The sort of Marshall 'arrgghh' sound, the sound that everyone else wants, that's what we try not to do. It's fine if you've just got one guitar track. But the moment you start adding other tracks, it just sounds a mess."

Woodroffe is similarly spartan in his use of effects. "I'll print choruses, but never delays on guitars," he says. "I'll put the delays on during a mix. On guitars, I will virtually not use reverb. It just muddies



things up too much. Compression, too, for that matter. Just take the high end off so it sounds duller and put a delay on."

Drums on Def Leppard records have mainly been programmed for several years now. In some cases, on *Euphoria*, drummer Rick Allen's Simmons electronic kit was used to input MIDI data to a Emagic Logic Audio sequencer program. "Also on this project," adds Woodroffe, "Rick would be in L.A. and I would send him an ADAT tape with the track we were working on, with a basic drum pattern that we'd done. A stereo pair would be the drums and another pair would be the other tracks. He could then play with it at home and send the tape back to me. I could put that into the computer, listen to his audio playing, and adjust our drums to be like his."

Another key participant in the project was celebrated producer Mutt Lange, the original architect of the *Pyromania/Hysteria* sound, and originator of many of the recording techniques under discussion here. Since the Leps were looking to return to that style of record, it was natural to get Lange involved. The producer co-wrote several of the songs and gave some production input.

"Mutt co-wrote 'Promises,' " Woodroffe details. "We sent him a backing track and said, 'Mutt, what can you do?' He wrote a melody and lyrics. At his own studio, he put down a guide vocal and some backing vocals. He sent the tape back, and we replaced the guide lead vocal and did all the guitar solos and stuff. Mutt was at Joe's house for only five days to work on the record. On the song 'All Night,' he sat at the back of the control room and said, 'Let's try this. Let's try that.' He was fantastic on that. Then one morning I just played him about half the album and said, 'What do you think?' I got a master class. He had a lot of arranging ideas: 'Try shortening this section. That intro's a bit too long.' For two weeks after that, I was cutting and pasting in Pro Tools."

Shaping arrangements was another use that Woodroffe and the band found for Pro Tools: "If someone said, 'I wonder what it would sound like if we took out that section,' I could edit the stereo mix in a matter of seconds. We'd listen to it, and if everyone said, 'Yeah, that's cool,' I could edit all the individual tracks within two hours — edit out a section with all those guitar tracks and everything, yet still keep the old arrangement in case we wanted to go back to it."

Fine tuning in Pro Tools continued right through the mix stage. "For instance, on one song we had a sequenced keyboard playing the bass part," says Woodroffe. "In the mix, right at the end, I thought, 'Wow, wouldn't it be great to have a real bass.' And all I had was a guide bass track that was done a year before and was completely out of time. It was just Rick goofing off. But I could just take it through Pro Tools and put it in time. There were sections where he hadn't even played. But I could find notes from other sections and paste them in."

Woodroffe mixed *Euphoria* on a 72-channel SSL G+ console at London's Townhouse studios. Mixing was completed in just a month's time. Although the entire project was recorded at the lead singer's house, it was not the kind of domestic recording that needed to be "rescued" during mixdown at a high-end commercial studio. Woodroffe and the band had done their homework.

"You put the fader up, you hear the guitars, and that's the sound," he says. "That's the trick: the precision playing of complicated arrangements. EQ

Heart Break House

Mike Campbell records the new Tom Petty album without leaving home

BY ALAN DI PERNA

PHOTOS BY JAY BLAKESBERG

HeartBreak House

"It was kind of an accident," says Mike Campbell of his newest role in Tom Petty and the Heartbreakers. Campbell has been the Heartbreakers' lead guitarist since the band's inception in the mid '70s. He has co-written some of the group's biggest hits, including "Refugee" and "I Won't Back Down." Along with all this, Campbell has co-produced every Petty album starting with the 1989 solo disc *Full Moon Fever*. And on *Echo*, the newest Tom Petty and the Heartbreakers album, Campbell found himself engineering most of the sessions as well. It all came about, he says, because he bought a vintage Neve 1024 console for his project studio in the L.A. suburb of Encino, CA.

"I have this studio at home that I mainly use for my own stuff. I'd just gotten this console in, and I invited the guys over to help me try it out. We made a tape, Tom took it home, and he called me back and said, 'Sounds pretty good. Let's just work here for a while.' The next thing you knew, we'd done most of the tracks for the album here."

The main room in Campbell's studio is spacious yet homey: cluttered with vintage guitars, amps, and recording gear. A large window looks out on magnolia blossoms and a wooded San Fernando Valley hillside. It was in this room that Petty, Campbell, bassist Howie Epstein, and "auxiliary Heartbreaker" Scott Thurston sat and strummed as *Echo* went down to tape. Keyboardist Benmont Tench and drummer Steve Ferrone were off in adjacent rooms.

"Everything seemed to sound good, so we ended up sitting in the same chairs for the next three months," Campbell laughs. "We didn't have an outside engineer or producer in the room, which was kind of interesting. It was the first time we ever did that for that long at the beginning of a record. It was kinda cool, because we didn't have too many opinions in the beginning — for a change."

The basic game plan was to record live-in-the-studio as much as possible. Coming off an extended residency at The Fillmore in San Francisco, the band was in top form. "The thing at The Fillmore was great," says Campbell. "So when we started this record, we wanted to bring that energy into the studio and keep it going. I would say that at least 90 percent of the guitar and keyboard solos on this record were done on the fly, as the basic track went down. In the old days we might have said, 'OK, let's just get a nice-sounding rhythm track. Then we'll go back and get other guitar sounds and overdub solos.' But this time it was just, 'Go for it. Make it up on the spot and hopefully you'll get something exciting.' "

"Live in the studio" generally connotes a cavernous recording room and lots of ambient miking. But Campbell was able to get what he needed with a modest-sized room and very simple close-miking techniques for guitars. "Almost all the time, it would be a [Shure SM] 57 right on the amp," says Campbell. "If we were doing an overdub, there might be a rare occasion where we'd put another mic in the room, further back, to add a little depth. But we didn't take much time with it. We were usually in a hurry. Sounds were never an issue. It was always, 'Is the song good? Do we have the arrangement right? Is the performance good?' My job as engineer was really just to get it down on tape without bad distortion, not to erase anything by accident, and to adjust everyone's headphone mix so they were comfortable. I'm not really a technical engineer. I couldn't tell you what frequencies those knobs are [pointing to console]. I go by ear."

In some cases, miked amp tracks were doubled by a direct track recorded via a Palmer speaker simulator. Along with Voxes, Marshalls, and other classic tube amps, Campbell also used an old, solid-state Kustom 250 amp. The Kustom's built-in tremolo became one of the album's signature guitar tones.

"I never would have thought to use that amp," says Campbell, "but it just happened to be in here one day and I plugged it in and thought, 'That tremolo has a nice sound.' That became my sound. What's funny about the tremolo on the record is that a lot of it came out of trying to play the whole song live. I would kick the tremolo on for a dynamic effect and then kick it off. This way I wouldn't have to overdub things for dynamics. So I started using that. Then, about halfway through the album, Tom goes, 'You know there's tremolo on every song.' We tried to cut back on it, but then it would be like, 'Where's the tremolo? We need it on this one.' So we ended up using a lot of it. This Kustom amp has tremolo and vibrato that you can mix together and get a nice rich tone."

What sounds like carefully layered acoustic guitars on songs like "Won't Last Long" and "Echo" were actually done live in one pass, Campbell reveals. "Tom was on a six-string acoustic and Scott was playing on top of that on an acoustic with a high-strung Nashville tuning. That's why it sounds layered. But it was actually laid down as a basic track with no overdubbing or layering of any sort."

On acoustic guitars, says Campbell, "we used condenser mics usually. Not too much

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HeartBreak House

in front of the soundhole, just six to eight inches back. I'd usually use a [Neumann U] 47 or a Sennheiser 451. Sometimes I'd use an AKG C12. Condenser mics usually pick up more of the ambience. They're not as unidirectional. We like to use them on acoustics. And, sometimes, if Tom sang live, you'd get a bit of bleed: acoustic guitar in the vocal mic and vice-versa. In some ways, it improves the acoustic guitar sound, makes it wider, having that extra mic there. The trick with that is you gotta sing and play really well. You can do punch ins to fix things, but it's hard to match the ambiences."

Campbell's studio is equipped with both Studer and Otari analog 24-track machines. "Everything ends up on digital when the project goes to CD, but it's good to start with analog," says Campbell. "Especially with electric guitars, it's good to put things in the red. That's where the fun is."

Toward the end of the recording sessions, the band called in engineer Richard Dodd to take over recording duties. Rick Rubin, who has co-produced the last three Petty albums, was also brought into the loop.

"Eventually, we reached a stage where we had a lot of songs and we kind of lost direction," says Campbell. "We needed somebody to help us get our focus back, like: 'These four songs you don't really need. I think you should stay with these over here.' 'Cause you can lose perspective when you produce yourself. Rick was very helpful at that stage."

When pressed, Campbell admits that there's a great deal of determination behind Petty's laid-back persona: "When it's time to get down to business, there's no fooling around. The bar is really high. Tom has always been like that. That's why he's the leader of the group. He's gonna fight for quality. And you know we won't stop until he believes what we're doing is really good." EQ

Buying a new digital recorder or workstation between now and September is sort of like buying a new turntable just before CD players came out.

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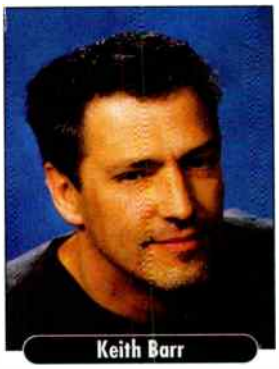
We've been hearing an awful lot about Y2K — probably too much. Still, the future of recording in the new millennium has yet to be discussed — until now, that is. We've asked a number of prominent recording industry professionals to share their thoughts on what the future holds for those who make their living through recording and performing music.

So break out the crystal balls and tarot cards and join us as we take a look into the future of audio.


Keith Barr


David Blackmer


Ed Cherney


Jack Douglas

Murray Allen, Former President of Universal Recording; Director of Audio and Video Production, Electronic Arts; Musician. I see music becoming more polarized — much as society has become. Those folks who like "sweet" music will still have those same five or six chords they have been listening to for the last 70 years. Some artists will sing in tune and others will still sing out of tune. We will still record multitrack and also release as multitrack so the listener can be interactive and make their own mix. Those who create "non-sweet" music will find new and better rhythms to addict their audiences. Schools will learn that music is really important and start teaching it again.

Keith Barr, President, Alesis, Electronic Engineer. Totally solid state recording in uncompressed form, automated mixing and mastering — hell, entire music productions could be generated automatically, with not just acceptable, but excellent results. We are musicians, we'll become sound designers. The odd fellow who can actually *play* will have all the tools at a minimal cost. Then again, it's possible that no one will be able to play in 50 years, and computers will provide yet more imaginative algorithmic results than any human...*Naaaa.*

David Blackmer, CEO, Earthworks, Inc. Time is of the essence, especially in human hearing. We can sense time relationships and waveshapes at a resolution of a few microseconds. This incredible capability in the time domain is responsible for the great difference we perceive between live and recorded sound.

Lord Raleigh's legacy of explaining sound perception primarily in terms of frequency components still persists in spite of efforts of Klemm (1920) and Von Bekesy (1929) and other pioneers to suggest that there were other important aspects to human hearing. This has led to disastrous mistakes such as inappropriate "perceptual encoding." These misconceptions will pass, and a new day will come, bringing more accurate sound reproductions.

Microphones, loudspeakers, and audio signal processing systems must have sufficiently accurate time resolution, which we measure with impulse response, to preserve the essence of sounds when they are captured and reproduced. The bandwidth required to do this is in excess of 40 kHz, possibly as much as 100 kHz. Here at Earthworks, we are striving to produce the tools which will allow you to achieve this perceptual perfection.

Marina Bosi, President, AES; Vice President, DTS; Consulting Professor, Stanford University. Audio technology has been evolving at an incredible pace in the last few decades. The recent introduction of new, high-capacity media, such as DVD and the Super Audio CD, in conjunction with the latest developments in digital signal processing, IC design, and digital distribution of audio, is leading to a wide-spread utilization of high-quality audio. While the consumer is now expecting "CD-quality" audio or more and enhancements in the spatial reproduction of sound, for example, the home theater entertainment systems or the multichannel CDs and DVD-Vs currently in the market, the professionals have a palette of new tools to explore and experiment with.

Digital broadcasting of multichannel audio is now starting in North America and is at the horizon in Japan and Europe. If we add to this picture the more and more important role the Internet and, specifically, audio on the Internet are playing in our everyday lives, and the next generation Internet developments, I see a lot of opportunities for the audio industry to grow stronger and stronger across platforms. More and more people will have access to audio and the means to produce audio. New business models are emerging as the results of this new, e-commerce era. I believe the AES will be consolidating new developments and will be the catalyst

CHASED, COMPILED, AND EDITED BY JANICE BROWN

nnium


Dust Brothers


Frank Fillipetti


Tomlinson Holman

among science, technology, and business on audio issues.

Ed Cherney, Grammy Award-Winning Recording Engineer/Producer (Bonnie Raitt, Eric Clapton, Jackson Browne, Bob Dylan). The next 50 years are going to bring profound changes as to how we go about doing our work. Certainly, the miniaturization of our tools, consoles, and storage mediums will enable us to work anywhere. Eventually, I see the time in the very near future that we will be working in virtual studios. A place where you log on, your music is stored, you can mix, record, overdub, collaborate, master, and deliver your music anywhere. We will be able to record and do overdubs from anywhere that has a transmission line, use musicians anywhere in the world, be able to synthesize acoustic spaces, microphone characteristics, amplifiers...there will be digital models of all of the good analog equipment we use today without leaving the friendly confines of the home, car, boat, office, etc. I also see the loss of certain engineering skills that have been honed through trial and error over the last 75 years. There are currently only a handful of seasoned veterans that know how to manipulate acoustic spaces using live instruments and microphones. I don't see, at this moment in time, many of those skills being passed down, or if these will be relevant skills in the virtual musical world to come.

Dust Brothers, Music Producers, Recording Artists (Beck, Beastie Boys, Rolling Stones). We predict that music, as we know it, will be replaced by fax/modem tones — also there will be fierce competition between recording artists to create the ultimate beeper sound. Analog metronomes will become the instrument of choice for live performers. Ultimately, sound recordings, in any form, will be deemed illegal (for no specific reason), which will, of course, eliminate voice-mail. A&R executives will become meteorologists and producer/engineers will be shot.

Jack Douglas, Grammy Award-Winning Producer (Aerosmith, John Lennon/Yoko Ono). Virtual producers will assist new bands. I imagine George Martin giving advice to a young musician/producer trying to reproduce Ringo's drum sound 50 years from now simply by accessing George's database at the "Virtual Producers" Web site. I have often heard record executives say, "Nobody wants to sit for hours in front of a computer screen!" I, myself, could easily take issue with that, but I'm sure in the future that all of these tools and services will be "on demand" and we will not be shackled to our screens and keyboards. We will simply state what we want and instantly have the results — "More bass please," or "3 dB at 80 cycles and 6 beats per minute faster HAL." Everything from music to film will be available just by asking for it.

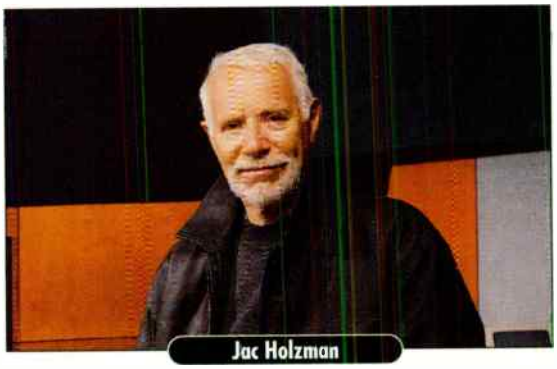
We will probably talk to our musical instruments as well. For example, we may simply hum a musical line or ask for a specific chord played by whatever we want from a sitar to an entire orchestra. We may even connect ourselves directly to our instruments using "bio-feedback" to relay particular emotions to our music. PS: Tape will be dead!

Larry Emerine, Precision Mastering, Los Angeles. The next millennium for mastering houses is about format proliferation and flexibility. People predicted that the CD would put mastering houses out of business in the '80s, but, more than ever, we are still busy and still cutting lots of lacquers. Our future will hold some kind of high-resolution format and a multichannel format, plus several compressed-file Internet formats. Our challenge will be to make the Internet formats as good as they can be, which, compared to the quality of many CD releases these days, may actually be quite comparable.

Frank Filipetti, Grammy Award-Winning Producer/Engineer (Barbra Streisand, James Taylor, Elton John). On the last two Barbra Streisand albums, I've been mixing at Right Track Studios in Manhattan with Arif Mardin on a Neve Capricorn digital console while Barbra has been monitoring my mixes in real time at her home on the beach in Malibu. As I am working on a fully automated console, I can reset a mix from one song to another in less time than it takes to wind the multitrack tape. More and more, I see this as the future of recording and mixing. An artist and/or producer can relax in their home environment and still stay totally on top of what's going on in the studio. The concept of tape storage is already being replaced by hard-disk systems. Soon, the delineation of console and storage medium will be replaced by a unified, totally integrated system. The console will be the storage medium and all recording/production/mixing functions will be done from this centralized source. This has already surfaced in the project arena, and will quite soon find its way into the professional mainstream.

Very soon, a studio musician will be able to play a part into his/her home computer, upload it onto the Internet, and download it onto the master tracks in a studio hundreds or thousands of miles away. All the while the artist can monitor the whole process from their vacation home in Fiji. And all this will happen in the next five to ten years! Beyond that, we may also see a blurring of video vs. audio. In fact, we may even see the formation of a whole new type of artist — the virtual artist, designed by focus groups for a particular demographic, and created in a computer.

Paul G. Gallo, President Miller Freeman, PSN. As we enter a new millennium for recording, we will undoubtedly see tremendous strides at all levels of technology. Gear will become more sophisticated and more expensive on one end, while it becomes more affordable at the other. Commercial studios at the high-end will continue to flourish, setting a standard for technical service. Meanwhile, project


Jac Holzman


Leslie Ann Jones


George Massenburg

rooms will proliferate, expanding access to recording technology for all forms of media. Still, as it has been for the past 25 years, creativity will remain the primary driving force. Engineers, producers, and musicians will utilize those tools that work — regardless of whether they are digital or analog. Of course, better sound quality will be the underlying theme. However, simplifying the creative process and expanding the creative potential of the individual will characterize the course of the recording industry well into the next century.

Tomlinson Holman, President of TMH Corporation; a Professor in the School of Cinema-Television at the University of Southern California; and Founding Editor of *Surround Professional* Magazine. The next 50 years will produce a far more immersive experience of reproduced sound, as composers and musicians come to grips with the creative possibilities offered by advances in spatial reproduction. While human hearing can be completely satisfied in frequency range and dynamic range by new media coming out now, only the surface of the "spatial range" has begun to be penetrated. We know that everyone equipped with normal hearing can hear the improvement from mono to stereo to 5.1 channels, and we have a 10.2-channel system working as a prototype for the future. Everyone who has heard it has remarked on the difference in reality compared to the 5.1-channel system. Where is the end? No one knows, but we're sure the number is greater than 10.

Jac Holzman, Chief Technologist, Warner Music Group; Founder, Former CEO, and Creative Head of Elektra Records and Nonesuch Records. Of all the near-term technologies that will impact the way we listen to music, multichannel surround sound may be the most important because it engages us totally. So much of our current listening leans toward overhearing music, and it is a lazy, Muzak-style tendency. High-resolution multichannel invites us to integrate ourselves emotionally, to be one with the music and to enjoy a profound connection to the creators who make the magic. We will acquire music in many different ways, including downloading, but until we have solid-state recording "blanks" in the home that do not have to function in an electro-mechanical mode, downloading will be centered on the fans of contemporary popular music. Music listening will skew to both ends of the activity spectrum; from highly involved interaction to a more passive and immersive experience, including wrap-around visuals and other stimulations that entertain and involve the entire physical human being. We are just at the beginning.

Leslie Ann Jones, Director of Music Recording and Scoring, Skywalker Sound. I think music makers have already come to the conclusion that live music with acoustic/electric instruments is a sound that is preferred to synthesizers. The further we isolate ourselves with computers, e-mail, and the Internet, the more we will realize the need for interaction, i.e., everyone playing in the room at the same time. My hope is that this will result in a higher standard of playing. That's the "social" part. From a technical and commercial standpoint, I think hard drives, as a storage medium, will become more readily available. Euphonix' R1 and Otari's Radar are two of the early examples. However, manufacturers will still have to chase the analog/reel-to-reel tail — it needs to work the same. Engineers will trade away only so much for speed and storage capacity. I think Internet distribution will become the record company of the next decade at least. It is already changing the way record business commerce behaves. Companies like Gary Gersh and John DeSilva's GAS are only the beginning.

Ray Kennedy, Engineer/Producer, Partners with Steve Earle in TwangTrust Studios (Lucinda Williams, Steve Earle, Cheri Knight). First, I see the analog/digital war continuing for a long time — and it's still a matter of personal taste. Because of this format dilemma, I predict a viable pro-dual application-type multitrack where you can choose on one machine which tracks will go to analog and which go to digital without all the SMPTE locking up problems. As far as mics and signal path gear, I see a continuation of cloning the past. I believe slick production will continue to take the place of great songs and performances until consumers can't stand it or won't buy it any more. And a quote from the great Townes Van Zandt: "There's only two kinds of music: the blues and zip-a-dee-doo-dah."

Al Kooper, Musician; Producer; *EQ* Columnist. I think the immediate future of music will be concerned with MP3. The record companies have overreacted, and, in that wake, a piece of hardware I have longed for will hopefully arrive. I call it NMP, which stands for "no moving parts." The buyer purchases a piece of software, not unlike the size of a sound card or credit card, upon which is digitally imprinted the album's sound and credit information. The hardware (player) reads the card and spits out the information tune-by-tune (with credits) with no moving parts at a hopefully high sampled rate that will show the MP3 for what it is: a valuable, stop-gap promotional tool. This breakthrough would calm the record companies down and serve the consumer with miniaturization of hardware and software, less chance of breakdown, and, hopefully, low prices. Let's all dream, shall we?

Hector La Torre, Executive Director, *EQ* Magazine. The question should be: "How will future technology affect the quality of music?" It's a good-news/bad-news scenario. Record companies are getting bigger, and the bigger a business gets, the more people it excludes. That's why we'll


Arif Mardin


Robert Moog


Phil Ramone

see more digital technology — more affordable recording gear, computers, software, faster but less costly communication lines — because that technology includes people. Musicians and producers will be able to create music at their own pace, and the Internet, at first, will advance the democratization of music by permitting more corporately disenfranchised musicians to pinpoint potential audiences. Beware, though: we shouldn't be too anxious to embrace the total democratization of music and its transmission, because not everybody should be a musician, engineer, producer, etc. Technology does not equal talent.

It'll be astounding to have players and engineers in different countries working on the same music piece; for example, Brazilians regularly working with Finns and Moroccans. However, will that lead to a global homogenization of music? All of this, of course, will eventually come around again, because the best music has always come from people sitting around in the same room screaming at each other and banging on something until magic happens.

Bob Ludwig, Gateway Mastering Studios. I can't predict anything *five* years from now let alone 50, but here are three things that I do know: [1] It will be cool; [2] New gear will start off being very expensive and before it can be amortized, it will be cheap enough for everyone to afford!; [3] Parents will be appalled by the music their children listen to.

Now that it is possible to do switching on the molecular level in the lab, future computers will be much faster than previously predicted. These machines will probably be more fun to hang out with than some humans. Improved sound transducers, combined with advanced DSP, should be able to produce totally holographic sound.

Arif Mardin, Grammy Award-Winning Producer (Bette Midler, Bee Gees, Aretha Franklin); Co-Chairman/Co-CEO, Atlantic Records. When I first started at Atlantic Records in the mid-'60s, I had a vision that someday they would invent a biological record. The listener would take one pill and the hear the music throughout their entire system. You could have a double-play pill, and, of course, if you drink a little wine, the tempo might go a little faster. This was a little unfinished short story at the time, but, now, I don't think it sounds so unbelievable. Of course, this will have drawbacks — if you have too much hot sauce in your food, it may affect the music as well. I also envisioned a recording studio in which you'd be showered by some kind of a beam instead of using microphones. Your voice and music would be enveloped by a shower of electronically charged ions or particles that would then be transferred into a computer or board, and recorded. These were my fantasies — a shower-like microphone and a biological record. I think the microphone idea is very possible, but I can imagine the biological record having too many drawbacks with the FDA.

George Massenburg, Producer, President, GML, Inc. Within 50 years, popular music recording continues to devolve. Artistically challenged "artists" continue to sample great R&B and pop music recordings, but record companies don't share the income with original producers, who look for work elsewhere. Sampled and sequenced pabulum will account for more and more of the product on "pop" radio. A few actual live performances are recorded, but they are pitch-corrected and time-corrected, scrubbed, and looped, and there is no soul or feeling whatsoever left in them. This does not bother the megalithic record companies. Surprisingly, in the face of daunting competition from guileless startup MP3 record companies, the majors, whose staffs are running like cockroaches caught in the light, still don't get it. My 11-year old son, Sam, thinks there will be more Fat Boy Slims. I hope so myself.

In the far distant future, music generated for no other reason than the love of the art form emerges, and musicians who sing tales and legends appear again. A new folk form rages. (Naturally, there will be a "folk scare" all over again anytime now.)

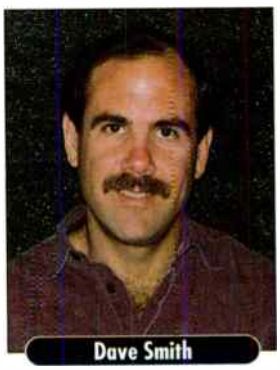
Robert Moog, Ph.D. Engineering Physics; Founder of Moog Music and Big Briar, Inc. Just as few people in 1950 could foresee ADATs, soft mixing consoles, and the Internet — it is virtually impossible for us to foresee the state of music recording and distribution in 2050. However, it does seem likely that music will be stored, processed, and distributed as so many files of information, easily and rapidly available anywhere in the world. Does this mean that hardware instruments, tools, and media are on their way out? Not necessarily. For musicians, the user interface will become more and more important as electronic sound production and processing become mature technologies. My prediction: the contemporary musician of 2050 will own a finely-crafted control device that will incorporate sophisticated multi-dimensional tactile feedback. Such control devices will combine mechanical, electronic, and software elements into a responsive device that is matched to the capabilities and preferences of the musician.

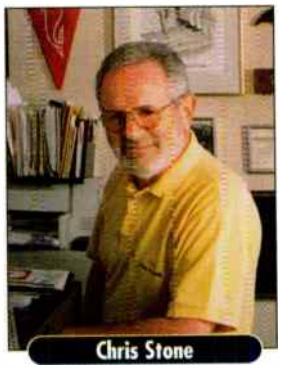
Russell Palmer, CEO, Event Electronics; Co-founder, Alesis Corporation. As a manufacturer of audio equipment, we're always going to rely on making sales through our dealers, despite the dramatic rise in Internet-based shopping. While you can certainly get customers to go to a Web site to gather technical and marketing information about a speaker product, for example, it's likely to be difficult to complete a sale because the customer can't hear how that speaker sounds over the 'Net. So we need our dealers to provide a showroom for our products, and we need their salespersons to provide one-on-one education of our customers.

I see computer workstations becoming more central to the processes of recording and performing. That's not to say that musicians will necessarily be dragging CPU


Al Schmitt


Allen Sides


Dave Smith


Chris Stone

towers and monitors to their gigs, but I believe that the basic nature of computers is going to change. They'll be more function-specific, and may ultimately sit as just another piece of gear in a musician's rack.

As for the immediate future, surround sound is the key to success for today's project studio. There's a tremendous opportunity — remastering of existing stereo albums in multichannel DTS, remixing movie soundtracks in Dolby 5.1, and, of course, breaking new ground with original surround music projects.

George Petersen, Editor, *Mix* Magazine. Music on the web is gonna change *everything*! The possibility of downloadable music — in either CD or MP3 form — not only opens avenues for new bands and new music, but allows record companies to release many older albums that are currently unavailable. Meanwhile, online stores, such as CD Now and Amazon.com, combined with band Web sites where consumers can listen to clips online, will open every garage band to a world audience. But, most importantly, as the millennium approaches don't forget that it's not too late to start planning for Y10K!

Phil Ramone, Grammy Award-Winning Producer (Billy Joel, Frank Sinatra, Paul Simon); Chairman, NARAS. In 1899, my great grandfather was pondering the millennium: electricity, phones, silent movies. To come: radio, TV, recorded music, LPs, surround sound, VHS, CDs, and DVDs. Now the great grandson ponders the next 50 years ahead. I am sitting in my new car as I download the Top 50 chart songs. As I drive, my in-dash screen lets me know if it is 5.1 or just plain stereo sound. The car takes off for work at the studio, programmed to avoid traffic jams. My language course will get my conversational Spanish; it's 15 minutes before my destination. Printouts of information and pictures of my listening session of the Top 50 are printed in the glove compartment.

At the studio, there is a session to produce using both digital mics and old tubes for texture. Hey guys, whatever happened to that old digi 48-track machine of the 1990s? "It's at the museum!" The words "infinite tracks" will be standard, studios will finally be designed to be interactive, it will be visually easy to be together making music. Technological functions as one with musicians and artists loving their working environment. Walls that move to not create boxes. Personal production devices for producers and engineers that find the newer consoles make it easier to capture the event. Along with digi cameras, remote-controlled, sensitive lighting. A place to enjoy. As I leave, I am given files to look at and listen to on my way home. Creative thoughts remain in the private domain of the artist and the producer.

Since we use many studios, private encoding, there's more usage of pre- and post-facilities hosted by studios around the world. More flexible space in studios much like motion picture lots. Rooms with ambient walls that can exclude or include mixes of musical sound. Orchestra vibrations, tonal walls to transpose to digital on future storage devices.

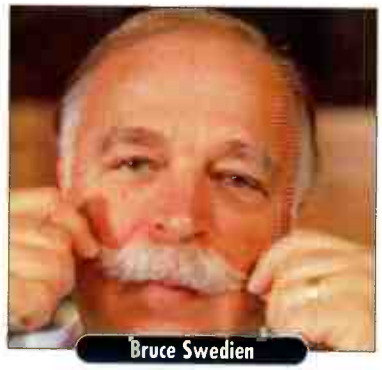
Your choice of space in the recording environment to change reverb from bright-to-dark reflections. What your mind wishes for will be available like your personal prescriptions: the EQ of speakers, personal earpiece monitoring, your personal profile from the way you enjoy working, to casting of engineers and assistants.

Finally, with all this luxury, all that is left is that old 20th Century philosophy: a great song capturing the artist and musicians or machines to make the hair on your arms stand up and the passions to flow. Not yet available in a jar. I cannot wait for the year 2000 to begin the next journey. Let's enjoy!

Al Schmitt, Grammy Award-Winning Producer (Frank Sinatra, Quincy Jones). Well, everything will be totally computerized. I don't think there will be any analog at all, or tape machines. Certainly, I'm hoping that digital will get it together so that it sounds just as good as analog. As far as music is concerned, electronic music will probably be at the scene's forefront in 50 years — more synthesized pieces making music.

Allen Sides, Owner of Ocean Way Recording in Nashville and Hollywood. As an engineer and producer, I see the short term future of severely data-compressed Internet music as quite depressing. But there is some hope that the new DVD-Audio disc may save us from a life of downward spiraling lo-fi. We can only hope that audio will be allotted more space and great sound may survive and prevail. On the positive side, the potential for direct distribution is amazing for the artists, particularly those who sell between 25,000 and 100,000 units. After the record company takes their cut, artists rarely see any royalties and must survive on whatever advances and musician fees they can collect during the production. The evolving Internet may offer greater rewards for creativity than ever possible before.

Dave Smith, Founder, Sequential Circuits; President, Seer Systems. It's obvious that all things audio will continue to advance with technology. Soon, everyone will stop talking about bit widths and sampling rates — when everything sounds so good that it will finally be a moot issue. Of course, everything will be digital. Of course, everything will be Web-based, though I think it will take longer than most people think. My guess (that's all we're doing here, folks) is that the big changes will not happen until at least one, probably two, complete generations of computer-literate kids grow up. These are the people who will be comfortable with the paperless office and media-less audio. The real work needed is not in basic digital audio processing, but rather in simplification, automation, and



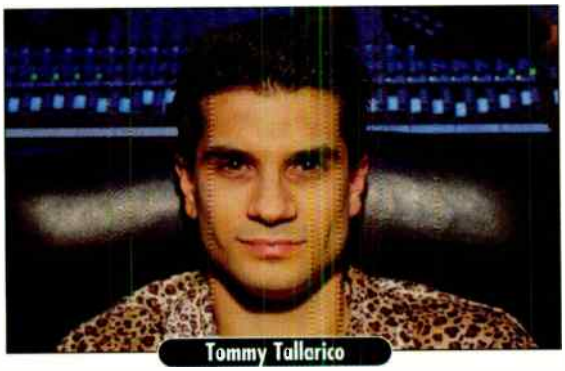
Bruce Swedien

control surfaces. Going ten years out, I don't see much of a change; when we're talking 50 years, though, it's night and day (think 1950...).

Chris Stone, President, MPGA. So many things impact the recording industry. For example, the first time in the U.S. that a satellite overdub was done, it cost tens of thousands of dollars, and, now, you can do an overdub between here and London for hundreds of dollars. Everything is moving very quickly, and it all will only get cheaper and clearer, and better. The rapid computer developments totally affect audio.

Fifteen years ago, I said, "Audio and Video are coming together." I believe that we will soon have music that is visual. We are really, finally going to have it. We've had CD-R, CD-V, and now DVD. I think it will get to the point within the next three-to-five years that the Internet is going to be able to download, quickly to millions of consumers because of the speed and viability of fiber optics. Any further than a five-year prediction is out of the question. Nobody knows because the technology hasn't been developed yet, and the practical things we're working on now will have become an important part of the consumer market by then. We're way far ahead with the Internet in the U.S. because of such a large percentage of the U.S. population having access to it. That's the way it has to go. There is no choice. The path is set. The Internet is it. Everything is always in a state of change in the music and recording industries based on developing technology.

Chuck Surack, President, Sweetwater Sound. Without a doubt, the most exciting thing happening in music technology is the ability to record and create music with a computer. It's mind-boggling to think where this field will go in the next 50 years. The Internet and other similar dis-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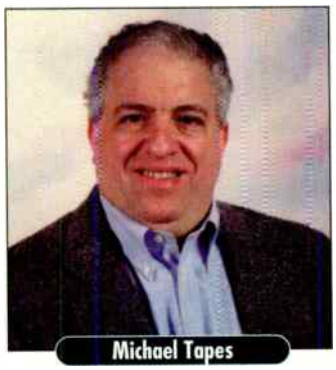
Michael Tapes

tribution channels will enable music to take on many different forms, maybe even becoming more interactive. For anyone who loves or creates music, this will lead to unprecedented diversity and new creative challenges. As this new technology becomes more affordable and more accessible, the means of making music will be available to more people. The great composers and musicians of the 21st century will create their masterpieces on a computer, and they won't need access to full orchestras, won't have to rely on a second party to arrange or conduct, won't need to have any outside influences on their music. Just sitting in front of the computer in their bedrooms or basements, the Beethovens and Mozarts of the next century will be able to bring their musical imagination to life.

Bruce Swedien, Grammy Award-Winning Producer (Michael Jackson, Quincy Jones, Duke Ellington). I wish I could tell you, with some degree of certainty, what the new millennium has in store for recorded music. What I can tell you with wholehearted confidence is that I am not frightened of the future. Never have been.

I've seen our beloved music recording business go through some critical changes. What I find most promising now is that musicians, bands, and composers have easy access to recording technology that is far better than at any time in the past. I have a strong belief that the music recording business is going to be put back in the hands of the people that truly love music for music's sake. Music has always seemed to be organic in myself. I think it's that way, to some degree, in the soul of every human being. That's why I'm confident that recorded music in the new millennium will emerge at least as strong and healthy as in the previous one.

Tommy Tallarico, Founder, Tommy Tallarico Studios; Multimedia Audio Producer (Earthworm Jim 1 & 2, Aladdin, The Terminator, Madden Football). Coming from the video game industry, here's what I feel will be the future of interactive audio: As huge bandwidths and DVDs become more commonplace for games, this will enable us to constantly be streaming in audio on the fly to mirror the action on screen. Instead of having a 24- or 32-channel MIDI file that you can branch off, mute tracks, change patches, etc. You will be able to do that with live instruments! The composer will have to write and record lots of different sections for the same song. As the action changes on screen, so will the live music. I feel that 5.1 audio will also be standard within the next 10 years with companies like DTS and Dolby leading the way. (Damn, I gotta re-buy every CD I have — I'm sure the record companies are gonna hate that!) As bandwidth increases, the Internet will be a huge source for buying and distributing music. My biggest prediction: "Greatest Hits" albums will become obsolete, because everyone will be making their own!

Michael Tapes, VP Product Development, Studer North America. Over the course of the next 50 years, I can only hope that there will be a wonderful balance of sound, picture, and solitude. My fear is that if the current mentality continues, we of the human species (not to mention our cats and dogs) will be bombarded with an unending stream of marginal-quality sound that assaults the senses rather than calm and soothe them. I hope and positively project that mainstream recorded sound in the future will be of an audio quality and content that will allow our brains to get audio cues that emulate those from nature. After all, "music calms the savage beast." Let us also hope that in the future we'll come to understand that the spaces between the notes are just as important as the notes themselves. The more we can appreciate the silence, the more emotional impact great recorded music and sound can have upon us. EQ

EQ Exclusive

Audio-Technica AT4047/SV

Microphone Review

BY MIKE SOKOL

Add some flavor to your mixes with Audio-Technica's tasty new mic

Last month, you saw my exclusive "First Look" of the new AT4047/SV microphone from Audio-Technica. Since that article, I've had the opportunity to see the Audio-Technica offices in Stow, Ohio, review their testing procedures, and, best of all, carry back a pair of new 4047's marked SN#00001 and #00002 for this exclusive *EQ* review. Therefore, let's delve more into the facts, features, and, most importantly, the sound of this new mic.

As I said in the last issue, the finish of the AT4047/SV is a non-glare, satin-silver instead of the expected black. The shock suspension is especially cool, offering a dual set of elastic isolation elements. An inner ring suspends the microphone with a pair of elastic bands, which itself is suspended within an outer ring — really a mount within a mount. It's a little tricky to position the mic at first, but, once assembled, it inspired confidence that nothing short of an earthquake is going to separate the mic from the mount. Furthermore, the whole mount assembly is of quality cast aluminum, rather than "bent metal." Shock mounts and filters are important for a studio mic — especially one that may end up on a 5.1 surround recording (think unwanted infrasonic garbage). (Incidentally, A-T is including a custom-designed, vintage tweed carrying case with the first 600 units of this microphone sold.)

A pair of switches on the mic body includes a 2nd-order (12 dB/octave), 80-Hz low-cut filter and a 10-dB pad that boosts the levels up to 159 dB SPL while engaged and 149 dB SPL when not engaged. Factor this into the incredibly low self-noise of 9 dB SPL, and you get a dynamic range of something like 140 decibels. (My only real regret is that I had to send back the Millennia Media HV-3B preamp I was testing last month, so I didn't get a chance to use them at the same time. But I'll wager the combination would have been awesome in the low-noise department.)

Now I'll say this about the testing lab at A-T: those people are fanatics (and I mean that in the best, non-terrorist way possible). I used

Audio-Technica AT4047/SV Microphone Review

to work for a company that manufactured parts for nuclear missiles, and while this doesn't quite put me into the same atomic engineering club as Roger Nichols, as a result, I do know electronic testing procedures. Of course, microphone meltdown won't start World War III like a failed guidance system might (*oops* — maybe that hasn't yet been declassified). Still, I've known singers that have gone "nuclear" when a recording doesn't go their way.

That's why you need a mic that offers an interesting perspective on a voice or instrument and that's as bulletproof as possible. This, by the way, is how Audio-Technica used to generate the impulse noise in their anechoic room. That is, they used to fire a 22-caliber pistol blank directly in front the microphone. The test is used to generate what's called a waterfall graph of the impulse response, which they say is more useful in determining the "sound" of a mic than the traditional frequency response chart. The procedure worked perfectly with their capsules (with an estimated 160 dB SPL), but it did tend to bottom out the elements in the competition's mics, which made data comparison impossible. Now A-T settles for the *playback* of a gunshot via a speaker, instead of live munitions. (As a side note, anechoic chambers are cool. If you ever get a chance to go in one, jump at the chance.)

You want to know the answer to the big question: What does this thing sound like? If you read my June *EQ* article titled "Love at First Listen," then you know my opinion on microphone selection: No single microphone is perfect for every application. Since every mic has its own unique sonic signature, there's always a coloration of the original sound involved. It's just the physics of transducer design — there is no such thing as perfect energy conversion. Now this doesn't mean that some talented designers haven't come close. Moreover, for lots of applications, perfect reproduction is not what you really want anyway. I sometimes like to think of the sounds of various mics as flavors of ice creams. If you don't believe me, try this simple test: pull out four or five different mics and use them to record the same instrument or vocalist in succession. Now, if you listen to the tracks one at a time, you'll discover just how different they all sound (or should that be "taste").

Some mics may be bland-sounding (sort of

NOTES FROM THE FIELD

I just finished a Fourth of July gig using a collection of Audio-Technica mics, including a pair of the new AT4047's. The show was, as mentioned in the main article, the Maryland Symphony Orchestra at the Antietam Battlefield in Maryland.

We close-miked an entire orchestra, then sent the mix to ten delay stacks positioned to cover 30–40,000 people. (I know you don't want to hear about the heat, but it was close to 100 degrees for the 3-day setup, and it rained heavily the day before the event.) We kept the mics in the trucks, not on the stage, until an hour before soundcheck. High heat and humidity are hard on any kind of gear, but especially so on amplifiers and microphones. I wouldn't normally trust studio-grade mics on a live stage under these kinds of conditions. However, since we had used Audio-Technica mics in similar situations in the past, we didn't worry too much about them and didn't use any special handling procedures. We needed windscreens (as for all outside events), but aside from having a viola player rapping a stand every so often, we had a minimum of unwanted noises.

We put up Audio-Technica mics on the entire string section (AT4041's and AT4051's over each pair of players), and reserved the AT4047's for the flute section. (I didn't want too shrill of a sounding mic for the all-important piccolo.) The AT4047 turned out to be an excellent choice for the flute position. The mic had lots of body for the midrange notes without getting too hot on the high register.

We received many compliments from the crowd on the overall fullness and warmth of the orchestra mix. In addition, after the show the maestro herself complimented the sound crew both personally and on televised news (and how often does that happen?).

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Audio-Technica AT4047/SV Microphone Review

non-exciting vanilla), while others might hit you with something richly unexpected, like pistachio-mint. I like a good mic that's got various "tasty" elements that can co-exist in ways that seem impossible. (Do you remember the first time you tried sweet-and-sour Chinese food? I would never have guessed that pickles and pineapples would work so well together, but there it is....) There's no way to really predict what will be the perfect source/transducer combination for a given situation. However, believe me, when you find it, you'll know it.

In the time I've had to try the AT4047/SV, it's been used in and outside my studio on an assortment of voices and instruments, including female singing, male narration, maracas, and nylon-string guitar. It was beautiful on female vocals without presenting too much of a high-end, sibilance boost. The male narration had a nice testosterone quality, but without too much chesty boom. A nice presence made the voice snuggle nicely into the mix, but didn't bludgeon you with a nasty peak. This could even be an excellent DJ mic for radio stations. In addition, it is versatile enough to handle nearly any studio or live chore — from high-SPL guitar amps to ensemble symphony instruments.

I like to do a maraca test to determine ultrasonic overload; some otherwise good-sounding mics will freak-out with this type of input. (Just be careful to use a good preamp, since maracas can wreak havoc down the sound chain.) The 4047 seemed very smooth on the top end, with not a hint of overload or strain. The nylon guitar, while nice, was not the best fit for this mic. I needed something with a little more aggressive top end to make the strings sing. I tried the 4047's on the Maryland Symphony Orchestra's concert at the Antietam battlefield in Maryland, where approximately 40,000 listeners attended. [*See sidebar for details.*]

The sound of the 4047 is like rich ice cream. Not the wimpy, watered-down, air-puffed stuff they sell out of cardboard boxes in the supermarket. No! If the 4047 had a sonic "flavor," it would be chocolate almond fudge. It's a little darker tasting than you might expect from a condenser mic, and there are these crunchy, whole almonds each dipped in chocolate and surrounded by the richest fudge ice cream you can imagine. It's sweet and smooth, but with lovely bursts of flavor in unexpected places. This is not a plain-vanilla mic, or, God forbid, some low-fat (or should that be low-*PHAT*?) ice-milk creation for the masses. This is a gourmet, high-calorie, high-PHAT, microphone flavor that simply makes your mouth water.

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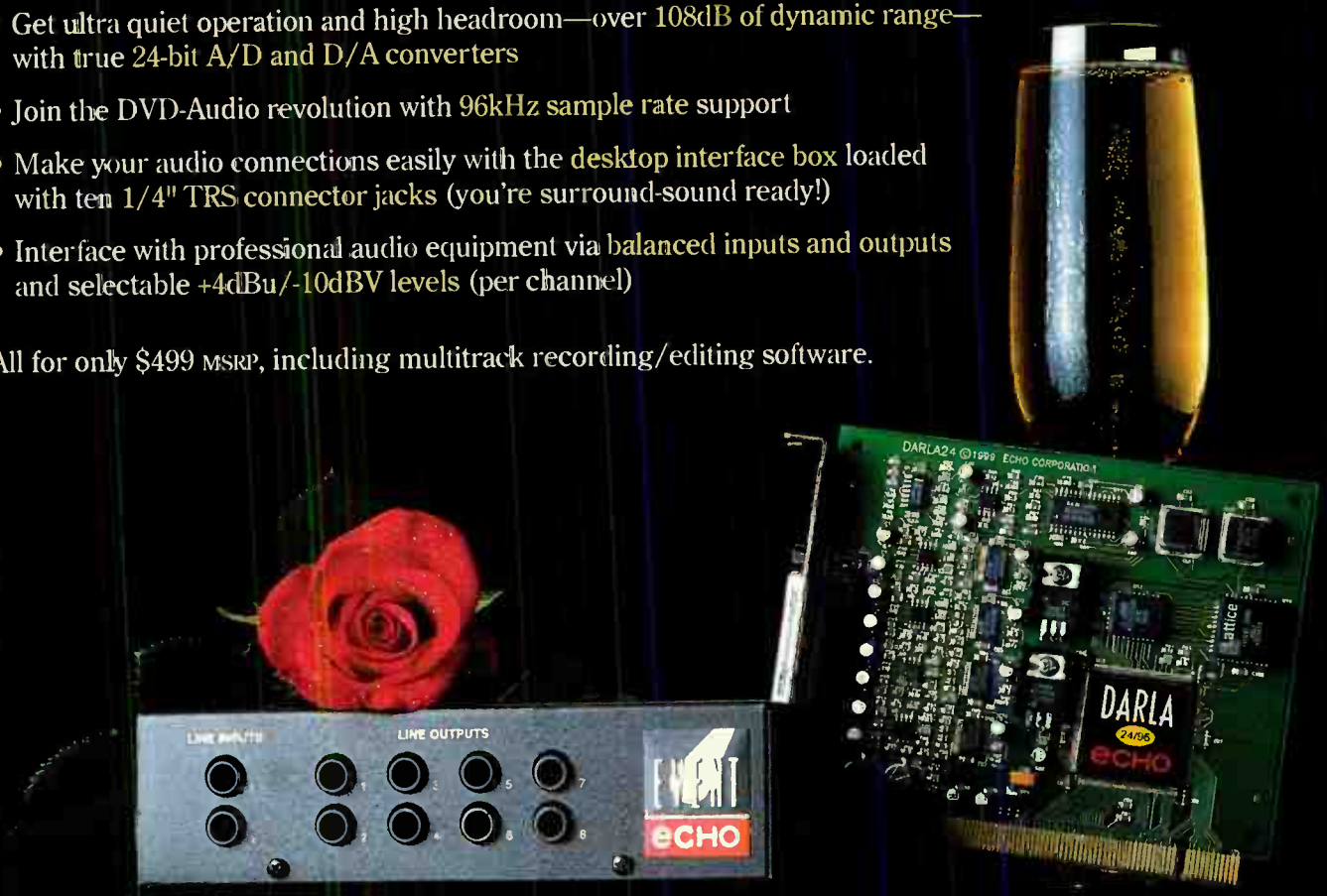

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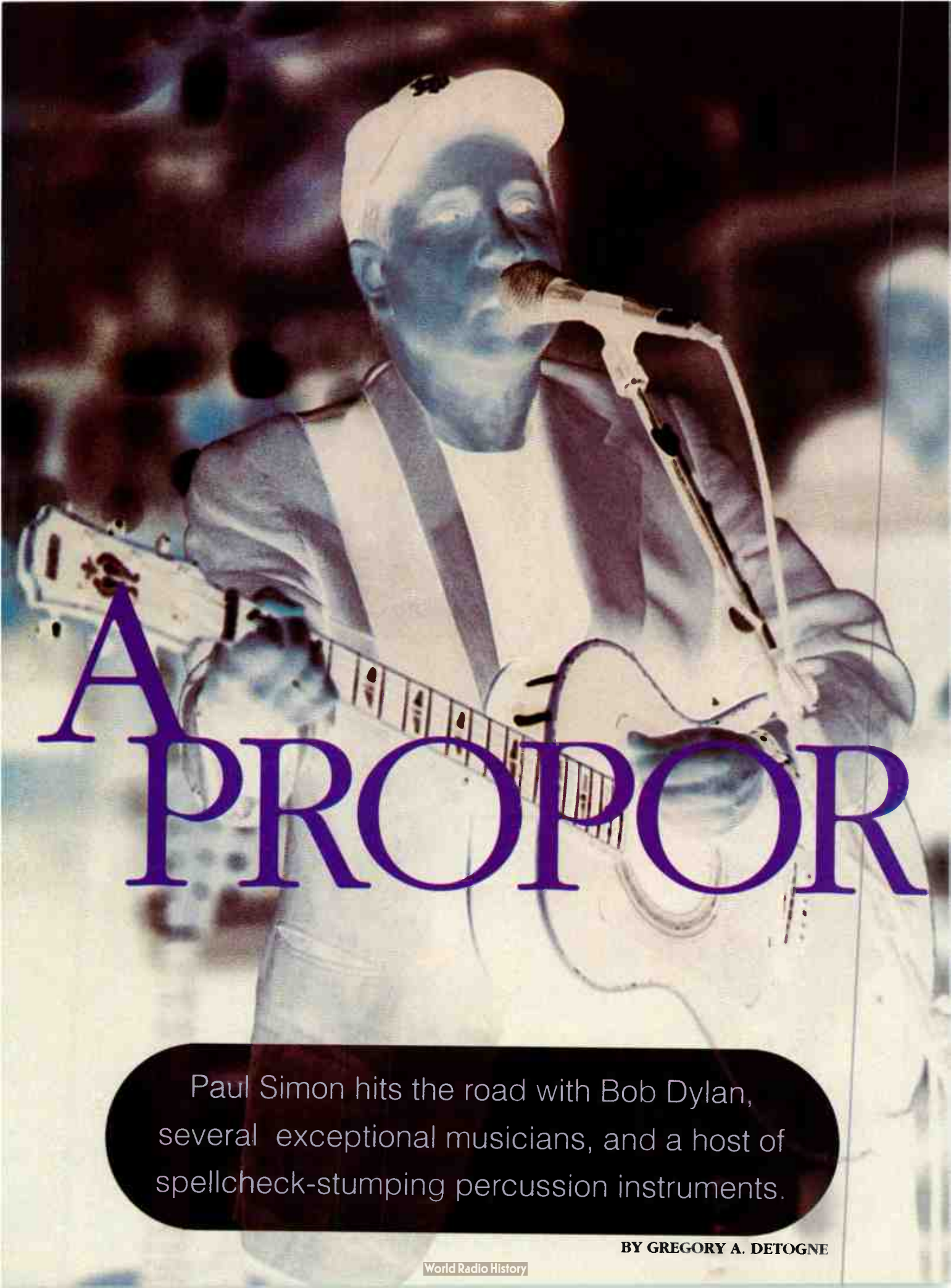
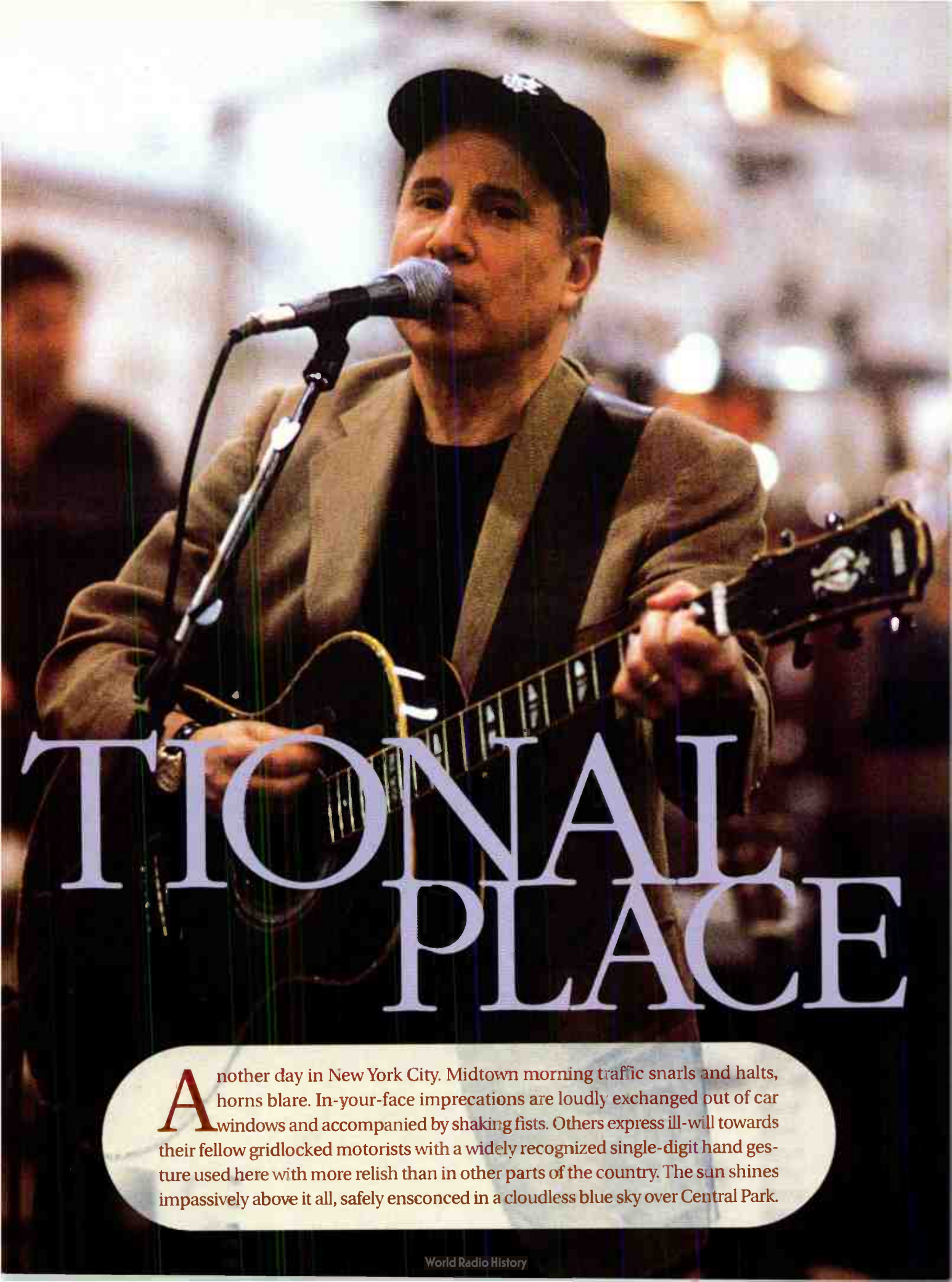


A PROPOR

Paul Simon hits the road with Bob Dylan, several exceptional musicians, and a host of spellcheck-stumping percussion instruments.

BY GREGORY A. DETOGNE



TIONAL PLACE

Another day in New York City. Midtown morning traffic snarls and halts, horns blare. In-your-face imprecations are loudly exchanged out of car windows and accompanied by shaking fists. Others express ill-will towards their fellow gridlocked motorists with a widely recognized single-digit hand gesture used here with more relish than in other parts of the country. The sun shines impassively above it all, safely ensconced in a cloudless blue sky over Central Park.

A PROPORTIONAL PLACE

PHOTOS BY PAUL NATKIN, PHOTO RESERVE, INC

Rehersal gets underway at Manhattan's SIR.

On this beautiful day in late May, you can quickly trade the din of the city for music of a different sort at 310 East 52nd Street. There, within a nondescript building wedged tightly among its taller neighbors, a free-form, pan-cultural medley is coming to life, fueled by the underpinning rhythmic strength of congas and timbales.

The sounds emanate from an open door leading to a large rehearsal space at the rear of the structure. Inside, road cases stacked high like cordwood just within the entry area thwart the visually curious. With a nod from the right person, you step around this obstacle and are swept into the vortex of efforts aimed at preparing Paul Simon for this summer's historic tour with Bob Dylan, which opened on June 6th in Colorado Springs, Colorado.

It's shortly before noon here at SIR, NYC's venerable redoubt of practice-makes-perfect for a glittering array of some of the best musicians in the world. Paul Simon, singer/songwriter, folk icon, and consummate professional known for his meticulous arrangements and abilities to seamlessly blend a polyglot of musical genres from around the world, is slated to arrive at any moment.

The rehearsal sounds echoing through SIR's corridors come from a group that includes diverse faces, some very familiar within the Simon camp: There's long-time associate Steve Gadd on drums. Bassist Bakithi Kumalo and Tony Cedras on keys and accordion; both regulars since *Graceland* days. Seated at stage left on the risers, Vincent Nguini calmly noodles on his guitar. To the left and right of Steve Gadd atop the risers are percussionists Jamey Haddad and Steve Shehan. Alain Mallet sits in at stage-right keys, just above the multi-instrumental Mark Stewart, who contributes his considerable skills on guitar, mandolin, banjo, and cello to the cause. Andy Snitzer, Chris Botti, and Jay Ashby, respectively, form the group's horn section on saxophone, trumpet, and trombone.

Simon enters the room with a large, instant presence belying his compact stature. Work begins on arranging "The Coast," a song from 1990's *The Rhythm of the Saints*. Heard on the disc as a fusion of Brazilian and West African melodies, this tour's incarnation starts with an Appalachian flair stemming from Mark Stewart's mandolin, swings wide into a samba groove fueled by the percussion section, and winds up on the notorious Vieux Carre with a trombone glissando that draws the rest of the horn section into New Orleans-like revelry.

Painstaking, mindful attention is given to each bar in the song while the arrangement takes shape. Clearly in charge, Simon nonetheless takes an egalitarian approach to the task, both soliciting and accepting ideas from the group. Joining "The Coast" on the tour's 15-song set list are time-honored classics such as "Bridge Over Troubled Water," "Mrs. Robinson," "Graceland," and "You Can Call Me Al." Two numbers are rehearsed for the encore: "Me and Julio Down by the Schoolyard" and "Still Crazy After All These Years."

Plans called for Simon and Dylan to each perform their individual sets out on the road, then take the stage together. "The trickiest thing about my set is that we jump back and forth between Simon & Garfunkel and my songs from the '70s, '80s, and '90s," Simon says of his musical selections. "The grooves change very quickly, but the whole sound comes together and makes its own musical environment. I don't think there's anything that's an imposition — everything has its proportional place."

The man measuring these proportions and then meting them out to the public is David Morgan, a seasoned industry vet who has occupied the FOH position for Simon since the first days of his *Graceland* Tour. Tall, lean, and given to wearing his long brown hair in a ponytail, Morgan is joined this day by monitor engineer Craig Van Tassel and Dave Skaff. Van Tassel comes to the gig from Broadway, where he mixed for Simon's production of *The Capeman*. As crew chief for Clair Brothers, Skaff is one of the driving forces that will help maintain the presence of Clair's new i4 line array technology on the tour.

Simon's sound this time around employs a rhythmic foundation built around the extensive percussion sections assembled by Jamey Haddad and Steve Shehan. By Morgan's count, there are 31 percussion inputs (plus 13 for drums) coming into his two Yamaha PM4000 consoles for instrumentation, including gongs, congas, timbales, djembe, bongos, frame drums, dumbeks, a cajon, and numerous small items like finger cymbals, all of which were lumped on the input list under the category "hand toys." Complementing the list are even more exotic instruments few would recognize such as a sordu, tumba, and a ceramic hadjimi handmade by Haddad.

Monitor engineer Craig Van Tassel

Morgan confides that he was a bit apprehensive at the outset of his assignment by the way the percussionists and Steve Gadd were approaching the music. "It seemed to me that their particular presentation was better suited for a quiet and intimate show in a theater or club," he recalls. "Not the large sheds and arenas we were scheduled to play. Their sound incorporated a lot of gentle fin-

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A PROPORTIONAL PLACE

ger-scraping across the membranes of different instruments, delicate cymbal parts, and the quiet rustling of shakers and triangles. The difficulty I saw was in how to deliver all of this vital nuance to crowds of 18–20,000 people without artificially coloring it or losing something in the translation."

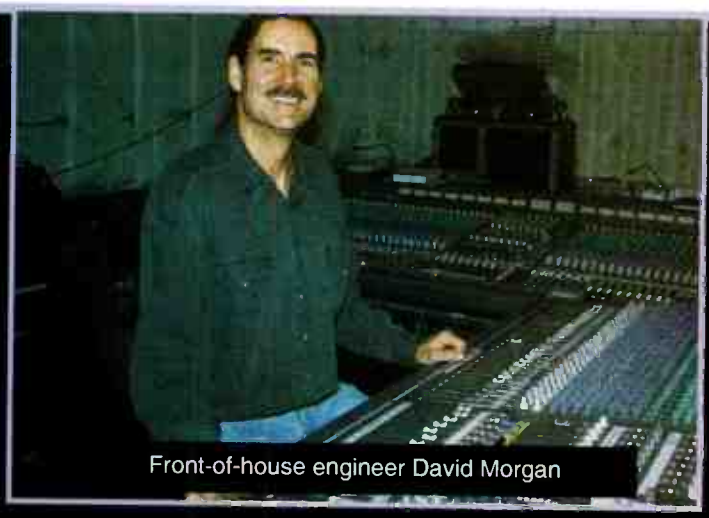
Dealing with dilemmas like this are what rehearsals are for. And that, according to Morgan, is one of the most satisfying parts of his job. Like many creative endeavors, Morgan began his task with research.

"Time permitting, the first thing I do in rehearsals is *not* setup all my gear," he says. "On the first three days of rehearsals for this show, I spent all of my time just listening."

Perched in a sweet spot he discovered on stage right downstage from Steve Gadd's drum kit, Morgan listened for how the band members heard themselves. "From that first spot, I could hear each of the percussionists with great clarity. Man, they sounded good — I was in heaven. Then I went and sat with everyone else. I listened to how Vincent heard himself coming out of his guitar cabinet, how the bass player heard himself from his amp, and so on down the line. This is a critical part of the process for me, because the last thing I want to do is change that vibe. I'm not here to make myself happy, I'm here to take what's happening onstage and place it out front in the most transparent fashion possible."

In meeting this goal, Morgan has a certain latitude of artistic license. After all, no one has yet come up with one huge microphone that can precisely pick up all of the band's natural sound. Such being the case, there is, in Morgan's estimation, "a little bridge you have to make" to achieve the same results. And along that bridge, one has to dig into the old toolbox to select the proper mics and effects, then combine them with the wisdom of experience and constant experimentation.

Armed with this philosophy, Morgan took to the risers and headed for the overflowing percussion sections with more microphones than Sam Ash could unload at half off in a Sunday clearance sale. Steve Gadd, Jamey Haddad, and Steve Shehan knew better than to get in the way as he flailed about with clamps and cables, trying different combinations. When the dust finally settled, a carefully crafted audio landscape emerged that was designed to respond to the myriad tactile pressures utilized by the musicians, while bringing precise patterns and highly tailored rejectional qualities to the fairly small area as well.



Front-of-house engineer David Morgan

"What I wound up doing was tuning each of the percussion sections as one big mic," Morgan points out. "One big, basically omnidirectional mic. This allowed the musicians to play in any part of the area without the input characteristics changing. In other words, it was one big microphone regardless of whether Steve moved around with a shaker in the middle of his area, or Jamey was madly thrashing on his timhales. This concept was especially important when it came to getting the various drums just right. When these guys are in the studio, they often use a mic on or close to the floor to capture the bottom end. I accomplished the same thing in this live situation by literally surrounding them."

Clair Bros. crew chief Dave Skaff

Starting at Steve Gadd's drum kit, Morgan deployed a scheme built with a Shure Beta 52/Beta 91 combination in the kick drum, Beta 56's at snare top and bottom, an SM81 at hi-hat, Beta 98's across the toms, and KSM32 overheads. An AKG C409 takes care of the cowbells.

At stage right, Steve Shehan's world included Sennheiser MD 409's on short booms for tumba, congas, quinto, and bongos. A Milab DC96 on a tall boom was put to work with the gong, while skin djembe numbers 1 and 2 were brought to the console with a Shure SM98A from and a Sennheiser MD 421, respectively. Another Shure KSM32 stepped into the fray for Shehan's hand toys. Shure SM81's took flight for overheads.

Jamey Haddad's "big drums" (a djembe and a dumbek) and cajon were corralled with Sennheiser MD 421's. Sordu and floor toms each spoke to an SM98 from Shure, while the rhythms from a pair of frame drums were directed to AKG C409's. The rest of Haddad's real estate included Beta 56's on bongos and "a-go-go" bells, and one more KSM32 for his personal hand toys.

[At the time of the rehearsals, both the Beta 91 condenser boundary and Beta 98 supercardioid microphones from Shure were so new that the units provided to Morgan were pulled from the first full-production trial runs for both models.]

With the percussion inputs organized, Morgan turned his attention downstage. Tube DIs custom-made by Nat Priest of NYC's Music Valve Electronics got the call for electric and acoustic bass, Mark Stewart's acoustic guitar, mandolin, and cello, plus Tony Cedras' accordion. An EV RE20 was put to use on Stewart's electric guitar cabinet. Shure Beta 87's were strapped to tall booms for vocals across the board — with the exception of Paul Simon, who became the beneficiary of a prototype cardioid version of Shure's Beta 87, the first ever to be used in a live performance situation. Why the necessity of such a device? "Paul writes difficult guitar parts that are hard for him to play while he's singing without looking to see where his left hand is on the fingerboard," Morgan explains. "The result is

that he has a tendency to wander off-axis on his vocal mic. I was looking for something that was more forgiving in these situations."

Wireless gear introduced into the Morgan design centered around Shure's U4 UHF receivers. Transmitters included Shure's U1 body-packs (used for Simon's guitars) and a pair of U2/Beta 87 handheld units, both of which were slated for use by Simon and Dylan when they joined each other's bands onstage.

The RF plan was an offshoot of another conceived and coordinated prior to tour rehearsals by Jim Corona, Simon's "technical ombudsman." Utilized with good results for Simon's performance of "Mrs. Robinson" at a Yankee Stadium tribute to Joe DiMaggio, Corona's ballpark wireless world was recreated in a 12-channel format for the shows with Dylan.

Any amount of time spent with David Morgan reveals his candid qualities. "If there is a secret to mixing Paul Simon's vocal," he says, "it's that there is no secret at all. There is little subterfuge going on in this show. These are just incredibly talented people playing great arrangements of great songs."

For the moments when some coloration is required, however, Morgan draws from an aural palette of effects housed in a pair of outboard racks. New among the group is an M3000 processor from TC Electronic that he first discovered at the Winter NAMM show earlier this year. Housing two engines, the device provides Morgan with a subtle doubling on horns, along with a very transparent reverb.

A Lexicon 480L digital effects system sees dual-duty as a vocal reverb on one side, and a percussion reverb on the other. A staple in Morgan's toolbox for longer than he cares to remember, a warhorse Lexicon PCM70 "is still the best snare reverb." Even older, an Advanced Music Systems rmx16 digital reverb is used for toms and timbales. "Maybe I'm showing my age a bit on this one," Morgan concedes, "but they really are good old troopers, and they do the job better than anything else I've found."

Another regular in Morgan's effects collection, an Eventide DSP 4000 Ultraharmonizer™ finds a place on Simon's vocals, along with a model 2290 digital delay from TC Electronic, which serves up vocal slap. A TubeTech CL2B stereo compressor is also part of Simon's vocal mix, while Morgan's choice for Simon's four acoustic guitars (two Martins, including a Paul Simon signature model, and a pair of Yamahas), is a Summit DCL-200 dual-channel compressor/limiter.

During the final days of rehearsals at SIR, it was still an open question as to whether Bob Dylan would show up or not. As a natural alter ego to Simon's preproduction preparedness, Dylan's informality brought a spontaneous air to the tour's possibilities.

"I don't know — we'll probably rehearse with Bob the day before the opening show in Colorado Springs," Morgan says with a shrug. "That one will be kind of a shoot-from-the-hip situation. As for me, I'm just about ready. All I have to do now is make sure that none of the flavors dominate, and everything occurs at the right time. It's like traveling on different rivers. You spend a little time on the Amazon, the Congo, the Mississippi, and then the Ohio. It's a pleasure to be here. Not only is Paul a great arranger of notes, he's a great assembler of human beings. These guys can really play. That's always the way it is."

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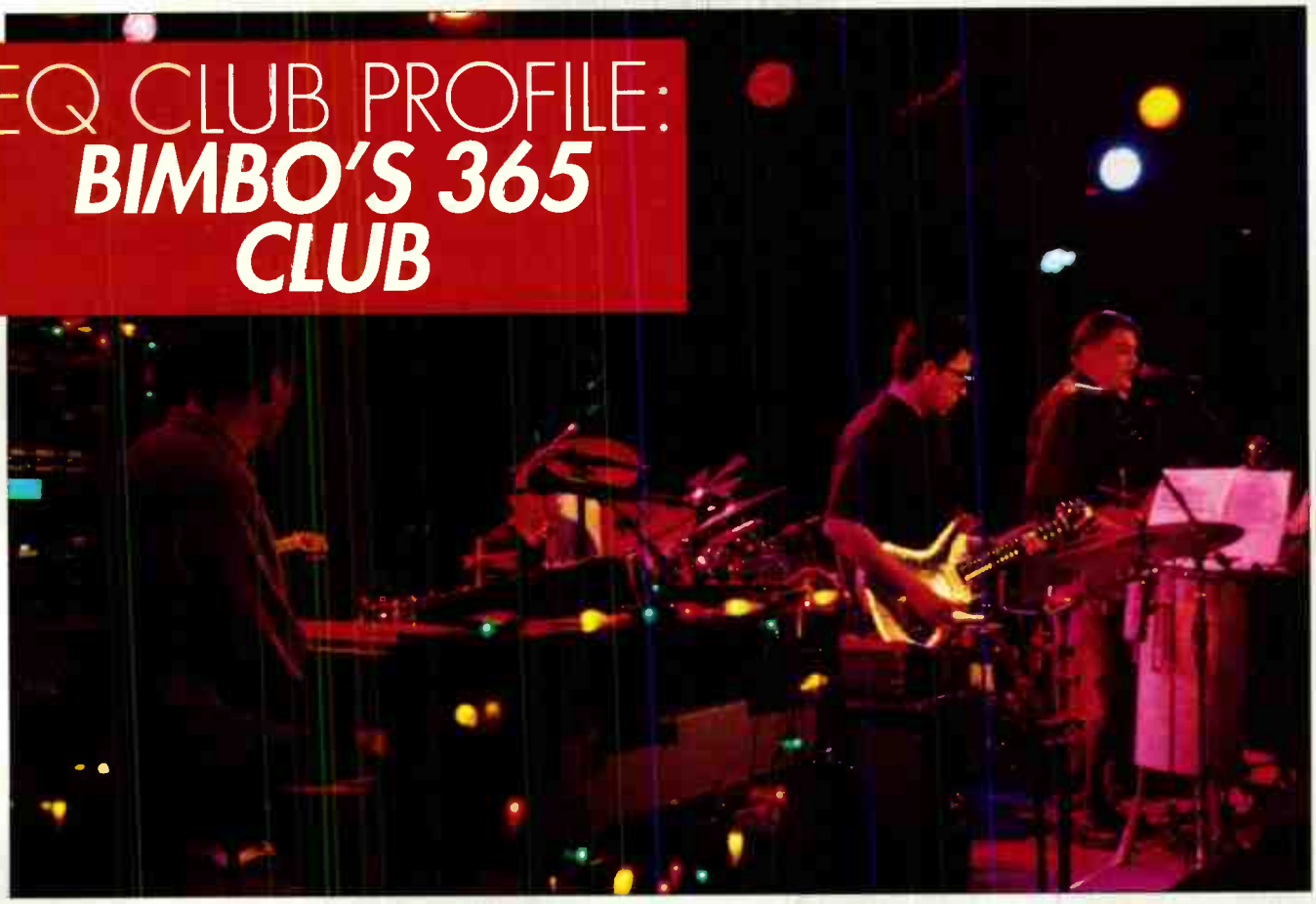

BY STEVE LA CERRA

Way, way back in 1931 — even before *EQ* was a glimmer in the audio industry's eyes — Agostino Giuntoli opened a club called the 365 Club on Market Street in San Francisco. Although it moved in 1951 to its present location, the 365 Club has remained owned by Giuntoli's descendants. Now, slightly renamed as Bimbo's 365 Club, the venue — with its formal decor including red velvet curtains, murals, and tuxedo-clad barmen — has recently hosted acts as diverse as Goldie, Skatalites, Shawn Colvin, Buzzcocks, Bootsy Collins's Rubber Band, Cassandra Wilson, and Chris Isaak. The staff at Bimbo's 365 has collectively logged in quite a few road hours, so they understand the needs of touring artists and crews — not only in terms of sound and lighting production, but also in terms of clean dressing rooms and showers.

The 29- x 33-foot stage at Bimbo's is built on three different levels, making the need for additional risers rare. At stage right of this main stage, there is a smaller stage that is sometimes used for opening acts — thus creating a minimum of mayhem during a set change. Electricity for production is supplied on a 200-amp service for lighting and a separate 100-amp service for the PA system, preventing electrical interference problems between the lighting rig and the in-house PA.

For the past six years, Scott Burke has been the production manager at Bimbo's 365. He explains that he's been "upgrading the PA on-the-fly ever since I arrived here. We started with a very small system that had a couple of old Rane EQs and one reverb." From the looks of the present system, it would appear those days are over. Located slightly off-center of the stage, about 70 feet from the downstage edge, is the centerpiece of Bimbo's audio system: a Midas XL-200 console with 44 mono inputs (each fed with a dedicated snake line from the stage), as well as four stereo modules (a quick side note: the XL-200 uses separate TRS jacks for the insert sends and returns, don't forget the appropriate cables if you're bringing in rack processors).

You're sure to get a warm reception at this San Fransisco club

A well-outfitted assortment of FOH toys includes TC Electronic M2000, Lexicon MXP-1 and LXP-5, Yamaha SPX-990, and Roland SDE3000 digital effect units. There's no shortage

ABOVE: Scott Burke, FOH engineer.
PREVIOUS PAGE: Latin Playboys take the stage.

of comps and gates either, with one dbx 1046 and 2 dbx 166 compressors, and two dbx 363X and two Aphex 622 gates. Also provided at the mix position are a CD player, cassette deck, and communications.

House PA for the 685-capacity room consists of six Meyer UPA cabinets (flown in an arrangement of two left, two center, and two right), four Meyer 650 subwoofers (situated on the floor nearby the stage), and two EAW cabinets (each with a 15-inch and a 2-inch driver) flown behind the mix position for fill purposes. The entire main PA is powered by Crest 6001 and 7001 amplifiers. Several EQs are available for tweaking frequency response of the PA, including a Klark-Teknik DN 360 stereo 1/3-octave for the Meyer UPA cabinets, an Ashley 1/3-octave unit for the delay system, and an Ashley 1/2-octave EQ for the subwoofers.

Located at the stage left side is the monitor mix position, featuring a Gamble SC32 console, 32-input/8-mix outputs. A TT patchbay is installed in the monitor rig, facilitating configuration changes for the system. A total of ten Sound On Stage floor wedges (each loaded with a JBL 2445 2-inch driver and a JBL E130 15-inch driver) are provided for the eight stage mixes, all of which are biamped with Crest amps and UREI active crossovers (crossover point is 1.2 kHz).

As you may have already guessed, Bimbo's offers a complete selection of mics, including two Audio-Technica AT4050's, three Shure Beta 56's, nine Shure SM57's, eight Shure SM58's, three Beta 58's, four Beta 87's, three SM81's, four AKG C418's, five Sennheiser MD421's, and two AKG D112's. Rounding out the complement of stage gear are six Whirlwind Director and four Countryman direct boxes, and 30 microphone stands (which, as noted in Bimbo's Web site, are all "in perfect working condition"). Piano aficionados will be happy to note that there's also a 6-foot Yamaha C3 baby grand piano.

One of the big factors in the rather dry acoustics of Bimbo's 365 is the ceiling — a drop-ceiling that is covered with a soft layer of felt facing the room. This is a feature that contributes to what Scott describes as "an incredible warmth in the sound of the room. I've been at a lot of clubs both here and abroad, and Bimbo's is one of the best rooms I've ever mixed. The ceiling keeps it from sounding cavernous, and it's generally positive for most engineers. Good engineers can have great nights in here, and great engineers can leave the audience weeping with joy by the end of the night." Depending upon the nature of the act, the floor in front of the stage can be cleared or set up with tables for sit-down shows.

As mentioned in the opening, much of the production staff at Bimbo's 365 have at some point or another in their careers spent time on the road. So it doesn't come as that much of a surprise to learn that the venue has two dressing rooms up on the second floor, stage right. The larger of the two dressing rooms has a shower ("with plenty of hot water!") and a smaller room nearby that can be used as a production office if needed.

Visit the Bimbo's 365 Club Web site at www.bimbos365club.com

"Q"ING A "TERP"

For the past several years, sign-language interpreters have been legally required at all large music and speaking events. (See the sidebar for the legal requirement from the ADA.) That's because many people with hearing loss attend these functions, and, at the very least, they need to be able to receive emergency instructions. However, interpreting the words of a song can make the experience much more meaningful for the deaf, as well as adding a visual element for the rest of us that can at times be quite beautiful. That's why you often see a "terp" (their own official slang for a sign language interpreter) on-stage at music events.

A quick and easy solution to cueing a sign-language interpreter at an event

Their normal position is next to the main FOH speakers for maximum audience visibility. But this position can be problematic for the interpreter to be able to hear and understand the words, and then sign them for the audience. Here's why: Have you ever stood directly to the side of the main speakers and tried to understand the words? The combination of the backline instrumentation and slapback from the rear wall usually renders any spoken or sung words unintelligible at that position. While you could add a separate floor wedge, that's a lot of extra weight and real estate on an already-cramped stage. Plus, a floor wedge will only add extra volume to the stage, which can cause feedback and intelligibility problems. It's not an optimum solution.

There is, though, a quick, lightweight, and simple solution to this dilemma: headphones. Yes, a pair of headphones helps block out the roar of a loud stage, while it allows interpreters to set their own comfortable volume level. But how do you provide someone with a headphone feed from the monitor console without bringing an extra amplifier channel, and how do you give them a separate volume control they can adjust themselves? After all, you just can't plug a set of headphones directly into a speaker or amplifier channel without bad things happening.

These questions went though our minds as Mike Scarfe (MHA Audio) and I were preparing the equipment list for the Western Maryland Blues Festival last month. We were contracted to provide headphone feeds on two different stages, and didn't want to rack up a separate amplifier just for that function. So he asked, "How about your Q-Box?" and I said, "Hey, I just used that as a headphone amp last month on a remote and it worked great." Since we each had one it was the perfect solution.

For those of you not familiar with it, the Q-Box is a piece of audio test gear made by Whirlwind. It can provide a 400-Hz tone, test a microphone or audio processor for output signal, or drive an internal speaker or set of headphones using anything from mic to line level. The street price is around what you might pay for a standard stage microphone — $150 or less. And, to top it off, it has a hook to hang on your waistband or belt. It seemed like the perfect solution, and, indeed, it was...

We simply took a separate XLR auxiliary feed from the monitor console, and had the sound tech put in only the vocals and emcee mics on that mix. Each stage had a Q-Box and headphones. We laid out an XLR feed with an extra 25 feet of cable to the interpreter's location and labeled it "terp." The Q-Box will accept either a male or female XLR cable, as well as a 1/4-inch TRS plug at any level you can possibly get out of a console, so it's pretty much a no-

BY MIKE SOKOL

brainer for hookup. I used some $20 Radio Shack (KOSS OEM) headphones that were around 50-ohm impedance so that the 9-volt battery in the Q-Box could give them enough kick for reasonable volume. I don't think my 600-ohm headphones would be loud enough with a 9-volt peak-to-peak swing. For very loud stages, closed headphones (rather than open-air) would be a better choice since they would offer more attenuation of the sound from the stage.

Another (albeit higher-priced) alternative to a Q-Box is a wireless in-ear monitor system. (I'm familiar, for example, with the Shure PSM600/PSM700.) This gives you the same headphone functions, but with the added advantage of being wireless. If you have the budget, it would be the high-class solution.

Once set up, we never had to worry about adjusting headphone levels for anyone since the Q-Box gave them their own volume control. It was a beautifully simple solution to the potentially messy problem of an additional floor wedge. And the "terps" just loved it. It certainly reduced their strain by the end of the day, and made our monitor engineers easy heroes. Such a deal.

AMERICANS WITH DISABILITIES ACT (ADA)

II-7.1000 Equally effective communication. *A public entity must ensure that its communications with individuals with disabilities are as effective as communications with others.*

In order to provide equal access, a public accommodation is required to make available appropriate auxiliary aids and services where necessary to ensure effective communication.

Examples of auxiliary aids and services for individuals who are deaf or hard of hearing include qualified interpreters...

If anyone in the audience is hard of hearing and requests an interpreter, promoters and facilities with public access are compelled, at their own expense, by federal and state law to provide an interpreter.

ROAD GEAR

32 FLAVORS

Samson's UHF Synth 32 system features SAW (Surface Acoustic Wave) filters to enhance reception and PLL synthesized VCO circuitry to benefit users in multi-system situations. The UHF Synth 32 system provides 32 selectable UHF frequencies in both the receiver and transmitter for expanded versatility. The system's R32 half-rack receiver features a large LCD on the front panel that indicates Channel, Group, Audio Level/RF Level, Diversity A/B antenna, and Mute. The R32 also includes discriminating Noise Squelch circuitry; balanced XLR and 1/4-inch outputs; a mic/line switch; tuned, molded antennas; and an optional rack-mount kit for touring and fixed installations. The T32 beltpack and H32 handheld transmitters feature a battery life indicator; a full complement of on-board controls; a 3-pin mini-XLR connector; a detachable guitar cable; and 9-volt battery operation. For more information, call Samson at 516-364-2244, fax them at 516-364-3888, or visit www.samsontech.com. Circle EQ free lit. #114.


Ramtech Ramlatch

SHOW STARTER

Sennheiser has introduced the next generation of its MKE2 Red Dot miniature lavalier microphone, the MKE2-4 Gold. Designed to virtually eliminate sweat-out problems commonly associated with the powerful stage lights, make-up, and heavy costumes of theatrical productions, the omnidirectional MKE2-4 Gold incorporates a dual diaphragm design whereby the top diaphragm protects the lower one from sweat-outs. Two new acoustic end caps offer frequency response contouring: a "short" cap adds 2 dB at 10 kHz, while a "long" cap increases the frequency by 8 dB at 10 kHz. Removal of the end cap ensures a flat frequency response. The retail price is $425. For more information, call Sennheiser at 860-434-9190, fax them at 860-434-1759, or visit www.sennheiserusa.com. Circle EQ free lit. #115.

COLORFUL CONNECTION

The world-famous Ramlatch connector from Ramtech Industries is now being offered in five colors: red, blue, yellow, white, and green. After the successful reverse engineering of the original gray AMP Quicklatch connector in 1994, and with its improved modifications (an indestructible black ABS shell, a 12-gauge steel chrome handle, and steel and aluminum machined handle cams), Ramtech has listened to the demands of the pro audio industry and responded accordingly. Including the Ramlatch's ability as a completely compatible connector to the old Quicklatch, it has become the choice connector for many touring rigs, TV studios, and theme parks. The new colors are being offered so users can color-code their interface audio system. The colorful connectors are available for $29.95. For more information, call Ramtech Industries at 800-817-2683 or fax them at 352-466-0906. Circle EQ free lit. #116.

AM-AZING

Lectrosonics's AM16 automatic matrix mixer has 16 mic/line inputs, a patented automatic mixing algorithm, and a 16 x 12 programmable matrix in a 2-space, 19-inch rack mount package. Versatile control options and built-in conveniences accommodate mix-minus sound reinforcement, recording, and teleconferencing. Set-up and operation is simplified with supplied LecNet software and RS-232 serial interface. The input section of the AM16 provides 16 low-noise preamps for microphone or line level balanced or unbalanced inputs. It also provides 17 programmable control input and 19 programmable control logic outputs. The unit also includes a built-in noise generator and 15 non-volatile memories to store multiple setups. Multiple AM16 mixers can be daisy-chained to add additional inputs. For more details, call Lectrosonics at 505-892-4501 or fax them at 505-892-6243. Circle EQ free lit. #117.


Lectrosonics AM16

Stage Accompany XL-bin

BASS BIN

The XL-bin from Stage Accompany is the first of a new range of low-frequency enclosures. This cabinet will deliver maximum bass without the use of a processor. With the cabinet's "Two Stage Tunnel"-loading (TST-loading), a double SA 15-inch configuration has a maximum SPL of 140 dB at 2000 watts peak. This new technology combines the sonic qualities of a front-loaded system with the high efficiency of a horn-loaded system. The air mass in front of the speakers is coupled to the loudspeaker/cone surface, generating a resonance over a wide frequency bandwidth. To increase this efficiency, the output of the reflex vents enters the tunnel at 1/3 of the tunnel length from the speaker baffle. For more information, call Stage Accompany in the Netherlands at +31 229 282930 or visit www.stageaccompany.com Circle EQ free lit. #118. EQ

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ALESIS

* Headphones or monitor speakers required

CIRCLE 93 ON FREE INFO CARD

QSC PLX 3002 POWER AMPLIFIER

BY STEVE LA CERRA

When QSC introduced its PowerLight™ Series of amplifiers several years ago, the live sound world was pleasantly surprised because the amps offered serious power capabilities *and* light weight — largely due to QSC's PowerWave™ audio technology. As often is the case with new technology, QSC has proceeded to employ PowerWave in subsequent amplifiers — including the PLX 3002 tested herein.

QSC's power performer is a lightweight in mass only

The PLX 3002 is a 2-channel amp capable of delivering as much as 3000 watts bridged into a 4-ohm load. While its capabilities into stereo loads are more modest, it can still deliver 900 watts stereo into 4 ohms across the audio frequency range. Housed in a 2-space rack chassis, the PLX 3002 is surprisingly compact for its power rating and weighs in at only 21 pounds.

Front-panel controls are minimal but functional, with a power switch, gain controls for each channel, and a series of LEDs showing amplifier status. When you first power up the amp, the green power indicator and red clip-protection indicators light until the amp has stabilized and safe operating conditions are sensed at the output. Additional LEDs show signal level, bridge mono, or parallel input mode. Also visible on the front panel are a series of vents for the internal fan. Note that the PLX 3002's variable-speed fan sucks air in from the rear vent and blows it *out* of the front panel.

Rear-panel features of the PLX 3002 include balanced XLR and TRS inputs, binding post and Speakon™ output connectors, and a series of DIP configuration switches. QSC has wisely chosen to print a reference chart pertaining to these switches right on the rear panel. Though the switches look a bit intimidating, they allow easy adjustment per channel for peak limiter on/off, LF rolloff on/off, and LF frequency 30 Hz or 50 Hz. Stereo or parallel-mono operation and bridging on/off are also controlled from these DIP switches. Recessed into the rear chassis, the switches are out of the way of inquiring hands.

ROAD Test

MANUFACTURER: QSC Audio Products, Inc. 1675 MacArthur Boulevard, Costa Mesa, CA 92626. Tel: 714-754-6175. Fax: 714-754-6174. Web: www.qscaudio.com.

SUMMARY: High-power audio amplifier using QSC's proprietary PowerWave audio technology.

APPLICATIONS: Live sound reinforcement, DJ systems, on-stage amplification for musicians.

STRENGTHS: Good bang for the buck; high headroom; offers stability into a variety of loads.

WEAKNESSES: No filter for ventilation fan; metal chassis is somewhat lightweight — we recommend careful rack mounting for road use.

PRICE: $1795 **EQ FREE LIT. #:** 119

We first used the PLX 3002 on the road in a monitor rig with the band Driving Blind. We placed the amp into the bottom two spaces of a non-shock-mounted road rack. If you're racking a bunch of PLX 3002's — or any amplifier for that matter — we strongly suggest using rear rail supports to prevent the front panel from twisting due to the weight of the amp. (Rear supports are available from QSC.)

On one particular run, the PLX 3002 sat overnight in a van when outside air temperature was about 0 degrees Fahrenheit. While we certainly don't make a habit of exposing electronic gear to these kinds of conditions, touring gear does often end up exposed to extremes during travel — putting the gear's mettle to the test. The cold didn't hamper operation of the PLX 3002 (we let it warm up for about 15 minutes before powering it up).

Since we knew the monitor mixes would not include drums or bass, we set LF rolloff to 50 Hz to keep any LF crud from interfering with system efficiency. The Electro-Voice cue wedges we were using produced plenty of clean, clear vocals and crisp acoustic guitar when powered by the PLX 3002. The circumstances didn't require high-volume monitoring, and the 3002 cruised through this gig.

Next, we pressed the PLX 3002 into service in a "PA-on-a-stick" system for voice reinforcement at a corporate meeting. Powering a pair of JBL MR925's, the amp produced plenty of clean power. Thus helping the speakers easily cover the audience of 350 people. Since we didn't have an outboard limiter on this system, we switched-in the amp's clip-protection limiter to prevent any "accidents." Although the PLX 3002's fan noise was noticeable to those of us who sat next to the gear, the audience members did not hear it. We'd say it was quieter than the average high-powered amplifier. In this situation, setting the LF rolloff to 50 Hz helped keep popping P's to a minimum, improving clarity of the voices.

One of the more unorthodox uses for this PLX 3002 was during another Driving Blind show where the bass player's amp blew up at the start of the second set. We pressed the PLX 3002 into service to power his 18-inch bass cabinet with one channel, while, at the same time, the other channel powered the monitors (now connected in parallel). The QSC amp didn't seem to mind this unusual arrangement of an 8-ohm load on one side and a 2-ohm load on the other. Both the bass rig and the monitors continued playing loud and clear. It was readily apparent that the PLX 3002 was pushing a lot more air from the bass cab than the bassist's original amplifier, and that it produced a low-end tightness and clarity that we hadn't heard before

Once we had solved Driving Blind's equipment problems, we used the PLX 3002 to power a pair of full-range, JBL TR225 PA cabinets. Compared to the amp we had been using — which, granted, was a rather old Carver model — using the QSC amp on the mix was like taking a layer of gauze away from in front of the JBL's. Transient response was excellent on acoustic guitar and vocals, and the PLX didn't run out of gas when bass and drums kicked in for an SPL that peaked at about 105 dB in the room. The PLX 3002 gave us the impression that it had the reserves to provide whatever power we might need for the occasion.

At one point we had a loudspeaker connector with + and – shorted. Initially, the PLX 3002 showed no indication of a problem. However, as soon as we applied signal to the input, the clip LED on that channel lit, telling us that a problem existed (which we then traced to the cable). Since the amp is capable of delivering so much output, this kind of protection is paramount in preventing harm to associated gear.

When we bridged the 3002 for mono operation, we thought we might have heard a bit more extension in the bottom end — but it could have been a mind-over-matter thing. A neat feature is that when you switch the amp into parallel operation, [the] input to channel 1 *or* 2 will drive both outputs. We found that the amp was quiet in any of these modes, emitting only a low-level hiss with outputs all the way up and no signal applied. In addition, turning the amp on or off never produced any noise (i.e., no clicks or pops) other than audio applied to the input.

We found the PLX 3002's back-to-front cooling arrangement to be a double-edged sword for rack mounting. Since hot air rises, there's a buildup of heat within any rack when air flows front to rear. By sucking hot air from the rear and blowing it out of the rack entirely, the PLX 3002's fan gets the heat out of the enclosure, keeping the amp happy. QSC intentionally designed the amp without a filter so that there'd be no maintenance issue with filter replacement, and no worry that a clogged filter would shut down the airflow and fry the amp. On the other hand, it's scary to imagine the kind of crap that'll be sucked in under permanent install conditions in a slimy club. Perhaps a "master" filter at the rear of the rack could be used to keep the rack interior dirt-free while not impeding airflow.

Power amplifiers are like the Rodney Dangerfields of live sound — I tell ya', they get no respect. That is, until something goes wrong with one and leaves you at a gig scratching your head. QSC's thorough protection scheme guards the PLX 3002 from short circuits, open circuits, thermal overload, and RF overload. Further, an 8-Hz high-pass filter at the input prevents DC input from passing, and if DC is sensed at the outputs, the amp mutes to protect your speakers. The PLX 3002 sounds good, delivers a lot of power, and should last for years. Definitely a contender! EQ

MANUFACTURER'S SPECIFICATIONS

Power Output in Stereo, Both Channels Driven: 550 watts into 8 ohms @ 0.03% THD (20 Hz to 20 kHz); 900 watts into 4 ohms @ 0.05% THD (20 Hz to 20 kHz)
Power Output in Bridged Mono: 2000 watts into 8 ohms @ 0.1% THD (20 Hz to 20 kHz); 3000 watts into 4 ohms @ 1% THD (1 kHz)
Frequency Response: 20 Hz to 20 kHz, ±0.2 dB. 8 Hz to 50 kHz, +0/-3 dB
Damping Factor: 500 or greater (@ 1 kHz and below)
Sensitivity: 1.7 Vrms (+6.6 dBu) for rated power @ 8 ohms; 1.5 Vrms (+5.7 dBu) for rated power @ 4 ohms
Physical: 19.0" W x 3.5" H x 13.25" D; net weight, 21 pounds

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CIRCLE 69 ON FREE INFO CARD

EQ Drive!

EQ DRIVE!

FIGURE 1: ACID on a G3.

RUNNING WINDOWS AUDIO SOFTWARE ON A MAC

BY MICHAEL ANTHONY

With Apple's remarkable resurgence, developers are migrating back to the Mac. Although many companies now provide Mac equivalents of their PC titles, not everyone is convinced just yet; some excellent Windows titles like Wavelab, Mixman, Sound Forge, B-Box, ACID, and various DirectX plug-ins still aren't available for the Mac. Or at least they aren't officially! But, after reading this article, Mac owners need no longer have "Windows Envy": not only will you be able to run all that great Mac software you know and love, but most Windows audio software as well.

Here's the gear needed to bring about such miracles.

- Mac with a 233 MHz processor or better (G3 preferred). Although a fast 604 might work, I wouldn't guarantee the results.
- A copy of SoftWindows 95 or (better yet) 98. Virtual PC isn't fast enough, doesn't seem to have proper 16-bit emulation, and there's no MIDI functionality. SoftWindows has MIDI Out, and I'm consulting with Insignia in the U.K. in an attempt to encourage them to implement MIDI In as well.
- Windows software of your choice. I've had little difficulty in running any programs, including top-drawer MIDI sequencers.
- Any PCI card with a Sound-Manager driver, including "old standbys" like the Korg 1212 I/O.
- QuickTime 3.0 or better (4.0 is now available, so go get it!)
- The old Sound Control Panel version 8.0.5 (to set I/O assignments). If you have Digidesign's AudioMedia III, Arboretum recommends using version 1.4.1 of their driver.
- Lots of RAM — at least 64 MB free beyond SoftWindows' requirements, so install a minimum of 150 MB. This means assigning around 87 MB to SoftWindows in the usual Mac info box. With my Apple 850 20-inch monitor, if I assign 4 MB to SoftWindows video, I can run as close to full screen as desired in thousands of colors mode (see fig. 1) with little detrimental effect on overall performance. Considering how well the system works with Wavelab, I believe my Mac is approaching or even surpassing 200 MHz MMX performance — not bad for an OS (Windows) overlaid on another OS (DOS) that itself is overlaid on another OS (Mac)!

Emulation is the sincerest form of flattery

GETTING STARTED

As Macs are true "plug and play," installing SoftWindows is easier and faster than installing Windows in a regular PC (and there are no ISA cards to worry about). Just follow the documentation and you'll be up and running in no time.

Do bear in mind that, like Windows, SoftWindows is fundamentally an application, not an operating system. That job is left to DOS. SoftWindows has DOS under there as well, but unless you go rooting around, it's unlikely you'll ever have to deal with it.

There is one important caution: After installing SoftWindows, making all your MS Control Panel adjustments, setting up the desktop, and so on, quit the program. Now go into your Mac preferences folder and make a copy of the SoftWindows preferences. Put this file in some safe place, as it may just save your butt someday if your main preferences file ever becomes corrupted. This has happened to me on one or two occasions, and it's not fun to start over from scratch.

SPEEDY SOLUTIONS

Since beginning my experiments, another company has released a new PC emulator called Blue Label (www.lismoresoft.com). I haven't checked it out thoroughly, but it does look promising — inexpensive, yet apparently complete enough to run audio software.

Meanwhile, because we are dealing with emulation, escape velocity speeds are essential to accommodate the emulator's bottlenecks, especially with DSP (Digital Signal Processor) emulation. The results I've had running my current system have been a little short of phenomenal. Courtesy of the folks at Channel Dynamics U.K./Power Logix, I'm presently blessed with a ZIF 400-MHz processor jacked up to 466 MHz. This level of processor speed means that with Wavelab, ACID, or Sound Forge, I've been able to run a fair number of both VST and Direct X plug-ins simultaneously, *and* in real time (fig. 2). This is very, very cool, but lightweight processors need not apply.

The point here isn't "mine's better than yours," but that mine can do both. This leads to other advantages, such as being able to "drag and drop" between the PC and Mac operating systems. For example, using ACID's exceptional loop tuning capabilities, I can do my adjustments in ACID, then drag the file over to the Mac side for further work in another application (e.g., Cubase), then return to the PC side.

Before closing, here are a few final recommendations and observations.

FIGURE 2: Wavelab on a G3.

If you have the hard-drive space, set up partitions. I've dedicated a 1 GB partition to PC programs. This keeps things organized and allows for smaller file sizes. Moreover, if you lose one partition, it's unlikely you'll lose them all. Believe me, this is a good thing!

Given the current low cost of computers, some may quite rightly ask, "Why not just buy a PC?" Well, I covet only a few Windows programs, and buying an entire truck just to haul an occasional load seems a major waste. Besides, this setup's level of quality is just fine, and I don't have to maintain two completely different operating systems. Furthermore, being the second banana does have its perks: Macs can usually deal with Windows-centric software (most files, Internet downloads, even their operating system). Running Mac software on a Windows machine is far more complicated and, in most cases, impossible.

Incidentally, the sound quality of the average Mac's built-in audio never ceases to amaze me. Although I have a range of excellent PCI sound cards, whenever I do lectures and shows I use SoundManager as well, mainly to give people a realistic idea of what they can expect. Although, while under PC control you'll be restricted to 16-bit operation, I've never found this the least bit confining. In fact, showing ACID running on a Mac never fails to bring down the house.

One other sacred cow that has also just been put out to pasture is the exclusivity of all those lovely SoundFonts (SF2 files). Formerly the privileged domain of lucky PC/Sound Blaster owners, this wealth of audio sound bytes is finally available to Mac owners thanks to Bitheadz's Unity DS-1 for the Mac.

For the moment, companies like Steinberg, Emagic, Cakewalk, MOTU, and Opcode (the list goes on) are on the right track — biplatform compatibility. Until there's a uniform hardware platform, the software must carry the weight of compatibility.

Eventually, though, all this concern with emulation and compatibility may be moot. I predict that, in the long run, a single, trim OS with full across-the-board compatibility will be the only really worthwhile option (actually, I've heard rumors that IBM is working on just such a beast). The only decision to be taken then will be based on the quality of available hardware. In the meantime, hit your Turbo Start buttons, and let's motor in both worlds!

Musician, producer, software developer, and former L.A. resident Michael Anthony resides in Ireland. Besides launching the label Citadel with his wife, Cara, he works as a multimedia consultant — primarily for Apple Sales Ireland, Trinity College Dublin, and the Irish Department of Education.

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CIRCLE 67 ON FREE INFO CARD

RAID: NOT JUST A BUNCH OF DISKS

Fewer things are more valuable than our sound and data files. Hard drives are becoming increasingly more reliable, but that shouldn't give anyone a false sense of security. And, yes, writeable CD-ROM is the most affordable and the most practical solution to the "long-term" storage problem. However, for studios that must store large amounts of data for multiple clients and ongoing projects, daily reliability (read: lack of downtime) is equally as important as long-term storage. If you're responsible for data integrity, ask yourself the following questions:

- How much hard-disk space would be "adequate" to save all the necessary jobs?
- What's the most effective protection against an unforeseen catastrophe?

Make sure that you never lose an important piece of data again

We all know that hard drives have become a lot more affordable, and new ways to backup your data seems to pop up on a monthly basis. However, you should know that one other option exists. One that can give you access to huge amounts of data, while offering data protection and backup schemes that can help you to avoid (or greatly reduce) any system downtime: RAID!

Standing for Redundant Array of Independent Disks, pros once called these multiple-drive systems JBOD (just a bunch of disks). Nowadays, large businesses, Web-site servers, and media production companies often link drives together as a single, huge-capacity drive. In addition to improving data integrity, a RAID system can also speed things up (more tracks, anyone?) by increasing transfers to and from "the disk." Depending on the level of security chosen, a RAID can even tolerate a single drive failure. That's to say, if one drive fails, there's enough redundancy on the other drives to reconstruct the data onto a newly installed drive (which can even be plugged-in while the system is online).

Typically, a RAID will consist of a minimum of two hard drives using a special HDD controller or a software-controlling environment (Windows NT has the ability — in a most rudimentary way — to format "two drives as one.") As usual, storage space is limited only by your pocketbook or credit card limit.

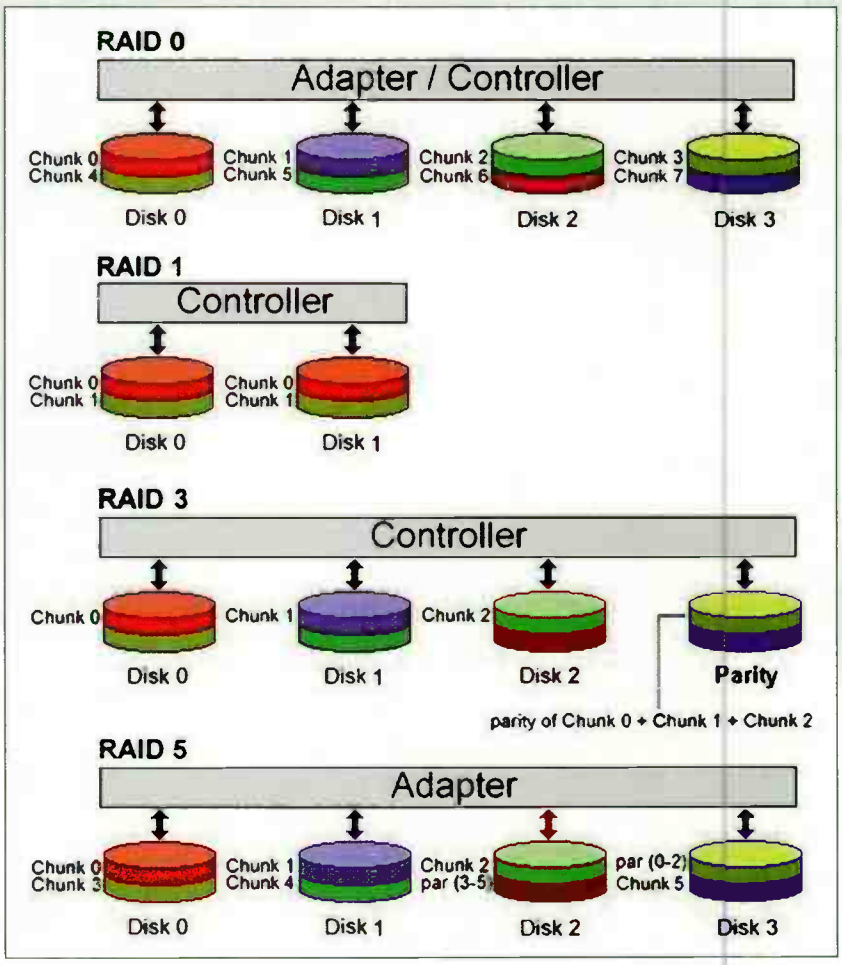
BY DAVID MILES HUBER

ON THE LEVEL

Currently, a RAID can be configured into seven "levels." There are several different ways of spreading data across multiple drives; each one depends on the parameters that must be optimized versus those that can be compromised. In short, the tortoise-and-hare issues are speed, reliability, capacity, and cost. Each level combines these four ingredients into a "data soup" that most suits your business taste. The most popular of the RAID levels are:

RAID 0: When using level 0, the data in a file is split into small "chunks," which can be evenly split across all the drives in an array (a process known as striping). This has the effect of speeding up the system's data throughput, as the various data chunks can be written to/or read from the separate disks at the same time. Since no redundant data is stored, the downfall would be that a failure in any disk would result in lost data. You should plan to use tape backups if you decide to use this level.

RAID 1: Level 1 protects the system against disk failure by storing the data onto two, separate hard disks or disk arrays (a process known as mirroring). In



the event that either disk fails, the remaining disk set can immediately be placed online, thereby preventing any downtime.

RAID 3: Level 3 distributes the data evenly across the disks in an array, while an additional disk is used to store parity information that can be used to restore lost data should any drive on the array fail. When a new drive is installed into the system, the data will be restored from the information on the parity drive, and you can get back to business.

RAID 5: This type offers the highest degree of protection against lost data, but it's usually an expensive option that's generally reserved for larger business institutions. In a nutshell, level 5 distributes both the data and the recovery (parity) information across all the disks in an array in real time. Since all the drives share the parity information, valuable data can still be recovered should "any" drive on the array fail. In addition to requiring expensive hardware controllers, level 5 is generally slower and often uses too much system-overhead to be considered a good choice for media production.

SCSI VS. IDE

Of course, the SCSI vs. IDE drive choice depends upon to whom you talk. SCSI drives have long been the preferred drive for high-end RAID systems where speed and optimal I/O performance is essential. However, it's also true that the newer generations of Ultra ATA IDE drives have narrowed the performance gap between the two types and, even more importantly, have brought RAID systems down to a fraction of their original price. Consequently, individuals or small companies (and not just large corporations) can afford arrayed drive systems.

One of the cool things about most RAID arrays is that they support "hot swapping." That's to say that a drive can be unplugged and a new drive put in its place at any time while the system is still running. Hot spares are also common, allowing a dead or faulty drive to be replaced while still online.

Although many RAID systems distribute file and parity data across the disks via a special hardware controller, some operating systems can support various levels of RAID in the software domain. For example, certain versions of Windows NT will support RAID level 0 in software, while others will support levels 0, 1, and 5. I could go on about the restrictions and advantages that certain operating systems offer, but it quickly gets complicated. If you feel you want to delve deeper into the subject, the Web is chock-full-o-useful tidbits. Before you decide to make changes to your system, however, I'd definitely recommend that you consult a pro. It would almost assuredly save you bucks, frustration, and potential downtime.

I don't expect many people to rush out and turn their systems into a huge data-storage tank. I don't intend to. For most of us, the newer generation of large-capacity drives, coupled with the many options for backing up data is enough to get the job done. On the other hand, with 24/96 kHz rates quickly becoming a reality (for better or worse) and surround already being a reality for many of us, maybe size does matter. For now, it simply doesn't hurt to know what your options for growth are. Just remember that it's a complicated subject. Carefully consider the cost, speed, and application. EQ

DRIVE! GEAR

STRIP-TEASE

The ChannelStrip from Metric Halo Labs allows Pro Tools users to mix the same way they would on a real console — by providing all of the essential channel processing functions you would expect to find in a top-of-the-line mixing console channel. ChannelStrip includes 6-band EQ with four fully parametric bands, plus high- and low-cut filters; ±24 dB input gain control; high-resolution metering for all processing sections; full-featured compressor with sidechain key input; expander/gate with sidechain key input; polarity reverse switch; direct manipulation of all parameter controls; and the ability to tweak multiple processing elements interactively and simultaneously. For more information, call Metric Halo Labs at 914-462-1230, fax them at 914-462-4865, or visit www.mhlabs.com. Circle EQ free lit. #120.

IMAGE IS EVERYTHING

The latest Microtech CD-R production system, ImageAutomator 150, features a new automated handler with a small footprint, up to 4x or 8x write drives, and up to a 400-disc capacity. ImageAutomator 150 handles premastering, verifying, and inline printing simultaneously. Up to four jobs can be run at the same time, without operator attendance. Options include ImageAligner, a combined hardware and software mechanism that aligns a printed image precisely on a silk-screened disc, and MYDisc software. For more information, call Microtech Systems at 800-223-3693, fax them at 650-596-1915, or visit www.microtech.com. Circle EQ free lit. #121.

STICK SHIFT

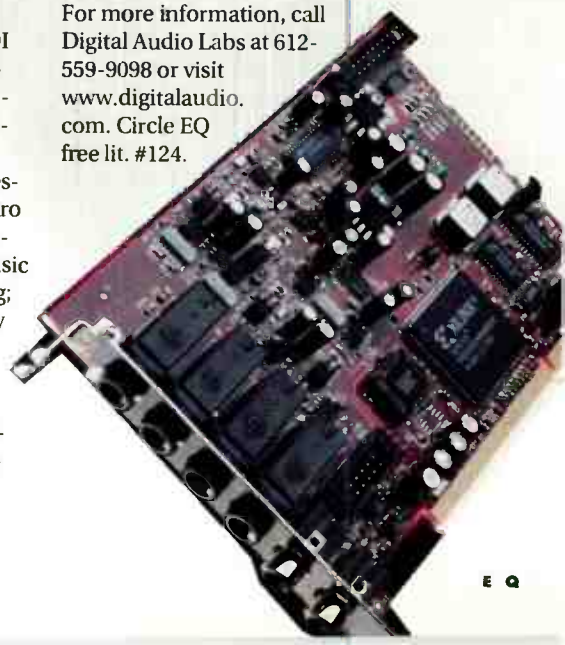
Kind of Loud Technologies will distribute Gallery's Axis joystick and software in North America. Axis is a professional joystick designed for smooth control of Kind of Loud's SmartPan Pro surround panning plug-in for the Pro Tools recording system. Available in both stealth black and sleek chrome, the Axis joystick is compatible with either ADB or USB Macintosh computers. Central to the joystick is the Axis software, which interfaces the joystick and other devices with SmartPan Pro. Other supported devices include Gallery's Mission Control panner, the JL Cooper MCS Panner, generic MIDI controllers, and Wacon pen tablets. The suggested retail prices are $795 for the black model and $895 for the chrome model. Axis software is also available through Kind of Loud for $299. For more information, call Kind of Loud at 831-466-3737, fax them at 831-466-3775, or visit www.kindofloud. com. Circle EQ free lit. #122.

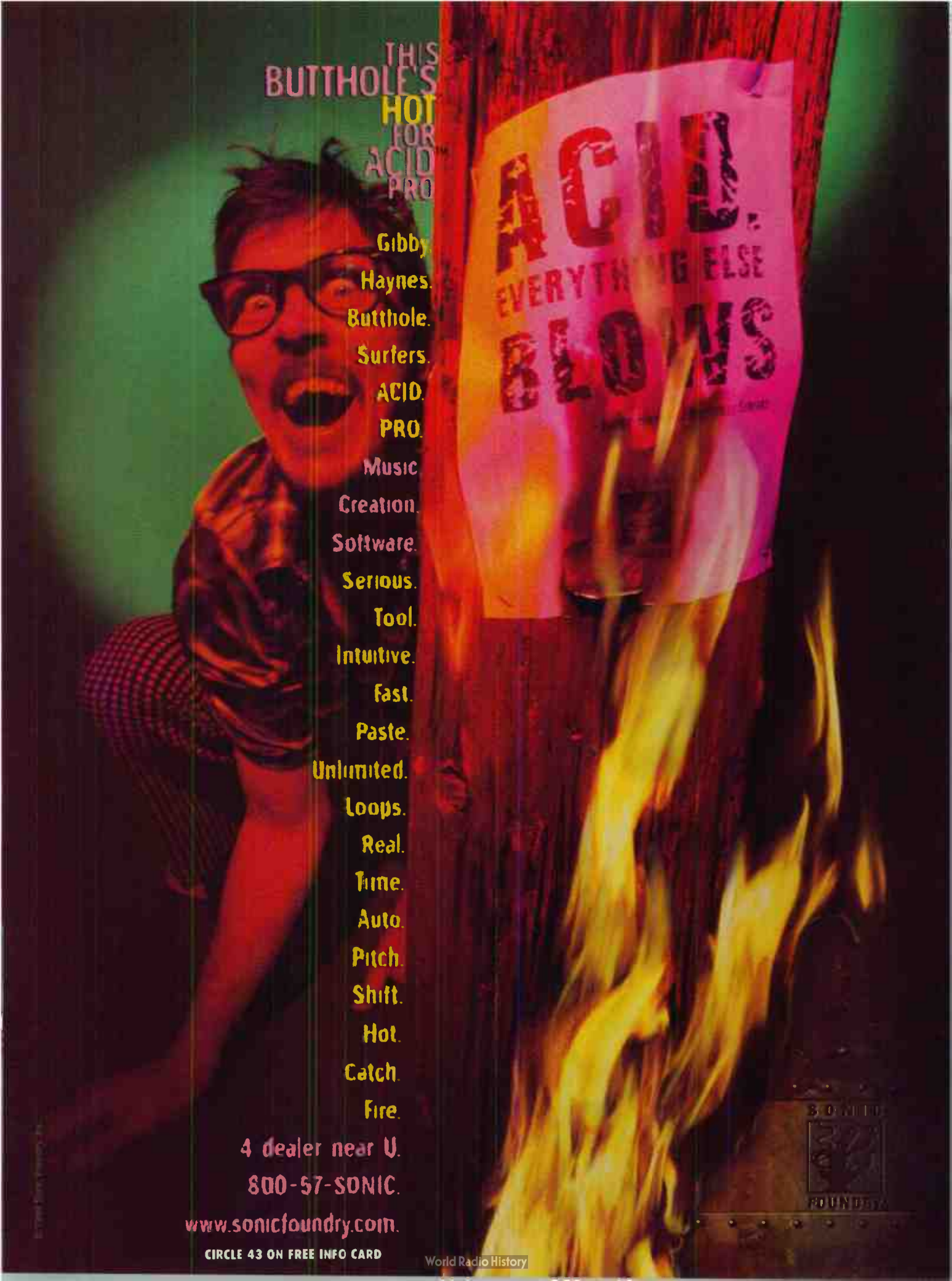
GO METRO

Cakewalk introduces Metro 5, a complete multitrack digital audio and MIDI studio for the Macintosh. With new effects automation, an enhanced user interface, and support for ASIO-compatible audio hardware, Metro 5 provides Mac users with an alternative for professional music and sound products. Metro 5 allows users to record any musical instrument or sound source; arrange music with flexible pattern-based sequencing; work with creative tools like NoteSpray and the Rhythm Explorer; mix in real time using automated audio effects; edit and print notation; and create soundtracks for film, video, and QuickTime movies. The manufacturer's suggested retail price is $249, and Metro users can upgrade to Metro 5 for about $59 (plus shipping and handling). For more information, call Cakewalk at 888-CAKEWALK or visit www.cakewalk.com. Circle EQ free lit. #123.

DIGITAL DELUXE

The Digital Audio Labs CardDeluxe is a PC-based interface that supports the latest 24-bit/96-kHz capable software, including Cool Edit Pro, SAW, Cakewalk Pro Audio, and more. The CardDeluxe is outfitted with both analog and digital stereo I/O connectors. For analog, 1/4-inch balanced connections are provided, and, on the digital end of the spectrum, a pair of S/PDIF I/O RCA connectors allow stereo digital transfers. Both the analog and digital I/O can be put to work simultaneously for 4-channel operation. For further channel expansion, Digital Audio Labs provides an inter-card sync connector, which can be utilized to slave multiple cards to a single sample clock. The suggested retail price is $595. For more information, call Digital Audio Labs at 612-559-9098 or visit www.digitalaudio.com. Circle EQ free lit. #124.





GUILLEMOT MAXI STUDIO ISIS SOUND CARD

TEST Drive

MANUFACTURER: Guillemot, 5505 St. Laurent Blvd., Suite 5000, Montreal, Quebec, QC H2T 1S6, Canada. Tel: 514-279-9960. Web: www.guillemot.com.

APPLICATION: Provides digital I/O, multichannel analog I/O, MIDI synthesis, and Sound Blaster-compatible functions for the Windows 95/98 platform.

SUMMARY: Despite a few rough spots, ISIS is an extremely cost-effective way to get into multiple aspects of hard-disk recording and sequencing.

HARDWARE REQUIREMENTS: 166 MHz Pentium, 32 MB RAM (basic I/O); 200 MHz Pentium with 64 MB RAM for Logic Pro Audio ISIS

STRENGTHS: Very useful bundled software; external breakout box for multiple I/O; S/PDIF I/O; good on-board MIDI synthesizer; expandable to 36 MB RAM for sound banks (doesn't use up host RAM); low latency with internal sounds; real, not just theoretical, plug and play.

WEAKNESSES: No ASIO or NT drivers yet; while definitely acceptable, analog I/O could be quieter; some quirks in initial software rev.

PRICE: $399 **EQ FREE LIT. #:** 125

ISIS (Interactive Sound Integration System) is a hardware/software sound card package for Windows 95/98 machines. The hardware consists of a PCI card with standard I/O (game port, line in, line out, mic in, and surround out), as well as a MIDI synth based around the French-made "Dream" synthesizer chip set. But what will pique the interest of musicians is an external, 1/2 rack-size breakout box with coax and optical S/PDIF I/O, eight 1/4-inch analog line ins, four 1/4-inch analog line outs (all analog I/O is unbalanced and uses 20-bit converters) and MIDI In/Out/Thru.

ISIS delivers a pro-quality sound card at a consumer card price

The card requires one slot; the breakout box connects via a multipin cable (included) to a second back-plate bracket. This doesn't take up a slot, but does need the back-panel space a card would normally use. (However, in some systems, you can remove the board from the backplate and install it in an unused parallel or serial port.)

The software bundle includes numerous Guillemot utilities, as accessed with the ISIS Manager program:

- Console 8/4 (fig. 1 shows breakout box ins, outs, and monitors, along with status and metering)
- Sound bank manager (compiles, opens, and auditions sound banks)
- Instrument manager (assigns samples to MIDI notes, shifts tuning, creates splits, etc.)
- Media station (WAV, MIDI, CD player with mixer)
- Gamer FX (for game freaks)

Other programs include Logic Audio Pro ISIS (based on Logic Audio Silver) with four inserts on each track, Cool Edit Pro SE (10 tracks maximum), ACID DJ, and various Cakewalk and Steinberg demos.

USING IT

Installation was truly plug-and-play. To use the system, first choose a configuration: either Multimedia (which turns ISIS into a Sound Blaster-type sound card) or Console 8/4 (which brings up the breakout box virtual console and is optimized for multitrack recording). Note that in console mode you lose the multimedia "gamer FX" DSP and surround functions.

I checked out S/PDIF I/O, the various analog ins/outs, and so on. It all worked after figuring out the various clock and assignment issues (sample rates are 32, 44.1, and 48 kHz; S/PDIF can slave or master). Note that the internal synth appears exclusively on outs 1 and 2.

Instrument Manager (fig. 2) is a comprehensive, if "texty," editor (e.g., there are no graphic envelope or keyboard tracking curves). Splits are easy to define and add; you can layer samples and do multiple splits, limited only by available memory. The basic idea is you turn samples into instruments, which you then assemble into Banks, then finally compile into something that can be downloaded into RAM. It's tedious, but doable.

BY CRAIG ANDERTON

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For the Masters, yesterday's best isn't good enough. Introducing genuine innovation in effects plug-ins with the Pro-FX bundle as well as the new TDM II & NPP II collections of Waves classics representing genuine value.

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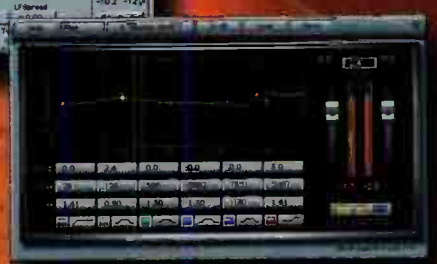

Renaissance EQ

ProFX is for TDM. Native versions in Q4 '99. Only TDM II contains PS22.

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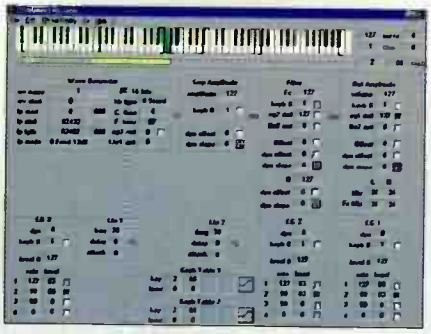

FIGURE 1 (left): The Media Station is in the foreground, with the Console 8/4 as the main window. ACID DJ is peeking out from the upper right corner.
FIGURE 2: The instrument manager window.

Cool Edit Pro, Logic ISIS, and ACID DJ all performed as expected, but also worked efficiently within the ISIS environment. For example, with my 450 MHz, Pentium II-class, 256 MB Q Systems computer, there was no trouble getting the maximum 16 audio tracks out of Logic, and still have plenty of room left over to run plug-ins and EQ.

I experienced some bugs in the initial release software. For example, booting the computer with a joystick plugged into the MIDI port gave a "card installed improperly" error, although all the audio programs still worked just fine (the fix is to disconnect the joystick while booting, then connect later). Another problem is that, although you can access the ISIS Manager from a Taskbar Tray icon, the menu extends only upward or to the left. Consequently, if you have your Taskbar located at the top or left side of the screen, you'll have to change it to the screen's bottom or right.

Other bugs included the 8/4 S/PDIF button sometimes showing up in the wrong place on the screen (a recognized bug fixed in the newest drivers). At first, I thought the console-input meters didn't work, but you need to arm a recording function in a program to wake them up. Without arming, though, the signals still came through; they just weren't registering on the meters. On balance, though, most quirks were more annoying than serious, and major crashes were relatively rare.

MIDI

Sounds are dealt with similarly to the Sound Blaster: you load sound banks into RAM, although SoundFonts are not supported. The card comes with 4 MB RAM onboard, but is expandable via a 32 MB SIMM to 36 MB total.

The 4-MB default GM bank, basically a Roland GM/GS set, is very good. It may not be up to some of the best 8-MB banks out there, but I found the sounds accurate and pleasing, as

continued on page 136

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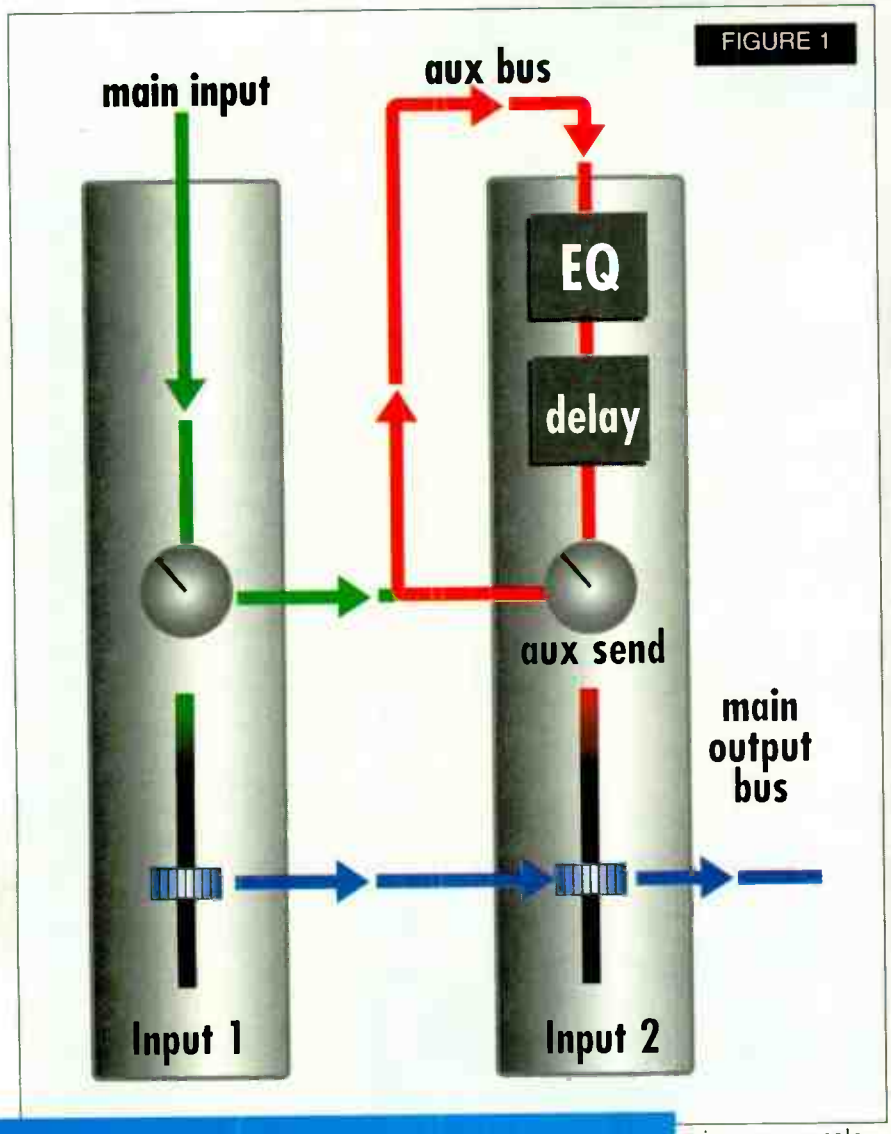
EQ Groove

EQ GROOVE



GOURMET GROOVE ECHO

BY CRAIG ANDERTON

Echo is an important complement to groove-oriented material, as it can add a rhythmic "push" when you need to send a loop or track into overdrive. Although most people just reach for a multieffects unit to do the job, there may be a better alternative sitting within your console.

First, though, some history. Before push-button signal processors, engineers "rolled their own" effects using whatever equipment was available. For example, patching a spare analog tape recorder into an aux bus could provide delay, because there was a delay between the time a signal was recorded at the record head and played back at the playback head. Lowering or raising the recorder's speed increased or decreased the echo time, respectively, with variable speed providing a limited degree of "fine tuning." For feedback, you'd return the aux bus output into an input channel strip, patch that to the same aux bus, then turn up the aux send to regulate the number of echoes.

Fast forward to today...and while it's cool to be able to patch in a multieffects and create echo at the push of a button, that's not the only option. In fact, for some applications, it may not even be the best option.

How to get the sound of the future through old-school techniques

With many of today's digital mixers providing per-channel delay functions, it might be time to revisit the idea of "constructing" your own effects. (You can also use this technique with analog mixers if you patch a delay line set for delayed sound only in the aux bus, as described later.) There are three main reasons:

- You can modify the delayed sound via EQ, create polyrhythmic echoes, mess with the panning, etc.
- If the mixer includes automation, you can automate the echo effects.
- For remix fans, you can "play" the echo parameters with long-throw faders. This alone can make "the gourmet echo unit" worth investigating.

THE CONNECTIONS

The connections (fig. 1) are essentially the same as for "old school" echo effects, and you still need to dedicate a separate channel strip to create the echo. In this example, the main signal feeds channel 1 and proceeds through an aux send control into an aux bus that's patched into input 2. The input 2 signal goes through EQ (optional,

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see later) and delay, then goes through its aux send into the same aux bus, setting up a feedback loop. (If you can't afford to give up an additional channel strip during mixdown, there's a workaround described later.) With an analog console, the delay line would patch between the aux bus output and channel 2 input.

With most digital mixers offering a blend of digital and analog I/O, there are various ways to route an aux bus output to a channel input. Route either a digital aux bus out to a digital input or an analog aux bus out to an analog input. To help keep feedback under control, initially set the aux bus sends for channels 1 and 2, as well as the aux bus master send, to unity gain.

Channel 2 has two crucial parameters other than delay time: aux send level and send pre-/postfader assignment. With prefader, the channel 2 aux send control determines the amount of feedback (note that exceeding unity gain can lead to runaway feedback), while the channel fader determines the echo level. Prefader works well for traditional echo effects, especially "set-and-forget" type applications where you don't work the faders.

With postfader selected, the aux send control sets a nominal feedback amount, but turning up the channel fader increases the feedback amount *and* echo level simultaneously. This is great for the remix and dub crowd, as you can create major echo mutations by playing the fader. And don't forget to play with the echo's panning.

OTHER MUTATIONS

Because the channel 2 equalizer is within the feedback path, any filtering is cumulative with each successive echo (assuming you've set the aux send to produce multiple echoes). For example, if you reduce the high-frequency response of the feedback path, each time the echo passes through the channel strip, it has somewhat fewer highs. This prevents older echoes from "stepping on" the original signal or newer echoes, and emulates the way tape echo works (which is important in this vintage-conscious era). A stage of high-shelf cut or low-pass EQ is ideal for this application.

However, reducing highs can sometimes lead to an overly "boomy" sound with lots of feedback. Therefore, you might want to introduce a low-shelf cut as well. These are just suggestions; the main point is that EQ can shape the sound in useful ways.

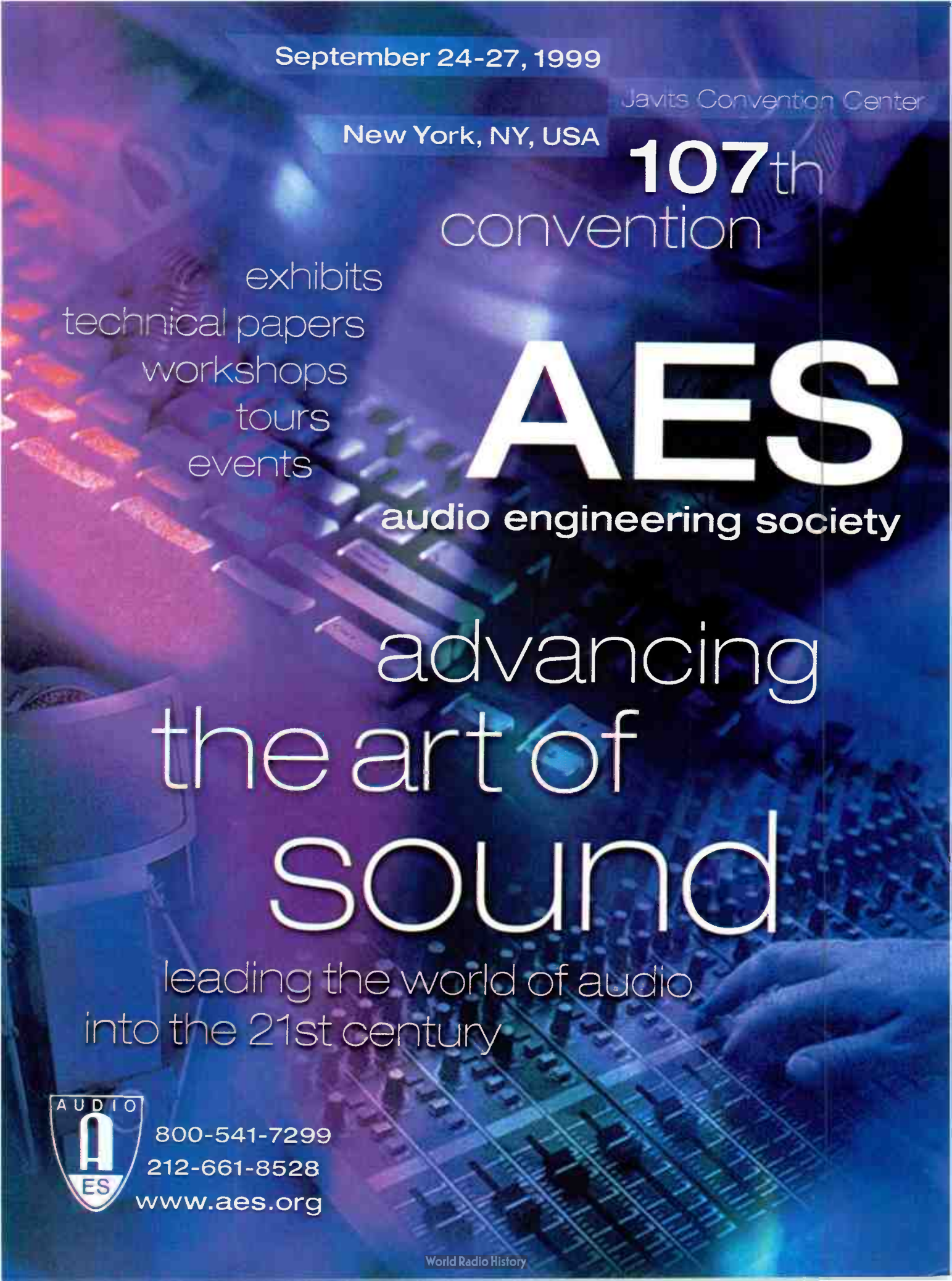
Now for something truly twisted: one mixer where I used this technique has a pink noise generator that can be assigned to the aux bus. Adding a bit of hiss in the background really nails the tape echo sound; as with tape echo, the noise becomes part of the feedback path, and changes character as it goes around the feedback path a few times.

BEYOND INDIVIDUAL ECHOES

For polyrhythmic echoes that ping-pong all over the stereo field, don't stop at one aux bus: assign the channel 1 signal to two, three, or more aux buses and bring those back into their own channel strips, which are also set up to create feedback. Furthermore, these strips can feed more than one aux bus, which creates "cross-echoes." In other words, suppose aux bus 1 feeds channel 2, which is assigned to aux bus 1 to create feedback, and aux bus 2 feeds channel 3, which is assigned to aux bus 2 to create feedback. Channel 2 could also send some signal to aux bus 2, and channel 3 could send some signal to aux bus 1, so echoes will circulate through both aux busses. The mind boggles...

If you have a limited number of aux busses and want to use them for something else (like reverb), no problem. In today's digital age, it's no big deal to bounce the composite sound over to an empty track, thus freeing up the aux busses for other, presumably less-interesting applications. Best of all, if you are into playing the faders, playing them during the bounce preserves your "performance."

So there it is — how to get the sound of the future through old-school techniques. Is the 20th century cool or what? (Well, there were two world wars, various natural disasters, the threat of nuclear annihilation, and the advent of the Spice Girls, but at least times have been good for the project studio.) Have fun with your new toy; you just might find that multiple echoes you can play in real time are the perfect touch to dress up your next mix or remix. EQ



GROOVE GEAR

MIX MASTER

Yamaha has introduced the RM1X MIDI Sequencer/Remixer for the DJ and dance production market. The unit's "Play FX" feature includes several real-time MIDI processes that can be applied dynamically to song data as it plays. These include: the harmonizing of parts with octave and unison parameters; clock-shifting to move instrument tracks forward or backward in time; note-duration quantizing; and the ability to modify note velocity. At the heart of the RM1X's power is Yamaha's AWM2 synthesis engine, providing 32 notes of polyphony divided among 16 timbral parts.

A sizable selection of 654 voices and 46 drum kits covers all major stylistic areas. Three on-board effects processors are also provided, allowing users to produce finished mixes using only the RM1X. The unit's sequencing section delivers a professional-level timing resolution of 480 ppq and a memory of 110,000 note events. The suggested price is $899. For more details, call Yamaha at 714-522-9011 or go to www.yamaha.com. Circle EQ free lit. #126.

TAKE IT TO THE HEAD

To solve DJs' two most common problems — poor headphone monitoring and the guesswork associated with mixing and cataloging tracks — RED Sound decided to roll two products into one. The MICRO BPM counter/headphone monitor amp is fed from the headphone output on any mixing desk. The unit delivers all the power and control needed for successful DJ monitoring, even in the loudest club environments. The BPM counter engine (taken from the company's VOYAGER 1 Beat Xtractor) constantly analyzes and calculates the tempo of the incoming audio signal. The BPM reading is clearly visible on the 4-digit LED, while a red beat indicator flashes on every downbeat to visually confirm the tempo of the track. The MICRO BPM comes complete with three mounting options offering permanent, semi-permanent, or free-standing placement for total integration into any DJ setup. For more details, call RED Sound's U.S. distributor Tracoman at 954-929-8999 or go to www.tracoman.com. Circle EQ free lit. #127.

REAL BOSS

The Boss DR-202 is not a glorified drum machine. This groove box is designed exclusively for groove/dance music production and features 10 style groups with multiple preset patterns in each, plus 100 user slots. The unit features 24-voice polyphony, 256 drum and bass sounds, 128 preset drum kits, 400 preset patterns, 64 user drum kits, and 100 user patterns. On-board sounds include classic TR-808, TR-909, and TB-303 sounds; vinyl sounds; Lo-Fi drum samples; various synth basses; sound effects; and more. Drum buttons are set up in an offset pattern similar to a piano keyboard, and they double as "keys" for inputting bass lines. Other features include bass boost or cut capabilities, RCA line outs, and a mini phone plug for headphones. For more information, call Boss at 323-685-5141, fax them at 323-722-9233, or visit www.rolandus.com. Circle EQ free lit. #128. EQ

RED VOYAGER 1 "BEAT XTRACTOR" V2

Many DJs, as well as recording engineers, would like to sync (and/or beatmatch) MIDI drum machines, sequencers, and syncable multieffects with audio tracks — and RED's Voyager 1 is billed as the missing link. It accepts two stereo, RCA phono audio inputs (each switchable between CD/line or phono levels; internal jumpers allow selecting two different line levels), displays their tempo in BPM on dual LEDs, and translates the selected channel's tempo into MIDI clocks. Large, illuminated buttons select whether the MIDI Out listens to input 1 or 2. Although all indicators use LEDs and can wash out under bright stage lights, that's not really an issue with the DJ market.

The missing link for sync

The latest software version upgrades the beat analysis engine to accommodate a wider range of tempos (60 to 230 BPM), and provide more reliable beat detection. However, with that comes extra responsibility, as Voyager 1 will look for the most consistent sound to which it can synchronize. This might be something like an offbeat hihat, but, fortunately, you can easily add or subtract a half-beat offset to bring Voyager 1 back into alignment (as well as "push" or "pull" the MIDI clock out in 1/12th of a beat increments).

Because it's impossible to generate song pointer when extracting from audio, Voyager 1 has two re-sync options: a Run/Pause button and a Beat 1 Reset button that returns a MIDI sequencer to its beginning. You have to release either button at precisely the same time to sync the MIDI to the audio, but hey — that's what DJs are all about, right?

Additional controls edit the overall tempo (useful if there's no audio present) or momentarily cut off the input if the music is about to get too arrhythmic, at which point the unit freewheels until the tempo settles down again. Another option is to "tap tempo" while the unit freewheels. Incidentally, the audio inputs are also paralleled with audio outs, so you can insert Voyager 1 between the sound sources and a mixer.

DOES IT REALLY WORK?

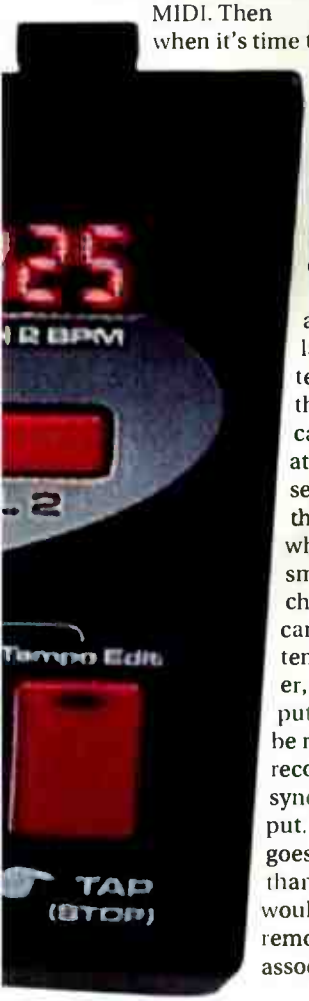
While a marked improvement over the original version, Voyager 1's accuracy still depends on a consistent, percussive rhythmic source. This can be a loop with a prominent kick and/or snare, a track with handclaps on the backbeat, and the like — a "four on the floor" house kick is the ideal source. Proper level is crucial, but it has to be adjusted at the source, and it's not always easy to find the initial optimum level. However, once you find the right level setting, you shouldn't need to change it too much with different material. The LEDs will alert you to an overload condition, but it would help if there were an LED meter showing optimum signal level.

With the right type of audio source and levels, Voyager 1 works like a champ. If it doesn't like the source material or levels, the tempo can waver (which you definitely don't want), or may freewheel at the wrong tempo. Beat detection seems particu-

BY CRAIG ANDERTON

larly problematic with parts that have heavy "swing," although version 2 can deal with a wider range of material than the original.

Running Voyager 1 parallel to existing audio is the toughest application, and one that is arguably of more interest to studio engineers than DJs. On the other hand, using Voyager 1 for beat-matching between vinyl and MIDI is a snap, because you can almost always grab enough tempo information from a piece of music and run Voyager 1 in the background without actually sending out MIDI. Then when it's time to switch from vinyl to MIDI, just hit the Run button, and Voyager 1 starts sending MIDI clocks to your MIDI device at the desired tempo.



The dual-input approach is particularly important for tempo changes. Although Voyager 1 can track minor variations, it takes a few seconds to register the new tempo, which demands smooth, gradual changes. It simply can't handle abrupt tempo shifts. However, with the two inputs, you'd typically be removing one record while Voyager 1 syncs to the other input. When the sound goes away for more than few seconds (as would happen when you remove the record), the associated input resets itself and waits for tempo information. As you cue up the next record, you let it play for a few seconds so Voyager 1 can grab the tempo information and retain it, even when you're "scrubbing" the needle to find the right section to cue up. When you're ready to switch back to the newly cued record, start the turntable, verify that Voyager 1 is sync'd, then hit the Run button to run MIDI and vinyl together.

If you're trying to track abrupt tempo changes using only one input, you're best off having Voyager 1 ignore the input for a few seconds (make the other channel active) to "reset" it, then re-sync to the new tempo.

STUDIO APPLICATIONS

Voyager 1 has studio applications as well as DJ'ing. (Finding suitable source material can also be easier, as you can isolate a specific track, like kick or a click track, to derive the tempo.) You can add sequenced parts over older, pre-MIDI tunes, sync to a pre-MIDI drum machine by using its output as a source, or generate a clock for driving a sequencer that changes multieffects parameters via MIDI.

So is it worth it? If you learn its quirks, choose the right audio material,

continued on page 137

IN THE Groove

MANUFACTURER: Distributed in the U.S. by Stanton Magnetics, 2821 Evans St., Hollywood, FL 33020. Tel: 954-929-8999. Web: www.stantonmagnetics.com.

APPLICATION: Translate rhythmic audio into MIDI clocks suitable for beat-matching MIDI devices to vinyl, or driving drum machines, sequencers, etc., "in parallel" with the audio.

SUMMARY: When it works, it's great — but try it with the material you plan to use *before* you go to the gig.

STRENGTHS: Dual-channel operation for switching between two different sets of audio material; very accurate with highly rhythmic source material; compact; tap tempo option.

WEAKNESSES: Response lag for tempo changes and lock to tempo; can't handle abrupt tempo shifts; no meter to indicate optimum incoming signal level.

PRICE: $399 ($499 for limited edition chrome model)

EQ FREE LIT. #: 129

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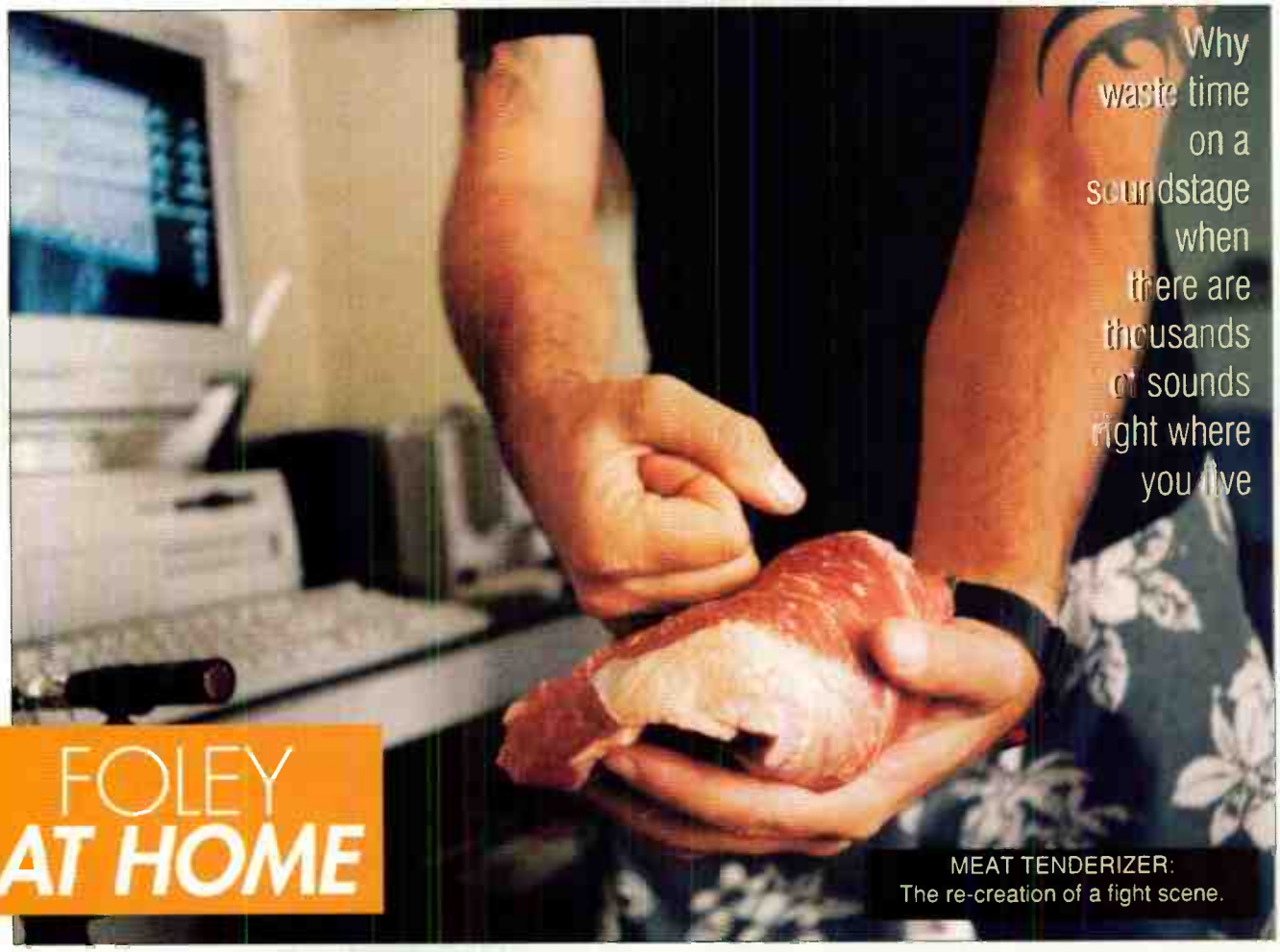

MEAT TENDERIZER: The re-creation of a fight scene.

FOLEY AT HOME

BY STEVE LA CERRA

When EQ *last dropped in on engineer Ronnie Kimball, he had just finished recording a live CD for Bad Religion — a band for which Ronnie has also been engineering live sound. Lately, Ronnie has been moving into a new realm, in particular with his audio postproduction work on CBS's* The Young And The Restless, *as well as an assortment of independently produced films. In his adventures of capturing sounds for picture, Ronnie has come up with some interesting Foley tips and techniques to share with the readers of* EQ.

"Any sound that you can find in your house can be applied to something. Whether it's realistic or not is irrelevant in Foley world." So begins engineer Ronnie Kimball when discussing his latest adventures in creating and recording sound for picture. "Something I'm just learning is that the sound you are actually using may have nothing to do with what you are seeing on-screen, and yet it may work perfectly with the visual. Basically, all you need is a DAT recorder and a microphone of your choice. I really encourage experimentation with different microphones for different sounds."

FOLEY IN THE BATHROOM

A recent project demonstrates Ronnie's use of various microphones. His task was to re-create the sound of a party going on next door, and then make it sound as if someone was walking over to the party house. Three mics were used to record the sound: an Electro-Voice RE20, an Audix SCX-1, and a PZM. Ronnie frequently uses the RE20 "on the floor for miking a room. For this scene, I placed the RE20 and a PZM on the floor of my bathroom and closed the door. Then I put on some rave music in the living room down the hall and cranked it up. Lo and behold, I got all the sub I wanted for 'the party next door.' Then we put an Audix SCX1 on a stand right inside the bathroom door, and when it was opened you could hear some of the top end of the music. The visual showed someone walking from their house, outside, then to the house next door. You

know — the type of thing where someone knocks on the door and says, 'What the heck is going on here?' When they opened the door, I needed to hear the music sound closer and it worked out really well." Because the bass content in the tiled bathroom was quite heavy, Ronnie placed foam underneath both the PZM and the RE20 to prevent the mics from rattling.

PILLOW FIGHTS?

For another recent project, Ronnie needed to create the sounds of a fight scene, including a bone-crunch sound. His solution was quite inventive: "We grabbed a few pillows and a giant steak (I don't remember what kind of steak it was). We put the piece of meat on a pillow — kind of like a tightly-stuffed couch pillow — made a fist, and hit this steak like we were hitting somebody. It had been in the 'fridge, so it was nice and firm. We once tried a frozen steak, but we ended up with sore knuckles! I don't know what the hell happened," Ronnie jokes.

"Another scene showed one guy kicking another in the leg and we needed the sound of a leg breaking. To create that

HEAVY METAL: Creating swordfight sounds.

sound, we took a head of cabbage and ripped it apart in front of the Audix SCX-1. When you put it to picture and you don't know what it is, it's amazing what you'll believe is that sound. If I see the cabbage breaking, I say, 'Oh that's cabbage breaking.' But if you don't know what the source is, anything is believable. Since most people have not heard bones breaking, it's not a common sound — so you can pull it off."

RONNIE PLAYS THE HORSES

In addition to recording shaking keys, his car stereo, the sound of a car seat being pulled forward, his son's squeaky swing set for a door hinge sound, bird sounds at a local pet shop, and his Harley Davidson roaring down the street, Kimball has also created the sound of swords clashing. "If you put swords together," he reveals, 'they don't sound like much of anything — just metal hitting metal. But you really want that sound you'd hear on a show like *Xena*. There was a strange fantasy scene where I used two pieces of metal banging together. One was a flat piece of metal which I'd hit it with this small, round metal rod, about 1/8-inch in diameter. It made this harmonic ring that sounds like what you'd expect swords to sound like. I'm not so concerned with what the sound is like when you only

listen to it. I'm thinking against the picture."

If you need the sound of a horse walking on sand, Ronnie has an idea for you. "I brought in real sand and put it in a box. The I took a plunger and a pushed it onto sand. For that we used the SCX-1, pointed it down toward the plunger a couple of inches away, and recorded a bunch of different takes." In this instance Kimball was recording to a Sony PCM-800 (DTRS) machine while watching the picture for timing.

BLOWIN' HOT AIR

One of the more difficult sounds to create or capture can be that of wind, and Ronnie explains that he's used "just about everything to get rid of wind noise: stockings, the zeppelin, and your basic pop filter that comes with the mic, as well as just putting the mic out and letting it go. A little table fan is great for creating wind, but you have to watch the angle between the mic and the fan. If you put the mic on-axis to the fan, you'll get a lot of 'boo, boo, boo, boo' at the frequency of the fan. You'll hear that popping at the frequency of the fan's rotation depending on how far away the mic is — proximity effect of the mic is going to play a big factor. Certain mics have no proximity effect, so you can get as close as you want (I have a bunch of different capsules for the Audix SCX-1, including cardioid, hypercardioid, and omni).

IT'S MY POTTY, er PARTY: A mic, a bathroom door, and a speaker help create a party atmosphere.

"Another way to capture wind is in the corner of a fence or wall area. Fences generally work better — they produce a little more low end for you to capture. You get this low-end pocket in the corner where air circulates and resonates as well as the fan."

POURING IT ON

Ronnie has learned that there's even an art to creating the sound of liquid pouring: "I'll pour from a glass or pitcher into a bowl with water, and hold the pitcher from a high distance because the liquid has to have some force behind it when it hits the destination. Otherwise it won't really sound like a pour. At a close distance it doesn't have the same force as from a couple of feet away. Also, it matters what liquid you use. I found that milk makes no sound. Water makes a great sound. Things with sugar in them like iced tea make good sounds. Coke pours well, especially into a glass with ice. Even the ice can make a difference in the sound. When you make your ice with cold water, it retains more air for some reason. If the Coke is not quite cold, the ice starts to crackle and break. But if you made those ice cubes from lukewarm water, that crackling would not necessarily happen. I happened on that one drunken night...while pouring drinks!"

ZIPPING UP

With all of these sounds floating around, archiving becomes a very important thing. Ronnie archives his sounds "either on my computer or to a Zip disc, and I keep them for future use. I usually dump the sound into my Power Mac G3 through the built-in audio input and save it as a Mac sound file at 16-bit, 44.1 kHz. I name everything that's logged into the computer, because, if you don't name it at the time you do it, you're in trouble. I log what the sound is, what I recorded it with, and a short description of what I can use it for. It's basic, but effective. The amount of sounds on tape builds up really fast because they're short samples and you can fit a lot of them on a single tape. It's kind of like being a kid with a can of spray paint!"

Ronnie Kimball may be reached at LonKimbar@aol.com

REEL SOLUTION TO A REEL PROBLEM

Don't throw out that analog tape machine! At least not yet — you never know when it might come in handy. Even in this age of the digital studio, there are still things that can't be done without analog tape. A recent audio-for-video project cemented this concept in my mind...

TASK AT HAND

Here's the scenario: a small-budget film comes into a studio for the addition of sound effects to some of the scenes. In particular, there's one scene that presents a very interesting challenge: it's an action scene where a guy driving an automobile is chasing a second person who is riding a horse(!). The mission? Make it sound real. The equipment available for recording the sound effects included a Yamaha 02R (version 2) with two TDIF I/O cards, TASCAM DA-88 and DA-38 recorders (the DA-88 had an SY-88 sync card installed for chase-lock capability), a rack of synths, an old Korg DSM-1 sampler (only 12-bit, but it sounds great), an Apple Power Mac 8600 with 160 MB RAM running Emagic Logic Audio Platinum, a Mitsubishi VCR, and a bunch of microphones for recording live sounds.

An unusual use for an analog reel-to-reel in the digital post production studio

Getting the whole system in sync required a bunch of connections, but nothing extraordinary. The Mitsubishi VCR was designated as the "master" since it had no way of chasing the other machines. SMPTE had already been printed on the mono audio track of the supplied VHS tape, so we ran audio output from the VCR to the timecode input of the SY-88 card. This allowed the TASCAM DA-88 to chase the VCR; the TASCAM DA-38 would chase the '88 via TASCAM's proprietary sync. Timecode output of the SY-88 was patched to the timecode input of an Opcode Studio 4 MIDI interface, enabling the computer (and thus Logic Audio Platinum) to lock to code as well.

Many of the sound effects required for this scene were "hits," easily dubbed in with some simple sampling. For example, there was one spot where the outside of the car door gets kicked. To get that sound, we miked up a broiling pan and kicked it while recording the sound into the DSM-1. We then set a MIDI track in Logic Audio Platinum to record MIDI data from a Roland A90-EX keyboard controller, which was also triggering the DSM-1. While watching the picture, we hit a key on the A90-EX in time with the kick. Logic Audio recorded this MIDI note; by playing back our one-note "sequence" we could judge whether or not the sound matched the picture. Many of the hits were recorded in this manner. In cases where our timing was a little "loose," we'd go into the list editor of Logic Audio Platinum and adjust the timing of the MIDI note until it matched the picture.

It certainly was interesting dubbing sounds in this manner, but the real sonic adventure came when the car keys hit the ignition. The key sound was easily produced with some house keys, but the engine sound took a bit more doing because we needed a continuous "bed" of engine noise under the entire scene. We had a garden-variety sound effect CD that had a car engine starting up and running for about 10 seconds, but the whole scene was several minutes long, so we ran out of engine audio way before the scene ended. A possibility was to sample a section of the engine sound and loop the sample so it'd keep repeating — but the sound effect CD was stereo and the DSM-1 is mono (we didn't want to give up stereo). Plus we didn't have enough memory in the sampler to capture a good chunk of the CD at a decent bandwidth. Houston, we have a problem.


PHOTO 1


PHOTO 2

BY STEVE LA CERRA

After a bit of head scratching, the solution came as I picked up the phone in the studio lounge and saw an old beast lurking in the corner. Its silver face gleamed like a shining pinball machine waiting to be awoken: a Scully 280. An analog beast that had been manufactured in 1966 and rebuilt by yours truly several years ago. It has a 1/2-inch transport, running at 15 and 7.5 ips, with a 4-track headstack in the block. Antiquated? Oh yeah (don't expect any subtle punch-ins), but, sonically, a nice analog machine. The idea? Make a tape loop of the engine sound and run the loop against the scene.

LOOP DE LOOP

First, we recorded a few seconds of the engine sound from the CD (we knew this piece would not be long enough). We threaded this tape through the tape path of the Scully 280, but went around the take-up reel directly to the supply reel. Using a white China marker, we marked the spot where the tape overlapped, and then spliced the head and tail together at this point. Now the tape ran through the path, and around the supply and take-up reels (see photo 1). By hitting play, the loop ran continuously, but the whole thing sounded kind of contrived. The problem was that the loop was too short — only about 2 seconds — and our ears could hear it repeating. We needed to make the loop longer, preferably as long as the original sound effect.

STAGE TWO

Doing a bit of math, we came up with the amount of tape required to record the entire engine sound from the CD: 10 seconds of tape at 15 ips equals 150 inches of tape. We measured a piece of 1/2-inch tape to this length, head- and tail-leadered it, and threaded it onto the Scully transport. Then we simply recorded the sound as you would make any recording on a reel-to-reel machine. The trick came in making a loop out of this piece of tape and getting it onto the machine. For starters we pulled the leader off from the head and tail of the tape and spliced the beginning of the recording to the end of the recording. Then we laid this loop on the floor to see how much space it would occupy, and began to think about how we could make a "tape path" to guide the loop for its journey. We decided to use two cymbal stands to support and guide the tape loop (see photo 2). The tape went through the supply swing arm, around a roller then to the headstack guides, over the heads, between the capstan and pinch roller, around the take-up swing arm, and then off to its journey around the cymbal stands. To keep the tape path level, we adjusted the height of the stands while moving them closer or farther from the machine to adjust the tape tension.

Luckily, the Scully 280 has a pretty strong capstan motor that handled the torque of pulling all that tape around. After recording this sound as a "bed" to the scene, we added a short sample of the engine in the DSM-1 modified by MIDI pitch bend to get the sound of the engine revving at key points in the scene. It was pretty odd to have a 33-year-old analog machine running in the company of all this digital stuff, but it certainly did the job. EQ

Fostex FD-8 Digital Multitracker

Fostex 8-track digital personal recorder will have you saying good riddance to analog cassette decks

BY EDDIE CILETTI

Last summer I reviewed the Fostex FD-4 Multitracker and was honestly surprised. Why? Because I found that a digital recorder is so superior to any analog, cassette-based, personal multitrack since it does more than just record. The quality is so good and there are so many features that you don't have to start over just in case the end result is "perfect except for the sound."

The FD-8 is more than four tracks better than the FD-4. With 8 real tracks and 16 virtual tracks, you can do multiple takes of vocals, guitars — whatever — create the ultimate "comp" and be able to audition a bounce before trashing the individual tracks. Worst case, all the tracks can be backed up to a number of optional formats.

EQ LAB REPORT

MANUFACTURER: Fostex Corporation of America, 15431 Blackburn Ave., Norwalk, CA 90650. Tel: 562-921-1112. Fax: 562-802-1964. Web: www.fostex.com.

APPLICATION: Personal digital multitrack recorder/mixer.

SUMMARY: Eight tracks, two sends, and two stereo returns. The DB-25 SCSI-II port supports various media. Review unit tested with 2.5-inch IDE drive.

STRENGTHS: MIDI I/O; 16 virtual tracks; digital patchbay; ADAT lightpipe and optical S/PDIF port; two XLR mic preamps.

WEAKNESSES: Scrub wheel still needs to be smoother; manual could be more concise.

PRICE: $899 without drive ($1199 with internal 2.5 IDE drive); Iomega Zip250 drive available for $199.95 through the dealer with one cartridge ($19.95).

EQ FREE LIT. #: 130

DRIVE ME HOME

Both the FD-4 and the FD-8 include a DB-25 connector for an external SCSI drive (your choice), as well as an internal IDE port for a 2.5-inch laptop-style drive. Last time, I used the SyQuest ezFlyer 230-MB removable drive. This time, SyQuest is no longer in biz, so Iomega Zip250 and Jaz drives are the most likely options. I requested (and installed) a 2 GB IDE internal drive. (There are 3.2 GB drives that are also available. IDE drives are now a "standard installed option.") The IDE drive does make a little noise and should be shock mounted to minimize transfer of noise into the case, something that wouldn't happen with an external drive.

Whether you are a weekend warrior or a full-time pro, you'll like the idea that the FD-8 can be self-contained. Hanging a drive off the back makes it a bit less portable (for tour bus or hotel use), but very useful for archiving and normal recording.

TIME TABLE

The FD-8 has three recording modes:

Normal (A.D.A.C.), Mastering (16-bit), and Backup. ("A.D.A.C." stands for Advanced Digital Acoustic Coding, a data compression algorithm developed by Matsushita and Fostex.) Normal mode is the most forgiving of removable drives, which are slower than "fixed" hard drives. Therefore, to avoid data compression, use a fixed drive — internal or external. (On the FD-4, I used the 32 kHz sample-rate with compression and survived!) Backup mode does not care how fast the drives are. While you may not be able to record without compression on slower drives, for storage purposes you can easily calculate recording time as 5 or 6 MB per mono track minute (respectively, based on 44.1 kHz or 48 kHz sample rates). Expect 17 to 20 track-minutes on an Iomega 100 MB Zip, 41 to 50 minutes on a 250 MB Zip, and 333 to 400 minutes on a 2 GB drive (IDE or SCSI) hard drive. You can even back up to a magneto-optical (MO) drive.

Track	Step 1	Step 2	Step 3	Step 4	Step 5	Step 6	Step 7	Step 8
1		Ld vox 1	Ld vox 1	Hi-bkg-1	Lo-bkg-1	Bass pno-1	Keys-1	**Mix-Left**
2		Ld vox 2	Ld vox 2	Hi-bkg-2	Lo-bkg-2	Bass pno-2	Keys-2	**Mix-Right**
3		Ld vox 3	**Ld vox Comp**	Ld vox	Ld vox	Ld vox	Ld vox	Ld vox
4		Ld vox 4	Ld vox 4	Hi-bkg-3	Lo-bkg-3	Bass pno-3	**Keys-cmp**	Keys-cmp
5		Ld vox 5	Ld vox 5	Hi-bkg-4	Lo-bkg-4	**Bass pno-cmp**	Bass cmp	Bass cmp
6			**BKG-L** *	**Insert hi-bkg**	bkg-cmp-L	bkg-cmp-L	bkg-cmp-L	bkg-cmp-L
7			**BKG-R** *	BKG-R	**Insert lo-bkg**	bkg-cmp-R	bkg-cmp-R	bkg-cmp-R
8	Pno/vox pilot track							

TABLE 1: The step-by-step for Mercedes Benz. Bold entries indicate the result of each "step." An asterisk (*) indicates pieces of Lead Vox that were bounced into, and became, background vocal parts.

MAN-NOT-SO-WELL

The manual is very thorough — in a way that borders on too much information — because it can occasionally be more confusing than helpful (especially if you've got that creative hat screwed on). For example, a step-by-step procedure is first provided as an overview, without detail. Then the detail is added, at which point I was already confused. In short, the FD-8 is easier to use than the manual suggests. (Perhaps the information could be more effectively delivered as linked hypertext — HTML — via CD-ROM.)

Once familiar with the FD-8, I would often press buttons before the machine was ready. To remedy this potential problem, I would like to suggest a built-in "speaker" so that the FD-8 could communicate with the user, beeping after completing a task, for example. (The speaker could also be used with the built-in metronome.)

MERCEDES BENZ: HOW I DROVE IT HOME

I'd been singing an *a cappella* version of Janis Joplin's "Mercedes Benz" for fun, and had even recorded a very rough version to use as an outgoing phone message. This time, I had plenty of tracks to make a fool of myself. Fortunately, the FD-8 has plenty of tools to help me choose the best performance. I started with a Wurlitzer electric piano and voice to get the feel and lay down a basic road map. (It was only after laying down a few tracks that I learned of the FD-8's built-in metronome.) I recorded five tracks of lead vocal using a Sennheiser MD-421 dynamic mic.

The most important button on the FD-8 is STORE because it captures a location point, which can then be entered into five possible locations: START, Punch-IN, Punch-OUT, END, Clipboard-IN, and Clipboard-OUT. There is even an adjustable pre-roll. I used the clipboard to copy the best vocal attempts and paste them into a composite on track 3. (See Table 1 for the step-by-step.) I chose this approach to keep the vocal in the digital domain. I did also bring all of the vocals up on faders to take advantage of (combining) an "accidental double." (You know, like the ones you could never do on purpose!)

I also used the clipboard, in stereo, to fly the backgrounds from the first chorus into the second chorus. Even during mixdown, the punch-in/punch-out feature was helpful as "automation," allowing the levels in each section to be optimized.

IN, OUT, AND EQ

The FD-8 has a generous supply of analog I/O ports plus one digital I/O (optical). Two mono effect sends and two stereo effect returns means somebody was thinking. Separate stereo (tracking and mix) and monitor (overdub) outputs plus a Monitor switch make it easy to hear what's going on. The switch allows you to listen to both outputs at the same time via the headphone jack or monitor out jacks. I was very surprised to see a pair of XLRs on the back panel for microphones (only on channels seven and eight). All eight inputs appear as 1/4-inch jacks on the front panel. Using them bypasses the balanced microphone circuit. The three-band equalizer has fixed bass and treble frequencies plus mid sweep.

INTER RE-ACTION

The display is where most of the "activity" occurs, including metering and — via the Setup button — access to parameter defaults. There are plenty of user tweaks including the choice of a timeline, represented by either bars or absolute time. Up to 99 songs can be named (if you write short songs or have a large-capacity drive).

Since the FD-4 and the FD-8 are identical save for the number of recordable channels, I did not attempt to repeat my previous tests, but chose instead to explore more of the feature-set. For example, last time I linked the FD-4 with my Soundscape workstation via MIDI and transferred the tracks through the optical digital interface. The latter doubles as both ADAT lightpipe (8-channel) and S/PDIF (stereo) I/O. This time I took more advantage of the FD-8's cut-and-paste features as well as using the internal IDE drive.

The FD-8 is incredibly powerful despite its somewhat cheesy feel, the analog mixer being especially *fromage*. Adding a grommet to the shaft of the "scrub" wheel made it smoother and faster — a crucially needed modification — because I relied so heavily on it to create precise cuts, pastes, and mutes (by erasing). The control surface, by contrast, feels more solid. It is compact yet functional.

Again, it was a real surprise to get such great sonic results. Even more than the FD-4, the FD-8 challenges the user's creativity. Unlike an analog cassette multitrack, there are no sonic excuses, no fear of bouncing "adjacent" tracks. Eight tracks on the best analog machine never gave you this kind of freedom to explore ideas — no matter how many tangents.

Both the Fostex FD-4 and the FD-8 show how much power technology can buy. Thinking of buying a used analog cassette multitrack? Think again! I repeat, "Good riddance to analog cassette decks!"

Wanna hear "Mercedes Benz"? It will be available for your listening pleasure at: www.tangible-technology.com.

Klein + Hummel O 198 Monitor

They may not be loud, but these monitors offer true sound reproduction

BY BOBBY OWSINSKI

Engineers who routinely spend thousands on a piece of outboard gear will spend hundreds or, even worse, nothing on speakers. Could it be that they're afraid of technology passing them by or, perhaps, investing in old technology? If someone spends a great amount of time working with his ears, any amount spent on speakers is a wise investment indeed. Which brings us to this month's audition, the German manufactured Klein + Hummel O 198.

The K+H O 198 is a 3-way powered monitor utilizing an 8.25-inch woofer coupled to a 100-watt amplifier, a 3-inch midrange fed by a 50-watt amp, and a 1-inch tweeter that's also married to a 50-watt amp. The drivers are crossed over by an 18-dB-per-octave active dividing network with turnover frequencies at 650 Hz and 3.3 kHz. A built-in overload protector alerts the mixer by flashing a red power-on front-panel LED (more on this later). The rear of each monitor contains an XLR input, an input level control, and a three-position equalization switch to adjust the low-frequency response for either free field, console meter bridge, or near-wall operation. K+H states that the frequency response measures 45 Hz to 20 kHz, ±2 dB and can provide up to 110 dB SPL at 1 meter short term and 105 long term (see below).

The cabinet, which is totally sealed and made of MDF, measures 8.25" by 13" by 10" and weighs in at 31 lbs. The manual is very complete, providing not only the basics one would expect, but also things like horizontal and vertical radiation plots, free field response, THD and group delay charts, and even circuit descriptions and PC layouts and a full parts list!

One of the best things about the K+H's was the road cases that they came in, which, I understand, may be a future option. Most aftermarket road cases give protection at the expense of a lot of unneeded additional bulk, but these cases were the perfect size — not only to carry and store, but also to easily fit in the trunk of any car.

When I first put the K+H's up, I must admit that I wasn't too impressed listening back to some mixes I'd just previously finished. The reason, I came to find out, is that the mixes weren't as good as they could have been and the K+H's were merely telling me the truth. This, of course, brings up a maxim that bears repeating once again: You can't accurately judge a set of speakers until you really use them! Had I judged the K+H's on projects I had previously finished, I would have been sorely mistaken, since the mixes I achieved using these units were some of the best I'd ever done. The K+H's allowed me to hear through the mix and give it that elusive third dimension of depth, as well as total frequency response accuracy.

There is a point about the K+H's, however, that warrants being brought up. Although I never experienced this myself (since I don't mix loud), I did hear from another mixer that he loved the speakers except for the fact that they didn't play loud enough for his tastes. After checking with Brad Lunde at Transamerica Audio Group (the U.S. distributor), I was told that this is due to the limiting circuit that K+H employs for driver protection. While this may prevent the monitors from playing as loud as other comparable loudspeakers, it turns out that it's also part of the secret of their sound.

Unlike other powered monitors, K+H uses a hard limiter for driver protection. The philosophy behind this is that a soft limiter (as used in most powered monitors) changes the tonal balance according to the playback level as it gradually engages on transients, but a hard limiter is out of the circuit completely until it engages. This means that it becomes very obvious to the engineer when the limiter has kicked in and he can adjust accordingly.

Another factor is that the monitor uses a sealed woofer cabinet, not a ported one like most of its competitors. While the sealed woofer offers greater detail and damping, and therefore more accurate bass, it doesn't have the maximum output capability of ported or bass reflex design of equal size. Once again, there was plenty of level for me with the K+H's, but if you need to fill a large control room with the wind of a thousand decibels, then they might not work for you.

For someone who makes their living with their ears, these speakers should be considered a long-term investment. Highly recommended!

MANUFACTURER: Klein + Hummel, Germany; distributed by Transamerica Audio Group, Inc., Tel. 805-241-4443. E-mail: sales@transaudiogroup.com.
PRICE: $4000 per pair.
EQ FREE LIT. #: 131

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113	Minnetonka Software	76	612-449-0187
138	Music Yellow Pages	XX	516-489-6514
79	Musician's Friend	37	800-776-5173
155	Neumann	32	860-434-5220
7	NHT Pro/Vergence Technology	38	707-751-0270
49	Panasonic Broadcast	XX	714-373-7277
23	Peavey	39	601-483-5365
41	Rockford/Hafler	40	800-366-1619
16-17	Roland	41	323-890-3700
127	Royer	61	818-760-8472
83	SAE Institute of Technology	72	212-944-9121
114	Samson	7	516-364-2244
28-29	Seasound	42	415-331-4978
57	Shure Brothers	28	800-25-SHURE
24-25	Sonic Foundry	43, 44	800-577-6642
138	Soundmirror	45	617-522-1412
15, 150-151	Sweetwater Sound	46, 64	219-432-8176
32	Tannoy North America	48	519-745-1158
55	TASCAM/TEAC America, Inc.	35	213-726-0303
43	TC Electronic	49	805-373-1828
139	Terrasonde	57	303-545-5848
107	Terratec Pro Audio	59	800-960-9440
91	The John Hardy Company	51	847-864-8060
133	The Recording Workshop	52	614-663-2544
111	Waves, Inc.	71	423-689-5395
45	Yamaha Corporation of America	53	714-522-9011

Power (Distribution) to the People

Isolated ground versus balanced power (...or both?)

BY EDDIE CILETTI

A recent e-mail posed the question, "Isolated ground or balanced power or both? Are they worth the investment?" The answer is very much a matter of time and money. Chasing hums and buzzes is a waste of resources. If you have the cash, choosing to have an isolated ground as well as balanced power distribution will improve your audio system's chances of being noise-free over the long term. That said, neither solution fixes what might be wrong with the equipment...

If all gear were balanced — in and out — and properly designed from the ground up, we would not be meeting like this. Digital audio confounds the issue because we can no longer hear the hums and buzzes, only their side effects. It is not always the design of the gear that is flawed; sometimes it's the careless compromises made in order to streamline the production process. Making equipment cheaper to assemble without regard to noise immunity results in gear that is inherently vulnerable to any power- or transmission-induced (RF/TV) interference.

Compromised digital gear will "see" power spikes and line noise. The "translation" can cause the momentary unlocking and noise-modulation of clocks (and data busses) generating random sonic glitches and the fuzz we normally associate with clock jitter. I do believe that newer gear is less susceptible, but no studio is comprised solely of late-model toys.

This article is as much for anyone thinking about building or rewiring a studio as it is for those studio owners, technicians, electricians, and designers who have experienced problems after attempting to do the right thing. Keep in mind that nearly all *new* installations are sonically free from hum and buzz (assuming they started out quiet). Don't kid yourself into thinking you got what you paid for, or got away with not paying for "that special treatment." Time has a way of teaching all of us a lesson. So, if you have more time than money, read on...

I am obligated to say the following. Saving money is not about compromising "legal" or safety issues. There are rules and regulations regarding the "implementation of electrical power," a.k.a., the electrical "code," which may vary from region to region. Sometimes it may seem that meeting "code" and achieving a quiet system are disparate goals. For example, the outlet-ground connection is simply to protect humans from the risk of shock. All of our mentors have warned against the dreaded "ground loop," which occurs when a device is "grounded" in more than one place. Using a ground-adapter as a ground "lifter" is, by code, illegal. Remember that whether the gear or the wiring is at fault, every temporary fix will eventually become an intermittent noise problem.

Finding a knowledgeable electrician, sensitive to the needs of audio dweebs like us, may require the expertise of a studio designer. Don't be afraid to spend a little money with an "experienced traveler" who has navigated the many waters to sonic peace. A studio designer will be familiar with acoustics, contractors, power and audio wiring, interior design, and air conditioning requirements. While their design charges may not be in your ballpark, many will offer consultation services for an hourly rate. When consulting multiple "experts," don't randomly combine ideas. Doing so may compromise system safety or performance. Still looking for the guru of ground? Consider contacting the chief engineer of a local TV or radio station. People who understand transmitters understand how to minimize interference.

Now I will show you where the demons are hiding, as well as the result of some experiments.

DEMON-1: GROUND CONTAMINATION

Power is delivered to the outlet from the breaker box as three wires — hot, neutral, and ground — all traveling through a common jacket — plastic, flexible metal, and/or conduit (pipe) — some number of feet. With plastic cable, the ground connection is made at each end with a dedicated wire attached to a conveniently located screw. Metal-jacketed BX cable has a somewhat less-substantial noninsulated ground wire. The ground connection is made via clamps, to the jacket, at the points of exit and entry. When metal conduit is used, the conduit itself makes the ground connection; although, from an audio per-

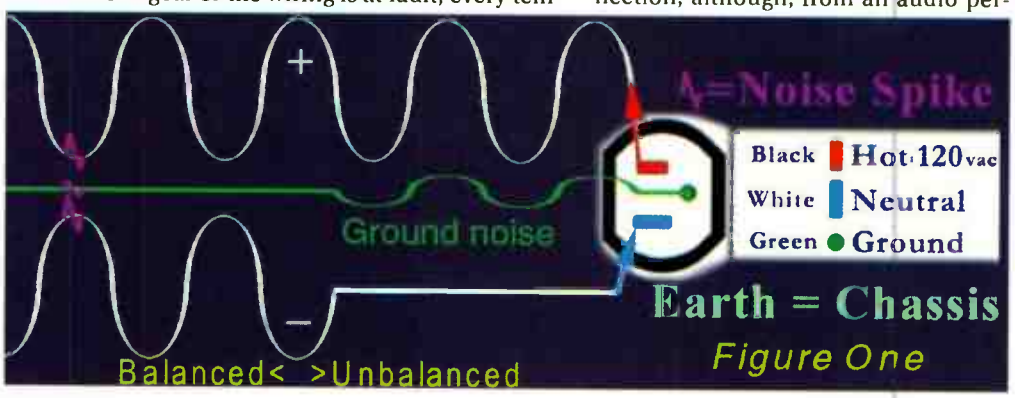

Figure One

spective, a dedicated ground wire provides piece of mind.

Following only the ground all the way through to a *common* household outlet, the connection to the "third-pin" is mechanically made when the outlet is screwed into the metal mounting box. (Plastic outlet boxes require the ground wire to go all the way to the outlet. Metal wall studs can also contribute to ground contamination.) Note that this method of ground distribution is satisfactory for safety purposes, the system meets code unless the ground is completely lost.

However, there are number of places where the ground might get interrupted enough so to become corrupted from an audio perspective. A loose clamp leaves room for oxidation to build up, increasing the resistance and, therefore, increasing the susceptibility that intermittent noises will be injected into the system as appliances are turned on and off and as power demand changes over time. (What happens to the other wires will be discussed at Demon-3. The short-term fix — with the power off — is to check *all* connections and tighten when loose.)

DEMON-2: "STAR" WORSHIP

It is generally thought that "star" grounding is the ultimate scheme for a quiet system. Running a separate ground wire from each piece of gear to a copper spike penetrating deep into damp earth is the goal. Rack-mounting a metal chassis with metal screws into metal rack rail will tie all of the cases together. You'd have to create a wooden rack rail, leave space between each chassis, and avoid multiple-outlet strips, all of which are simply impractical.

There is almost no way to achieve a perfect "star" ground. Some run a ground wire from each rack rail to a common ground point, but what happens to all of the power cable ground pins? Power plug castration is *not* legal! Part of the solution is for all gear to have balanced inputs and outputs. This means getting "adapters" to make unbalanced gear conform.

SHAREWARE-1: PROTECTING THE ISOLATED GROUND

Many people choose to plant a spike and distribute an isolated ground wire to each "hospital-grade" outlet. (The third-pin ground connection, in an orange, hospital-grade outlet, is isolated from the mounting hardware.) The problem with distributing an isolated ground is that there are so many other places where your best efforts will be thwarted.

If you have gone to the trouble and expense of planting a spike, running the cable and installing the orange outlets *only* to be rewarded with an imperfectly noisy system, here's the first clue. That ground wire is in the same "jacket" as the power cables. The more current is flowing in the power cables, the more hum and noise will be induced into what is no longer a clean ground. Guess what? All of your hard work and money went down the tubes. The following options might ease the pain.

One solution is to find a legal way of delivering ground to the outlet with a separate wire — not in the jacket and preferably not parallel with any power lines. Because code varies from state to state, discuss the matter with a licensed electrician. I am not an expert on "code," but this is not a theory. In an installation where a video monitor showed "hum bars" when connected to a ground-isolated outlet, simply re-routing the ground wire away from any "power radiation" eliminated the hum.

IN-FLIGHT SIDEBAR: BALANCED POWER

The other fix is balanced power. Normal outlets are wired as shown in fig. 1, where white is neutral, black is hot, and green (or the metal-to-metal connection) is ground. With respect to ground, only the black wire is "hot," the white wire provides a "return" to what is at near-ground potential, or neutral. Again, the ground wire provides safety by making the case or "chassis" the same potential as the earth, good old terra firma.

Balanced power distribution starts by installing a power transformer with a pair of 60-volt secondary windings. The junction where the two windings are joined is called the "center-tap," the mid-point or "reference" that is tied to ground. Now, the hot and formerly neutral connections are each 60 volts (60-0-60) with respect to ground. With balanced power, the closer the match between the two secondary windings, the better the noise cancellation.

On the audio side, a mic preamp has a differential input designed specifically to amplify the balanced signal from the microphone while rejecting common-mode noise (as might be induced by running the cable next to a wall-wart transformer). The ability to discriminate between signal and noise is

continued on page 136

The Backspace *Project*

PHOTO BY STEVE JENNINGS

How a London company is creating equity over access

BY JON LUINI AND ALLEN WHITMAN

"If you don't like the news, go out and make some of your own." —Scoop Nisk

In our travels, we've seen a lot of Web-based organizations gorged on hype. There's money to be made, and most everybody in the Internet business community will do anything to make it. But what about the community of artists? What about the actual creative connections the Internet can provide? The old story of the lowly musician, writing alone at home...how can she use the Internet to raise herself up to some kind of visibility or, at least, meet like-minded players?

We've all seen the big portals, search engines, and retail sites. We've seen guys in sport jackets with no musical experience posture themselves as online messiahs to the largely ignored independent musician. We've seen hacks-who-would-be-CEOs with fire in their eyes hot on the spoor of an IPO. But who is actually creating [an] emotionally and artistically equitable community (besides your faithful FezGuys)?

Meet James Stevens, proprietor of the Backspace Project in London, an online access and arts community that exists for no other reason than to be a community. A combination of the physical and virtual worlds where artists of all shades can meet, commune, observe, collaborate, and express themselves. Created out of shoestring and salvage, the Backspace Project has become a vital, viable community that isn't interested in finding The Gimmick that will provide a quick cashout in hypermarketed Internetland.

We catch up to Stevens between interviews with the BBC and various European periodicals. He describes the Backspace Project and his hopes for success based solely on its own resources. Peppering his conversation with ear-catching words and phrases like "slacktivist" and "non-hierarchical representation of the collective state of mind," he expresses visionary dreams about a self-perpetuating community based on the fertile ground of imagination. A place where creative work can "sidestep commercial and cultural co-option, and the interruption of corporate concerns."

All this might sound like a tryptifunk groove looped out of rehashed political samples from the Marxist utopia at the beginning of the 20th century, but Stevens displays a remarkable ability to cull raw materials, power, connectivity, hardware, and even rent out of thin air, intelligently delegating responsibility for aggregating these resources. He then stands back to let the thing succeed or fail on its own terms.

Online and off, Stevens describes the three-year-old Backspace Project as an "under the counterculture" experimentation with media, tech, and "the bits in-between." For £25 (about $37 U.S.) per month, as the introductory blurb states, a Backspace subscriber gets: "activities and interests of subscribers and friends promoting access, learning, discussion, and development. Subscribers gain a full Internet access dial-up account, quality access to the Web, an e-mail account, and an open invitation to drop in any time for as long as you like, come down to events, meet up with friends, or just chill out by the riverside." Stevens points out that, while there are many "virtual community" projects

OFFICE SPACE FOR BACKSPACE: The small office belies the activities for the multi-national (including spin-off sites) company.

available, there are few that operate in a physical space as well. He observes: "A physical space means that computers, audio and video libraries, editing equipment, video projectors, etc...can be brought together to use."

Assembling the nuts and bolts to roll a going concern requires a lot of legwork. Backspace is funded entirely by the subscriptions of its users, with a little help from some synchronous relationships. Situated in a 200-year-old wharf building on the Thames just west of London bridge, Backspace thrives in the fragile web of these interdependent relationships. Web serving is hosted free by nacamar.net.uk (an originally grassroots, now big-time, ISP in Germany that still appreciates the little guy). Its 512 kbps connectivity (about a third of a T1 or quadruple ISDN) is handled by others in the building who pay for and control the datastreams in exchange for free access to Backspace resources. Stevens explains that he has bartered and negotiated the costs of running Backspace down to only the rent. "We haven't paid any power costs for over a year because of confusion about the ownership of the building. The current owners are highly incompetent, which is good, and we're currently looking at a way to buy the building. It would be the most way-cool thing we could pull off in this century."

Stevens handles maintenance and acquisition of hardware, as well. Having supplied high-tech gear to many of the design and interactive businesses in London, he knows a lot of people. A sort of telecommunications hardware ragman who could be said to wander the districts in a cart overflowing with boxes, wires, and advice, he barters and trades his way through town, generating interest in Backspace by meeting people and discussing broader interests then merely the Internet. Physical community is the "keystone activity" of his realm, and "gritty day-to-day life" comes first. This existence usually works well for the 37-year-old father of two. Though Backspace has "died two or three times," Stevens's proprietorship emerges more "sinuous and flexible after each death."

The online presence of the Backspace Project resembles an intriguingly designed, colorful, and spacious virtual manufacturing facility, converted to digitally airy and well-lit lofts where behind each icon is a different and colorful gateway to another world. Within the site can be found many pages featuring

BACKSPACE BUILDER: James Stevens. proprietor of the Backspace Project.

audio and video real-time jamming, resources for creation and manufacture of pirate radio, and a panoply of art, ideas, diaries, games, lists, suggestions, and rants that remind one of the familiar and welcome feeling of a large family. It is community in the truest sense of the word; those at the center sourcing organization and logistics, users and contributors dropping in and out, and camp followers and hangers-on, eager to get a word in edgewise in a vital and energetic civilizational soup of the most fascinating kind. Stevens comments: "It's a place where anybody can become involved and interfere with the system."

What about the audio side of this Rube Goldberg contraption? Though coming out of a rock 'n' roll background (he started a booking agency called BRAG and also handled lighting design, roadie duties, driving, haggling with club owners, dealing with business, etc.) Stevens makes clear that, at Backspace, audio is "as important as anything else." Ninja Tune (a loose amalgamation of DJs, samplers, producers, and remixers) is in the building, DJ Coldcut (one of the more visible contributors of the so-called "flip-flop" scene) does some semi-regular performances, and there are numerous ongoing Internet radio programs. Though only 5 percent of Backspace content is audio, that figure takes up fully half of the disk space.

"The whole point is that people can come and use as much of the resources as possible," offers Stevens. "There are few rules. The main rule is: no one should have to wait more than half an hour to get on a machine. There are physical resources on location you can come and use anytime, whether you come everyday or just visit London occasionally. A person can join from anywhere in the world, whether they need RealAudio servers, decent disk space, or just somewhere to put their ideas into reality. Backspace is a physical space first, but open to anyone in the world. The online world lacks true community without a physical space," comments Stevens. "Backspace is a place to come down and actually interact." It's a good idea. Imagine a network of physical spaces working together in which members can share online resources as they travel around the world.

What does the future hold for the Backspace Project? "There are half a dozen spin-off sites in Europe," reports Stevens, "and we are encouraging more. New subdomains, new URLs, more content...we're pushing the space into perpetual transmission of audio and video streams. It's a pilot space, a collaborative effort, and hopefully the project will give confidence to others. As we take on more and more subdomains, I expect to lose control, which is good!"

It's refreshing to remember that hordes of venture capital locusts don't necessarily make the difference in the perception of success or failure. Content still takes precedence over form.

BACKSPACE URLS:
bak.spc.org/vacuum
bak.spc.org/iod/destructo/
bak.spc.org/everything
bak.spc.org/downlode
bak.spc.org/j18 bak.spc.org/vt
Subdomains include:
dms.spc.org http://rad.spc.org
Spin-off sites:
www.lowtech.org
www.test.org.uk/
www.idea.org.uk/splitshift/
www.ohos.org.uk
bak.spc.org/gallery37
www.irational.org/cube
www.okchicken.com/

The FezGuys welcome your comments. Offer them by visiting www.fezguys.com.

MAINTENANCE

continued from page 133

called the "common mode rejection ratio," or CMRR. With balanced power, the ground wire — isolated or not — will have less noise.

Ever notice that when flying at 30,000 feet you can have no sensation of motion? To a certain extent, this analogy can be applied to ground noise. So long as every device is "flying" on the same ground (no matter how noisy), no device "sees" the noise. You might say this is "The Ground Plane." (You are allowed to laugh while reading this column.)

Here's one more analogy: With one foot on the floor and one foot in an elevator, it is obvious that the elevator floor is not as steady as the structural floor. This correlates to what happens when a direct box inadvertently links the system ground (via the recording console) with a bass amp plugged into a "utility" outlet that is not connected to the officially sanctioned ground. The "point" here is to translate what we already know — the sound of hum and buzz — with what is actually happening. The two "grounds" are not at the same potential relative to the earth or to each other.

SHAREWARE-2: THE POWER TREE RECIPE

One way to troubleshoot a small audio system for noises is to distribute power from one outlet. Assuming that you know this one outlet to be the only outlet connected to a 20 AMP breaker, you can safely connect a "system" whose power requirements should max out at 15 AMPS or 75 percent of the breaker's rating.

Connect two power strips to one outlet, and to those power strips connect the rest in the most symmetrical way possible. (I have also been known to use a two-to-six outlet adapter that screws into a standard wall outlet. All strips then plug into that.) Remove all "ground lift adapters." Power up slowly, using a current probe at the breaker box to monitor demand. Your system should be quiet, and, if not, find all the wall-warts and make sure none of the audio cables go near them. This is only a test. If "the tree" solves the noise problem, have your facility's outlets rewired in a similarly symmetrical fashion.

DEMON-3: THE OTHER WIRES

When electricians normally install home and commercial wiring, there are two options when attaching wires to an outlet. Either the wire is placed under a screw and tightened (preferred) or the wire is inserted into a pressure-fit hole (provided to reduce installation time). Either way is considered "to code," but it can create problems down the road...

In addition, it is also quite common to daisy-chain multiple outlets, looping the power through two or more outlet boxes. The combination of poor physical connections and high current generates heat. Expansion and contraction over time increases resistance at every junction — making each hot, neutral and ground connection a potential noisemaker. Power wiring in such a state will become especially vulnerable to devices such as air conditioners and elevators with momentary high-current demands. As each device snaps "online," the weak link will get hot and become noisy. In addition, the power supplies in some gear will reflect noise back into the power lines.

Again, this practice is considered "to code." The short-term solution — with an electrician's supervision — is to turn off the power at the breaker box and inspect all connections from the breaker box to the outlets. Tighten screws and rewire any outlets using the pressure-fit connections. Wires should be under screws.

Running a dedicated ground wire from a central point to each outlet may not eliminate system noise, but, at the very least, it provides a bit more confidence in the integrity of the ground. It is also preferred that the power be distributed in a similarly dedicated fashion. Of course, as your system grows, so will its problems. Start with a good foundation.

HERE'S YOUR "TO-DO" LIST

1. Have an electrician tighten all of the connections from the breaker box to the outlet and routinely check them approximately every year, or whenever you make changes or additions.

2. Know your power distribution. How many outlets are on the same breaker? How are they linked?

3. If you have installed an isolated ground, keep all cables — from the outlets to the gear — as short as possible. Belden Wire and Cable [765-983-5200; www.belden.com] makes a short IEC power cable.

4. It is better to "fan out" power strips than to daisy-chain them.

Drop by for a virtual visit at www.tangible-technology.com or e-mail edaudio@tangible-technology.com. E.C. thanks Dave Huber, Hector La Torre, Craig Anderton, Joe Hannigan, Dan Kennedy, Dave Hill, and the folks at Ampria Research and Design for their input.

GUILLEMOT REVIEW

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long as you played in the intended instrument range. Even some of the sounds that don't come off too well with other sets — accordion, voices, choirs, distorted guitar — were great. Given that this is a sound card and not a keyboard synthesizer, I'd have to say that the instrument chip set is a real overachiever. What's more, latency when playing the synth is negligible; and, using the ISIS DirectSound drivers, the MIDI response with Seer's SurReal software synth was equally snappy. If you've ever experienced the "latency molasses" that comes from running software synths with standard Windows drivers, you'll really appreciate what DirectSound can do.

THE BOTTOM LINE

There's a definite learning curve because of all the bundled software; you have to constantly switch among the various pieces of documentation (only the ISIS User Guide and Logic Audio Pro ISIS have paper manuals — the rest are PDF files). Still, spending a little time with the system is really all you need to get up and running.

ISIS is clearly a package for those who want to move considerably past consumer sound cards without spending a bunch of money. The hardware audio performance is not exactly Pro Tools, but is reasonably quiet and far better than what you might expect for the price. The bundled software is excellent; Logic Audio is a great program anyway, but when you throw in Cool Edit Pro SE and ACID DJ, you have almost all bases covered. S/PDIF I/O is a nice touch, as is the ability to record on eight separate ins, with full-duplex operation (ASIO drivers are slated for inclusion in the next software rev, which will help minimize latency when monitoring post-card while playing).

The more I used ISIS, the more I appreciated the synergy of the various elements, until it got to the point where I genuinely enjoyed working with the system. For the bucks, ISIS is an outstanding deal, and with the upcoming release of new drivers, presumably some of the rough edges will be smoothed. Although there's a lot of competition in the sound-card field, ISIS manages to straddle the world of consumer electronics and the project studio, delivering a serious recording/synthesis system but with a consumer price tag. EQ

ROOM WITH A VU

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The atmosphere of a big studio can feel sterile, but since Gold Key is in the first floor of a Colonial house, it has a very comfortable feel. We don't have the standard glass window looking into the live room. Instead, we use video cameras and monitors to allow everyone to see each other and maintain contact. It allows for great isolation at the same time.

EQUIPMENT NOTES: Bob is an electrical engineer, so he maintains all of his own gear, and all those years of school pay off every time the studio is used. If a band brings in gear that isn't working properly or is causing problems, the problem can be fixed on-the-spot. This keeps the outboard gear used in the mix from being a Band-Aid™ for a mistake made earlier in the production.

PRODUCTION NOTES: Grosso continues: The feel for every song dictates what it needs. To record everything "perfectly" doesn't make a great song. The best thing to do is listen to the song for the feel and try to record that feel — not just the sound of the instruments. EQ

BONZAI BEAT

continued from page 54

vocals — on helium!

Who do you think you were in past lives?

This nutty guy who didn't believe in reincarnation.

What animal do you identify with?

Eric Burdon

What makes a great producer?

A great producer is one who is himself musically aware, totally in touch with the artist's vision, and, at the same time, able to tactfully take control when necessary. He or she must also be a motivated businessperson who has the personality and confidence to reassure the people who are putting up the big bucks for the record. A great producer is someone who can serve the artist and the record company without compromising either. Is that really possible?

Have you ever witnessed a miracle?

Once, in New Mexico, I saw a reverse alien abduction: the alien was abducted by some military guys. Does that count? Oh — and the other day I bought some new software and it actually worked without me having to update everything else on my computer.

What is the biggest mistake of your life?

Investing heavily in that slide-rule company. It wasn't a mistake — it was a "learning experience."

Who is the Great Impostor?

I could tell you, but she'd have to kill me.

What would you like Santa to bring you this year?

A month's worth of rampant inspiration.

Any advice for getting a good start in the music business?

It's gotta be fun and something you love doing, and if you can make some money — great. EQ

RED VOYAGER

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and remember to disable the input when the tempo gets too confusing, Voyager 1 delivers on its promise. In the right context, it can be downright magical. It's not flawless by any means, and you do have to pay attention to it (remember to disable the input when the tempo goes away). Nevertheless, under the right conditions, it can be a useful addition to an engineer's or DJ's bag of tricks. EQ

ACROSS THE BOARD

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big trouble if the tracks were not locked together?

THE 96 KHZ ALLIGATOR

OK, so you now are riding on the bleeding edge of technology. You are even transferring your cassettes to 96 kHz/24-bit. Life is good...almost. With an upper frequency response limit of 40 kHz you can make excellent dog-whistle recordings. I'm just finishing up my first DVD, entitled *Studies in Parallel Harmonies Above 20 kHz*. You could play it to entertain your pets without disturbing any humans. Another project idea is to record everything an octave below where I really want it, then pitch shift every instrument up an octave so I will have plenty of ultrasonic overtones.

Even if you recorded everything at 48 kHz on your ADATs, the act of mixing, EQ'ing, and adding reverb and effects will generate program content above 20 kHz. That is the good news. The bad news is that you can't check to see what is up there because you can't hear it. You can look at it with a spectrum analyzer, or zoom in and look at the waveform, but you can't really tell by looking what effect it will have on your hearing.

Audio equipment is designed for a fairly flat response from 20 Hz to 20 kHz. We have known for a long time that there are problems with recording information below the 20 Hz limit. DC components must be filtered out, capacitor noise and power supply ripple must be eliminated, and does anybody remember "turntable rumble"?

With 96 kHz recordings, I have run into some ultrasonic problems that you should watch out for. Remember I said that ultrasonic material is generated during the mixing process. Sometimes these can be pretty healthy transients generated by the music. Other times they can be harmonic impulses that are caused by the console EQ. In the 20 Hz to 20 kHz world there is no problem, because if something causes a click, you usually hear it and fix it. Ultrasonic transients go undetected. I have some mixes that contain some of these ultrasonic transients. When I play back the mix in the studio, or on my studio quality gear at home, everything is fine. When I play it back through a consumer power amp and speakers, there is a giant click. When I listen on headphones powered by a very expensive amp, everything is fine, but when I plug the headphones into a $500 receiver, the click makes my nose bleed.

Low-price amplifiers contain circuits that cannot change the voltage fast enough for the high frequencies coming from the 96 kHz material. It is kind of like the click you hear when you have the bass turned up too loud and your speakers hit the stops. (My daughter, Ashlee, actually likes the extra click added to the kick drum attack. Oh well, she's out of my will.)

THE ANSWER

I talked to one mastering engineer about the problem and he said he just rolls off the stuff above about 22 kHz so he won't have that problem, and as long as the end product says 96 kHz, who will know? Wait a minute. Doesn't this negate the need for 96 kHz? Have I wasted all of my money again?

I guess a good comparison would be owning a Dodge Viper. You can't find many places to drive 180 MPH, but it sure impresses people who see it in your driveway. I guess the Apogees and Genex and Mytek and Alesis and TC 96k stuff looks good in my rack, so maybe everything will turn out OK. I'll have to think about this and get back to you. EQ

DIGITAL MIXERS

MACKIE

New

Digital 8 Bus Mixing Console

Everything you've been waiting for and more!!! The new digital 8 bus from Mackie features great sound quality, full recording and mixdown capabilities, motorized faders and an array of digital features geared to take you flying into the next century. See for yourself what the entire industry is raving about.

FEATURES-

- 48 channels of automated compression, gating, EQ and delay
- Built in 3-way meter display keeps you on top of your mix.
- Built-in meter bridge.
- Ultramix II automation for complete control, hook up an S-VGA monitor and you'll feel like you spent a lot more money.
- All functions can be automated, not just levels and mutes. Store EQ, reverb, compression, gating and even Aux send information.
- Fast SCENE automation allows you to change parameter snapshots on every beat.
- Reads Standard MIDI tempo maps, displaying clock info on the built-in position counter.
- Truly the cutting edge of mixing technology.

* monitor, mouse & keyboard sold separately

Panasonic

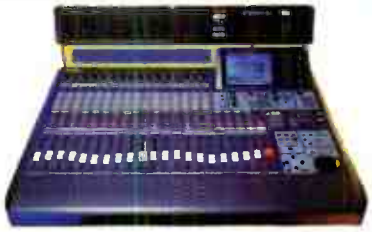
New

WR-DA7 Digital Mixing Console

Stop dreaming about your digital future, it's here! The Panasonic WR-DA7 digital mixer features 32-bit internal processing combined with 24-bit A/D and D/A converters as well as moving faders, instant recall, surround sound capabilities, and much more. Best of all, it's from Panasonic.

FEATURES-

- 32 Inputs/6 AUX send/returns
- 24-bit converters
- Large backlit LCD screen displays EQ, bus and aux assignments, and dynamic/delay settings.
- 4-band parametric EQ
- Choice of Gate/Compressor/Limiter or Expander on each channel
- 5.1 channel surround sound in three modes on the bus outputs
- Output MMC
- Optional MIDI joystick



* meter bridge optional

TASCAM

TMD1000 Digital Mixing Console

You want to see what all the digital mixing buzz is about? The NEW TMD100 from Tascam will have you smilin' & automatin' in no time. It features fully automated EQ, levels, muting, panning and more in an attractive digital board with an analog 'feel'. Your digital future never looked, or sounded, so clear.

FEATURES-

- 4 XLR mic inputs, 8 1/4" balanced TRS inputs.
- 20-bit A/D D/A conversion, 64x oversampling on input, 128x on output.
- Store all settings fully MIDI compatible.
- ***Optional IF-TD1000*** adds another 8 channels of TDIF and a 2-channel sample rate converter.
- ***Optional FX-1000*** Fx board adds another 4 dynamic processors and another pair of stereo effects.

New

DIGITAL RECORDING

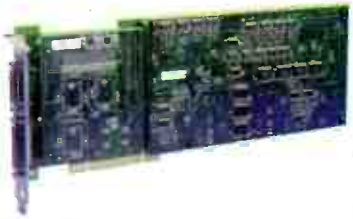
Lexicon

Lexicon Studio Recording System

The Lexicon Studio System interfaces with your favorite digital audio software for a complete hard disk recording package. Supporting both PC and Mac, Lexicon Studio can be expanded up to 32 voices from a variety of I/O options. For recording, editing, mixing and DSP, Lexicon Studio is here.

FEATURES-

- The Core-32 System PCI-Card is capable of supporting 32 audio streams simultaneously. It can also be used as a time code or clock master or slave.
- The PC-90 Digital Reverb daughterboard attaches to the Core-32 providing 2 discrete stereo reverbs.
- The LDI-12T delivers up to 12 channels of simultaneous I/O supporting analog (+4 XLR and -10 RCA), spdif, and ADAT
- Direct support of Steinberg Cubase VST and many other software programs.



OPTIONS-

- The LDI-16S provides 8 channels of +4 XLR balanced analog I/O, and 8 channels digitally through TDIF. 3 option packages are:
  AES-8, 8 channels of AES/EBU digital I/O
  ADT-8, 8 channels of ADAT digital I/O and sync
  STC-1, Post option including read and write of LTC, read and generate of VITC Window-burn, House Sync and a General Purpose Interface for triggering external devices from the system).

EFFECTS PROCESSING

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ULTIMATE SOUND MACHINES

New

Finalizer Express

The Finalizer Express is a fast and efficient way to turn your mix into a Professional Master! Based upon TC's Multi-Award winning Finalizer Mastering Technology, it delivers the finishing touches of clarity, warmth and punch to your mixes, putting the world of professional mastering within your reach.

FEATURES-

- 24-bit resolution A/D & D/A converters
- 16 & 20 bit dithering
- TC's unique Multiband Comp & Limiter Algorithms
- Boost and cut over three bands with the Spectral Balance Controls
- Soft Clipping and Look Ahead Delay.
- Finalize Matrix for 25 variations in style and rate
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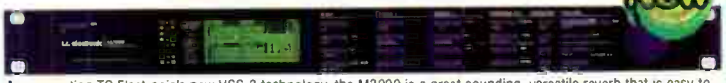
- Extra compression in each band using Emphasis keys
- Record fades from the built-in Digital Fader or the optional TC Master Fader via MIDI.
- Connections include AES/EBU, S/PDIF, Optical Toslink & MIDI I/O's.
- High Res LED Metering of I/O & multi-band gain reduction

M3000 Professional Reverb

New



Incorporating TC Electronic's new VSS-3 technology, the M3000 is a great sounding, versatile reverb that is easy to use. Combining ultimate control of early reflections with a transparent reverb tail, the art of reverberation is brought to a new level. Whether it's a phone booth, cave or concert hall, the M3000 delivers high-quality ambience

FEATURES-

- VSS-3, VSS-3 Gate, C.O.R.E. & REV-3 reverbs as well as Delay, Pitch, EQ, Chorus, Flanger, Tremolo, Phaser, Expander/Gate, Compressor and De-Esser
- 300 high-grade factory presets including Halls, Rooms, Plates, Ambience, Gated Reverbs, and more
- Up to 300 user presets in internal RAM and 300 more using an optional PCMCIA card.
- Dual engine configuration featuring 24-bit A/D/D/As.
- Connections include AES/EBU, Coaxial S/PDIF, Optical Tos-Link/ADAT & analog XLR I/Os, MIDI IN/OUT/THRU, Clock Sync and External Control.

Lexicon MPX1 Multi-Effects Processor

The MPX-1 is truly an outstanding multi-effects device. Using Lexicon's Lexchip, it offers outstanding reverb or ambience as well as a separate processor for effects for awesome power in the studio or on the road.

FEATURES-

- Intuitive user interface for easy editing, built-in help.
- Balanced Analog I/O (1/4" & XLR)
- 56 effect algorithms
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COMPRESSORS

PreSonus

ACP88 8 Channel Compressor

Stemming from their popular ACP8, the ACP88 comprises eight channels of compression, limiting and noise gating for a variety of studio applications. It features individual side chain for each channel and it's attractive blue anodized finished lets you show your true sonic colors.

New

FEATURES-

- 8 separate compressors/gates with individual controls.
- Servo balanced or unbalanced inputs & floating balanced or unbalanced outputs.
- Individual side chain jacks for spectral compression and a separate sidechain jack for gate processing.
- Each channel boasts full gain reduction metering, compression threshold indication & gate open/close.
- Front panel buttons include hard/soft knee compression, peak/auto compression, bypass, gate range and link.
- Link feature uses a unique summing bus for multiple combinations of master/slave link setups.

dbx Model 566 "Silver Series" Compressor

The new Silver Series introduces a 2 vacuum tube circuit design making the 566 no ordinary compressor. Loaded with features including custom designed analog VU meters that monitor tube level, gain reduction, or output levels. Full sidechain functionality, including sidechain monitor, Contour function allows low frequency material to pass through the threshold without triggering un-musical compression effects.

FEATURES-

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- Drive control for a wide variety of great tube effects
- PeakPlus limiter on each channel
- Complete sidechain, OverEasy, and Auto function
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MOTU 2408/1224 hard disk recording

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EQ

PROJECT STUDIO OF THE YEAR AWARD

Nominations

Finally, there is award recognition for some of the most creative and productive facilities in business today — project studios. This is your chance to choose who will win these new, but already important, awards. The following nominations have been compiled from *EQ*'s editors and a large gathering of industry professionals, but, as always, *EQ* is as much your book as it is ours, and so we've left an "Other" option in each category so you can bring our attention to something or someone we might have missed. So please take the time to fill out this form and fax it back to us at 516-767-1745. You can also fill it out online by visiting www.eqmag.com, or snail mail it to us at: EQ Magazine, Project Studio of the Year, 6 Manhasset Ave., Port Washington, NY 11050. The lucky winners will be announced in our AES issue.

.....

(Please choose only one from each category, ***or*** *fill in your own choice in the "Other" section.)*

Producer/Engineer-Owned Studios

- ☐ Dust Brothers, PCP Labs, L.A., CA
- ☐ Steve Albini, Chicago, IL
- ☐ Daniel Lanois, Kingsway, New Orleans, LA
- ☐ Mick Guzauski, Barking Doctor, Mount Kisco, NY

Other: _____

Musician-Owned Studio

- ☐ Brian Wilson, Chicago, IL
- ☐ TLC, Atlanta, GA
- ☐ The Artist, Paisley Park, Minneapolis, MN
- ☐ Butch Vig, Smart Studios, Madison, WI

Other: _____

Indie Record Label Studio

- ☐ Nothing Records, Nothing Studios, New Orleans, LA
- ☐ dmp Records, Tom Jung, Stamford, CT
- ☐ Sumthing Distribution, Nile Rodgers, NY, NY
- ☐ Bad Boy Records, Daddy's House, Puff Daddy, NY, NY

Other: _____

Film/TV Scoring and Soundtracks

- ☐ The Attic, Michael Whalen, Boston, MA
- ☐ Media Ventures, Hans Zimmer, L.A., CA
- ☐ Mutato Muzika, Mark Mothersbaugh, L.A., CA
- ☐ Asymmetrical Sound, David Lynch, L.A., CA

Other: _____

Video Game/Multimedia Studio

- ☐ Tommy Tallarico Studios, L.A., CA
- ☐ Electronic Arts, Redwood City, CA
- ☐ Darryl Duncan, GameBeat, Lake Mary, FL
- ☐ Mark Waldrep, AIX Entertainment, West Hollywood, CA

Other: _____

Project Record of the Year

- ☐ Brian Wilson, Imagination
- ☐ Cher, Believe
- ☐ Beastie Boys, Hello Nasty
- ☐ The Fixx, One Thing Leads to Another [remix]

Other: _____

Project Studio on Stage

- ☐ Erasure
- ☐ Fatboy Slim
- ☐ Air Liquide
- ☐ Chemical Brothers

Other: _____

Project Studio Legacy Award

- ☐ Peter Gabriel, Real World, England
- ☐ Frank Zappa, Barking Pumpkin, L.A., CA
- ☐ Jimi Hendrix, Electric Lady, NY, NY
- ☐ Roger Waters, The Billiard Room, England

Other: _____

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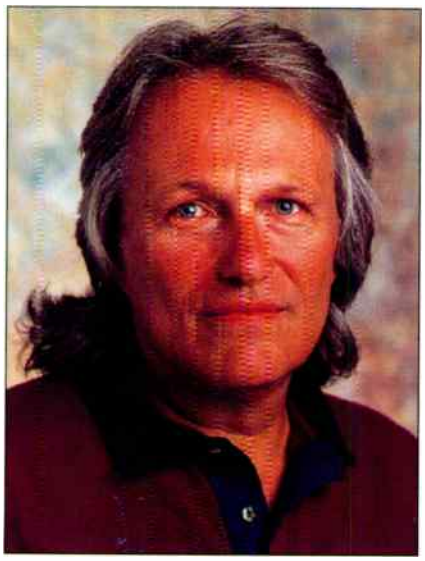
So You Wanna *Do 96k*



The "hole" story of recording in 96 kHz/24-bit

BY ROGER NICHOLS

So, 44.1 kHz is for wimps, you say? Real men mix to 96 kHz/24-bits, or 192 kHz/24-bit, or maybe even DSD (Direct Stream Digital). More and more 96 kHz equipment is becoming available, from converters to recorders. But figuring out what works with what makes the phrase "Would you like fries with that?" seem like a less distant alternative.

TWO-HOLE, OR NOT TWO-HOLE

When 96k first reared its ugly head, there was no standard for getting the digital audio stream from one piece of equipment to another. The use of an XLR-terminated AES cable to carry the double-speed bit stream was going to be pushing the limits. The smart guys wanted to use coax cable designed for the new frequency requirements. Coax cable would easily carry 192 kHz/24-bit signals as well. Somewhere along the line the idea was dropped because coax doesn't look like it is carrying audio, but a nice XLR connector is familiar to audio facilities. This is the same mistake that was made the first time around.

So now the XLR digital cable is the data carrier. Each piece of digital equipment has an interface chip that turns the digital audio bit stream into a signal that is designed to withstand the hostile environment of a length of wire. Similar to a normal RS-422 serial connection, the driver and receiver chips are robust enough to work over varying cable lengths. The current implementation, however, would not handle the double-speed data.

Some manufacturers decided to split up the signal and send it out a pair of AES connections. This is called the "Double Wide" format. Normally, one AES cable carries the left and right channel of a 44.1 kHz or 48 kHz signal in one bit stream. (There is already room for 24 bits in AES data.) The receiving equipment splits the data back up into the left and right components and sends it on its way. Two AES cables carry twice as much data; just what is needed for a 96 kHz signal. Each audio channel of the 96 kHz signal is split in half. The odd-numbered samples are sent on the AES port's left bit stream, and even-numbered samples are sent on the right bit stream. Each AES cable is thus carrying all of the data necessary for one audio channel at 96 kHz/24 bits. Apogee uses the two-hole interface for their PSX-100, 96 kHz/24-bit converters. Genex uses the two-hole interface for the GX-8000 MO recorder. The Genex GX-8500 supports both one-hole and two-hole formats (as well as four-hole for 192 kHz/24-bit).

Some manufacturers decided that one hole was enough. In the beginning, it took expensive handpicked driver chips to spit out the double-speed data stream. Now chip manufacturers are making faster drivers that are designed to work with the high-speed data, so extra cost in manufacturing is no longer a factor. This format is known as the "Double Fast" format. One reason for deciding on the one-hole approach was that interconnection would be the same as for existing AES hookups. One cable is all you need to get from device "A" to device "B." The drawbacks are that you need to record on a 96 kHz/one-hole recorder, and you must be very careful with cable selection and length. The new Alesis MasterLink ML-9600 hard disk/ CD-R recorder is a one-hole machine. TC Electronic's new Finalizer-96k is also a one-hole box.

The kicker is that very few devices give you a choice between one-hole and two-hole interconnects. This means that I cannot use my Apogee PSX-100 converters to feed my TC Finalizer-96k. I could use the 96 kHz/24-bit converters built-in to the Finalizer-96k (which are great, by the way), but then I can't get into the Genex GX-8000, or into my Pro Tools. What's a guy to do? Nobody makes a one-hole-to-two-hole converter box.

LET'S SPLIT!

This two-wire method works very well when trying to record 96 kHz data on existing 48 kHz/24-bit recorders. You just use four tracks to record on and everything is perfect. I record 96 kHz information into four channels of my Pro Tools system. It works great. The Apogee PSX-100 has additional bit-splitting capabilities for multitrack recorders that can only handle 16-bit data. There is an Alesis ADAT I/O and a TASCAM TDIF I/O located on the back panel. For 96 kHz/24-bit recordings, all eight tracks are used for the various pieces of the split-up signal. When you play back the eight tracks of data, all of the pieces are put back together and sent to the D/A converter as 96 kHz/24-bit information.

If you have a 16-bit 8-track or better hard-disk recording system that supports the ADAT lightpipe, you could record the 96 kHz/24-bit audio and play it back with no problems. You should be able to do hard edits to all eight tracks, but fades or level changes will screw up the data and rip your speakers right off of the wall when you play back the resulting digital audio train wreck. In a 24-bit Pro Tools system with the 96 kHz/24-bit data stored as a 4-track stream, you can edit, perform fades, and change levels as long as you do it to both halves of each channel at the same time. Make sure the tracks are grouped together so that everything gets edited together. Did I say that there would be big trouble if

continued on page 138

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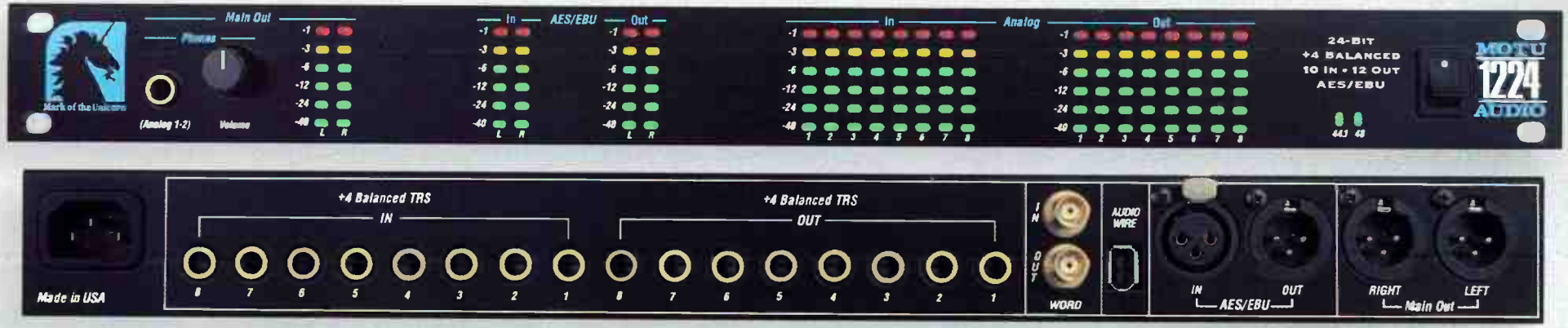

24-bit balanced +4 analog I/O - with 116 dB dynamic range - for your Mac or PC

- The 1224 is a 24-bit hard disk recording system for Mac OS and Windows 95/98.
- 10 independent inputs and 12 independent outputs.
- 24-bit A/D and D/A Converters for 24-bit recording and playback at 44.1 or 48 KHz.
- 116 dB dynamic range (A-weighted) on inputs and main outputs.
- 8 balanced +4 dB TRS analog inputs and outputs.
- Stereo balanced +4 dB XLR main outputs.
- Stereo AES/EBU digital I/O (24-bit).
- Word Clock In and Out.
- Available as a stand-alone system or as an expander for the 2408 system.
- Expandable — connect extra 1224 & 2408 expansion I/Os. Add up to 48 additional inputs and outputs to a core 1224 system.
- Includes AudioDesk™ sample-accurate workstation software for Macintosh with recording, editing, mixing, real-time 32-bit effects processing & sample-accurate sync.
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1224 expansion I/O for the 2408 system: $995

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