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Gear 'Slut' Extraordinaire
Roger Nichols'...*

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PROJECT RECORDING
& SOUND TECHNIQUES
VOLUME 10, ISSUE 10
OCTOBER 1999

ON THE COVER:
Roger Nichols in his pool-side studio. Photo by Deborah Gray Mitchell.

IN AN EFFORT TO AVOID ANY MULTI-FUNCTION COMPARISON TO THAT OF A SWISS ARMY KNIFE, WE HAVE ELIMINATED THE TOOTHPICK.

M-00

The comparison would have been flattering, however, the M-00 is worthy of high praise on its own merit. Not only is the M-00 extremely versatile, articulate and accurate, it delivers tremendous output, along with surprising bass and clarity for a monitor of any size. And while the M-00 is built to the construction standards of a polar ice breaker, its compact nature makes simple duty out of schlepping it from one session to the next. And because the M-00 is magnetically shielded, it is ideal for use with PC based workstations. Further, it is sold separately so you can easily gang together 5.1 systems or daisy chain up to 10 M-00's per channel for fixed installations. As for the toothpick, all considered, we simply figured it was something you could learn to live without. **www.nhtpro.com**

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Lend **a Hand**

Some observations about the AES convention

At press time, we are getting ready for AES. By the time you are reading this, the show is a memory. Here are some trends I'm already finding out:

In the coming months, be prepared for lots o' legit, new products, not just glossy retreads. Check to see if the big companies make a move into "one-stop shopping." Some of them, I believe, will take the full-service approach with their new lines, telling us: "Your shopping needs stop here at XYZ." These companies have always produced many different types of gear, but now their product lines will be engineered to support each other and work together, as with a turnkey studio design. The question for the market and users is this: Will we be willing to commit to only one brand? It's a philosophy professionals have avoided in the past. We have always had a roving eye when it comes to gear.

On the other side of the money-fence, smaller companies will be looking to mate their latest with existing products from the aforementioned larger firms — the third-party approach — as it gets more difficult to break into a crowded arena.

I was recently discussing the topic of mentoring with your favorite maintenance columnist, Eddie Ciletti. It's a heartfelt subject E.C. and I address every so often. (See Mr. Ciletti's column in this issue.) The idea of passing on help, information, and tales of audio woe seems, on its surface, a noble one. However, it should appeal even to the less noble among us. Seeing others arrive at solutions, or make great recordings, thanks to your information, can't be all bad. Hey, it might even feel good. Now, I am sure some out there are muttering, "Whoa, H., you're talking about givin' away my hard-earned sonic voodoo. No can do. I fought hard to get it — I ain't givin' it up."

If so, think back. Surely someone helped you along the way, or is helping you now. Pay the good deed back. Not to just anyone, of course. Choose the recipient carefully, so you can give the advice freely, knowing your time and knowledge are appreciated. Share it, and you leave a worthwhile legacy as you move on to other projects. Keep to yourself, and your sacred, sonic voodoo becomes self-indulgent, sonic doodoo.

Back in the day, mentoring was common in the studio world. Now, though, the technology that allowed the growth of the project studio environment has created a structure whereby more people work alone. It's become a closed world of the individual, a computer, digital gear, and telephone lines. Alone together. The mentoring tradition is not fostered in that atmosphere. That's why we need to reach out to younger, less experienced players and engineers. They can't find us as easily as in the past.

Mentoring is about sharing information; it is real-world audio education. Education is something I have supported since my days as director of studios for the Recording Institute of America. It's why I believed in creating magazines like EQ and GIG, and why I work with folks such as Craig Anderton, Eddie Ciletti, David Frangioni, David Miles Huber, Roger Nichols, Mike Sokol, Wade McGregor, et al.

Many readers know I organize — under the EQ aegis — workshops and seminars for university and independent recording programs and music retailers throughout North America. Doing so keeps me aware of what the real technology needs and problems are for those working in sound. It's mentoring on a broad scale. Speaking of which…

Manufacturers can also be mentors. In fact, it is good business. Despite that, however, most companies ignore the process. From speaking with students and professionals around the country, as well as from my own recording projects, I know that the more complex our audio tools become, the less adequate owner's manuals, on-line help, and 'phone tech help become. Manufacturers need to support the end user by getting out and showing folks about a new technology. That is why I appreciate companies such as Panasonic, NHTPro, and Minnetonka Audio — sponsors of the latest EQ Surround Sound Road Tour this September and October. They grasp the concept of education, and put their money where their mouths are when it comes to lifting up the knowledge base. I will be organizing many more Road Tours over the next months — miking, FOH & monitor mixing, audio coding, surround for small systems, and more — so keep reading and stay in touch.

Until then, enjoy AES, and remember, share the wisdom!

—Hector G. La Torre, Executive Director

Why sound flat...

SOUL SEARCHING

I just received my August issue. I read with delight, the articles on Mitchell Froom and Mike Campbell. Great musicians playing great instruments in a room with a great engineer recording it! What a concept! I might be a little biased as Mitchell Froom and Tchad Blake are two of my favorite producers.

I then read the article on Def Leppard. *God that sounds awful!* Do young producers and engineers that are coming up now think that this is the way you make records?

Sounds like no one was ever in a room playing with each other! The drummer was in a different country! The bass player played a track to one version of a song, someone took Pro Tools to "create" a bass track for a different version! Guitar players playing through speaker emulators...on and on.

Where is the expression, passion, and emotional involvement in that? Plus, they said it only took three months to mix! As if that's some kind of accomplishment.

Froom in return, if I'm not mistaken, spoke of mixing three [songs] in a day. Are some forgetting that the first Beatles album was recorded in a day — *and* it kicks ass!

What's happening to many in the industry? Let's not forget that we are here because we like music and wanted to be a part of recording music for *music's* sake. Technology is great if we use it to enhance, not dictate, the creative process.

If an artist can't play their instruments or they sing so terribly that they need to get an engineer to copy and paste all the choruses or use a box to auto correct pitch, then, in my view, they have no business being in the music business. Every time we auto correct or fix these video-capable people, we are shutting out some truly great artists that may not be so video capable, but are artists! We have enough Britney Spears. We are buying into all this crap the record companies are pushing on us by creating these half-assed artists with our plug-ins and auto-correct crap!

I don't feel that sitting in front of a computer screen and scrubbing and drawing is what Hendrix had in mind when he picked up a guitar. I have a rough time hearing Coltrane say, "Copy and paste that section in all choruses, plus I think I heard a small valve noise that wasn't suppose to be there. Can you scrub that out?"

I feel that they learned their instruments inside and out — had something to say and had the passion to say it. Got a Kramer or Thiele to record it.

Play your instruments guys, and let someone that knows what they are doing put the appropriate mic in front of a box that moves air. Play it right so that you don't spend two weeks in Pro Tools putting everything on the beat.

Knox
Arcadia Recording Studio
Producer/Writer
via Internet

KNOW THE SCORE

In regards to Ned Creed's letter in the August issue, there are many occasions in the professional environment when it will be to your benefit to have musical training. I have engineered sessions where score reading was a necessity because a musician wanted to punch in on the G-sharp in bar 34, for instance. Conversely, I have produced sessions with hired players with an engineer who couldn't read a score, and when it came time to do fixes, and his, "I can do that punch if you show me where it is," only earned him a, "Hand over the remote." There's no time to waste when the clock is running and clients are watching!

While it's true you can find work that you won't need these skills for, like voice overs, live engineering, and techno music perhaps, you'll be removing yourself from a lot of film-type work and acoustic work.

I would recommend at minimum being able to read treble and bass clefs, be able to navigate a simple ensemble score, and be able to carry a tune confidently and without embarrassment. Also, regarding tuning, it's a good idea to be able to tune drums and to be able to hear when pitched instruments are out of tune!

Jack Curtis Dubowsky
via Internet

SAFETY FIRST

Regarding your diagram on page 14 of the July issue: *Do not wire your white wire to the green screw on an AC outlet.* The diagram on that page instructs us to do so. Someone could get seriously hurt doing so.

Andrew J. Harris
via Internet

[Our apologies for the typo. The last sentence in the answer should have read: "In reality, the wiring to the outlet is green, white, and black, respectively. —Ed.]

INSIDE BRIAN'S MIND

I just wanted to attempt a reply to Al Kooper's comment in his "There's No Denyin' Brian Survivin'" article in the August issue that "today, 33 years after *Pet Sounds*' release, [he] still can't figure out the opening chord changes to 'Don't Talk.' " I have also been enjoying *Pet Sounds* for years, but never bothered to analyze any of the music. His comment inspired me to take an analytical look at the album — that song in particular. To my best ability, the opening chord progression goes something like this:

Ebm7   Ebm7/Gb   F7
Abm7/Cb   Bb7
Ebm/Db   Ebm/Dbb
(or Dbb-halfdim7...semantics...)
Ab-halfdim7/Cb   Gb/Bb
Abm   Db7
Abm   Db7 (etc.)

The real genius of this progression is that, despite some simple changes (mostly tonic, subdominant, and dominant with an eventual move to the relative major), Brian Wilson creates a complex sense of harmony with an inventive bass line of first inversions and a chromatic descent. Some other notable moments include the short two-chord sequence after the opening Ebm7 and the parallel minor mixture of the Ab-halfdim7 chord near the modulation to the relative major. Truly, such an example reminds me that the tools of genius are but simple ones used in an extraordinary way. I hope this little analysis is helpful (I hate it when I can't figure something out).

Trevor de Clercq
via Internet

WRITE TO US

Send your letters to:
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E-mail: EQMagazine@aol.com
Web: www.eqmag.com

World Radio History

INTERNATIONAL INTERFACE

Q Hola. *I'm a freelance sound engineer who has been reading Eddie Ciletti's articles in* EQ *magazine for a long time. They were very helpful when I was working as the studio manager and first engineer in a studio outfitted with DA-88's, SV-3700's, and other digital devices he seems to know so well.*

I'm trying to find information regarding TDIF protocol (data chains, rates, voltages...). I'm wondering if it's possible to implement a 2-channel TDIF/AES interface. Where can I find this information? I've contacted the Spanish distributor for TASCAM, but they won't give me this "secret" information.

Jose M. Rosillo
Madrid, Spain
via Internet

A You would have to contact TASCAM as a "developer" in order to receive TDIF protocol information. [TASCAM/TEAC America, Inc., 323-726-0303.] However, the "shortcut," at least for the hardware, would be to obtain the service manual for the IF-88AE and IF88-SD — the TDIF-to-AES and TDIF-to-S/PDIF interface boxes, respectively, made by TASCAM.

Eddie Ciletti
Contributing Editor
EQ magazine

ASK DR. DIGITAL

Q *What are the differences between the AES/EBU and S/PDIF digital formats? Do AES/EBU and S/PDIF each require special cables? Can we use an analog XLR, balanced cable in an AES/EBU plug? What is their impedance rating?*

Alex Chen
via Internet

A Here's the official stuff: Table 1 shows the standard IEC958 "digital audio interface" from the EBU (European Broadcasting Union).

Basically, there are two versions of IEC958: consumer and professional. We call the consumer version S/PDIF; it's the one on the back of our CD players. It can be either in 75-ohm coax or plastic fiber optics (called TOSLINK) and has a voltage lev-

el around 0.5 volts. The professional version is AES/EBU (sometimes called AES-3). It uses exactly the same data structure as S/PDIF, but with a higher voltage of around 5 volts using balanced circuitry, XLR connectors, and twisted-pair cable with an impedance of 110 ohms. The relationships are very similar to analog audio with –10 unbalanced and +4 balanced lines. Also, note that S/PDIF includes the SCMS copy protection bit, while AES/EBU ignores it and uses an ASCII ID text instead.

However, you probably couldn't care less about this technical wordplay. My highly developed powers of observation tell me you just want to hook this stuff up. Therefore, here's the bottom line. Yes, you can use regular XLR mic lines to hook up AES/EBU connections, but it's not recommended. I've used it for short runs with low-bit rate audio (16/48) without any trouble, but long runs are impossible, and you'll likely introduce extra errors and jitter in the signal. You really need balanced twisted-pair cable rated with 110-ohm impedance to do it properly and not degrade your signal. Moreover, if you want to play with 24/96 rates, then everything must be perfect for it to work at all. For consumer S/PDIF, just get regular 75-ohm video cables with RCA connectors from your local video store and you'll be fine. However, don't use RCA audio cables for all the same reasons mentioned about the XLR cables.

You can pad down the levels from an AES/EBU output to work going into an S/PDIF input. You can do this by inserting a 330-ohm resistor in series with pin-2 and putting a 91-ohm terminating resistor across the input, then tie pin-3 to pin-1 in the XLR connector. This is similar to how you would passively convert a balanced +4 dBu analog signal to unbalanced –10 dBu audio. Going the other way requires some active circuitry to boost the signal level, but it's well within standard breadboard techniques. If you don't want to roll your own, Behringer makes an inexpensive box called the ULTRA-MATCH that will go from anything to anything, and will also take care of the SCMS copy protection bit. Check out www.samson-tech.com/behringer/digital.html for more information.

Mike Sokol [call him Dr. Digital]
Contributing Editor
EQ magazine

IF THE SPIRIT MOVES YOU

Q *I am considering getting a Spirit 328 for my project studio, and plan to operate it with two ADAT XT20's for 16 tracks. I know that the 328 is suitable for interconnection with ADAT recorders through fiber optic connections; however, I am not real sure about how to provide foldback (cue) to the talent while recording. Currently, I am using a Seck 1282 console, which provides a separate monitoring strip with level, pan and effects injection for each track. However, does this capability exist with the Spirit console? Maybe I'm overlooking something, but I just can't figure how, with the Spirit, I am going to provide the talent with a mixed, panned foldback with reverb, etc., as I can with the Seck. For one thing, it seems that, within the Spirit 328, the (digital) signals to tape are postfader. If this is the case, how can I control individual foldback levels and effects to the talent while recording? I don't want to shell out the bucks for the 328 until I have some idea how to provide this foldback audio. How would you do it?*

George Hodgkiss
Phoenix Productions
via Internet

A The 328 has four aux busses plus two FX busses for the internal reverbs. Each aux bus can be universally assigned as either pre- or postfader. That is, all aux 1 sends can be pre or post, but you can't single out a channel strip and make one input channel different from the rest. What I've done in my studio is set aux 1 and aux 2 as prefader and used them to drive headphone cues for the talent. Then I've set aux 3 and 4 as postfader to drive external reverbs (my beloved REV-7, for example), and then FX 1 and FX 2 can be used to drive the internal Lexicon effects. If you return your external effects through a channel strip rather than an aux return, then you can add a little confidence reverb to the talent's cue mix without printing it to tape. I've actually done this live with the 328 a few times with cue wedges, and while it's not quite as quick as having a dedicated analog monitor board with everything laid out on a physical surface, it does all the functions required to route the signals properly.

Mike Sokol
Contributing Editor
EQ magazine

World Radio History

VM-3100

Roland

VM-3100

TABLE 1

	AES/EBU	S/PDIF
Cabling	110 ohm shielded TP	75 ohm coaxial or fiber
Connector	3-pin XLR	RCA or BNC
Signal Level	3 to 10 volts	0.5 to 1 volt
Modulation	biphase-mark-code	biphase-mark-code
Subcode information	ASCII ID text	SCMS copy protection info
Max. Resolution	24 bits	20 bits (24 bit optional)

FUNNY BUSINESS

Q *I would like to make quality recordings of telephone conversations for possible use later as a comedy album (not a major commercial release). I'm not an engineer, but I used to be a musician, so I am familiar with basic recording concepts. What is the best way to make extremely good-quality recordings of telephone conversations?*

Dan Gerawan
via Internet

A While it seems pretty simple in concept, getting audio on and off the phone lines has its own set of unique challenges. If all you're attempting to do is get a good recording of the remote caller without consideration of the quality of the local audio, then a simple 600-ohm transformer-based coupler will do the job. For instance, JK Audio makes something called a QuickTap Handset Tap. This connects between the telephone and the handset, so you can use it on any single or multi-line analog or digital telephone. (Cost on the street is around $120.) This will get you the needed DC blockage and produce a mic-level signal from the telephone line. However, this sort of device doesn't provide any isolation between the send and receive audio, which means the level of local voice will be a lot louder than the remote voice.

This is an artifact of how the phone system squeezes both ends of a conversation down a single pair of wires. Another (albeit more expensive) option is something called a Hybrid Coupler. This unit hooks to the POTS (Plain Old Telephone System) line itself, and can seize the line and do the dialing for you. Gentner is one of the best know makers of hybrid couplers. Their simple analog versions provide about 10 dB of isolation between send and receive, while the more comprehensive digital hybrids can give you 50 dB or more with automatic recalibration of the nulling effect with every call. Prices range from around $200 for a simple analog unit, up to several thousand dollars for a digital unit that does it all. This level of sophistication only gets important if you're doing a live call-in show because you not only need to capture the remote caller, you also need to feed them high-quality audio from the host's mic feed. That's where something called a mix-minus bus comes in.

If you need to feed a remote telephone listener a mix of the broadcast in progress, then it's important to keep the remote caller's voice out of that same mix. If you don't, there will be extra echoes and feedback from the cross-connection between the two-wire phone connection and your mixing console. A console with a mix-minus bus provides a cue mix for the remote phone listener that nulls out the signal coming from the remote phone itself, while allowing them to hear the voice of the host and any music or SFX cues going out on the air. A good source to get this sort of gear is Markertek Video Supply (800-522-2025). You also can get more information directly from the Gentner Web site at www.gentner.com.

[All the information herein presupposes that the callers have been informed they are being recorded.]

Mike Sokol
Contributing Editor
EQ magazine

A [*Dale Smith, applications engineer, Gentner Communications Corp., adds:* You have properly addressed the issue of isolation. This is an important issue to consider when selecting a hybrid. If your application is simplex (one direction audio), then isolation is not a concern. You only have to worry about the proper level and quality of audio. If your application is half-duplex (two-way audio, only one path active at a time), then, again, isolation is not a concern. However, matching the level of the send and receive audio is. If the application is full-duplex (two-way audio, both paths simultaneously active), then isolation and level matching is a big concern.

There are two parts of isolation to consider when using a hybrid. First is the hybrid null. If a hybrid does not provide sufficient isolation between the send and receive paths of the phone line, then phasing and/or feedback can occur as the receive audio is mixed with the send audio at the console. Second is a mix-minus bus. If the audio console does not remove the caller's voice from the return mix, then they'll have a hard time conversing as their own audio is sent back to them after a noticeable delay. This is due to the inherent lag time in all communications, especially ones that take a satellite hop.]

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NEWS

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Surround 2000 Conference Offers 5.1 Audio Techniques

Leading names in audio to highlight two-day conference program

PORT WASHINGTON, NEW YORK — The leading names in music and audio have targeted November 5-6, 1999 as the launch date for the next era in multichannel audio. At that time, Surround 2000, the 1st International Conference & Technology Showcase, will explore the latest surround sound technologies, production techniques, and marketing challenges.

The event is scheduled for November 5-6, 1999, at the Beverly Hilton Hotel in Beverly Hills, California.

"With the introduction of new multichannel audio carriers such as DVD-Audio, SACD, and DTV this fall, it is time that the professional audio and music communities thoroughly investigate the current state of surround sound technologies and begin to evaluate ways for their widespread introduction to the mass market," explains Tomlinson Holman, conferencechaiman.

Attendees will include professionals involved in the creation, production, or installation of multichannel audio systems. Surround 2000 is sponsored by leading surround technology companies, industry associations, and publications.

Corporate sponsors include JBL Professional, Dolby Laboratories, DTS, and TMH Corporation.

Association sponsors include: the International Alliance of Multichannel Music (IAMM), the Music Producers Guild of the Americas (MPGA), the Consumer Electronics Manufacturers Association (CEMA), the International Recording Media Association (IRMA), and the Society of Professional Audio Recording Studios (SPARS).

Publication sponsors include Pro Sound News, EQ, Surround Professional, Systems Contractor News, Videography, Television Broadcast, Audio, Widescreen Review, and Music Connection magazines.

The conference program includes two-day's worth of tracks on all facets of surround technologies and production techniques. A separate business conference will discuss ways to market surround to the consumer, and a basics course will introduce L.A.-area musicians and recording artists to the art of surround sound composing and recording. Hands-on mixing training, surround auditioning sessions, and technology showcases will accompany the tutorial program.
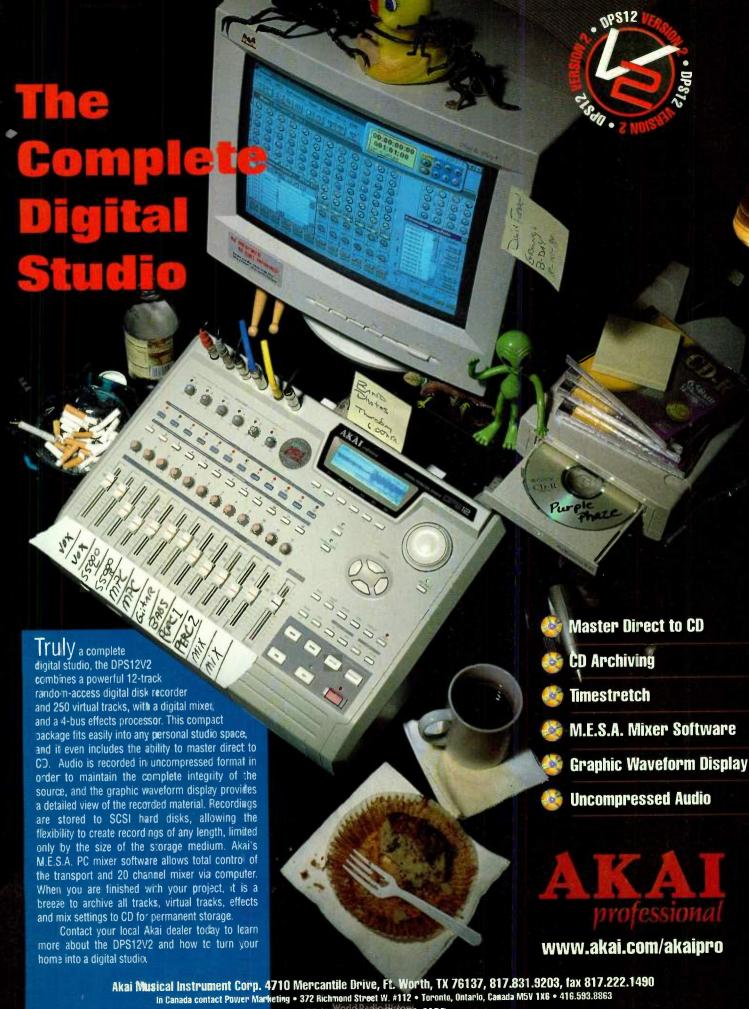
The leading names in audio for music, film, video and broadcast have been named as Conference Program Advisors for Surround 2000. The Surround 2000 Conference Advisory Board includes:

Murray Allen, Electronic Arts
Ed Cherney, Record Producer
John Eargle, JME Consulting
Michael Frondelli, Capitol Records
Herbie Hancock, Recording Artist
Tomlinson Holman, TMH Corporation
Jac Holzman, Warner Music Group
Robert King, Entertainment Technologies
Bob Ludwig, Gateway Mastering Studios
Robert Margouleff, Record Producer
George Massenburg, GML
Bobby Owsinski, Surround Associates
Phil Ramone, Record Producer
Gary Rydstrom, Skywalker Sound
Steve Thompson, Cinram/POP
Paul West, Universal Music.

Admission to the two-day conference program will cost $395. One-day passes will be available for $225. To register and for full conference information, visit the Web site at www.surroundpro.com

1st International surround 2000

November 5-6, 1999 • Beverly Hills Hotel Beverly Hills, California
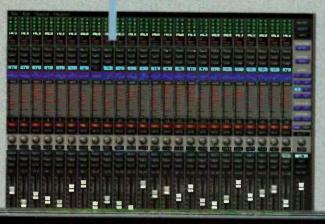
3 **56 INPUTS AND 72 CHANNELS.** It's harder to impress clients when your console isn't twenty feet long, but the D8B gives you the same performance. One-touch switching gives you 24 tracking channels (Chs 1-24), 24 tape return channels (Chs. 25-48), 16 FX returns, 8 alt returns, 8 virtual fader groups, 8 bus masters and 8 control faders.

4 **EQ, COMPRESSOR AND GATE ON EVERY AUDIO CHANNEL (W/O ANY CHANNEL LATENCY).** Most other digital mixers or work stations "bus" your audio to a DSP pool for EQ and dynamics. The resulting propagation delay causes phasing problems with your audio. The D8B has dedicated DSPs for each channel pair, so you don't run out of DSP and you incur no DSP latency. (One reason it sounds so good.)

5 **A CHANNEL AT A GLANCE VIA ON-SCREEN "FAT CHANNEL" DISPLAY.** Even million-buck mondo mega consoles don't have displays this informative and easy to use.

11 **EASY-ACCESS TRANSPORT CONTROLS.** Run the transport controls of any tape or hard disk recorder that has MMC. Add cue points anywhere. Create song start and stop points or loop between two points. The D8B speaks SMPTE or bars/beats/ticks. You can use the keypad to dial in to any point with frame accuracy.

12 **AUTOMATION EDITING MADE SIMPLE.** The Digital 8•Bus' Version 2.0 software provides you with the most comprehensive automation and automation editing tools ever found on an automated console of any size. View levels and mutes on a fullsize pop-up Mix Editor window that scrolls in real time with the audio. Then make changes graphically with the flourish of a mouse — or by typing in exact time code.

12 TWELVE REASONS WHY

48 Windows, Walter Afanasieff, Arista Records, David Arnold, The Bomb Factory, Jeff Bridges, Ed Cherney, Cinemuse, Columbia /Black Music Division, Michael DeLorenzo, Tim Dubois, The Dungeon, Earth, Wind & Fire, Ed Green, Mick Guzauski, Scott Hendricks, Islandlife Records, Wayne Linsey, Little River Band, Machine Head Post Production Studios, Binky Mack, Bobby Mackston, MXPX, Keith Olsen, Glen Phillips, Poke, Trent Reznor, Lance Rubin, Michael Score, Soundelux, South Beach Studios, Chester Thompson, Virgin Records, Bruce Willis, Dwight Yoakam, & the CIA are among those who

OWN A MACKIE DIGITAL 8•BUS*.

(Thousands in use around the world everyday.)

*Mention in this excessively wordy headline indicates ownership only as reported to Mackie Designs and is not intended in any way to denote endorsement, official or otherwise, by the individuals, companies or organizations listed above. Especially the CIA.

Ocean Way Nashville Adds AMS Neve VR96

Ocean Way Nashville has added to their facility a 96-channel AMS Neve VR Series audio console with GML automation. Now residing in Ocean Way Nashville's B Studio, the new AMS Neve VR replaces the Sony Oxford, which was recently moved to their Studio C. The Neve VR96 is also 5.1 compatible, with an eight-channel film monitoring section. Additional modifications by MAD Labs' John Musgrove are also featured.

After consulting with a select group of top music producers, Ocean Way Nashville Owners Gary Belz and Allen Sides selected the AMS Neve VR console.

Belz explains, "The AMS Neve VR Series has an excellent sound and features that our clients here really respond to." With regard to the console's appeal to Ocean Way's clientele, Belz adds, "We've made significant modifications to the console, and it is truly unique to this region — as is Studio A."

Ocean Way Nashville's Studio A has the world's largest 80-channel AMS Neve 8078 console with a 60-channel monitoring section and GML automation. The size of the console, its surround capability, and Ocean Way's customized monitoring system will set Studio B even further ahead.

Belz discusses future plans for Ocean Way, suggesting, "We anticipate putting the new VR to work on 5.1 music mixes and potentially feature films as well. The size of the console should lend itself to numerous projects, and our customized Ocean Way monitoring gives us a truly distinctive sound. With all of that, plus our extensive outboard equipment inventory, Ocean Way Nashville offers an unparalleled recording experience."

For more information on AMS Neve products, call 212-965-1400 or visit www.ams-neve.com.

Arboretum Realizer Improves MP3 Sound Quality for Macs

Arboretum Realizer, the audio enhancer for Internet music listeners, is now available on the Mac platform as a plug-in for the Casady & Greene SoundJam MP player/encoder.

"We are thrilled to include Arboretum Realizer in SoundJam MP," says Donald Beirdneau, VP for Product Development at Casady & Greene, Inc. "Arboretum Realizer provides a noticeable improvement in music quality and is especially effective when streaming music over the Internet. The combination of SoundJam MP and Realizer provides a dynamic duo that is impossible to beat!"

Unlike conventional equalizers and bass/treble controls, Realizer adds missing low-end frequencies, expands the stereo image, and synthesizes new harmonics to recreate the realism of the original recording.

It restores much of the sound quality lost to compression and lets users hear better audio with their current speakers and sound card.

"Millions are using MP3 players to listen to music from the Internet, but MP3 files and streams just don't sound as good as the original CD-quality recordings," says Georges Jaroslaw, Arboretum Systems' founder and CEO. "We decided to use a little studio magic — DSP technology originally designed for recording engineers — to improve the Internet listening experience."

SoundJam MP (www.soundjam.com) is the first Mac player/encoder to feature streaming MP3 support. Arboretum Realizer is included in every SoundJam MP package.

Complete information and free downloads can be found at www.arboretum.com/realizer.

Guitar Center Launches Web Affiliate Program

Guitar Center, Inc. recently announced that its wholly owned subsidiary, Musician's Friend, Inc., has launched an affiliate program at its Web site, www.musiciansfriend.com/affiliate. Musician's Friend, Inc. is a leading direct mail and e-commerce provider of musical gear, acquired by Guitar Center, Inc. earlier this year.

Affiliated Web sites, through a link to the Musician's Friend home page, can collect an advertising referral fee on any orders that originated from their site. The program will allow affiliates to receive advertising fees from sales of $5 guitar strings to $50,000 recording studios, and will include no cost limit or quota.

Rob Eastman, CEO of Musician's Friend, comments, "We believe this new affiliate program will drive more traffic to our Web site and will contribute to the growth we are already seeing in our Internet orders." Eastman continues, "The average order through our Web site is well over $200. Individual affiliates will have an opportunity to see real income from this program."

The site combines a rich mix of content and commerce, where visitors can search for gear, shop, order catalogs or gift certificates, see and hear individual products, download free software, read an industry newsletter filled with helpful tips, review product reports, or track recent or-

ders. The site also features a broad selection and presentation of music gear covering thousands of products and including photos and specifications for more in-depth product information.

The Musician's Friend affiliate program is 'e-nabled' by Be Free, Inc., and the program earns affiliates a commission on net sales from all purchases made through a link to Musician's Friend. This "pay-for-performance" business model dictates that Musician's Friend only pays a commission when a sale is generated.

Check out the award-winning Web site at www.musiciansfriend.com for more details on the new affiliate program.

AKG **C60**

A look at AKG's tube-based predecessor to the C451 and C460 microphones

MICROPHONE NAME: AKG C60

FROM THE COLLECTION OF: Ken Avant, Avant and Associates, Upland, CA

PRICE WHEN NEW: Approximately $400

YEAR OF MANUFACTURE: 1960 through 1969

TYPE OF MIC: Vacuum tube condenser

POLAR PATTERN: Cardioid

FREQUENCY RESPONSE: 30 to 18,000 Hz +/– 2.5 dB; 20 to 20,000 Hz when used with omnidirectional capsule

SENSITIVITY: –62 dBV (1 volt/dyne/square centimeter)

RATED SOURCE IMPEDANCE: 50 or 200 ohms with N 60A power supply
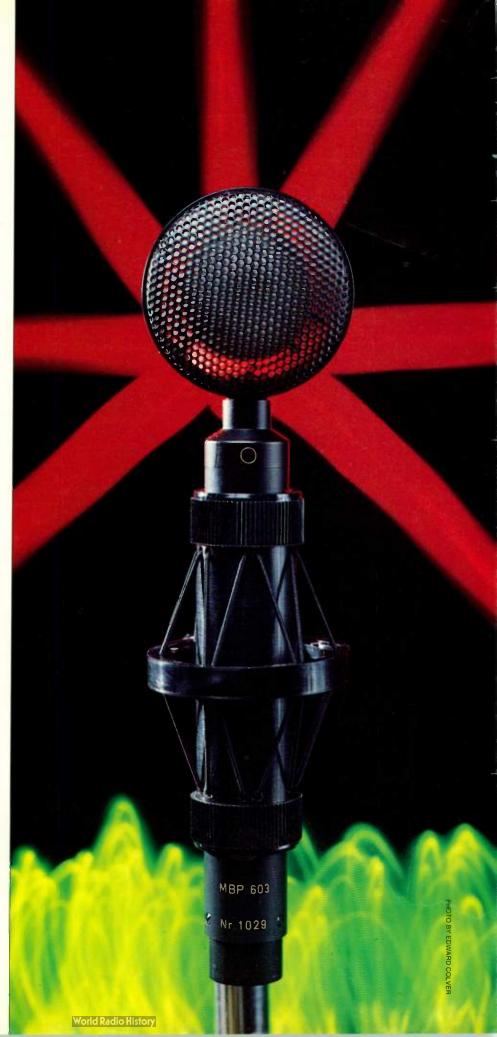
TUBE: AC 701k

FILAMENT VOLTAGE: 4 volts dc at 105 milliamps

PLATE VOLTAGE: 120 volts dc at 0.6 milliamps

RESIDUAL NOISE: less than 20 dB SPL

SOUND PRESSURE LIMIT: For 0.5% harmonic distortion, 117.5 dB SPL

DIMENSIONS: 20 mm diameter x 130 mm length

MIC NOTES: Originally distributed in the United States by Norelco (a division of North American Philips Company), the AKG C60 was the tube-based predecessor to AKG's extremely successful C451 and (later) C460 microphones. Initially, the C60 was supplied with the CK28A cardioid capsule, which employed an aluminum diaphragm. The CK28A was later revised to the CK1 cardioid capsule,° which featured a gold-deposited mylar diaphragm; the CK1 was subsequently used for the C451, and became a popular choice among studio and live sound engineers. An optional omni-directional capsule — the CK26A — was also available. The C60 was discontinued by AKG in 1969.

USER TIPS: Since response of the C60 extends down to 20 Hz, the microphone includes a low-frequency rolloff switch with positions for 0, –7, and –12 dB at 50 Hz. AKG recommends the –12 dB position for vocal or solo use. Power to the C60 is delivered by the AKG N 60A outboard power supply, which connects to the microphone via AKG's MK 60 cable.   **EQ**

MBP 603

Nr 1029

EQ PRODUCT VIEWS

MAX OUTPUT

Dbx Professional introduces two new digital output cards — the 704X and 504X — which bring the sonic performance of digital to dbx's popular Blue and Silver Series processors. The 704X is designed for installation in a 160S/160SL compressor or dbx 786 microphone preamplifier. The card features sample rates at 44.1, 48, 88.2, or 96 kHz; a 48-bit internal signal path for increased headroom and low level resolution; a Type IV A/D conversion system; two user-selectable noise-shaping algorithms for a lower perceived noise floor; and more. The 504X is designed for use in dbx's Silver Series products — the 566 compressor, 576 microphone preamplifier/compressor, and 586 microphone preamplifier. Its features include sample rates at 44.1 or 48 kHz; a Type IV A/D conversion system; and AES/EBU format on gold-plated Neutrik XLR and S/PDIF format on a gold-plated RCA coaxial connector. The 704X costs $799.95 and the 504X costs $399.95. For more details, call dbx Professional at 801-568-7660, fax them at 801-568-7662, or visit www.dbxpro.com. Circle EQ free lit. #111.

WORLD CLASS

The WK2 Oriental is the newest addition to Generalmusic's range of keyboards dedicated to the music of the Mediterranean and Middle East. In addition to adding all the sophisticated technical features of Generalmusic's WK2 Arranger keyboard, the WK2 Oriental offers a vast array of traditional sounds and rhythms inspired by the fascinating world of Arabic, Greek, and Turkish music, created in collaboration with leading musicians from those regions. The WK2 Oriental features 61 keys with MIDI aftertouch, 32-note polyphony, and 16-part multitimbrality. An internal sound library of 460 sounds — (including 16 Oriental sounds) and 17 drum kits (including one Arabic drum kit) — is available for all different types of music. The keyboard may be programmed to feature a split, and three sounds may be layered together to create complex sound textures. It is also equipped with two digital processors offering 22 reverbs and 22 modulation effects. For more details, call Generalmusic Corp. at 630-766-8230, fax them at 630-766-8281, or visit www.generalmusic.com. Circle EQ free lit. #112.

CENTRAL FIGURE

Genelec's 1034BC Center-Channel Monitor is specifically designed for three-channel and surround sound systems. Complementing a two-speaker Genelec 1034B Monitoring System, the new 1034BC ensures tonal compatibility by using the same mid and treble drivers as Genelec's 1034B. The monitor also features Genelec's Directivity Control Waveguide (DCW) technology. The 1034BC allows for optimum placement, and can be placed above or below a video screen, or vertically if the DCW is rotated accordingly. While it can be used free-standing, the center speaker is designed for flush mounting and the separate amplifier is built into a rackmount chassis, easily fitting into a standard 19-inch 7U equipment rack, with vibration isolators that double as quick release hinges. All drivers in the 1034BC center monitor are magnetically shielded to minimize interference at video monitors. For more information, call Genelec at 508-652-0900 or visit www.genelec.com. Circle EQ free lit. #113.

EQ PRODUCT VIEWS

WIZ-BANG

Allen & Heath's WZ20S console — part of its MixWizard Series — is packed with fully-featured dual stereo inputs, making it ideal for many AV and PA applications demanding numerous stereo sources, and for professional keyboard and effects submixing. The WZ20S has four mic/line inputs, with six auxes and acclaimed four-band, two-sweep MixWizard EQ. The unit also includes eight stereo inputs, each with a four-band EQ and six aux sends. The stereo channels have dual A and B inputs with phono and TRS balanced jacks, respectively, allowing two stereo sources to be permanently connected and eliminating the need for cable adapters. Sources A and B can be selected individually or mixed together with independent gain control. Also, there are two assignable stereo mix busses, Mix1 and Mix2, each with inserts, metering, and balanced XLR output. For more information, call Allen & Heath at 801-568-7660, fax them at 801-568-7662, or visit www.allen-heath.com. Circle EQ free lit. #114.

TRUE TEST

The TEST 1-2-3 from New Frontier Electronics is an easy-to-use and effective diagnostic tool for troubleshooting instruments, microphones, audio and video components, speaker systems, and entire signal chains. Housed in a rugged anodized aluminum enclosure, measuring only 4 1/2 inches long, the TEST 1-2-3 features a 1/4-inch jack on one end and three LEDs on the other. The LEDs — amber, green, and red — can easily be seen in the dark or in low-light conditions, and provide accurate indication of three distinct signal levels. The TEST 1-2-3's compact design fits all instruments, as well as effects pedals and pedalboard systems, making it a "must" for every guitar tech. With a low-Z balanced transformer adapter, the TEST 1-2-3 can be used to quickly check microphones, snakes, and mixers. The tester has an operating battery life of 200 hours and is available for $89.95. For more information, call New Frontier at 215-862-9344, fax them at 215-862-0270, or visit www.frontierelec.com. Circle EQ free lit. #115.

World Radio History

NEW FROM EVENT!

PS5 Biamplified Direct Field Monitors

Not everyone has the space for full-size direct field monitors. But everyone has the need for monitors that provide pure, accurate, detailed sound—regardless of their size. And that's exactly what the PS5 delivers: wide range, non-fatiguing, reference monitoring in a compact, small profile format. Oh, yes, and sound that's just plain fun to listen to.

We get that sound through the use of custom drivers, crossover components, and amplifiers. That's right, amplifiers—one for the woofer, and one for the tweeter. Giving each driver its own amplifier means the amps can be far more efficient, since each one is dedicated to a specific frequency range.

The result: Increased dynamic range. Higher SPL. Greater transient response. Improved damping. Smoother phase response. Lower intermodulation distortion.

Or to put it another way:
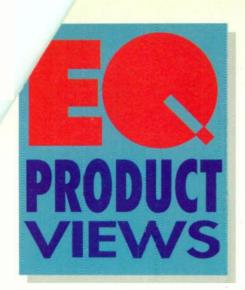
Mixes You Can Trust.

Actual Size

PROJECT . STUDIO 5

E V E N T

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ELECTRONICS

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Fax: 805-966-7771
E-mail: info@event1.com
Web: www.event1.com

EQ PRODUCT VIEWS

FEEDBACK FOILED

Sabine has announced Version 2 of its acclaimed and informative "bible" of feedback control. The new Positive Feedback booklet explains the cause of feedback, and offers solutions ranging from acoustical to signal processing. The 20-page booklet features the complete lineup of Sabine FBX products, including the company's ground-breaking entry into the wireless microphone market, the True Mobility Wireless Microphone system. Positive Feedback Version 2 is available by mail from Sabine, or can be downloaded in Adobe Portable Document Format (PDF) from the Sabine Web site at www.SabineUSA.com. For more details, call Sabine at 904-418-2000 or fax them at 904-418-2001. Circle EQ free lit. #116.

MAINTAIN FOCUS
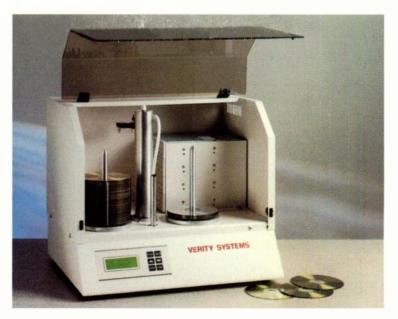
New from Studiomaster, the Focus 608 is an 8-channel rack-mount stereo powered mixer. The Focus 608 is powered by dual 300-watt internal amplifiers. Features include six mono mic/line channels with 3-band mid sweep EQ, two stereo line channels with 3-band fixed EQ, channel PFL "listen," active channel pan controls, and 60 mm faders. An onboard 32-bit digital processor provides digital reverb and multi-effects with 79 preset programs and four user memories. The master section includes dual 7-band graphic equalizers, amp split to monitor switch, output inserts, stereo record/playback bus, headphone bus, stereo auxiliary return, 60 mm faders, and much more. The suggested retail price is $1149. For more information, call Studiomaster at 800-878-7883, fax them at 714-998-2086, or visit www.studiomaster.com. Circle EQ free lit. #117.

PROFESSIONAL DUPLICATOR

Verity Systems has recently extended its already considerable range of CD duplication products. The VS6000 Automatic CD-R Duplicator is a fully automatic, standalone unit capable of duplicating up to 100 CDs. This is accomplished in just over four hours; the VS6000 can even handle PQ and ISBC codes, and is 80-minute-media compatible. With an internal 4 GB hard drive and upgradeable firmware, the unit is one of the most adaptable and future-proof CD-R duplicators in its range. For more information, call Verity Systems at 800-642-5151, fax them at 530-626-9395, or visit www.veritysystems.com. Circle EQ free lit. #118.

VOLUME CONTROL

Russound has introduced a free-standing tabletop volume control for use in situations where in-wall control is not desirable. The TBL-75 is a stereo control that is a combination standard control and impedance-matching device. The high-quality autoformer and rotary knob controls the signal going to the speakers and the headphone jack. The control handles up to two pairs of 4- or 8-ohm speakers and can handle up to 16 pairs of speakers connected to a 4-ohm amplifier output, when used as part of a Russound Ultra Match system. The TBL-75 has a total power-handling capacity of 126 watts — 42 watts/channel — RMS. The rotary knob operates through 12 detent steps, including the "off" position, and provides a maximum of 43 dB attenuation. The price is $113. For more information, call Russound at 800-638-8055, fax them at 800-915-5519, or visit www.russound.com. Circle EQ free lit. #119.

ONYX DUPLICATES

Discmatic's innovative multi-drive CD duplicator, ONYX, features a proprietary autoloading mechanism that provides fast, safe, and reliable disc handling. ONYX is a standalone 100-disc duplicator that supports up to four 8X drives. It offers on-the-fly CD-to-CD copying, batch copying, and the option of adding a dedicated CD-ROM reader. With the help of its four 8X drives, ONYX can duplicate 24 full CDs (74 minutes of audio/650 MB of data) per hour. To ensure reliable operation, ONYX uses the Discmatic EZ-ONE controller technology that has proven itself over the years in a range of the company's products. With a 4 GB internal hard drive, the unit can store a large volume of data, while Discmatic's Multiple CD Image Management system makes it easy to manage that data for a wide variety of applications. ONYX supports all major CD formats. For more information, call Discmatic at 800-422-6707 or visit www.discmatic.com. Circle EQ free lit. #120.

MORE MINIDISC

HHB's MD80 is the first 80-minute MiniDisc available outside Japan. The disc achieves its extended recording time of 80 minutes stereo (160 minutes mono) by an ingenious reduction of 0.1 microns in the groove width (from 1.2 to 1.1 microns) and by a similar reduction in the pitch (from 1.6 microns to 1.5 microns). Despite these developments, the MD80 remains entirely compatible with all MiniDisc recorders and players, both old and new. Performance specifications for the MD80 remain the same as the widely used HHB MD74, including the critical area of block error rates in which HHB Professional MiniDiscs outperform some consumer discs by a factor of 10. For more information, call HHB Communications USA at 310-319-1111, fax them at 310-319-1311, or visit them at www.hhb.co.uk. Circle EQ free lit. #121.

POWER DISTRIBUTION
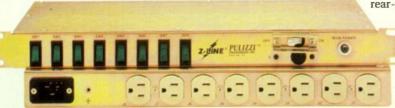
Pulizzi Engineering introduces the T8S series of power distribution systems that provide individual on/off control over eight outlets in a 1U x 19-inch rackmount steel chassis. The front-mounting panel has a main power "on" indicator light that illuminates when the over-current supplementary protector is engaged. Also on the front panel are eight lighted on/off rocker switches that control the individual power to the eight rear-panel NEMA 5-15R (125V/15A) or 6-15R (250V/15A) outlets. The power connector is an IEC 60320 C20 inlet located on the rear panel. This configuration allows the use of any type power cable and plug rated up to 125V/20A or 250V/16A. The price starts at $199. For more information, call 714-540-4229, fax them at 714-641-9062, or visit www.pulizzi.com. Circle EQ free lit. #122.

DRUM MAJOR

Audio-Technica has introduced the new AKP-DRUMS KitPack set of microphones designed for the precise reproduction of drum and percussion instruments in live and recording applications. The Snare/Tom mic features a moving coil dynamic design with cardioid polar pattern for accurate reproduction of snare and rack toms. The Kick/Tom mic also features the moving coil dynamic design and has a frequency response of 60 Hz to 12 kHz. The suggested price is $259. For more details, call Audio-Technica at 330-686-2600, fax them at 330-688-3752, or visit www.audio-technica.com. Circle EQ free lit. #123.

SURROUNDED BY BASS

The new KRK S10 powered subwoofer features a cast-frame 10-inch woofer with a special Kevlar cone. The built-in electronic crossover is designed for both LCR and 5.1 systems. The S10 has both vertical and horizontal internal braces, and the ports are covered with sound-deadening material. The input and output connections are 2-channel XLR and the built-in amplifier is capable of producing 125 watts. The frequency response is 30 Hz to 50 kHz, and the maximum SPL @ 1m is 110 dB. Controls include a System Gain (+6 dB to —30 dB variable), a Low Pass Variable (50 Hz to 130 Hz), and a Phase Adjust Switch (0 or 180 degrees). For more details, call 714-841-1600, fax them at 714-375-6496, or visit www.krksys.com. Circle EQ free lit. #124.

JBL LSR25P Studio Monitor

JBL takes aim at the workstations, edit suites, and small control rooms with their latest LSR Series monitors

BY BOBBY OWSINSKI

The introduction of the LSR Series monitors re-established JBL's reputation as an innovator in speaker technology. Now, with the introduction of the new LSR25P, JBL extends that tradition. Realizing the high demand in the marketplace for a small, high-quality, nearfield monitor, JBL has focused the efforts of its prodigious design team to develop just such an accurate reference monitor for workstations, edit suites, and small control rooms. This exclusive EQ First Look gives us a view of the newest member in the much-touted LSR Series monitor family.

FROM THE GROUND UP

Not content to rest on its laurels and simply use off-the-shelf components, JBL built a unit totally from scratch. The LSR25P combines JBL's latest in transducer and system technology with many of the design philosophies of its big brother LSR studio monitor systems. The Linear Spatial Reference (LSR) philosophy is based on a set of design goals that attempts to control the overall performance of the system in a variety of acoustic spaces by providing much better control over dispersion via transducer selection and crossover frequency design. Due to the incorporation of LSR into the system design requirements, placement rules are relaxed, a more stable stereo image is maintained, and off-axis coloration is minimized.

THE SPECS

The LSR25P is a two-way, biamped system featuring a 5.25-inch woofer powered by a 100-watt amplifier and a 1-inch composite tweeter powered by a 50-watt amp. The shielded 5.25-inch polypropylene woofer is based on JBL's research into low-distortion/high-excursion designs, and provides a substantial increase in dynamic range over other typical transducer designs. The woofer also features a tempered paper cone and a cast aluminum basket for advanced heat dissipation.

The high-frequency transducer is a 1-inch, titanium-composite diaphragm integrated with an Elliptical Oblate Spheroidal (EOS) Waveguide with 60- x 120-degree dispersion (–6 dB off-axis measurements), which is critical to the smooth spatial response required in today's working environments. Active 24-dB-octave crossover circuitry at 2.3 kHz provides a smooth transition in both the frequency and time domain response between low- and high-frequency devices. JBL states that the frequency response of the 25P is 70 Hz—20 kHz, +1/-2 dB, with a maximum continuous SPL of 106 dB @ 1 meter.

The front panel features a level control, power switch, and a bi-color LED that indicates power and any onset of clipping in either amplifier section. The rear panel includes balanced and unbalanced XLR and RCA input connections, and high- and low-frequency response adjustments to compensate for typical desktop boundary effects for use with workstations. Other features include integrated mounting points for horizontal or vertical orientation and a built-in high-pass filter for use with an optional subwoofer.

Price is $399 each. For more information, contact JBL, 8500 Balboa Blvd., Northridge, CA 91329. Tel: 818-894-8850. Web: www.jblpro.com. Circle EQ free lit. #125.

Capture Every Exhilarating **Detail**

"The sound quality of Lucid converters is excellent. They give a sheen to the sound without artificially coloring it."

Marcus Miller Producer, composer

"Lucid converters add a depth to the mix that you can't get going straight into a DAT. You can hear the difference."

Jason Miles Producer, arranger, composer

"With the 8824, we get extremely clean sound on our Sonic system. Even the people in the dubbing room are satisfied."

Jim Henrikson Music editor for James Horner
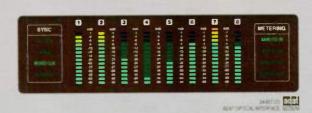
Get the most pristine sound from all your audio gear using 24-bit converters from Lucid Technology. These units preserve the smallest sonic details, revealing the true depth, richness, and clarity of your recordings. So you can count on uncompromising quality on every project.

"The best part about this converter pair is how they sound. The conversion is first rate, with attention to detail not only in the digital domain, but also in the analog circuitry."

George Petersen Mix Magazine September 1999

ADA8824 (ADAT)
Your computer becomes a pro recording studio with this audio interface for ADAT-optical lightpipe gear. Transfers eight channels of digital and analog I/O simultaneously.

ADA8824 (Sonic)
Same capabilities as above, except designed specifically for SonicStudio workstations.

AD9624 and DA9624
First-class converters with superior imaging and realism for use during recording, mastering, and post. Each supports 96kHz sampling.

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Professional Digital Audio Converters

EMU/Ensoniq PARIS Hard-Disk System

Getting the most from this versatile modular DAW

BY STEPHEN ST. CROIX, INTELLIGENT DEVICES, AND DERK HAGEDORN, MARKETING, EMU/ENSONIQ

Since the introduction of PARIS two years ago, there have been frequent software and hardware developments — most recently Version 2.0 software and the SMPTE MEC expansion card. After Ensoniq's merger with EMU last year, the new company, EMU/Ensoniq, is more focused than ever on providing professional audio solutions by concentrating development efforts on the PARIS Bundle 3 System (EDS PCI card, 16-fader control surface w/transport controls, software, Modular Expansion Chassis). The modularity of this system offers almost unlimited expandability via 24-bit MEC I/O expansion cards and additional PCI cards (each offering 16 additional tracks and supporting an additional MEC), and best utilizes the power and flexibility of the PARIS software and hardware architecture.

What is "Free-form" editing, and why is it so special?

We know that some of you have not yet tried PARIS's Free-form Editing Mode. Try it! It is a far better way to build, audition, and fine-tune complex composite edits. It is a dramatic improvement over using tracks or older, "layer"-type systems. You can use the special Free-form Editing Mode for more than just assembling complex takes. You can assign two versions of an edited track (or even a complex flex-track composite) to two different instruments and switch between the two on-the-fly from the Mixer window. This way, you can A/B composite performances while using all of your reverbs, equalization, and other built-in processing, along with VST plug-ins and external inserts.

Does PARIS allow me to use my external processors during the mastering process?

Yes. PARIS allows you to apply external mastering processing (in addition to any internal processing, of course) by simply creating a stereo external insert loop on the master output bus, routing the insert sends to an external hardware effects box (like the TC Finalizer) via the analog or digital outputs of the PARIS interface, and then returning the signal into PARIS after processing. You can do this entire insert loop with a 24-bit path if you use the MEC's analog and digital I/O capabilities.

Am I able to edit my audio during playback?

You can edit any track while playback (or even recording) is in progress. For example, you can change EQ settings and cut, copy, paste, slip, and nudge objects on tracks 15 and 16 while recording on tracks 4, 5, 6, and 7, while listening to all the other tracks, without any interruption of audio. Playback and record functions will not be affected on any tracks other than the ones being edited. This is also true if you create inserts or turn on effects while playback/record is in progress.

Can I play back more than 128 tracks using only one card?

Yes. You can use Virtual Submixes for playing back more than 128 tracks with only one EDS card. If you have an older computer or older hard disk that isn't fast enough to play as many tracks as you wish, you can always use Virtual Submixes to increase the actual track playback capability of your system. Keep in mind that a Virtual Submix will playback exactly the same audio mix as if it were being mixed live on an EDS card, but reduces the disk bandwidth required from 16 additional tracks to only 2. This means that a huge 64-track playback or mixdown will work perfectly on a disk that can only handle 22 actual tracks of audio data, as long as you use 48 virtual tracks (three submixes)! Of course, you can always add more EDS cards — each one gives you 16 more live tracks and another 64 EQs, eight more effects busses, and another entire bank of DSP effects.

For answers to commonly asked questions and for further information on the PARIS DAW, please visit the EMU/Ensoniq Web site at www.emu.com.

MP3 Changes **the Industry**

The compression-heavy file format threatens to revolutionize the way music is distributed

BY CRAIG ANDERTON

Aside from being a wonderful event in general, the Interactive Music Expo in New York last August was a showcase for the much-touted "convergence" of computers, the 'Net, and music. And, without question, the star of the show was MP3, the ubiquitous file format that strikes fear into record companies, hope into unsigned bands, and raises a lot of questions for anyone into recording.

In case you've been living in a cave for the past few years, MP3 is the data-compression format that shrinks digital audio files by ratios of (typically) 5:1 to 22:1; more compression means lower audio quality. Compressed by

10:1, a 30-MB pop tune becomes a much more download-friendly 3 MB — or less than 1.5 MB if you can handle "FM-quality" sound (see fig. 1).

To say the public has embraced MP3 would be an understatement, especially at college campuses where the combination of T1-fueled Internet access, limited financial resources, and a love of music has turbocharged the MP3 phenomenon. Although considered mostly as a way to download music from the 'Net, MP3 has two other significant, emerging applications:

• "High-density" music distribution. It's no problem to fit a couple *hundred* songs on a CD-ROM, a far cry from the couple of dozen you'll find on today's audio CDs. In a way, MP3 has beaten DVD-Audio at its own game by making it possible to include huge amounts of data on a standard CD that plays back on your existing computer.

• Computer-driven "jukeboxes." Today's MP3 players are starting to include database and search functions, making it easy to locate individual songs according to artist, title, genre, or mood. Many consumers are buying a big hard drive (like 6 GB), hooking it up to their computer, digitizing cuts from their favorite CDs, and storing literally thousands of tunes for easy access or background music.

These trends, while welcome in the sense that they get people

involved with music, have many significant ramifications.

• When you have a couple hundred songs on a CD, that's a lot of mechanical royalties if you're working from a conventional music business economic model.

• Alternate mixes and outtakes — formerly the elements that drove sales of CD box sets — are easy to include. In fact, there's no reason why artists can't do mixed-media, "enhanced" CDs with the "real" cuts as audio, and the live cuts, outtakes, and alternate mixes as MP3 file "bonuses."

• Sampler CDs, with cuts from various artists, will take on a whole new dimension. Back in the '80s, sampler CDs were a staple of the "boutique" new age labels, and often outsold CDs from the regular artist roster. In the '90s, there have been tons of dance/house/techno samplers. With dance music's typically limited fidelity (being often based on funky samples and mastering that squashes everything into the top 10 dB of dynamic range), this type of music is an ideal candidate for the MP3 treatment.

• MP3's success is a setback to the drive for better audio quality (24-bit/96 kHz, surround, uncompressed DVD-Audio, etc.). With MP3, super-high fidelity is not part of the equation. Yes, MP3 can sound very good at low-
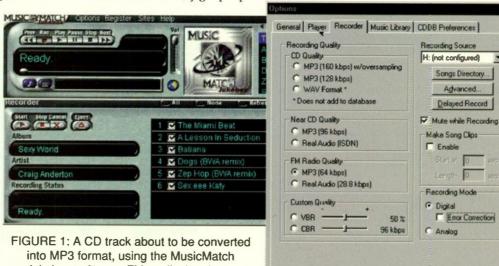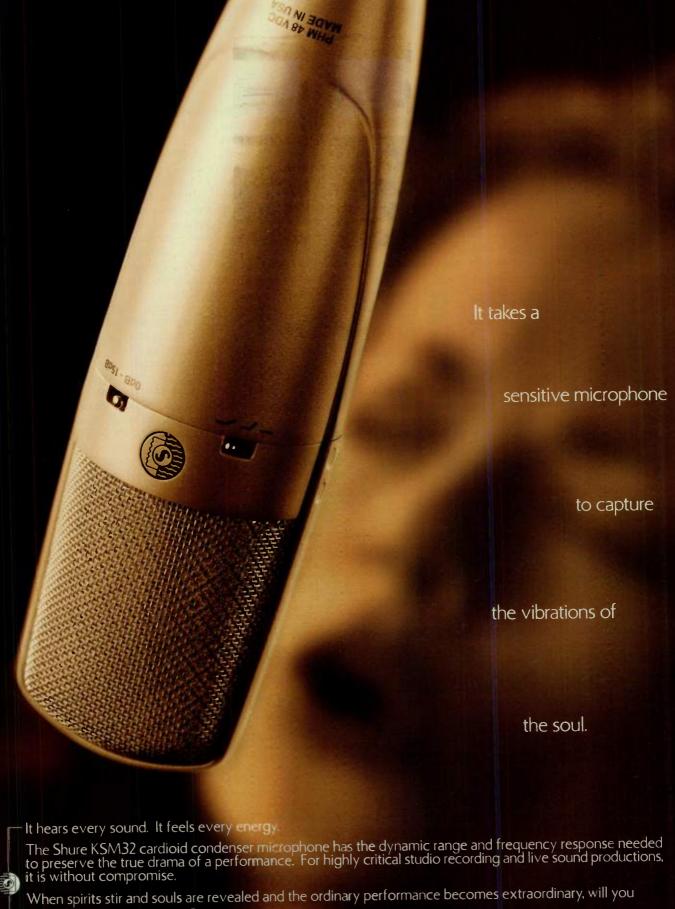
FIGURE 1: A CD track about to be converted into MP3 format, using the MusicMatch jukebox software. FM quality has been selected to conserve file size.

World Radio History

er compression ratios, and there's no doubt that future implementations will sound even better. However, companies expecting to sell the public on upgrading their gear to a better digital audio standard may have a tough time. Consumers are saying that convenience, variety, and customized music collections mean more to them than audio quality.

• Unfortunately, Sony can probably kiss MiniDisc goodbye, at least in the United States. I still love mine for re-mote recording, testing out different song orders, and doing block repeats for songwriting. Although MD remains a far more powerful tool for musicians than the current crop of MP3-based portable playback devices, most consumers just want to listen to music. They'll opt for the smaller size, lower cost, and inherent skip-free performance (no moving parts!) of MP3 playback devices like the Creative Labs Nomad or Diamond Multimedia Rio. Moreover, we're still dealing with first-generation MP3 devices;

there's no reason why future models can't have the same kind of editing bells and whistles as MiniDisc.

• I've been saying for a long time that the Web would bring back the single — a compromised-fidelity teaser for a full-length album — and it looks like MP3 is making that happen. One of the reasons consumers like downloading music is that they can obtain just the song they want to hear and stick it in their MP3 jukebox, rather than have to buy the entire CD. However, if they really like a tune, it could translate into a sold CD.

• Many tunes can be compressed to a small enough size to fit on a high-density floppy disk. Interestingly, Public Enemy released their latest CD in three different formats: physical CD, download (for $8.95), or on Zip disk (!) for $16.95. Because MP3 isn't tied to any particular storage medium, we might see the day when the entire yearly output of a label appears in MP3 format on a single DVD.

• "Transportable" mixes will take on a whole new meaning. It used to be that one of the goals of mixing was to create a mix that could survive being played over anything, from a mono transistor radio to an audiophile's dream system. Now the goal will be to create a mix that can not only survive any delivery medium, but any amount of data compression as well. You'll see more and more engineers searching for the right combination of dynamics processing and EQ to make music sound good, even with high data compression ratios.

Of course, there's always the issue of how artists are going to make money of off all this, but that's a whole other story. Meanwhile, love it or loathe it, data compression is here to stay: MP3 is now the most popular file format on the Web, and there's no doubt it's changing the music world — and that process of change is by no means complete.

Craig Anderton is the author of Home Recording for Musicians *and* Do It Yourself Projects for Guitarist *(available from Amazon.com). He's putting the finishing touches on his next CD, titled* Sexy World, *which will also be released in Sonic Foundry loop library format.*

Audio **Airbrushing**

PHOTO BY JONATHAN POSTAL

All right you audio frauds, I know who you are and I'm comin' to get ya

BY AL KOOPER

So I'm waiting for *Sex And The City* to come on HBO, and therefore am forced to watch the last five minutes of the *Cher Live In Las Vegas* concert. As she begins her encore of "Believe," I notice she is amazingly on pitch in the opening low section. I look to see if she's lip-sync'ing and, if she is, she's damn good at it. Then, as she gets into the next part of the song, I hear her voice glitching all over the place and I recognize the sound of auto-pitch correction.

Awww, that ain't fair!

Believe, indeed! *What can ya believe?* First off, the glitching sounds queer (later I was told that sound is on the actual record(!) — they probably said, "Hey, that's a great gimmick," and just left it on there), and, secondly, I'm not really hearing sharp Cher live. Technology is impeding the live experience. I heard a rumor that Stephen

Stills can no longer hit the high E note in "Judy Blue Eyes" anymore ("It's my *heart*, that's sufferin' "...the note on *heart*.) I'm told the keyboard player has Stephen's voice sampled on that note and he just drops off on "heart" and the sample hits it for him. Awwww, if that rumor is true...that ain't fair! That's not Stephen Stills live!

In this day and age, when people sit in the middle of large audiences to watch DJs spin records, I guess anything can happen, but you gotta ask yourself: *What hath Milli Vannilli wrought?*

I remember watching The Eagles work on their live album in post-production, and I can vouch that *some* of the drums were live and that's probably about it. We are used to live albums being worked on after the fact. It's a pop way of live in the '90s. But now the millennium demands that we're gonna get the alterations *right there on the gig!* I'm getting paranoid. Who *can* ya trust, nowadays? Is that Elton really playing and singing, or is it as synthetic as the hair on his head? Now I have to ponder if that's Janet Jackson's real voice at the same time as I ponder the same thing about various parts of her anatomy.

It's time for a new kind of RAP — the Real Audio Police. All connections have to be checked at concerts to see if they go directly to the mixing console or make a covert sidetrip to a black box. Penalties have to be enforced or confessions have to be made. "Ricky Martin's vocals will be performed through Antares pitch-correctors during

tonight's show" would be a good one. Either that or a meter has to be put on the pitch-corrector to tally up how many times it kicks in and a dollar has to be sent to Amnesty International for each hit. Yeah, that's good. I like that.

Is that Don Henley playing or is that a drum loop? Fine him $100 for each loop used in a song. Another acre could ostensibly be added to Walden Woods after each show. Let's not let these cheaters get away with it — we demand to know the truth.
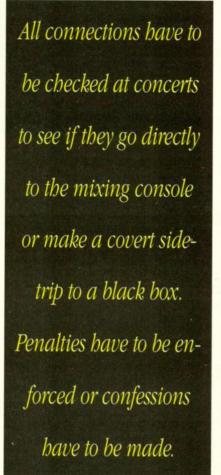
The Stones — now here's a target that's not even moving fast. Pull all the FX and tapes and loops, and it would be like watching guys with no hair and teeth strummin' and gummin' up there. Some of you paid $300+ a ticket for that? I'd pay $97 to see them play *sans* all the chicanery — that could be the actual draw.

So who's cool? I gotta say, love him or hate him, Bob Dylan's up to no trickery live. That controversial voice is immune to FX or tricks. It is what it is, God bless 'em. And his band never really knows *what's* gonna happen up there every night cause Bob is the official poster boy for spontaneity. His show is downright refreshing in the face of all this gadgetry. When Dylan resorts to this sort of thing, Armageddon won't be far behind.

So you readers with the big ears out there — I want you to report in from arenas all over the world and bust these fakers. Join the Real Audio Police today and keep the world safe from audio dishonesty. **EQ**

> *All connections have to be checked at concerts to see if they go directly to the mixing console or make a covert sidetrip to a black box. Penalties have to be enforced or confessions have to be made.*

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Dave **Pensado**

The red-headed megahit mixer lets the cat out of the bag

BY MR. BONZAI

Bonzai: Why did you become an engineer?
I wanted to make records, and my guitar playing had taken me about as far as I was going to go.
Who were your mentors?
Herb Trawick, Phil Benton, Ed Seay, Paul Davis, Larry Turner, and Wolf and Epic.
What was your first big commercial success?

As a mixer, "Do Me Baby-Remix," BBD.
How did you become known primarily as a mixer?
Our industry likes you to be known for *one* thing, and that is what I enjoy the most.
What specialty gear do you own?
Forat F-16, Gates Sta-Level, and a Shimano Calcutta 400 with 15# line.
What console do you prefer?
SSL J9000, serial #4940.
Could you give me a step-by-step description of how you do a mix?
I first like to hear a rough mix. There is always something sacred about the rough — that gives me a direction. A lot of times there will be an essential effect that is unduplicatable with all our expensive gear, so I will have an assistant run to Guitar Center and pick up that piece of gear. So, over the years, I have accumulated quite a lot of stuff.

I then do a very quick mix, or Dylan, my assistant, does one, so I can further get ideas. At this point, I start work-

Suspect: Dave Pensado

Ancestry: Spanish, descendant of Velasquez

Occupation: Mixer

Birthplace: Tampa, Florida

Residence: Woodland Hills, California

E-mail: fdpen@ix.netcom.com

Diet: Four main studio food groups: caffeine, burgers, donuts, and beer

Identifying Marks: Hair (a lot)

Pet Peeve: "A tech who tells me it's operator error."

Credits: K-Ci & JoJo, "All My Life" (tied Beatles for biggest jump to #1 on charts); Bell, Biv, DeVoe remixes, "Do Me Baby" and "Thought It Was Me"; Ice Cube & Krayzie Bone, "Sunshine"; Brian McKnight, most of new album; Christina Aguilera, several.

Notes: Suspect was photographed at The Enterprise in L.A. in a custom room primarily occupied by Mr. Pensado and his toys.

ing on the drums and bass. Then I'll fill in the rhythm tracks, and, finally, the vocals.
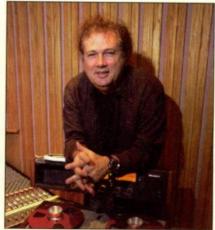
If the drums are bangin' on tape, I'll use a little dbx 160X and API 550. If they need help beyond that, I'll trigger from the Forat F-16. I use a lot of Pro Tools plug-ins as outboard gear. I love McDSP's Filter Bank on just about anything, and their new Compressor Bank is awesome. I also use the Waves Renaissance and L1 compressors, as well as their de-esser. I like to run my bass through the Distressor and an Alembic SF2.

The vocal chain is usually an NTI EQ3 and McDSP compressor on the backgrounds, and sometimes the Waves stuff. On leads, I split up the vocals to at least two tracks and run one side through Filter Bank and L1 and the other side through a Gates Sta-Level compressor (circa 1956) and GML 8200 or Neve 1073. I blend the tracks as needed on every syllable to make it sound good.

Along the way, I'm sure to use a few of my old favorite pieces of gear: Ensoniq DP-4, Behringer Dualfex II and Edison (thanks Bob), Boss EH50 and AW1, Moog Parametric EQ, Boss SDE 330 (thanks Dex), Korg A1, and Rocksonics MB-5x compressor.

At this point, I start getting all the balances between the tracks right. And usually at this point I'm convinced I will never work again due to the horrible mess I've created! But I keep tweaking, dump it on to some GP9, and it usually comes out OK.

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

What's wrong with the music industry?
The huge gap in taste between the public and the industry elite. Also, there are not enough African Americans in executive positions that count.

What music would you like played at your funeral?
"Sitting on the Dock of the Bay."

What do you listen to while you're driving?

CIRCLE 98 ON FREE INFO CARD

Old funk like Cameo, Brick, and Parliment.

If you could go back in time before the birth of recording, what would you like to hear?
The sound of a T-Rex crunching mammals like tater tots.

Who were your heroes when you were getting started?
Quincy Jones, EWF, Cameo (Larry Blackmon, Charlie Singleton, and Kevin Kendrick).

Who do you respect and admire today?

Herb Trawick, Ron Fair, Keith Andes, and Rhett Lawrence.

Who are some other mixers you like?
Jon Gass, Dexter Simmons, Tony Maserati, Kevin Davis, Rob Chicarelli, Dave Reitzas, Bob Brockman, and Jean Marie Horvatz.

What is your strangest characteristic as a human being?
I've had some things for 30 years. I keep stuff (including friends) forever.

Do you know any interesting business tricks?

Try to mix for talented people, and trust them.

What was your most ridiculous experience in a recording studio?
It wasn't exactly mine, but a buddy of mine came to an early session to finish engineering the project. The problem was he was still drunk from the previous night of partying. Well, after putting on the 2-inch masters and watching them whirl round and round, he got sick and barfed all over the tape and the machines. It gets worse — he then tried to fast forward the tapes off. Well, I'll leave the rest to your imagination.

Who is the most amazing artist you've worked with?
Brian McKnight.

Have you ever witnessed a miracle?
Sure, at church.

What is the biggest mistake of your life?
Waiting so long to engineer. I just never thought I'd be good enough for anyone to want to pay me for this.

Any advice for getting a good start as a mixer?
Go to the record store. You can learn more there than at recording school. It's much easier for me to show a kid how to get a great sound than to teach him what a great sound is.

Has anyone ever disliked one of your mixes?
Sure, if that doesn't happen once in a while you're not growing and trying new things.

What would you like Santa to bring you this year?
I've got a great friend I'd like to see receive a much needed organ transplant. **EQ**

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Doin' It **His Way**

New artist Owsley shopped his project studio-created finished album to record companies — and got signed virtually as is

BY HOWARD MASSEY

If the project studio industry ever needs a poster boy, Owsley is their man. In the tradition of self-assured artists such as Prince [the-artist-who-used-to-be-but-doesn't-want-to-be-known-as-anybody-anymore], Beck, and Lenny Kravitz, the youthful guitarist/singer/songwriter's eponymously-titled debut CD was recorded and produced almost entirely in his modest project studio in Anniston, Alabama (sardonically called "The Outhouse"). The album was then shopped to the major labels as a complete, finished product — a risky strategy indeed in this era of specialist producers. A few months ago, Giant Records picked up the gauntlet and released the album, with the sole caveat that mixing expert Tom Lord-Alge brought in to sprinkle his fairy dust on a few key tracks. The result is a strong collection of surprisingly mature, hook-laden power-pop.

To call Owsley a recording enthusiast would be a supreme understatement. An almost obsessive student of production techniques, as well as an avid Gear Slut, his love for the studio is unequivocal. "Man, you don't know how excited I am to be talking to EQ magazine," he told us. "As far as I'm concerned, this is the most important interview I'm ever going to do!" Hey, as much as we sincerely hope he's wrong, ya gotta love the guy's attitude...

EQ: The story behind your album makes a strong statement about artistic control.
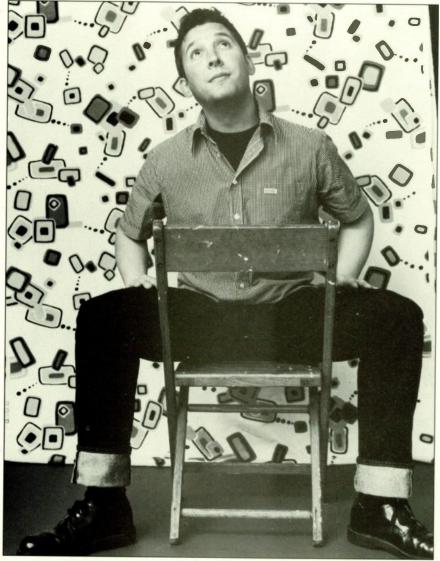
Owsley: Well, I'm glad my hard-headedness didn't screw me in the end. Every time I would get disheartened, I would just go back, listen to the tapes, and think, "There ain't nothing wrong with this. This is printed correctly...these performances are fine." I spent two years making this thing, and it's exactly what I want, so why would I want to redo it all? I was in a group in the early '90s called The Semantics, signed to Geffen, and that happened to us. We had incredible demos, but our A&R guy said, go back and remake the record. However, we never got it any better. We just chased the demo the entire time. So I said to myself, next time I'm going to make the record I want and then get an attorney to shop it already finished.

Plus, I wanted to prove to the record industry that I was capable of producing myself — that I didn't need a big-name producer to make a good record.

Presumably, it was the record company's idea to bring in Tom Lord-Alge to do some remixes.

Yes, the record company asked me whom I wanted to use. I thought about it a long time, and it came down to Bob Clearmountain and Tom Lord-Alge. I love Clearmountain's work, but my stuff could tend to be on the pretty side. Therefore, the more I thought about it, the more I thought Tom was the perfect person for the job. So, I went to meet him.

PHOTO BY CHRISTY BUSH

World Radio History

Everybody told me beforehand, "Oh, Owsley, you're a control freak; you'll hate Tom Lord-Alge." However, I loved him, we got along just great; our personalities just meshed. The first tune he mixed was "Oh No The Radio." He made the drums sound so incredible, I just said, "Hey, I'm out of here, I'm going to go shopping." He said, "No, stay here, it'll go faster." However, I was so confident after hearing the drum sound, what he turned them into, I was just blown away, I was ready to leave. It was one of the most incredible musical experiences I've ever had — working with Tom and getting to sit right beside him. Not that I helped that much — I mean, the guy's incredible, he knew just what to do.

So why didn't he mix the whole album?
To be honest, it came down to money. If money were no object, he would have done the whole thing. My thinking was, if this record does incredibly well, we'll have maybe six singles, so we picked all the tracks we thought had any chance of being a single and we gave them to Tom. Working with my favorite engineer (J. R. McNeeley) in Nashville, I felt I could get as close as I needed to get with the other tracks. In retrospect, though, I wish we'd done the whole thing with Tom. Not that I'm displeased with the other mixes, but I think next time I'll probably do the whole thing with Tom. **There is a qualitative difference between the tracks he mixed and those he didn't...**
Yeah, he's ballsy when it comes to EQ, when he grabs that treble knob... Some people would be scared to turn it that far, but Tom's not. All of his mixes sound aggressive, even on the slower tunes. There's a real 3D aspect to his mixes; they're really deep. Some people give him hell for over-compressing things, but Tom is a very smart guy, and all his compression is very fast and all his ratios are very low.
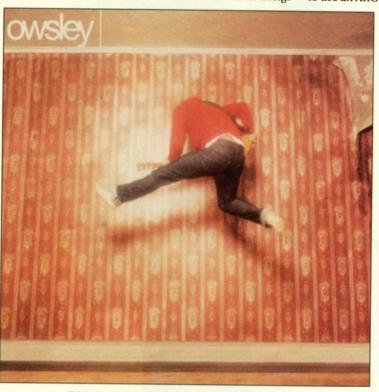
What's the coolest trick you learned from him?
Oh, man, I learned a ton of stuff from him. As I said, he compresses a lot, but he compresses at very low ratios. All his compressors are set at 1.5:1, not even 4:1 like you'd find on an 1176. He doesn't like 1176's, or LA-2A's, or Fairchilds — says they're too slow. He uses the dbx 160X.

His snare drum sounds are certainly very compressed, very fat.
That snare drum came up in five different places! He'd use gates to separate parts of the sound. For example, he'd have the very attack of the snare drum going through one compressor, the middle part of the sound through another, and the end of the sound through yet another — plus the ambient mics going through their own compressors. But he sets the compressor release times as fast as the damn things

FOR THE RECORD: Owsley's debut album.

would go — that's why he likes to use fast compressors. Even the very fastest setting of the 1176 is too slow for him. This way, you really don't hear the pumping of the compressors, the way you do on many British recordings, for example.

Another thing he did that I'd never seen done before was using tape flange [on the end of "Oh No The Radio"]. He routed the entire mix to a Studer 2-track, flanged it, and then flew it back to the multitrack. That way we always had the flange available on a couple of faders — you want more, just push the faders up some more. It was glorious!

What were the main pieces of gear you used for tracking at The Outhouse?
The main recorder was a Studer A80 16-track — I just love it. I also used a couple of TASCAM DA-88's. Most of the songs that Tom mixed were recorded almost entirely on the A80. I tried to keep the overdubbing down to a minimum because I think the song comes through better. When you have so much information coming across to a listener, it's hard to keep up, to take it all in. The listener may miss the point of the song if you get too crazy with the overdubs.

The board was a Mackie 24•8, but I didn't use its preamps. Instead, I used outboard APIs and Neve 1073's. I was going to use an AKG C12 VER reissue on my vocals, but it just sounded too nice, so we ended up using a [Shure] Beta 57. Just Beta 57 into the 1073, then into a blackface 1176 with all four buttons pressed in — it was instantly edgy. Tom loved the vocal sound — all he had to do was clean up the tracks a little, taking out the breaths and lip-smacking.

We were very limited for tracks, so we did a lot of bouncing down. For instance, there's a triple-tracked slide guitar solo on "Coming Up Roses," and we had to bounce that down to one track. When Tom pulled it up on one fader, he busted out laughing.
It's a great sound, very reminiscent of [Boston's] Tom Scholz.
All it was was a wah-wah pedal pushed halfway down in that sweet spot where it sounds very out of phase. Actually, I was really trying to go for a Brian May thing, but I can see where you hear the Scholz influence. A lot of the sounds were dictated by the lack of tracks — for example, on most of the drum tracks there's only a mono overhead. But it kicks ass.

I would hope my record proves that there ain't any rules. You can do anything as long as it sounds good, as long as it makes you laugh. It's better to use fewer tracks and to commit sooner. It makes you stoke a little harder, but that's a good thing. **EQ**

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CIRCLE 72 ON FREE INFO CARD

The Right Tools

A look inside the world's first commercial Pro Tools-based studio

BY DAVID FRANGIONI

John Haggins and Doc Wiley are true visionaries. They set out to build the world's greatest commercial Pro Tools-based studio and they didn't stop until it was done. In fact, their burning desire for this room to come to fruition was so great that they did it in less than two months — start to finish. Read on to find out how we approached the technical goals of the room (acoustics and room design are another article!).

John Haggins made his mark on the world of technology many years ago when he successfully marketed the C-Plus programming language. His success in marketing that product culminated in its sale to Peter Norton for about $20 million. John has always had a passion for music and technology, and this year decided to build a world-class studio to serve the fast-growing music community of Miami. Enter Doc Wiley. Doc was based out of South Beach studios and worked with the likes of U2, Ricky Martin, and many others. Doc and John shared the same enthusiasm for the newest offerings from Digidesign; Pro Tools|24 Mix Plus and Pro Control. They figured, why not build our studio around this technology? If it is done right, they could offer virtually any service imaginable to their clients — the power of the system is there. What about if they needed lots of inputs for keyboards or drum tracks? Well, then why not add a Mackie Digital 8•Bus?

Ironically, this gear is most often associated with project studios, not commercial studios. That's just the point. Most project studios need a tuned, properly setup mixing and mastering environment for their final product. It's cool to be able to record and edit everything at your own studio, but then where to you go? To DigiNote, Doc and John hope. That's exactly why they decided to offer full transfer capabilities in case a client wants to enter or leave the Pro Tools world and switch to another format. But which formats will be offered, how should the Pro Tools system be configured, and how does this all get integrated? Enter Audio One.

John, Doc, and I sat down and put a layout together for what the desired goals of the system had to be. How much mixing power did they want (that would determine DSP needs), how many tracks of transfer were necessary (three rounds of eight, or one straight transfer of all 24 tracks), what formats were to be offered, etc.? The heart of the system is Pro Tools and therefore that needed to be sorted out perfectly. We decided that a 32-channel Pro Control was a must since Doc would be navigating the system all day long and the Pro Control offered a user interface that really made the entire studio a reality. Had a mouse been the only way of operating Pro Tools, DigiNote might have been different (much different).

That said, DSP was the next consideration. Each Pro Tools Mix card offers a certain amount of real-time DSP. How much did Doc need? He settled on six Mix Cards and two DSP farms. This offered an incredibly powerful system with enough horsepower to effectively mix up to 64 audio tracks with lots of effects processing (EQ, compression, reverb, delay, and more). As many of you know, Pro Tools uses DSP cards to do the actual processing of all real-time effects. The more cards that you have, the more DSP is available. Up to ten cards total can be present in a system, but anything over five is truly awesome for serious mixing, depending on the amount of audio tracks that require processing.
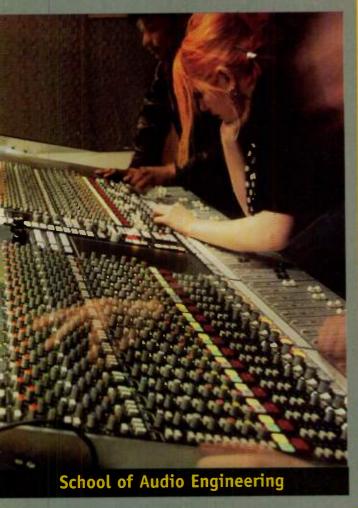
Now that the DSP needs were settled, the actual plug-ins that were going to be installed into the system needed to be determined. In typical fashion, Doc and John decided to add virtually every plug-in available for Pro Tools. They figured that if DigiNote were to be a world-class Pro Tools studio, then they needed all the possible plug-ins available. Waves, DUY, Wave Mechanics, Focusrite, TC Electronic, Lexicon, and many more found their way into the prospectus.

Upon completing the Pro Tools system requirements, the time had come to focus the direction of the transfer part of the facility. What if a client wanted to take 24 tracks of audio and go from one format to another? Especially Pro Tools to any other

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format? It was decided that there would be 24 tracks of TASCAM format (one DA-98 and two DA3-8 machines), 24 tracks of ADAT (three XT20 machines with a BRC), and 2-inch 24-track (Sony APR-24). At this point, the considerations were how was the audio going to digitally go from one format to another, and how does the 2-inch machine retain all of its warmth and character? The Apogee AD8000 was found to be the answer. It was the only interface that offered the necessary digital format conversion, sounded really great, *and* was fully upgradeable. All of these features were necessary due to the fact that four interfaces were going to be used! It would have been more cumbersome if multiple interfaces had to be used for each of the three goals. The AD8000 would handle digital conversion among Pro Tools, ADAT, and TASCAM protocols. It would also offer the A/D-D/A that was required to and from the 24-track machine. As an added benefit, it even allowed multiple format conversions to be done at the same time. That way a client could transfer 24 tracks of ADAT to both Pro Tools and DA-88 in one pass.

The ability to transfer tracks digitally and with high fidelity was only one consideration. A stable, reliable clock needed to be the foundation of the entire studio. An Aardvark AardSync II was setup as the word clock master for the entire studio. It resolved (and filtered) to a house sync, blackburst generator. The Sony APR-24 also resolved to the blackburst generator. All digital devices resolved to the AardSync, and, therefore, all "machines" were sample-accurate and properly resolved. Doc would never have to deal with the associated clocking problems of what device was a slave and what device was a master. All devices were slaves to the AardSync. Should another analog deck be installed (Beta, a second 24-track, etc.), it would be plug-and-play to lock it with the entire studio.

As a side note, the digital machines sounded a bit (no pun intended) better resolving to the AardSync than using their internal crystals. Other outboard gear was assembled based on Doc's past experience, such as a Sony C800 microphone, Focusrite mic pres, and lots more. Now that the equipment was chosen, a blueprint of the integration needed to be solidified, created, and installed. Let the good times roll.
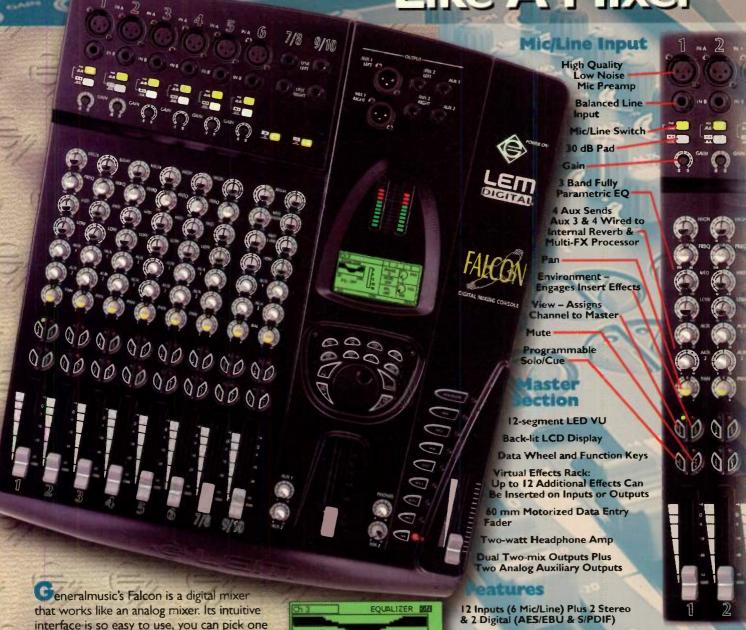
Upon dictating the physical panel requirements, Doc and John left most of the wiring-parts decisions up to me with approval necessary before actually cutting the cable. I chose Monster Pro Link multi-pair cable for the analog cabling, Monster Studio Pro digital AES cable and S/PDIF cable, and Monster TT patch cable. They wanted the best, so I used the Monster Cable. It really does sound better and reject noise better than any other cable that I have used. We wired all analog points to Signal Transport TT96 patchbays and used Neutrik connectors to complete the signal flow. All points were custom-made and normalled to Doc's specifications. We worked non-stop for three weeks to bring the entire wiring installation from start to finish. At the end of the installation, Doc and John were very pleased, and the room sounded great (Ross Alexander did a great job designing the room, as did Danny Diaz putting it all together).

David Frangioni has built studios for Aerosmith, Desmond Child, Kike Santander, Olivia Newton-John, and hundreds more. He can be reached at www.audio-one.com.

A Digital Mixer That Works Like A Mixer

Mic/Line Input

- High Quality Low Noise Mic Preamp
- Balanced Line Input
- Mic/Line Switch
- 30 dB Pad
- Gain
- 3 Band Fully Parametric EQ
- 4 Aux Sends Aux 3 & 4 Wired to Internal Reverb & Multi-FX Processor
- Pan
- Environment – Engages Insert Effects
- View – Assigns Channel to Master
- Mute
- Programmable Solo/Cue

Master Section

- 12-segment LED VU
- Back-lit LCD Display
- Data Wheel and Function Keys
- Virtual Effects Rack: Up to 12 Additional Effects Can Be Inserted on Inputs or Outputs
- 60 mm Motorized Data Entry Fader
- Two-watt Headphone Amp
- Dual Two-mix Outputs Plus Two Analog Auxiliary Outputs

Features

- 12 Inputs (6 Mic/Line) Plus 2 Stereo & 2 Digital (AES/EBU & S/PDIF)
- RED 208 Proprietary DSP w/ 40-bit Internal Processing
- 20-bit ADs & DAs w/ 64x Oversampling
- Internal Reverb & Multi FX on Auxes, Plus Up to 12 Additional Insert Effects, Included Standard
- 128 Snap Shots, EQ & Effects Library
- AES/EBU & S/PDIF Ins & Outs
- Dynamic Automation From Channel Knobs
- Optional AES/EBU & ADAT Interface cards
- $1,495 MSRP

Generalmusic's Falcon is a digital mixer that works like an analog mixer. Its intuitive interface is so easy to use, you can pick one up on the way to your next gig. If you're comfortable with an analog mixer, the Falcon will be a snap. And, at a list price of only $1,495, the Falcon provides processing power that rivals digital mixers costing several times more. Equally at home in live sound and recording applications, the feature-packed Falcon is a real value, delivering the performance you would expect from Generalmusic, the industry's most innovative manufacturer.

Generalmusic Corp.
1164 Tower Lane, Bensenville, IL 60106
Tel: (800) 323-0280 • Fax: (630) 766-8281

www.generalmusic.com

Plug-In **Power**

What plug-ins are and how they are helping some high-profile projects get the job done

BY RICH TOZZOLI

Webster's Dictionary defines a "plug-in" as…oh, wait, it's not there. Anyway, a plug-in is simply a software module that operates within its host computer's audio hardware. This article will focus specifically on the Digidesign family of plug-ins, what they are, where they came from, and how professionals are using them every day. In later articles, we will examine the many other companies that feature plug-ins in their software, but I wanted to focus on one company at a time.

Ed Gray, manager of Digidesign's development program, recently discussed the origins of plug-ins and how they apply to Pro Tools. "Plug-ins were developed based on our TDM (Time Domain Multiplexing) bus, which is dedicated hardware that has no purpose other than to support large mixes and plug-in processing. Digidesign's developer programs offer an API (Application Programming Interface) that allows companies outside of Digidesign to use their expertise to develop their own feature sound. We provide them sample code, documentation, and engineering support, enabling them to create the plug-in."

Gray continues, "Dave Froker began the Developer program by initiating contact with companies such as Waves, Focusrite, Lexicon, and TC Electronic, among others." These companies, using their years of individual knowledge and experience, developed the first group of TDM software-based plug-ins. "Now companies approach us at a rate of two to three per day, and we have to be very selective. We still love hearing from developers with good ideas though," Gray comments. Digidesign now has a full staff of engineers who does nothing but support all the third-party developers they work with.
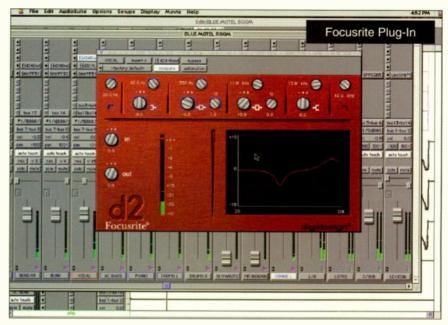
"Our plug-ins feature full dynamic, frame-accurate automation," comments Digidesign product manager Andy Cook. "Due to the wide variety of options and combinations possible in Pro Tools, audio engineers have more power to be creative on the production side. Instant total recall makes it easy to experiment with different plug-ins and to try taking your audio in new directions. People in this industry have some wild ideas, and Pro Tools allows them to harness that creative energy."
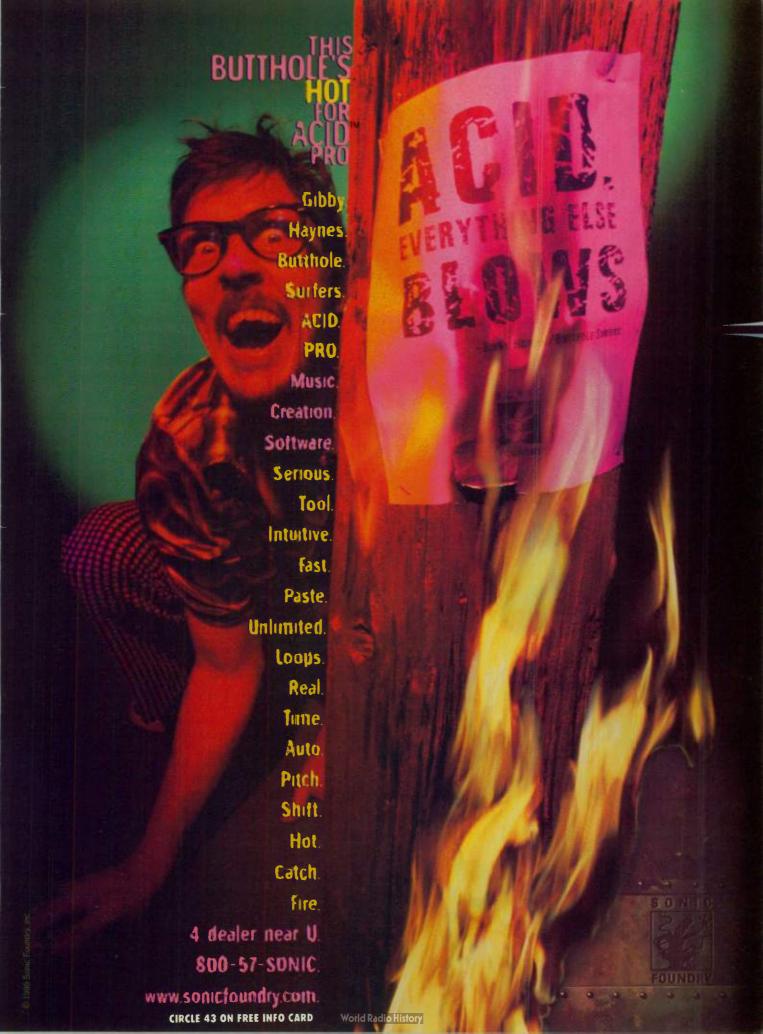
What makes plug-ins so powerful is the ability, with just the click of a mouse, to call up the right tool for your audio job — with no patch cables or external hardware. You can simply click on a Focusrite Red Range equalizer to fatten up that kick drum, or maybe a Lexicon plate reverb for the guitarist, or an Antares Auto-Tuner to fix that pitchy vocal (I won't tell you whom I've fixed this way). All the while on your stereo output bus you've applied a TC Electronic Master X mastering plug-in to add some top-end sheen and an Drawmer Dynamics compressor to tighten the thing up a bit. If you want to check your phase, balance, and levels just bring up Metric Halo's Spectra Foo metering software.
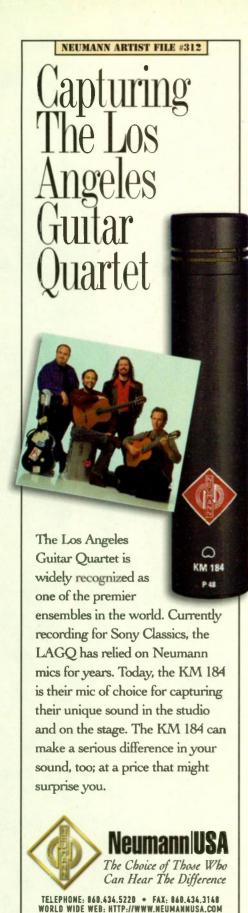
Regarding plug-in automation, almost every parameter can be programmed for total recall. You simply click on the automation button at the top of the plug-in and select the parameters you want to work with. Those same parameters will then be highlighted, and ready for real-time tweaking. In addition, graphic editing of all automation data is available for fine-tuning a quick reverb setting or an EQ sweep. Add to this libraries of presets (different for each manufacturer's product), and the scope of your engineering and production capabilities have just expanded greatly. I have found without question that, by using Pro Tools|24 and its arsenal of plug-ins, I have done mixes that I could not have done on any console and rack of outboard gear, regardless of price. That is not to say they don't have their place in our biz, they certainly do, but software-based mix systems with plug-ins are simply an extremely effective tool for today's production needs.

Digidesign features two separate plug-in applications: TDM and AudioSuite. TDM was developed from technology used in the telecommunications industry. It allows the sending of multiple streams of information down one path. Cook comments, "We advanced

Kind of Loud SmartPan Plus

SmartPan Pro

Lexicon Plug-In

TDM for the needs of audio signal processing. It all comes down to power on demand, and TDM has become the professional standard due to the unlimited power it provides." AudioSuite is internal-file-based processing. Cook continues, "Some aspects of audio production are more suited for file-based production, like normalization and/or time compression/expansion." AudioSuite also allows people with non-TDM hardware to have additional processing capabilities.

Plug-ins have expanded beyond the traditional and fundamental signal processing options to offer a new thread of creativity. Amp Farm by Line 6 offers a palette of great vintage and new guitar amps to apply to your sound. DUY's DSPider, Wave Mechanics Sound-Blender, and Digidesign's Bruno/Reso and SoundReplacer are just some of the plug-ins that offer functionality not possible without software. For surround sound, you have Dolby's Surround Tools or Kind Of Loud's Smart Pan Pro, which easily handles 5.1, 7.1, and LCRS mixes.

Award-winning sound designer Scott Martin Gershin of Soundelux comments, "I think of plug-ins as a hy-

brid between consoles and outboard gear. I can customize how I'm going to work not only on a track-by-track basis, but on a sound-by-sound basis, with the added feature of being able to recall all of those settings instantaneously. I could have five different types of EQ within a session or be able to create custom complex DSP chains that are automatable on each track, each with their own settings to accomplish a specific design or sound effect."

Producer/engineer David Tickle, who just finished mixing Sting's *Ten Somner's Tales* in 5.1, comments, "Probably 80 percent of my mix stays within Pro Tools, using plug-ins such as gates, EQs, compressors, and reverbs. I have a rack of outboard gear that I still use, but most of my work is in the computer. For Sting's record, we used Kind of Loud's surround plug-in, which was very effective and unusually phase coherent."

Producer/engineer Andrew Scheps said he thinks of plug-ins like guitar pedals. "I link them together, sometimes five or six at a time. I will take a chorus into a reverb into a delay. I also like to compress and de-ess the sends to my reverbs, which is easy using Pro Tools." Engineer/Producer Will Alexander is using plug-ins to remaster Keith Emerson's back catalog and new releases. "I'm using Arboretum's Ray Gun and Ionizer for noise reduction, removing analog tape hiss from recordings originally mixed to 1/2- and 1/4-inch analog tape. I then process with Waves Q10 equalizer, then get punchy dynamics with the L1 Ultramaximizer. This also was very effective for delivering high-fidelity digital downloads at LiquidAudio.com, which I am posting much of Keith's solo catalog to. Plug-ins simply give me fantastic results."   **EQ**

Babalooo! **Where Are You?**

Among the chaos of Cuba, Abdala Studios stands ready to make music

BY C. REEDER

Congas and claves punctuate the air with a pulsating salsa beat when the door swings open to Studio 1, mixing with the rich smoke of a Cohiba cigar and vapors from my *café Cubano*, a coffee substance, not unlike car oil, that my American friends affectionately call legalized speed. I am in Cuba, sitting in the lounge of the recording studio, Abdala, chatting with Katia and Sandra. Katia is an attorney for the studio and Sandra is a client liaison and, much to my relief, has a B.A. in English.

People on the street look at me with bemused horror when I try to speak what I can only call Spanglish. Things like, "Bueno, mi gusta...¿Old Havana?" I sputter as I jump into a taxi, while waving and pointing to the place on the map where I want to go. Gesturing and facial expressions can only get you so far. I wasn't sure I wanted to test my less than stellar foreign language skills on humorless looking guys with guns in army fatigues standing in these little wooden boxes [one sees] all over the place.

English is not the first language in Cuba.

In *Habana Vieja* (Old Havana), a big tourist stop, pot-holed streets are jammed with scooters, Chinese bicycles, Russian Ladas, new Korean cars, pedestrians from cruise ships, and huge American-made metal relics from the '50s in various stages of disrepair, spewing miasmic fumes from exhaust pipes.

Nuemáticos, hombres who fish from inner tubes, balance tires on their heads as they march down the Malécon and disappear over the historic seawall. This is the Cuban version of fast food.

Only 200 physical miles from my house, but light years from the world I know, I strain to find some common ground in this enigma called Cuba. My desire for wanting to be here is all mixed up with a love of all kinds of Latin music and images of myself as a very little girl in a living room with her family in Columbus, Ohio watching the colorful Ricky Ricardo sing "Babalooo" out of a little black-and-white television box.

The common ground is, of course, ¡Música!...and at Abdala Studios, music rules.

Abdala Studios is the brainchild of the gifted Cuban musician and songwriter Silvio Rodriguez, and built with the support of the government-owned CIMEX Corporation. With international interest in a cha-cha-cha over Cuban music, et al., the time is right. *¡Ya es tiempo!*

Opened for a year and nestled in the Miramar District of Havana, Abdala Studios is over 16,000 square feet, with three recording rooms, one large enough for an orchestra, a MIDI room, mastering room, and state-of-the-art SSLs in a compound totally self contained (see sidebar).

One can get delicious *cubano* food from short-order cooks and drinks from the bar, like a Cristal beer or *Havana Club y Coca cola*. CDs of the artists who have recorded there are for sale, or you can take home studio paraphernalia like hats, t-shirts, etc. The U.S. dollar is the currency.

Abdala is a short drive from sparkling five-star hotels on the water, through neighborhoods of crumbling apartment buildings and dilapidated Andalusian-era style mansions. Along the way, I pass by the lonely looking Russian Embassy with all 30 concrete stories hovering over its neighbors and sending off morgue-like vibes.

The U.S. Treasury Department prohibits U.S. citizens from spending mon-

GOING SOUTH: Abdala Studios provides a self-contained musical oasis in the hear of Cuba.

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HOW TO CONTACT DIGIDESIGN

World Radio History

I WANT IT ALL,
AND I WANT IT NOW!

Pro Tools 5.0 promises to let studio engineers put all their music and sound elements in one package

BY ROGER NICHOLS

Pro Tools has changed the way we make records forever — the same way multitrack recording did 40 years ago. There are hit records that were completely recorded and mixed in Pro Tools. I just finished a project that took 15 months in the studio, which is half of what it would have taken without Pro Tools.

I have noticed that the brand name "Pro Tools" is changing from a noun to verb — just like the word Xerox changed decades ago. "That vocal will be great, we'll just Pro-Tools it." "Yeah, we can just Pro-Tools that guitar to get it a little tighter with the keyboard." Pro Tools has become the Super Glue of the studio. You can stick anything to anything else and make it sound like it was played that way.

I want it all, and I want it now. This is the battle cry of today's audio professional. We want to go from concept all the way to distribution, and we want everything within our grasp to make it happen. What's more, we want the process to be as effortless and streamlined as possible.

The cry will be answered soon, as Digidesign gets set to release its much-anticipated Pro Tools version 5.0 software for MacOS and Windows.

As the creative roles of audio professionals begin to blur closer together, Digidesign understands the need to put all elements of music and sound production into one transparent package. The once-linear process of recording, editing, sequencing, processing, mixing, mastering, and distribution is becoming as random-access as hard-disk recording. Nothing is done in a straight line anymore. In order to make this process smooth, our audio production *tools* need to evolve into an audio production *tool*.

With Pro Tools 5.0, Digidesign is fully embracing this single-system philosophy. No new hardware is required to benefit from this new release. Instead, the company has put its efforts into boosting the features in Pro Tools software, while also offering new optional I/O and video peripherals. As an added bonus, Pro Tools version 5.0 will be free for anyone who purchased a Pro Tools|24 or Pro Tools|24 Mix TDM-based system on or after April 19, 1999.

Perhaps the most notable news regarding v5.0 is that the industry's leading digital audio workstation manufacturer has integrated professional MIDI sequencing and editing directly into Pro Tools. Version 5.0 will include an exciting set of powerful MIDI features, reducing the need to switch to a separate sequencer. At the same time, the application will maintain a sense

WAVES
GOES FOR THE GOLD

Way back in 1993, Waves made the first 3rd-party audio Plug-In for Sound Designer II — the Q10 Parametric Equalizer. They've never stopped working to provide the best audio tools for desktop workstations.

The Waves Gold bundle is simply the best buy for Digidesign TDM power users, with 15 audio processors. Not merely a collection of 15 Plug-Ins, the Gold bundle — indeed, the entire Waves product line — serves as a massive collection of processes and processors, both essential tools, and creative effects alike, but with the highest attention to quality throughout.

Over 300 processes are available quickly and easily through Waves "component" Plug-Ins; each Plug-In breaks into smaller pieces for optimal DSP usage, so DSP power is not wasted. Set-up libraries and factory presets provide a huge audio resource pool to streamline the user's production process. For instance if you need a de-popper for a vocal track, what better than to simply load the C1 compressor and select De-Popper from the Load menu? Voila It's de-popped. Have a DAT with pre-emphasis and can't make it play back properly? Insert the Q1O and load the CD/DAT de-emphasis setup and you're done.
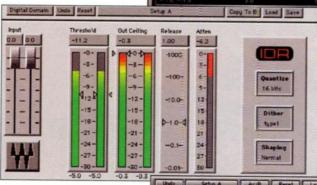
It is simply staggering to look at the list of tools these processors provide. Far more than what you might guess

you'd get for 15 Plug-Ins. For example: mastering limiters, IDR dithering, deessers, soft high-frequency limiters, bass compressors, de-poppers, mastering EQs, Plateau and tilt filters, SuperNotches, pre/de-emphasis

curves, real-room reverb emulation, stereo-image manipulation and mastering, MS processing, 68-band real-time analysis, classic tape flanging,

formant corrected harmony generation, and much more.

The contents of the Gold Bundle are: Q10—10-band ParaGraphic Equalizer; L1 Ultramaximizer mastering limiter with IDR dither; C1—Parametric Compander; S1— Stereo Imager; TrueVerb virtual room simulator/reverb; PAZ real-time analyzer; Renaissance Compressor (TEC-award nominee); PS22 Pseudostereo tool; MaxxBass patented bass exciter; DeEsser; AudioTrack (4-band EQ, comp/gate), MetaFlanger–classic tape flanging and more; UltraPitch 6-voice formant corrected pitch shifting; MondoMod AM, FM, and Stereo Rotation modulator; and SuperTap – 6-voice, 6-second mulititap delay.

Distinguished users of the Waves processors include Douglas Murray, Oscar-winning Sound effects editor for *The English Patient*; Nine Inch Nails; LucasArts; Interplay; Eric Schilling; Bob Ludwig; Rhett Lawrence; Joe Franco; Mick Guzausky; Mutato Muzika; Phil Ramone; Bela Fleck; Dave Frangioni; Bil Vorndick; Juan Patino; and many more.

Demos of almost all Plug-Ins and processors, plus lists of dealers worldwide, are all on the Waves Website. Find out what the Masters of DAW have known for some time: Waves processors are simply essential tools for the pro.

- -

Contact: Waves
Tel: 423-689-5395
Fax: 423-688-4260
Web site: www.waves.com.

CM AUTOMATION
KEEPS UP

CM Automation concerns itself with making digital and analog technology comfortably productive, ruggedly affordable, and as flexible as the software it integrates. The company came into being because of the desire to integrate science, physics, and the love of music. The goal is a clean approach to the integration of traditional, time proven audio production tools with the rapidly maturing PC-based digital audio technology.

The results are the Motor Mix (TM) Digital Mixer Worksurface, the PM 64 Automated Router/Level Controller, and the PM216 Automated Router/Mixer.

The Motor Mix Digital Mixer Worksurface started out to be a universal controller for digital audio mixing software. While the software itself has grown quite powerful, it seemed to require multiple power-hungry space-hogging video monitors and the repetitive motion monotony of a mouse. Discreet controls for analog mixing have a respected heritage.

They provide elegant access to specific audio parameters. Software companies have done a great job of capturing the functions of audio mixing and overcome many of the limitations of analog. Digidesign stands out for its development of the ProTools digital audio production environment. It encompasses a wide variety of tasks in the compact world of software and the CPU. CM Automation created Motor Mix to provide a compact reliable worksurface that would be the obvious choice over a

mouse & keyboard setup.

ProTools users discovered the graceful Motor Mix Worksurface. Their response was pleasantly surprising. A single Motor Mix unit is quite adequate to operate an infinite number of audio channels. ProTools users were taking them three at a time in order to access banks of 24 tracks at a time (4 units/32 track banks are possible). The solution proves to be a beautiful working environment in roughly the same space as a keyboard and mouse setup. It provides real-time control of all tracks and their parameters.

So now there is a solution for serious digital audio mixing. Space and motion are compact. Operation is intuitive and obvious. Now digital audio can be every bit as much fun as analog always has been…with Motor Mix.
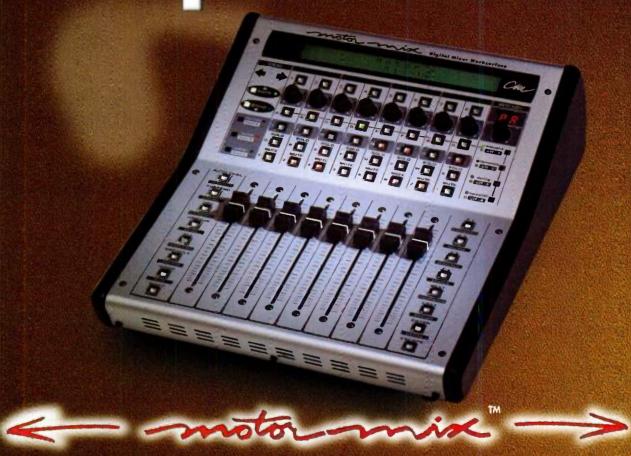
Sam Purkin, independent composer, arranger, and Gold record recipient says: "What can I say. . . Motor Mix rocks! I've always wanted one control surface for handling the automation of my mixes in ProTools. I wanted something that worked just like the SSL's and Neve's I work on all the time. Motor Mix combines all this func-

tionality and more in a compact, well designed worksurface. One of my favorite things is the ability to put the thing on your lap, shut your eyes and concentrate on the mix. CM Automation has managed to put so much power in such a small package! The faders are responsive and control switches are logically laid out. The time it takes to get up a great mix has been reduced ten-fold. After having done many mixes using my monitor and mouse, Motor Mix has become the most useful, welcome addition to my studio in a very long time. I can't recommend it enough."

CM Automation supports its products through professional audio dealers. The Web site provides further detail into Motor Mix and their other obvious audio solutions. Go to www.cmautomation.com to find out more about Motor Mix Digital Mixer Worksurface, the PM64 Automated Router/Level Controller, and the PM216 Automated Router/Mixer.

Contact: CM Automation
Tel : 888-588-6434
Fax: 818-709-4039
Web: www.cmautomation.com

SERATO PITCH 'N TIME

Some common questions about Serato's Pitch 'n Time Shed Light on its innovative new techniques and methods

What is Serato Pitch 'n Time?
Serato's audio research lab has developed a totally unique method for time scale and pitch modification. This new technology enables the tempo of music to be changed independently of pitch (or vice versa) without any of the distortion such processing is infamous for. We are making this groundbreaking product available to the professional users — Pro Tools users — who can appreciate its unprecedented processing quality. The Plug-In takes advantage of Digidesign's host-based AudioSuite architecture, and with a qualified C P U , changes can be previewed in real time. Pitch 'n Time will change your concept of sound manipulation.

What are the deficiencies of current time and pitch manipulation products?
The biggest problem with existing products is the limited range of material that can be processed without introducing objectionable distortion. Existing time compression/expansion products only work adequately on audio that's "one note at a time," and even then have difficulty working reliably. In the real world, audio doesn't come optimized for working within a Plug-In's restrictions.

Serato Pitch 'n Time has no confusing settings to tweak, and no nasty surprises. With Pitch 'n Time you can process a wider variety of material than ever before and always know you're going to get the best sound quality. Anything from a bass line to guitar chords to drums, or even a complete stereo mix!
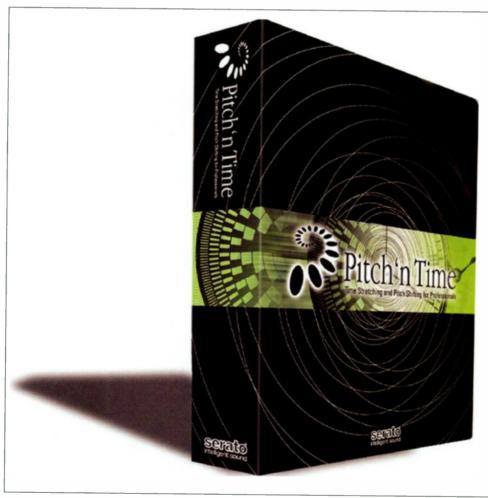
Why is Serato Pitch 'n Time different?
The answer is simple. Up until now researchers have looked at the problem of time stretching as a mathematical one, and applied objective mathematical techniques to the problem. They've failed, because they haven't properly defined what they're trying to solve.

It may seem obvious, but the perfect time stretch is one where the output sounds the same as the input, only faster or slower. The key word here is "sounds," which means the problem isn't about waveforms or mathematical definitions — it's all about hearing.

Serato Audio Research has developed a sophisticated model of the human auditory system. In a sense, the software "listens" to the music, performing a sophisticated auditory scene analysis. Only by listening can it determine what "sounds" the same, but faster or slower.

Normally using such a sophisticated model of the human auditory system would be computationally prohibitive, so novel mathematical optimizations have been developed by Serato to make this process feasible in real time.

What are the technical differences between Serato's new algorithm and other techniques for time stretching and pitch shifting?
The existing methods fall into two broad classes: Time-domain methods (micro editing) — these manipulate audio samples in their raw form, repeating or dropping blocks of samples in an attempt to maintain constant pitch while speeding up or slowing down audio. These approaches suffer from some very serious problems. By repeating blocks of samples, echoes, doubled notes, and other timing drifts are introduced. Processing polyphonic material results in severe harmonic distortion, as the algorithms struggle to track the pitch of the audio.

Frequency-domain methods (phase vocoder) — based on the Fast Fourier Transform (FFT), analyze the frequency content of the audio signal, and attempt to manipulate it in the frequency domain. The FFT must measure the frequency content of a signal over some fixed time period. When that time period is large, a high resolution in frequency is obtained but transients are smeared out in time, resulting in a reconstructed signal that is reverberant, lacking in "punch."

Alternatively, using a short analysis time results in a lack of frequency resolution, and so cannot discriminate

unrelated tones in the original signal. This causes an unpleasant inter-modulation effect in the reconstructed signal, most noticeable on bass notes.

Clearly the shortcomings of both these approaches makes them unsuitable for professional use. By taking a totally innovative approach to analyzing and processing digital audio,

Serato has crafted an algorithm that finally offers professional audio workers the quality they need.

How does the Pitch 'n Time Plug-In work within the Pro Tools environment?
The interface is divided into three areas, each one designed with an application in mind — tempo, length, and pitch. Although ultimately you're affecting only two variables (relative pitch and relative time), each area provides you with real-world parameters for expressing changes. The Tempo section works with Bars, Beats, and BPM. The BPM can be typed in, or calculated automatically by selecting a section of music and entering the corresponding number of bars and

beats. The desired BPM can then be entered, and the necessary expansion ratio will be calculated.

Or for more intuitive control, you can hit preview and use the slider or jog wheel controls to vary the tempo in real-time.

The Length section gives you direct control over the ratio. You can specify the time compression or expansion as a percentage factor, or choose the desired length in samples, minutes, and seconds, or feet and frames. Pitch 'n Time can also memorize a selection as the "target" length, and will then automatically set the ratio for matching another source to it.

The Pitch section also has slider and jog wheel controls, and allows the pitch change to be specified as a percentage frequency change, or as semitone and cent shift. Pitch 'n Time also features a built-in reference tone generator with note and cents settings for fine tuning audio by ear.

Obviously, this has huge ramifications for music recording, composing, and remixing, but what about other applications?
Pitch 'n Time provides unique and time-saving advantages in film and television post-production applications. It's now feasible to match unrelated musical pieces across scene changes, stretch vocal takes to fit, compress, and expand ambience, or pitch and stretch sound effects, allowing for unprecedented flexibility without compromising on quality.

Pitch 'n Time is Serato's first plug-In for Pro Tools. For more information regarding Serato Pitch 'n Time you can visit their Web site at www.serato.com.

BOMBFACTORY TECHNOLOGY
CREATES VINTAGE SOUNDS

VOCE SPIN AND CHORUS/VIBRATO

If you think bombfactory™ Plug Ins look great, wait until you hear them! Authentic, great-sounding digital versions of classic studio gear. Professional tools for Pro Tools - imagine that! Voce Spin™ provides the most accurate simulation of the well-loved rotating speaker, including speaker crossover, horn resonance—even different motor speeds and pulley lengths! Fifteen classic recording setups feature varying mic placements, motor accelerations, even an authentic "Memphis" sound with the lower drum's slow motor unplugged. No belts to tighten, no tubes to wear out. Just fire up Spin and click the switch—what could be easier?

MOOGERFOOGER LOWPASS FILTER & RING MODULATOR

Bob Moog goes digital! Moogerfoogers incorporate authentic analog designs from the electronic music legend, and feature bombfactory™ digital technology. Combining the best features of audio effects and vintage synth modules, Moogerfoogers open new frontiers for sonic exploration. The Lowpass Filter features a 2-pole/4-pole variable resonance filter with envelope follower. Use it to achieve classic 60s and 70s sounds on bass and electric guitar, or just dial in some warm, fat analog resonance when you need it. The **Ring Modulator** provides a wide-range carrier oscillator and dual sine/square waveform LFO. Add motion to rhythm tracks and achieve radical lo-fidelity textures—you set the limits!

BOMBFACTORY CLASSIC COMPRESSORS

bombfactory™ Classic Compressors are the most authentic vintage compressors available. No tricks, no gimmicks, just meticulously-crafted digital versions of the LA-2A and 1176—the most popular vintage compressors used in top pro studios. Whether you're just learning to use compression or a seasoned professional polishing a final mix, you'll immediately feel at home in front of these time-proven designs. Mindblowing analog modeling technology captures each unit's signature sound. They look, sound, and work just like the real thing!

SANSAMP PSA-1

Co-developed with Tech 21, the SansAmp™ PSA-1 preserves every audio nuance and sonic subtlety of the original. SansAmp provides the widest range of amplifier, harmonic generation, cabinet simulation, and equalization tone shaping options available. That's why it's used by top producers on everything from guitars and basses to vocals, drums, and tubas! All 49 original SansAmp presets faithfully and authentically recreated.

Add warmth and punch to existing tracks, or record guitars direct into Pro Tools the right way using the **Tech 21 XDI** recording interface. Do the right thing in the analog domain and capture guitar free of muddy sound degradation or excess noise. Then go digital and dial up any tone imaginable!

Contact: Bomb Factory Digital, Inc.
Tel: 818-558-7171
Fax: 818-558-1611
Web site: www.bombfactory.com.

SOUND BLENDER
SHAKES UP TDM EFFECTS
PROCESSING

Most serious Pro Tools users have had to rely on outboard gear like Eventide's legendary H3000 for high-quality effects processing – that is, until now. SoundBlender, by Wave Mechanics, brings to the TDM world the same level of creativity, sound quality, and depth previously found only in dedicated hardware boxes like the H3000. And this is no coincidence. The engineers of Wave Mechanics first made their mark as designers for Eventide, developing its award-winning Ultra-Harmonizer line of effect processors.

At first glance, SoundBlender doesn't seem that unusual, combining pitch shifting, delay, filtering, and modulation. But one listen makes it obvious that SoundBlender is very different indeed.

Wave Mechanics calls Sound-Blender a 'radical effects processor.' It easily lives up to its billing by turning ordinary audio tracks into trance-inducing sonic orgies that'll make your friends wonder why you haven't been out of the studio for days. Sound-Blender is truly fun, and is seriously addictive. But its range runs much deeper than weird effects. Sound-Blender is superb at creating rich chorus and flange effects, smooth autopanning, warm delays, super-fat resonant filtering, and many more effects that will get used every day in most studios. The lineage of its creators is clearly evident here – even some of the preset names are vaguely reminiscent of your trusty rack-mount processor.

SoundBlender accomplishes its magic by combining a set of very high-quality audio processing blocks in some very unique ways. The user interface of SoundBlender appears deceptively simple, with a set of six faders appearing in the main area of the Plug-In. However, by clicking on the small triangles in the display, many more pages of parameters are accessible, making it the most programmable TDM Plug-In ever created.

As an example of the depth of programmability of SoundBlender, the two-channel pitch shifter has a feature called pitch mapping, which can be used to create intelligent harmonies and arpeggios. The pitch mapper has over 50 settings, allowing for harmonies as diverse as diatonic, blues, West African, and Iranian (to name just a few).

The filters in SoundBlender are just as flexible, and are switchable between resonant lowpass, bandpass, highpass, and notch filtering. These filters are very analog-sounding, and would make great Plug-Ins on their own.

Soft-clipping is a big part of the SoundBlender sound, and it makes this processor sound and feel much more analog, even when using highly resonant filters or when pushing the input into the red zone of the on-screen meters.

Another key part of this processor is its organic sound – many of the effects are constantly moving and changing. The movement is created by SoundBlender's very extensive modulation section. Almost every parameter in SoundBlender can be modulated by a combination of three modulation sources. Each mod source can be programmed as a continuous oscillator with selectable waveform, a random number generator, or as an envelope detector or gate.

If all this programmability sounds a bit overwhelming, don't worry. SoundBlender comes packed with over 200 very good, and very useful patches. If you feel compelled to create your own effects, these are a great starting point. If not, there are plenty of effects to keep you happy for years to come.

It's no surprise that SoundBlender is currently the hot TDM Plug-In.

SoundBlender is available for $495, or as part of the UltraTools bundle, $895 (includes SoundBlender, PurePitch, and PitchDoctor).

World Radio History

SOFTWARE SYNTHESIS BY BITHEADZ

UNITY BRINGS VERSATILITY

BitHeadz Unity DS-1 Digital Sampler for Mac and Windows turns your computer into a powerful real-time sampling sound module. Unity DS-1 gives you all the features of hardware samplers, plus a complete integrated Digital Audio Editor, and now, Unity supports sampling rates of up to 96 kHz and bit depths of 8, 16, 24, or 32-bit floating point. Unity DS-1 allows you to recreate the sounds of acoustic instruments, and its extensive MIDI implementation allows real-time control of all parameters and easy integration into existing MIDI setups.

CONVERSION BY OSMOSIS

Osmosis easily converts Akai S1000, S3000, or Roland format CD Roms into Utility DS-1 or SampleCell format, using any Macintosh or PC with a CD-ROM drive. Unity DS-1 owners will benefit from a more convenient conversion of the Akai parameters, as well as the ability to use Roland Sample libraries.

TICKLING VIRTUAL IVORIES

Also based on the technology developed for the Unity DS-1 Sampler, "Black & Whites" is new Virtual Piano software for your Computer. Containing digital stereo recordings of each note, Black & Whites features breakthrough technology that allows you to actually play back those notes from your PowerMacintosh or PC. Attach any MIDI keyboard, and you can have the same musical control over each note as you would with a real piano!

RETRO STYLE

You can have the sounds of all of the classic synths at a fraction of the cost with the Retro AS-1 analog software synthesizer from BitHeadz. Retro goes way beyond the capabilities of the early synths with features like complete MIDI and NRPN support (you can control every parameter via MIDI) and built-in effects and sound quality that have to be heard to be believed!

DIRECTCONNECT FOR PRO TOOLS

BitHeadz is proud to be one of the first software synthesizer companies to implement support for Digidesigns' new "DirectConnect" technology. DirectConnect allows up to 32 individual streams of audio from Unity DS-1 or Retro AS-1 to be routed to the TDM bus in Pro Tools, making Retro and Unity the perfect synth and sampler to use with the Mix or Mix Plus systems. Imagine, your sampled sounds remain in the Digital domain throughout the entire signal path, and with Unity's 24-bit sound quality, you won't believe your ears!

Contact: BitHeadz
Tel: (831) 465-9898
Fax: (831) 465-9899
Web: www.BitHeadz.com.

ARGOSY'S SPECIALIZED FURNISHINGS

Argosy Console champions the last hurdle of satisfaction with the ultimate, must-have accessory. True, full-featured consoles, designed specifically for ProControl/ProTools systems, virtually eliminate any apprehension over "what clients will think." Argosy's revolutionary studio furnishings are changing the rules about what is considered the big league. Any 90 Series console for Digidesign ProControl can be configured to accommodate virtually any combination of ProControl modules, with or without other on-board rack-mount equipment. Because of an exclusive, (patented) expandability feature, Argosy's 90 Series mainframe can be extended from 90- to 120- to 150-inches or more, and still retain its sleek, integrated appearance. Creature comforts like a full padded armrest, steel leg supports, clean lines, and exact fit provide protection and add value to your ProControl system.

For ProTools-based systems, Argosy's DUAL 15 is ideal for a universal workstation. This comfortable ergonomic design features a padded armrest splayed before a two-level work surface, which is flanked by 19-inch equipment racks. The recessed monitor platform will easily accommodate up to two 21-inch CRT monitors, and the standard rack units have enough space to hold a 4 r.u. rack-mount CPU and/or a variety of ProTools modules, all within easy reach. Dual15 is available in black or gray thermofused finish with optional end panels in solid mahogany.

Argosy's Console enclosures start at under $1000. The Dual 15 workstation is available for $1199.95.

Contact: Argosy Console Inc.
Tel: 800-315-0878
Fax: 573-348-2769
Web site: www.argosyconsole.com.

Anyone who has spent time flipping through the pages of recent recording magazines can't help but notice the increased focus on microphones. From the proliferation of exotic new mics to the almost cult-like following of certain historical classics, the range of choice has never been greater. But amassing a substantial collection of high-end mics is financially prohibitive for all but the most well-heeled studios. Now, with the AMM-1, any Pro Tools owner can afford to record every individual track through a model of the specific mic that will best produce the sound he or she is looking for.

Using their patented Spectral Shaping Tool (SST) technology, Antares engineers have created precise digital models of a wide variety of

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Contact: AnTares Systems
Tel: 408-399-0008
Fax: 408-399-0036
Web: www.antarestech.com.

microphones. The user simply tells the AMM-1 what microphone they are actually using and what mic they would like it to sound like. The AMM-1 then references the stored models of both the source and target mics and processes the input to create the sound of the desired mic. The AMM-1 can also be used during mixdown to effectively change the mic on an already recorded track.

In addition to reproducing all of the subtle sonic characteristics that make each microphone unique, the AMM-1 also gives the user control of each mic's specific options.

Other items like mic placement and wind screens are also modeled. You can even mix-and-match mics, combining one mic's bass characteristics with another's treble characteristics.

To ensure that the AMM-1 always remains up-to-date with the state of the microphone art, new mic models will be available for download from the Antares Web site.

ANTARES AUDIO TECHNOLOGIES ANNOUNCES THE AMM-1 MICROPHONE MODELER FOR PRO TOOLS

New TDM Plug-In precisely models the sound of virtually any microphone

DIRECT FROM DIGIDESIGN:
PLUG-INS RIGHT FROM THE FARM

MAXIM PEAK LIMITER AND SOUND LEVEL MAXIMIZER

If you're mixing or mastering in Pro Tools, add Maxim to your list of "must

have" Plug-Ins. More than just a world-class peak limiter, Maxim optimizes the overall level of the audio input while preserving the original sound. Maxim is perfect for the stereo master of a mix, but it's also flexible enough to be used as a dynamics processor on any channel in the Pro Tools mixing environment. Available for both the real-time TDM and file-based AudioSuite environments, Maxim works with all Pro Tools systems.

The key to Maxim's smooth sound is an advanced, proprietary technique: Rather than clipping audio peaks, Maxim "looks ahead" to anticipate peaks in the audio file and then reduces peaks (relative to the track's lower level audio). The result is transparent or "perfect" peak limiting that preserves the sonic characteristics of the audio in a way that's not possible with analog limiters. In addition, Maxim automatically adjusts the overall gain of the audio to meet the ceiling as limiting is applied, so it's like

getting two Plug-Ins in one (a peak limiter and a normalizer). Maxim also offers built-in dithering, on-line help, and a full-color Histogram. When it comes time to mix or master audio tracks that you spent hours upon hours recording, don't settle for run-of-the-mill limiters than can distort or color your sound. Choose Digidesign's world-class, audio level maximizer: Maxim. Mac OS and Windows NT compatible.

BRUNO/RESO CROSS-SYNTHESIS PLUG-IN DUO

Take your audio into creative new realms of sound with Bruno/Reso — two TDM Plug-Ins for Pro Tools.

Both Bruno and Reso create rich, unique sonic textures using new cross-synthesis techniques to "synthesize" existing audio in real time. Specifically, Bruno uses time-slicing, a technique whereby timbres are extracted from the source audio during playback and crossfaded together, and Reso (as the name implies) uses a resonance generator to add harmonic overtones to audio.

Bruno and Reso can be played interactively via MIDI using an external controller. Play a chord progression or a scale on your keyboard, and Bruno or Reso will apply the performance to your audio. In addition, an on-screen keyboard allows you to "latch" the keys to create chords or play a scale right on the screen. With up to 24 voices possible, you can create a deep layer of adjustable resonant tones. Shaping and modulation parameters can be adjusted with a wide selection of on-screen control knobs. Timbre, amplitude, pitch, stereo spread, low-pass filter, "Q," and follower characteristics can be fully automated.

Bruno/Reso is a seriously advanced piece of software, but it isn't rocket science to use. Just grab the knobs, start dialing, and have fun. Mac OS and Windows NT compatible.

SOUNDREPLACER DRUM KIT AND SOUND REPLACEMENT

You know the story. Yesterday you recorded what you thought was a killer drum track only to find out this morning that it's anything but. Time to endure the lengthy task of fixing it in the mix. Or you could just let SoundReplacer do all the work.

SoundReplacer is a truly unique

AudioSuite (file-based) Plug-In that allows you to replace or mix an existing audio track with new samples from

your sound library. SoundReplacer allows you to retain the original "feel" by matching the timing and automatically adjusting the dynamics of the replacement sound to match the changing levels of the original performance. Up to three separate samples can be used to blend with or completely replace the original performance. Each sample is assigned to its own separate adjustable threshold zone. Variations in amplitude within the performance determine which sample is triggered at a given time. For example, you could assign a soft snare hit to a low trigger threshold, a standard snare hit to the medium range, and a rim shot snare hit to trigger on only the loudest peaks.

Although SoundReplacer is ideal for repairing weak mixes, it's also a powerful tool for sound design and post-production. Sound effects designers can morph up to three different sounds — such as different gunshot characteristics — to a single effect. It also creates tons of new possibilities for re-mixers. Turn a snare drum into a tin can, or maybe a cannon — or both! With SoundReplacer in your arsenal of tools, you may never again have to settle for a sub-standard mix.

D-FI™ FAMILY OF PLUG-INS

One day Digidesign decided to unleash some of its engineers from their fanatical commitment to fidelity, and asked them to come up with something downright "anti-fidelity." A few months

later, the engineers unleashed their response: D-Fi, a family of four Plug-Ins (Lo-Fi, Sci-Fi, Recti-Fi, and Vari-Fi) for both the TDM and AudioSuite environments. This unique group of Plug-Ins acts as a sound designer's retro dream tool — without the expense, hassle, and unreliability of actual retro hardware.

D-Fi combines a huge range of grungy, weird, and other retro-sounding effects with the flexibility and reliability of cutting-edge digital audio technology. Lo-Fi diminishes the audio quality through bit-rate reduction and good, old-fashioned noise and distortion. Sci-Fi contorts the audio signals through analog synth-type ring modulators and resonators. Recti-Fi warps the signal by providing super- and sub-harmonic synthesis, and Vari-Fi (AudioSuite only) simulates a tape player or turntable stating-up or slowing-down while playing back audio.

Without D-Fi, you would literally have to re-sample audio files through an 8-bit sampler or run a file through a modular analog synth to enjoy similar effects. Of course, then you wouldn't have the added benefit of total recall of all parameters or automation either (TDM only).
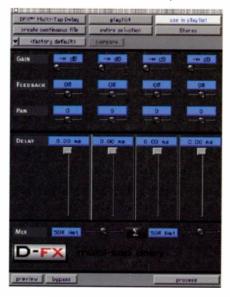
The bottom line is that D-Fi makes retro processing easier, quicker, and more cost-effective than ever before. It's the wave of the future for creating cool sounds of the past.
Mac OS and Windows NT compatible.

D-FX™ AUDIOSUITE PLUG-IN

Who says that Pro Tools|24 and Pro Tools|24 MIX users get to have all the fun? Digidesign always intended Plug-Ins to increase creative options for everyone from modest home studio users to megastar producers. It's in this spirit of Plug-In democratization that Digidesign introduced the D-fx AudioSuite Plug-In package. D-fx includes a bunch of carefully-selected,

great-sounding effects including reverb, chorus, flanger, multi-tap delay, and ping-pong delay. An AudioSuite version of the popular D-Verb Plug-In provides the reverb and ambiance processing.

Because D-fx doesn't require any special hardware I/O cards, interfaces, or DSP Farms, it makes an ideal companion to any system running Pro Tools 4.x software or higher. With its file-based processing, D-fx is also an excellent addition to any Pro Tools TDM system with limited DSP power. You can choose between "stereo" mode for processing left and right channels separately, or use the summation key to combine the two signals for a mono in/stereo out effect. In addition, each D-fx Plug-In features an intuitive, easy-to-use interface. All the parameters are displayed on one screen, making it easy to fine-tune settings with the click of a mouse.

Like all AudioSuite Plug-Ins, D-fx offers a handy preview feature that lets you hear the effects you've applied before processing, and all parameter settings are stored with your Pro Tools session for instant recall. Mac OS and Windows NT compatible.

DPP-1™ PITCH PROCESSOR

The Digidesign DPP-1 Pitch Processor brings high-quality, 24-bit pitch and delay processing to Pro Tools TDM systems. It operates in either mono or stereo modes and supplies up to four octaves of stereo pitch transposition. The intuitive user interface features musical staff notes and octave switches. Simply click on a note to quickly navigate to new pitches. In addition to pitch change, the DPP-1 can add up to 125 milliseconds of delay per channel, with positive and negative feedback. With easy on-screen editing and program storage, the DPP-1 delivers all the power of multiple pitch processors for the price of one software Plug-In. The DPP-1 is Mac OS and Windows NT compatible.
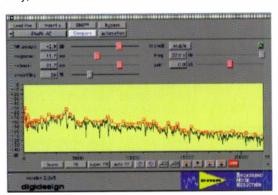
DINR™ (DIGIDESIGN® INTELLIGENT NOISE REDUCTION)

When noise is your nemesis, Digidesign Intelligent Noise Reduction (DINR) is your ally. An award-winning Plug-In, DINR effectively reduces unwanted noise — including conditioner rumble, tape hiss, or guitar-amp buzz — for cleaner, more professional

sounding music and film soundtracks. Originally available for TDM systems only, DINR now includes Broadband

Noise Reduction for AudioSuite (BNR-A/S).

DINR offers two modes — one tailored for broadband noise, such as tape hiss, and the other for more "pitched" noise, such as hum. For broadband noise reduction, DINR intelligently subtracts the noise from the digital audio file after analyzing an example of the noise within the source material. For cancellation of pitched noise, such as buzz and hum, DINR offers specialized digital filters to isolate and render the noise virtually inaudible. (Note: BNR-A/S is Broadband Noise Reduction only.) All parameter settings can be saved and recalled for use on other source audio files affected with similar noise. Of course, with DINR-TDM, all parameter settings, as well as noise signatures, are stored with your Pro Tools session for instant recall.

Since DINR analyzes and subtracts noise entirely within the digital realm, the results can be virtually free of side-effects, such as distortion, dynamic modulation (pumping and breathing), and the undesired fluctuations in frequency response associated with conventional noise reduction systems. What's more, DINR doesn't require any intensive training for high-quality results. Once installed, it takes just a matter of minutes to learn to use it.
DINR is Macintosh & Windows NT compatible.

PRO QUALITY D-VERB™

From spacious halls to intimate rooms, D-Verb brings professional-quality, 24-bit reverb and ambiance processing to both the TDM and AudioSuite environment. With D-Verb, you can choose between seven different algorithms (Hall, Church, Plate, Room 1, Room 2, Ambiance, and Non-Linear) and a variety of parameters for each (size, decay, pre-delay, diffusion, and more). Besides letting you create, store, and instantly recall thousands of different program settings, D-Verb can be used on multiple tracks, as an auxiliary input, in group masters, in mono or stereo, and more. With all its functionality and su-

perb audio quality, D-Verb does the work of multiple stand alone reverb processors at a far lower cost. In addition, D-Verb has recently been optimized to allow twice the instance count on a MIX Core/Farm Card.
Mac OS and Windows NT compatible.

digidesign
DEVELOPMENT
PARTNERS

SURROUND THEM!...WITH DOLBY SURROUND TOOLS

DOLBY
SURROUND

With over 31 million home systems equipped for Dolby Surround playback, demand for surround programming is skyrocketing. To satisfy this large consumer base, Dolby Surround mixes are also needed for such new media as DVD and DTV.

With Dolby Surround Tools, you can produce surround mixes for everything from TV shows and ad spots to CDs and video games entirely within the Digidesign Pro Tools TDM environment. They provide all the functionality of Dolby's hardware units, plus unique features, such as surround panners and special functions for video game developers.

Dolby Surround Tools can also be used to preview results of the encoding process ("4-2-4 monitoring") on discrete four-channel audio elements (L, C ,R, S) for movie soundtracks that will be mixed later on a film dubbing stage.* And your clients can take full marketing advantage of the renowned Dolby Surround trademark on their productions and packaging, with an easy-to-obtain, royalty-free license from Dolby Laboratories. Dolby Surround Tools will help you surround your clients and their audiences with the world's pre-eminent multichannel audio technology.

TDM Plug-Ins (bundled together):
 •Built-in surround panners**
 •Game Mode Encoder and Game Mode Positioner functions
 •Dolby Surround, mono and stereo monitoring
 *Pro Tools 4.0 or higher required for automation of Plug-Ins
 **Dolby Surround Tools are not designed for the final mixing of matrix-encoded theatrical film soundtracks; Dolby Laboratories supplies producers with dedicated systems and services for this purpose.

Contact: Digidesign
See page 2 for
contact information

DRAWMER DYNAMICS TDM PLUG-IN

The Drawmer Dynamics TDM Plug-In brings Drawmer's industry standard dynamic signal processing to the Digidesign Pro Tools environment and is part of a new range of digital products to emerge from Drawmer's ongoing digital research and development program.

The 'Drawmer Dynamics' module provides the Pro Tools user with frequency-conscious noise gating, expansion, ultra-smooth compression, and transparent variable threshold 'brick wall' limiting. Further innovative features include the ability to trigger the noise gate and/or compressor from any other audio track within Pro Tools and attenuate the frequency response of the trigger signal.

The dynamics package is based on the highly successful DS201 Noise Gate and DL241/251 Compressor/Limiter hardware units and has been designed to retain the same intuitive interface.

Features include variable low-pass and high-pass filters for frequency-conscious gating; full envelope control 'attack,' 'hold,' and 'decay'; variable frequency trigger source; key input with 'Key Listen' facility for internal or external trigger source; fast attack time to preserve natural attack of program material; Traffic Light LED display i.e., Green, Yellow/Amber, Red; smooth expansion ideally suited for vocals and other complex dynamic waveforms; variable ratio control; high-resolution metering; selectable 'manual' or 'fully automatic' attack and release times; and much more.

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Based on the popular DS201 Noise Gate and DL241/251 Compressor/Limiter

FOCUSRITE d2/d3

Through collaborative engineering development, including precision measurement and modeling, Digidesign and Focusrite have created a software-based digital equalizer worthy of the Focusrite lineage: The d2 multi-band equalizer is based on the Red Range 2 Dual EQ and the d3 Dual Compressor/Limiter is based on the Red Range 3 Dual Compressor/Limiter.

Features for the d2 include:

• 6-band, 4-band, dual/single band, stereo or mono modules.

Focusrite-quality equalization and compression is available for the Pro Tools environment

• EQ curve display supplies visual feedback of parameters.

• Left/right controls can be adjusted independently, then linked to retain the offset.

• Three different mono or stereo modules: 6-band, 4-band and dual/single band. These versatile configurations allow DSP power to be used where it is needed most.

Features for the d3 include:

• Two separate Plug-In configurations for maximum DSP efficiency.

• AudioSuite Plug-In included.

• On a single DSP chip, the d3 can be used up to six times on a PCI machine and four times on a NuBus.

• External side chain control.

MacOS and Windows NT compatible.

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Amp Farm, the guitar recording plug-in for TDM, uses Line 6's revolutionary new, patent-pending physical modeling technology called TubeTone to bring the warmth and feel of classic tube guitar amplifiers to Pro Tools TDM systems. Process a direct guitar signal as you record, or process pre-recorded tracks and tweak your amp sound right up to the final mix. It's the easiest, fastest, and most powerful way to record guitar with a TDM system.

WHAT'S IT GOT? The Amp Farm software Plug-In includes a truly incredible collection of guitar tones, modeled from a collection of specific, sought-after amplifiers. The TubeTone Amp models were developed from in-depth studies of Fender Twin; Fender Blackface Deluxe Reverb; Fender Bassman; Marshall JCM 800; Marshall

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See page 2 for
contact information**

Plexi; Marshall JTM45; Vox AC 30 (with and without top boost); and many more with the latest version 1.5!

HOW'S IT WORK? Plug your guitar right into your Pro Tools interface — no miking up amps, no special pickups or other gimmicks. Pull up the Amp Farm TDM Plug-In in your Pro Tools session, and you get your amp on-screen. Set the knobs the way you like 'em, and lay down your tracks. Then, any time you want, you can go back to those same tracks, and adjust your Amp Farm amp set-up, with full automation. Switch amps on the fly. Tweak your tone. Right up to the final mix. And every move you make is stored right in your Pro Tools session.

Macintosh & Windows NT compatible.

LINE 6 AMPFARM FOR GUITAR RECORDING

The easiest, fastest, and most powerful way to record guitar with a TDM system.

SYNCHRO ARTS VOCALIGN, TOOLBELT, TITAN SAVE THE DAY

Users of VocALign have said it's the "most powerful" and "astonishing" piece of software they have ever used. In simple terms, VocALign automatically edits one audio signal to match its modulations to another signal.

So it can be used to tighten double tracked or backing vocals and instruments, provide lip-sync dialog, mold sound effects together, etc. The stunning results are not only impossible to achieve by hand, but take only seconds to produce.

ToolBelt contains three essential non-real-time functions: A Chaos theory-based Audio Generator for producing short samples, longer and natural sounding "fills" or atmospheres for dialog and effects editors; TimeMod, a very fast, high-quality mono and stereo time compression/expansion process; and lastly, a Loop editor for lightning-fast edit manipulation and the creation of backward and forward patterns in seconds. The user has complete con-

Automatically edit one audio signal to match its modulations to another signal.

trol over the naming, time, and track positioning of the new audio in the Pro Tools session.

TITAN has three functions that save days of editing on film and TV productions. It's a stand alone program that processes entire Pro Tools Sessions. With the "Fix Sync" function, TITAN takes a Session containing conformed audio and, in minutes, automatically corrects the sync of hundreds of regions to exactly match to the work track.

The NEW "Cut and Move" or Re-conform function takes only seconds to create automatically a new version of a Session using an EDL-format Change List. And lastly, the "Cut Only" function creates and names regions in transferred audio using information from an EDL.

All of these programs can be tried out in full functionality for a time limited period.

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Contact: Digidesign
See page 2 for
contact informaiton

Aphex Aural Exciter Detail Enhancing Plug-In

Since 1975, more than a million channels of the patented Aural Exciter have been used successfully in a variety of professional applications. The Aural Exciter TDM Plug-In brings more clarity, detail, and presence to recordings, film, commercials, broadcasts, and live concerts. The latest Type IIIpi provides a wide range of controls, including fader, flow, and switch. By adding musically and dynamically related harmonics to the input signal, the Aural Exciter brings recorded or amplified audio closer to its natural sound. While it's great for restoring the originality of recordings, it can also be used make your audio sound even better than the original. The Aural Exciter Plug-In has even been optimized for Pro Tools|24 MIX system compatibility.

Aphex Big Bottom Pro Bass Enhancing Plug-In

Big Bottom Pro is a proven sound enhancement TDM Plug-In for Pro Tools. Modeled after the Big Bottom circuit first made available in Aphex Systems' Model 104, it is a unique and stunning example of Aphex patented technology. Big Bottom Pro works on the bass-end frequencies and adds low-end presence and punch without adding peak level, so you can pack more bass into your sound without overloading recorders and amps or blowing up speakers. It can be used as a standalone product or in conjunction with the Aphex Aural Exciter Type IIIpi and is optimized for Pro Tools|24 MIX system compatibility.

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APHEX AURAL EXCITER DETAIL ENHANCING PLUG-IN & BIG BOTTOM PRO

PRO KEYS INCREASES SPEED AND PRODUCTIVITY

ProKeys is a color-coded overlay with more than 75 preprogrammed shortcuts using Quickeys® by CE Software, that allow you to:

ProKeys
The Speedway to ProTools

• Access Pro Tools preferences, settings, and frequently used functions up to four times faster than with standard mouse navigation.
• Save time with shortcuts that provide easy access to repetitive tasks. (e.g. one key stroke will select previous region and create a fade in.)
• Become a Power User by accessing multiple functions with a single key stroke.
• Learn Pro Tools faster by viewing the Pro Tools functions right in front of you.

ProKeys can be purchased separately or bundled with Quickeys®.

Contact: PowerKeys
Tel: 323-655-4282
Email: Pkinfo@powerkeys.com
Web: www.powerkeys.com

METRIC HALO'S CHANNELSTRIP: WORLD-CLASS MIXING CONSOLE PROCESSING FOR PRO TOOLS

ChannelStrip is a TDM Plug-In that provides the essential processing functions of the best world class mixing consoles. It offers exceptional audio quality, incredible DSP efficiency, and a user interface that allows you to work with Pro Tools as efficiently and interactively as you would with a dedicated mixing console. ChannelStrip supports PT|MIX and MIX Farm cards and it can process six channels per MIX DSP chip. Now you can perform a 24-channel mix on a MIX core system with gate, dynamics, 48-bit 6-band EQ and delay on every channel. Download a demo at www.channelstrip.com/eq and experience "the future of digital audio."

Contact: Metric Halo Laboratories, Inc.
Tel: 888-638-4527; 914-831-8600
Fax: 914-831-4827
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ey in Cuba, so most of the clientele are artists and record companies from Europe, particularly Spain or Germany, Japan, South America, Mexico, and Canada.

The current project pumping sounds out of Studio 1 is a three-CD set called *Tocando Tierra* (playing/touching the earth) produced by Don Grusin and Frank Quintero, engineered by Roger Nichols, and executive produced by Samuel Quiros and Alejandro Zalles for Latin World Productions, S.A.

Tocando Tierra is an eclectic feast of over 30 musicians from various musical genres and sizzling with talent like the Habana Ensemble, Bela Fleck, Abraham Laboriel, Alex Acuña, Jerry Hay, and singers such as Ilan, Luis Enrique, and Soledad.

Sandra explains that Abdala Studios was named after a heroic character (i.e., died in battle) from an early epic poem by revered José Martí. Martí was a revolutionary hero who himself died fighting in a battle to free Cuba from Spanish colonialism at the turn of the last century, a mirror image of the character he created decades earlier.

Unlike the dead fictional Abdala, Abdala Studios is very much alive and kicking. Martí wrote in Our America, "The thought is the father to the deed." To that end, Abdala Studios is the new proud parent of recordings by Aldo Lopez, Carlos Puebla, José Maria Vitier,

Amaury Pérez, and a televised tribute to the barbarian of rhythm, Benny Moré, for RTV called, *A Benny*. A publishing company is in the works and Unicornio Publishing will handle all catalogs in house.

I find out later from other sources that Sandra and Katia are paid the equivalent of around $20 a month, and

are not allowed to meet me for a drink to chat at my hotel. I am told Cubans are not allowed in hotels where *turistas* stay.

The word on the street is that life for the average Cuban "got a lot worse" when the economy converted to dollars. Since the current rate is 22 Cuban pesos to one dollar, it's not hard to figure out why. The Cuban government doesn't allow Cubans to earn U.S. dollars directly, although I was told this is changing — at least for some musicians.

Well, and why not? In my global community the musician is usually at the bottom of the money food chain.

My approach to international politics consists of repeating my Cicero mantra, may-I-have-your-leave-not-to-know-what-I-do-not-know. Both sides of the almost 40-year-old U.S./Cuba embargo make a point. At what point does the point not matter anymore?

As the Mambo King neared death "...he heard the heavy bronze bells of the cathedrals of Santiago and Havana ringing simultaneously, he heard the tt-tling-tttling of a bicycle and blinked and saw the Havana night, shoots of light in the sky, a thousand trumpets and drums in the distance, cars honking, and the low murmur, like an ocean, of nighttime crowds." (Oscar Hijuelos©1989)

Well, so what if I turn into a big mush about Cuba.

From Silvio, who is in the habit of "discovering amazing things" comes this lyric, "Only love engenders miracles, only love turns clay into miracles."

Abdala Studios is a miracle in the making and a labor of love. Anyone with a heart drummed by a Salsa beat or an emotion emblazoned by a bolero would agree.

Find out more by visiting www.ab-dala.cubaweb.cu/indexi.htm or e-mailing abdala@imagenes.get.cma.net.

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Monitors: Control Studio
JBL 4412
AKG 270 Headphones
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Lexicon 300
Lexicon PCM 90
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Lexicon MPX 1
TC Electronic M 5000
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Alesis XT with 24 track + BRC
Monitors: Control Room
Genelec 1031 A
Genelec 1035 B
Yamaha NS 10 Pro
Monitors: Control Studio
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Well, here it is. Ten years of slaving over a hot computer. I hope you enjoy some of these little tidbits as much as I enjoyed writing them.
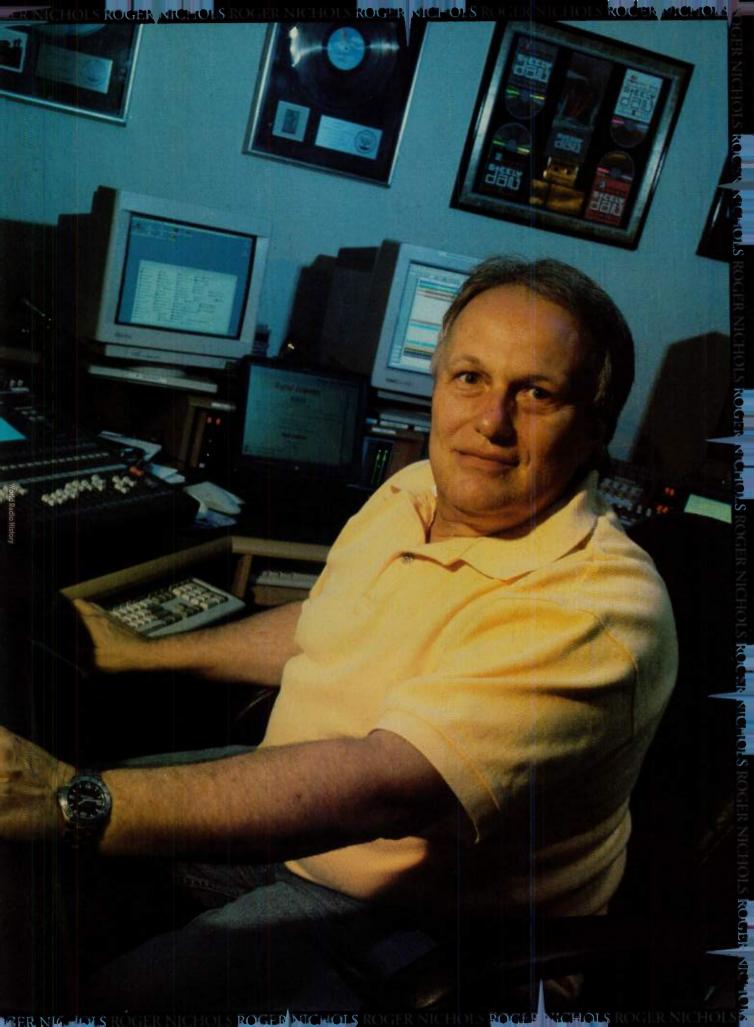
THE SIDE EFFECTS OF TECHNOLOGY

High-tech solutions solve high-tech problems...most of the time. More often than not, it seems that high tech just digs the hole deeper. Console automation makes you spend more time in the studio and hard-disk-based audio editing takes twice as long as conventional digital audio editing, which takes twice as long as razor-blade editing. The list is endless. At least once a day I find myself uttering what has become my motto of late: "It's always something!" It seems as if nothing ever does what it is supposed to do, or as fast as it is supposed to do it. This month I am going to grind my digital multitrack ax.

...As my little recording project progressed, I realized that I needed to use a tune that had been previously recorded on a 32-track Mitsubishi X-850. The thought of transferring the tape analog between the two digital machines went against everything in my gene pool. Keeping two digital machines around went against my budget. Saved by the bell. The studio I was working in told me that they had a new box from Otari that hooks on the back of their DTR-900 digital multitrack and will allow digital domain transfers between PD- and DASH-format machines, in both directions. I used it to transfer a tape that was recorded on a Mitsubishi X-850 to the Sony PCM-3348, and it worked flawlessly. This little box actually does what it is supposed to do. Oh well, every once in a while something slips through.

I CAN'T FIND MY MASTERS!

Remember when the multitrack masters for an album wouldn't all fit in the trunk of your car? A reel of 2-inch analog tape recorded at 30 ips could only hold two or maybe three tunes. Four or five of these plus the reel with the alignment tones in your trunk at one time would make your rear bumper drag over

the railroad tracks on the way to the studio. Not to mention that the price of one roll of this media would go a long way towards making your monthly car payment. The 2-track masters resulting from the mixdown sessions were substantial, also. Holding thousands of dollars worth of work in your hands felt good. It had some mass to it. It felt like it was worth a lot.

Well, all of that is behind us now. Two-track masters of mega-buck recordings are being handed in to record companies on tapes not much bigger than the microcassette from a dictating machine. I have heard reports that record companies are putting these R-DAT masters in 10-1/2-inch boxes or videocassette cases to keep from losing them.

Dis-R-Dat

I think that 1990 will be the year of the R-DAT editors. Panasonic has been talking about a new R-DAT editing system, and Fostex has the FAME editing software for their D-20 R-DAT recorders. The Fostex D-20 has a third track that is used for SMPTE timecode. The timecode can be recorded separately from the audio. This allows you to add timecode to a prerecorded tape, or add audio to a tape with pre-striped SMPTE. I have used the Fostex many times for flying instruments from one part of a song to another.

During the mixing of the Rickie Lee Jones album, there were some effects that had been used on one of the demos that she wanted to match in the final version of the tune. I transferred the demo to the Fostex R-DAT with timecode and locked it up to the 32-track digital machine during the mixing. The Fostex chased its little brains out. As the mix progressed, I could compare what I was doing with the demo just by punching one button to monitor the output of the Fostex. It was always there, perfectly synchronized with the mix. If I wanted to roll back just a little to hear the same chorus again, the Fostex chased and locked right up. It was like having two extra tracks on the multitrack. About two years ago, in an interview, I said that R-DAT would make it possible for an artist to record a complete album at home and then bring a stack of little R-DAT tapes to a studio where they would be locked together and transferred to a digital multitrack machine for final mixing. Well, it has happened.

I'd Like My Overs Easy, Thank You!

I remember when it was very easy to set the record and playback levels on analog 2-track recorders. All of the analog machines, whether they were mono, 2-track, 16-track, or 40-track, had these little pieces of cardboard with lines painted on them. Just in front of the cardboard was a little rubber needle-like pointer that aimed in the general direction of the painted lines. I know that the pointers

were rubber because I never saw one that wasn't bouncing all around while I was trying to adjust it. To set the proper level with analog meters, all you had to do was turn a knob until the rubber pointer hovered in the vicinity of the required reference mark on the cardboard background. This reference point was 3-percent total harmonic distortion of the audio signal stored on the tape. This point is called "Zero." Any levels deviating from this reference were calibrated in dB. Plus one, plus two, and so on for levels above the reference and minus values in dB for deviations below the reference. I never was quite clear as to whether minus levels meant that you didn't have enough distortion in your recording. If the reading on the meter wasn't just exactly what you thought it should be, just tap the meter with your finger until the needle moved to the correct position. Trying to be very accurate during these adjustments meant about as much as deciding where to stick the garden hose in your swimming pool so that the deep end won't get more water than the shallow end.

Then came digital recording. The references that we had to go by were thrown out the window. The "Zero" reference on the digital meters means just that. You have zero room left in the digital storage word. If you turn the signal up any louder going in, it won't be any louder coming out, it will be clipped, cut off, ugly, unusable. It is exactly like trying to pump two gallons of gas into a one gallon can. It just don't fit!
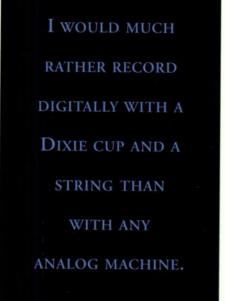
...Sony makes a stand-alone digital meter called the DMU-30 that can accept an AES/EBU (or S/PDIF) digital signal and display levels in the digital domain. I plug this meter in to whatever digital machine I am using. I have checked it with a computer-generated audio signal, and it is perfect. There are switches inside the unit to set the number of full level samples that will trigger an over indication. It's time to tell every digital audio accessory manufacturer to make one of these gems. We need them. This little goody makes life very easy. You know the old saying, "If God had meant man to fly, he would have bought us all Lear Jets." Well if He had meant us to get the proper levels on digital tape, He would have bought us all DMU-30's.

I Can't Believe my Ears, or is it my Eyes

After 10 years of recording digitally, I feel as though I am finally getting used to the detail that is available in the digital domain. When I first started recording digitally, anything digital was so much better than anything analog that it didn't make any difference whose machine I used. I would much rather record digitally with a Dixie cup and a string than with any analog machine.

Now that I have made up my mind that it will be digital forever, I can't decide what analog-to-digital con-

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verters to use. Every digital machine that I listen to sounds different. Every digital machine has a different philosophy as to how to treat the conversion from the analog domain. Some manufacturers concentrate too much on the bells and whistles that have been added to the machine and not enough on the quality of the audio it records. Professional 2-track machines that cost more than $30,000 apiece do not record audio as well as some consumer DAT machines that cost under $2000.

…What it all boils down to is that consumer pressure for higher quality audio is backing the tide up against the professional studios. We should make sure that the machines we use in the studio sound at least as good as the CD and DAT players that are being used to play back the music we produce. So when you are trying to decide which digital machine to mix to, give a long hard listen to the analog performance. If it is not quite what you think it should be, try an outboard converter box. It is not cheating, remember, all is fair in love and digital audio.

Welcome to the Workstation Blues

Christmas is just around the corner, and it is waiting to mug you, beat you senseless, and take all of your hard-earned money. If you can't think of anything else to waste your money on, how about the quietest consoles known to man or multiple synchronized digital multitrack machines that will allow you to record so many tracks that it will be impossible to find a place to mix them.

…How about this little scenario. You have decided to mix directly to your new hard-disk-based digital audio workstation with a 650 MB hard disk so that you can store over an hour's worth of material. Great, so now you take all of your new equipment into the studio to start mixing. Everything is going just fine. You finish mixing the first half of the album and then realize that you don't have enough room on your hard disk for the whole album. You forgot to allot space for the two different versions of one of the tunes or the TV mixes of all of the tunes. Boy, this stuff eats up hard-disk space pretty quickly. Well you guess that you are going to have to offload some of the tunes to make room for the ones that you haven't mixed yet. Your choices are to copy the mixes to a DAT machine or some other digital audio 2-track machine that has the appropriate digital interface, backup everything to a streaming tape backup device compatible with your system, copy all of the data to a bunch of removable hard disks, copy everything to a removable optical disc drive, or backup onto 600 or so floppy disks. I think one of the questions they ask when you call the suicide hotline now is, "Are you attempting to backup digital audio data onto floppy disks?"

…This is just the beginning of your problems. After you have spent half your life getting your digital information in and out of your workstation and have managed to get your album edited and sequenced, how are you going to get it to mastering? A lot of mastering facilities have some sort of hard-disk-based editing system, but I'll bet two tickets to the next Steely Dan concert that the hard-disk-based system at Bernie Grundman's or Bob Ludwig's is not the same brand as the one that you bought. So, once again, you have to download your data to a format that can be used by the mastering facility. This time-saving device has just cost you another hour or two. Was it all worth it? Only time will tell.

…Let me assure you that I am not against digital audio workstations, quite the contrary. I have been using a hard-disk-based editing system since 1981. I needed it so badly that I had to build it myself. It was based around an S-100 computer system with a Micropolis 32 MB 8-inch hard-disk system and a digital interface to the 3M digital multitrack…Its name was Wendel. I used it on the Donald Fagen *Nightfly* album to do things that couldn't be accomplished any other way.

…So when you sit on Santa's lap (pervert) and he asks you which workstation you want, tell him that Roger said that you should wait until you can perform some edits on your Digidesign system, take the optical disc to your friend's Sonic Solutions to clean up some background noise, and then go on to the mastering facility where they have a D.A.R. SoundStation II for mastering. Now you've got something!

Are Flutes Analog?

Well, another couple of months has gone by and another vocal has been completed on Donald Fagen's new album. I went to my local stationary store to get one of those calendar planners for 1992 so that I could start scheduling my time to finish the album. My choices were "Day at a Glance," "Week at a Glance," or "Month at a Glance." They do not yet make a version called "The Rest of Your Life at a Glance," which I think I may need before we get done.

…Hovering around at the level of acceptance that has been set for this project, we find that microseconds are tossed around instead of milliseconds, tempos have decimal points in them, tuning values are expressed in microcents, and comments on the groove feeling are something like, "It felt pretty good on that beat right there." My life outside the studio has started to imitate the reality inside the studio. I now only order egg salad sandwiches if the eggs came from Frizzle Chickens (a mutant chicken whose feathers are on backwards), and only wear clothes made from seedless cotton.

In the middle of the Donald Fagen month last October in New York, Walter Becker and I took a one week break from Donald's project to record a couple more jazz albums. One of the artists is a piano player named Dave Kikowski. The other artist is a flute player named Jeremy Steig. There was only one fly in the ointment. Jeremy is an amazing flute player. He is also, as it turned out, an amazing negotiator. It says in his recording contract with the label that his flute will be recorded analog. Walter and I could do whatever we like to the rest of the musicians (recording-wise), but we must keep the flute analog.

…At the last minute, Walter convinced Jeremy that the sound of the flute was just fine on the digital machine. Jeremy conceded and we went on with the recording. Jeremy's request was a valid one, I guess. But I would definitely put that request in the same category as a Scotsman who preferred to wear kilts.

Walter explained it very well when he called this switching back and forth between formats "Audio Cross-Dressing." Analog being the female analogy and digital being male. If you are mostly an analog person, then nobody

cares much if every once in a while you dabble in a little of the digital domain. But if you're a "digital-kind-of-guy," people point at you and laugh if they catch you messing with those frilly analog things. I am just glad that Jeremy didn't like the sound of wire recordings or the optical tracks on film. Imagine using a soldering iron to do edits or waiting for the film to come back from Fotomat between playbacks.

So now it is the beginning of February, 1992. We are in Hawaii (Donald, Walter, and myself) at Walter's studio working our little fingers to the bone, slaving over a hot 48-track. We have all but one of the tunes on tape. The last one is going to be printed tomorrow. Next week we are going to have a sacrificial ceremony and Donald is going to offer a couple of his digital delays to the Hawaiian God of Increments by throwing them off a cliff into the ocean. Walter is going to video tape the ceremony for posterity. Leroy Clouden is coming from New York to play real drums to replace the sequenced drums on tape. The vocals are going just fine, and we should be mixing by September. Hang on a second, I think my tick is coming back, back, back.

WISHFUL THINKING

I always seem to wait until the last minute to write my column. I like to blame it on the fact that all of the neat equipment comes out just before my deadline. Because of the lead time at *EQ* and the fact that the magazine only comes out every other month, a piece of equipment could be introduced and become obsolete before you even read about it. On the other hand, I still stop at my local Radio Shack once a week to find out if the $400 Tandy recordable CD they promised five years ago is out yet.

GEAR SLUTS

Well, it's official. This week I am president of Gear Sluts. Last week Walter Becker was president. What is Gear Sluts? I'm glad you asked. Walter and I formed Gear Sluts. Gear Sluts is a club whose members will do anything for equip-

ment. We have come up with a crest and a motto and everything. The motto, in Latin, of course, is, "We're only here for the gear." Sort of catchy, huh?

To become a member of our exclusive club, you have to spend at least 20 percent of your income on equipment, and in the process you must have not been able to get some item that was budgeted as a necessity. Let me explain. You couldn't afford to buy a new backpack for your kid to carry his school books because you just purchased a new B&M blower and fuel injection for your Z-28. How about if your neighbor's dog just had a litter of puppies because on the day you were supposed to get your dog fixed you were down at the music store plunking a deposit down so you would have the first ADAT in your neighborhood.

There are other symptoms that potential members display. The newest and biggest hard disk comes out and you run down and buy one at three times what the going rate will be in six months. A year later, you buy another one of the same hard disks because it is on sale at 1/20th of what you originally paid. You don't really need it, it was just such a good deal and you should have one for a spare. You never take it out of the box, and now both of them are out of production and obsolete. But wait a minute, check out this new erasable optical disc!

Someone who is a "loner" is not really as inclined to become a Gear Slut. Part of the fun of getting all this "stuff" is showing it to your friends. Do you think for a second that all of those kids with the 5000-watt subsonic car stereo systems have them because they enjoy them? No way! It is so you can enjoy them every time they drive by your house. The kids in the car aren't enjoying themselves at all. With that much subsonic energy, they are bound to be having uncontrollable bowel movements.

…Someone I know went with John Denver in his Lear Jet (talk about Gear Slut!) to Canada to get the first U.S. version DAT machine. We (oops) got it there because the Canadian machines don't have to have those stupid little

transformers on the AC cord like the Japanese versions do. John would book a tour in Japan just so he would have an excuse to go pick up the latest unreleased Nikon lenses.

To make it a little easier on potential Gear Sluts, Walter and I have decided to review equipment that would be worthy of purchase by a Gear Sluts member. In essence, we would be giving the "Gear Sluts Stamp of Approval" to equipment that should not be passed up as long as your credit card is under the limit. We have acquired the appropriate test equipment and access to the facilities necessary to run each piece of equipment through the Gear Sluts Gauntlet.

BLAH BLAH BLAH...BLAH

...I've started a weight training program. I now carry 100 lbs of batteries with me on plane trips so that I can use my Powerbook 180c longer than just the obligatory turn on at airport security. I think the twitching condition you get from watching your computer reboot should be called "Appleplexy."

EGG TOO YOUNG, AND OTHER SCATTERED WISDOMS

Here we go again. I just got home from the Steely Dan tour and I am off to another part of the world. I am leaving in the morning on a 23-1/2-hour plane trip from Nashville to Singapore. I will spend two days in Singapore, two days in Hong Kong, and then fly 19 hours back to Nashville. At first I was going to tell you that I was having a craving for 1000-year-old eggs, but I don't have time to wait for them.

...**GPS:** How in the Hell is he going to work audio into a paragraph about the Global Positioning System, you ask? Well, I don't know yet, but I'll think of something.

I took a bicycle on the Steely Dan tour so that I could ride around the venues during the time between soundcheck and the show. Some of the shows were a long way from nowhere, and I would ride 20 to 25 miles in places where I had no business being. The first couple of times I got lost and had to ask someone how to get back to the venue. The guy I asked for directions didn't even know what music was, let alone where Steely Dan was playing.

A bunch of companies make handheld GPS receivers with built-in moving map displays. They make models for boating and models for flying. The boating versions contain a database of all of the navigable (I like that word) waterways with lighthouses, buoys, channels, obstacles, marinas, and Jack In The Box restaurants. The flying version contains all airports with runways over 1000 ft., VORs, NDBs, airways, intersections, and special use airspaces.

Because I fly, I bought the flying version of the Garmin 95 XL. It comes with a mount that connects to the yoke (steering wheel) of an airplane. It also works fine as a bicycle mount. At the next venue, which was Irvine Meadows, south of Los Angeles, I programmed in the location of the venue (you push one button that says "start here") and then pedaled off into the sunset. As you move away from your starting point, the GPS unit tells you how far you have gone, how fast you are going, and which way to turn to get back. On the way back, the GPS receiver tells you when you are within ten minutes of your destination and when you have arrived. If you get really lost, the press of a button will instantly show the location of the nearest airport. (Remember, I said this was meant for flying).

...Oh, I forgot to mention that I saw an ad for a console that comes equipped with a GPS receiver and a LoJack transmitter. This is so that, when your console is stolen, I guess it will know where it is and tell the police to come and get it.

Winding Down: Well I have to run to catch my plane to China. If it were *EQ* Airlines, they might at least think about waiting for me, but United won't. Maybe I'll bring along my GPS unit just to make sure they don't stray over North Korean airspace on the way to Singapore. If you get one of those, "Help, I am being held prisoner in a Chinese fortune cookie factory" messages, please come and get me.

Everything You Own Is Obsolete (So Just Send It All To Me)

I have spent the last few months trying to go through my basement and garage trying to create space. Space, as everyone knows, is a three-dimensional area with nothing in it. I guess I read the wrong dictionary, because I thought space was there to fill up with your "stuff." If there is an empty space in your rack, you get something to put in that space. If you need to add another item and there is no room, then you buy another empty rack to create more space that you can fill with more stuff. An empty space on your desk means that you are not doing enough work, because, if you were, the space would have something in it. A garage is not for parking a car. Cars are just used as a measuring device to tell people how big the space should be to fill with "stuff." In California, houses are built with three-car garages so that you can at least use one car as a place holder to remind you where next year's "stuff" will end up.

OK, so I have ended up with a lot of "stuff" over the years. Some of it I think I can now part with, such as the second gas tank for my 1970 Lotus Europa. Dual gas tanks were hard to come by, and I thought that I should keep it in case I ever got another 1970 Lotus Europa. It has been 25 years. I guess I'm not going to get another one. How about some old S-100 computers with 4 MHz Z-80 processors, 56 kilobytes of memory, 315 k floppy drives, and 5 MB hard disk? I have three of them. They were $8500 in 1978, but now they are worth nothing. Time to go in the trash. How about a Compupro S-100 system with 8 MHz 80286, 8-inch floppies, and an 8-inch 30 MB hard disk that set me back about $20,000. I just finished paying it off, and I can't even give them away for the power supplies. How about old CO2 lasers, extra hard-disk drives, an old laptop computer from Data General, a Synclavier graphics terminal, old printers, old multi-pen plotters, an old B&M blower for a Chevy (maybe I'll hang on to that just a little longer), PROM burners, Sony F-1's, PCM 701, PCM 601, Betamax decks, old VHS decks, old bicycles, a Furman spring reverb, parts for cameras I haven't owned for ten years, SCUBA gear from 1968, an old set of JBL studio speakers the size of refrigerators that I haven't heard since 1974 (but I did have the woofers re-coned about five years ago), an old Pulsar digital watch from 1970 that would last almost a month before you had to change batteries. Get the picture?

Alive At Last

The Mixing : They told me that this issue was alive issue and could I please write with that in mind. First of all, I couldn't write with anything in mind if I wasn't alive, so what the hell were they talking about? Alive issue…does that mean they are going to cover issues about life in general, or life after death, or life before birth, or life after 50, or *Life* magazine? Oh, alive concert. Is that the opposite of a "Dead" concert? I guess I have wrung all I can out of that bad pun, almost.

It just so happens that, as I type this column, I am one week away from finishing the mixes for the Steely Dan live album. Well, almost live. I have mentioned before that we went into the studio and "cleaned up" a few things. C'-mon now, it is a Steely Dan album, and we couldn't let any afterthought recordings of raggedy-ass playing by a bunch of old farts (besides the Eagles) out into the public, now could we? Steely Dan fans have come to expect a certain quality control that could only be achieved by spending lots of money in a recording studio.
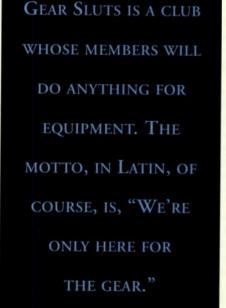
…The Part of the Column That Will Get Me Free Back Stage Passes: Sure, I have done some live recordings over the years. I started out in 1960 recording a band with Stuff Smith (jazz violin player) in the lounge of The Bowlium in Pomona, California. After working with house mixers like Dave Kob and Tripp Khalaf, I feel as though I really know that my place is in the studio where I can always say, "Oops, sorry, can we just do that part one more time?" Live guys have to know their stuff (no pun intended, but I reserve the right to use this in the future).

There is no place for error. You have 20,000 drooling fans listening to everything you are doing. If the lead guitar player comes sliding in from off stage ripping off the perfect solo and you forgot to turn up his microphone, you could be sorry that you are right in the middle of the audience with only your mixing console to protect you from the mob. I will confess, however, that during the last two summers out on the road with Steely Dan, people would come up to me to comment about the house sound. If they said it sounded great, I would smile and say, "Thanks, I'm glad you enjoyed it." If they had any complaints I would say, "I am just recording, that guy over there is responsible for the sound in the house," pointing vigorously toward Dave or Tripp. When someone complains to them about this live album and says, "I was there and that is not the same guitar solo as on my bootleg tape," they can point the finger at me.

I Am My Own Big Bang

I was down in my basement a couple of days ago when my wife Connie wandered in to ask me something. I was transferring 1630 tapes to Sound Designer II and then burning

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some CDs with Toast CD-DA. Anyway, as she approached, I was looking for some piece of equipment that I had set down on my work bench just six months ago, and I couldn't seem to find it. She said, "This room looks like an expanding universe." I was stunned for a second, like right after the initial TASER jolt when you approach some girl you thought you knew in a poorly lit parking lot. I replied, "Yeah, I guess I am my own Big Bang." I looked around and everything I owned seemed to be multiplying and spawning some new piece of gear.

...I guess I ultimately owe all of this to Stan Freeberg, a comedian who was famous in the '50s. He had an album titled *The Child's Garden of Freeberg*. On this particular album was the saga of Herman Horn, Son of Hi-Fi. Herman had turned his whole house into a giant speaker. His stereo cabinet rolled around on mink wheels. The turntable (remember them?) was packed in a mythical material called iderup, which was twice as soft as iderdown. His kids had to do without new shoes because he needed some new stereo gear. Well, Herman Horn is who I wanted to be when I grew up, and I guess I am almost there.

The only part of the Big Bang theory that bothers me is that my universe is supposed to be expanding. The equipment is multiplying, all right, but my house is still the same size it was when I moved in. Get Stephen Hawking on the phone, something is wrong here.

I HAVE NOTHING TO SAY

I am still recovering from the brain tumor I must have received from the heavy use of my cellular phone last month. It was nice of them to use the 900 MHz band, which just happens to be the resonant frequency of the brain. Have you ever noticed how your ear gets a little hot when you talk for too long? Is it coming from the phone getting hot or from your brain getting hot? Technology can bite back if you are not careful.

When I went to Cuba last month, I was outfitted like the high-tech road warrior I have always wanted to be. I think I must have looked like one of those guys on a street corner in New York City who opens his trench coat to display the stolen kitchenware he wants you to buy. I had my cell phone, 800 beeper, Mac laptop with internal modem, cordless razor, HP palmtop for phone numbers, battery operated color printer, radiation monitor, electric toothbrush, pocket camera, portable DAT recorder, MiniDisc recorder, scanner, Ham radio, air-band transceiver, altimeter/depth gauge watch, and portable GPS receiver with moving map display. (I had to see if it included Cuba. It did.) Oh yes, and one pair of underwear because that is all I had room for. Most of the toys were not charged up because they have been sitting around waiting for a good excuse to be used.

I was sort of hoping that the C-5A transport that we traveled on would have a massive electrical failure, losing all navigation and communication. Then I could say, "You can borrow my GPS, altimeter watch, and air-band radio if you let me drive," but that never happened. Our flight down was delayed six hours, however, because of a failure of one of the three INS (Inertial Navigation System) computers. My hopes were raised. I performed a radio check with the airport

tower and got the current barometric pressure to set into my altimeter watch. It turned out that we were staying on a cruise ship that was docked at Guantanamo for use by the military for R & R. We checked into our cabins and I unpacked all of my electrical goodies so that I could charge them up to prepare for the next possible emergency. A sticker by one of the wall outlets stated "WARNING 110 VOLTS DC." You're kidding! We were nowhere near D.C. Foiled again!

I ended up using nothing I took with me, well, except for the electric razor to shave the hair off my tongue after an all night party with the band and crew. I think that somehow technology knows you shouldn't be using it and dares you to accomplish anything. Recently, I had a power outage at home and my wife Connie had a song idea. No studio, no tape recorder (the cassette batteries weren't charged up), no MIDI sequencer, no synthesizer, nothing. Later on during the day, I saw her sitting at the acoustic piano (remember those) and writing her ideas on staff paper with a pencil. I heard an expletive and walked over to see that she had just written four bars of music on the wrong ledger line. I said, "Where's your Undo key?" She looked up at me, smiled, and proceeded to use the eraser on the other end of the pencil.

LAME AND MORE LAMER

I have been cruising the Internet. It is just like cruising the drive-in restaurants in the '60s. Up and down the boulevards with your windows rolled down and the radio blasting, hoping to be noticed by some cuties of the opposite sex doing the same thing.

I have noticed that there is a new way to ascertain if your opinion on a particular subject is valid or not. You post your opinion on the appropriate newsgroup, and then wait to see how many respondents agree or disagree with you. If you post to the rec.audio.pro newsgroup that hard disks sound better if they are in blue cabinets and you get no replies for awhile, it means that either everyone thinks that you may have a valid argument, or that everyone is busy painting their hard-disk cabinets blue to see for themselves.

I have been sucked into this mess myself. I remember buying blue Sharpies to coat the edge of my CDs to see if they actually sounded better. I won't admit it to anyone but you, but I bought one of those Radio Shack clocks that was supposed to improve the sound of your stereo if you plugged it in anywhere in your house. I won't tell you if it made any difference, you have to try it yourself.

WHAT THE HELL'S THE DANG DEAL?

Questions from readers have been piling up for some time now. E-mail, snail-mail, female, questions from all corners of the universe. The questions this month center around CD mastering. I will attempt to straighten out a few things right here, right now.

...*"I work for a company that invests in startup companies. We are currently considering investing $8 million in a company that says they have a black box that can be inserted into the digital audio chain and makes digital audio sound 'warmer.' What the Hell's the dang deal?" Signed Deep Pockets*

Dear Deep: First of all, let me say that I really loved your movie with Nick Nolte and Jacqueline Bisset. Second

of all, I will build you the same box for $7 million. These guys are really pulling your chain. I hear stories like this all of the time. "You should only work here because we have a special box that will make any kid that listens to this CD want to do their homework." Gadgets like this are not going to make your mixes sound better. You can make digital audio sound as warm as you want. If you take your final mixes to Doug Sax or Bob Ludwig or Bernie Grundman or Scott Hull or Greg Calbi, they can warm up your mixes because they know what they are doing and can tell by "listening" whether a mix sounds good or not. Guys like these don't need, and probably wouldn't use, a magical box like this. Now, if there was a box that made your record a guaranteed hit by running your mixes through it, that's another story.

BACK UP! (OR I'LL RUN OVER YOU)

I like trying to invent things. I like trying to improve on old processes by using the latest technology. That's why I just invented "Dead Sea Backup," a way to store your digital audio data so that it will be around in 2000 years when people can finally appreciate what you were trying to do.

I saw something on Discovery Channel about some stone tablets that were dated back thousands of years. This happened just as I was reading about the shelf life of CD-R. It hit me like a ton of bricks. Why not carve the digital audio information onto stone tablets instead of CDs? There was no time to waste!

I borrowed a Roland "3D plotter" from a friend. It is basically a three-dimensional plotter with a little Dremmel tool instead of a pen. The X and Y axis works pretty much like any other plotter. You just connect it to your computer, insert a sharpened router bit, clamp in a piece of marble, and press return.

My first calculations showed that to cut the 1s and 0s into 12-inch by 12-inch pieces of stone using a 12-point OCR font to store the 20-bit stereo data from the Steely Dan Live album would take 1,543,500 stone tablets. I thought this might be a little much, so I decided that Level II storage technology for stone tablets would have to be used. The difference is that, instead of 1s and 0s, the plotter would carve hex digits instead. One character stands for four bits of data. This would bring the storage requirements down to a reasonable 385,875 tablets. If the tablets were 1/2-inch thick, which would make them stack rather nicely, then with FAA approval the stack would reach 16,078 feet altitude. The trade-off isn't too bad, storage requirements vs. longevity.

EVERYONE IS ON SMACK! EXCEPT ME, AND I'M NOT SURE ABOUT YOU!

It seems as though there must be alien beings putting something in the water supply. No one can remember any-

> I WILL REFRAIN FROM LEAVING PAMPHLETS AND SPEC SHEETS FOR EXPENSIVE AUDIO GEAR ON THE PRODUCER'S DESK OR ANYPLACE ELSE WHERE IT WILL LIKELY BE SEEN BY WALTER OR DONALD.

thing, and everyone walks around like they have one of those brain implants you see in low-grade science fiction movies. I called a local music store to ask if they had any ADAT tapes. They said that they had plenty of them, so I jumped in my car and cruised downtown to get the tapes.

When I got to the music store, I went in and told the salesperson behind the counter that I called about some ADAT tapes. He said, "Do you want blank ones?" I said it would be nice if they were blank. He said, "Yeah, well we have lots of them that we use here in the studio for demos, but we haven't had any blank ones for over a week." Talk about "on smack," I would have liked to smack this guy a few times.

I had a mastering client come in with his DAT tapes and I proceeded to load them into Sonic Solutions for mastering. After the third tune that ended abruptly because the DAT machine that recorded it was apparently stopped before the fade had finished, I asked him if this was the condition of all his masters. He said, "These aren't my masters, these are just ruff mixes. I was hoping that you could master these, and when I get the real mixes done you could just substitute the good mixes." Now this guy was definitely on smack.

MISCELLANEOUS NOTES FROM MY BACK POCKET

On A Gum Wrapper — "Run For President": Campaign promises used to be, "A chicken in every pot and a car in every driveway." As the millennium approaches, we have attained that goal, sort of. There is a Kentucky Fried Chicken on every corner and there is a car in every driveway. The car might not be yours, though. It could be the police or the IRS. For this year, my slogan is going to be, "I promise 3 dB more headroom and a quieter noise floor for everyone." I think that just about sums up what we all need. Write me in and send me all of your campaign contributions.

While I'm at it, I think that everyone should own a computer. You guys are thinking, "I have a computer, so everyone else must have one, too." Not quite, CPU breath. I talk to a lot of project studio owners who don't own any computer at all. They insist on going into the next century completely ignorant of Web sites like hotsex.com and whatever else there is.

On The Refrigerator Door — "Return John Denver's Call": I engineered and produced John Denver records from 1980 until 1990. John then changed record companies a few times, and they supplied the producers and engineers, so I was free at last. A couple of weeks ago, he called me and said that he needed to cut four or five tunes and remix two previous recordings for an album to be released in Scandinavian countries. The problem was time. He had to start on Tuesday after Labor Day weekend and
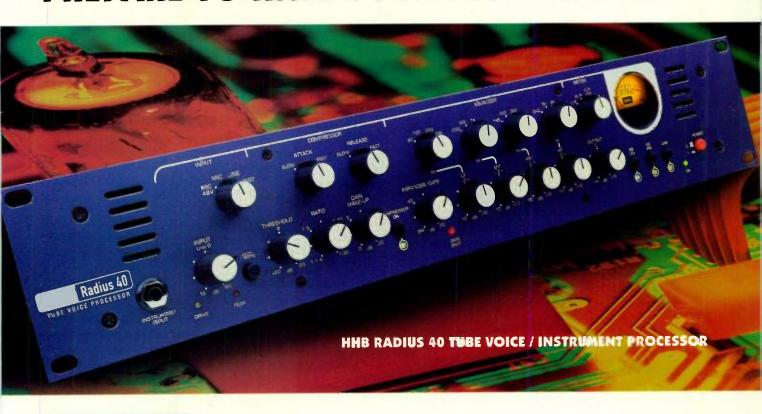
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PREPARE TO HAVE YOUR SPINE TINGLED.

HHB RADIUS 40 TUBE VOICE / INSTRUMENT PROCESSOR

HHb Radius Nothing beats the unique sound of the vacuum tube and nothing beats the extraordinary value for money of HHB's Radius tube processors. At a fraction of the cost normally associated with high end tube outboard, you can add real warmth and presence to your recordings with a range of five professional tube processors designed and built in Britain, the home of vintage tube audio. Thought you couldn't afford tubes in your mix? Check out HHB Radius tube processors at your nearest dealer and prepare to have your spine tingled.

HHB Radius Tube Processors (left).
Also available: HHB Classic Tube Processors (right).

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FIRST WE LISTEN

Don't forget to record on the best: HHB Advanced Media Products

be finished by Thursday night. I was supposed to be in Banff, Canada that Tuesday and wouldn't get home until midnight. John said he would send a Lear Jet to get me, but that it would cost $10,000 that would eat into my piece of the budget. I said I would hitchhike if I had to, but I would get to Nashville by 3 PM so we could cut the tracks in the afternoon and evening. We agreed that it could be done, and the studios and musicians were booked.

The race was on. I had to rent a car in Banff, get up at 3 AM, drive about 100 miles to Calgary, and change to an early flight back to Nashville. Since American Airlines dissolved its hub operation in Nashville, you can't get there from anywhere, so I had to fly to Dallas and change planes to get to Nashville.

As is usual when you are in a hurry, everything happens to slow you down. The flight from Calgary was late leaving because two trucks ran into each other right behind our plane and we couldn't push back out of the gate until they got all of the pieces cleared out of the way. Then, just as we were about to touch the runway in Dallas, all of a sudden the pilot poured on the power, raised the landing gear, and executed a missed approach. The pilot got on the intercom and said that there was another plane still on our runway and there wasn't room for both of us. Good choice. It took 40 minutes for us to go around, get back in line, and finally get on the ground.

Instead of an hour between flights, I now had only eight minutes to get to the other side of the Dallas terminal complex (this should have been an Olympic event). The only reason I made it is because the door was stuck on the flight to Nashville and the maintenance crew was working on it when I got there. They closed the door and we were on our way. The winds were in our favor and we got to Nashville on time, taxied up to the gate, and just sat there. The newly repaired door would not open. It took over an hour for them to fix it so we could get off of the plane.

I got to the studio at 4:30. As I walked in the door, all of the musicians were in position, eagerly awaiting my arrival. I had another engineer I work with go in and get everything set up, get cue mixes, and be ready to go when I got there. Sort of "tag team engineering." I went into the studio, checked some mic positioning, then into the control room, scooted up to the SSL 4000G, and "Ah one, ah two, ah one, two, tres, quattro." We were waxing.

This is the good news. As it turned out, John was a little off on the amount of work that needed to be done. We ended up cutting 18 tunes. Six that night, eight Wednesday, and four Thursday morning. Thursday afternoon I had two studios going at once. We needed to finish up some overdubs and fix some vocals, but if I didn't start mixing, we wouldn't make our deadline of mixed, mastered CDs in L.A. by Monday afternoon. So my assistant (Jason Lavine) worked on the overdubs while I started mixing.

I mixed Thursday, Friday, and Sunday. Don't ask me about Saturday, I can't remember Saturday. I finished mixing the last tune about 10 PM Sunday night, packed up my gear, went home, and started compiling and mastering mixes. I then had to cut six CDs for delivery to John, his manager, and the record company. I finished the last CD at 5 AM, packed my bag, and left at 6 AM to catch my flight to L.A. I dropped the CDs off on time Monday af-ternoon. We did it. With a lot of teamwork and little sleep, we pulled off what started out to be an impossible task. I attribute this successful project to all I have learned by reading my columns in *EQ* magazine.

SIRIUS AS A HEART ATTACK

…During the session one of the guitar players wanted to replace a guitar solo that everyone thought was "genius." I asked him if he was serious, and he answered, "I'm as serious as a heart attack." What more could I say. I punched in the replacement solo, which wasn't any better than the original solo, but it made him happy.

Since I seem to have an affinity for puns, I said, "Speaking of Sirius, recording analog makes me about as lethargic as the Dog Days of Summer." All I got was a room full of blank stares. I waited 2000 years for that pun. Ya see, Sirius is the brightest star in the sky. During the summer, the Sun is in the constellation Canis Major, where Sirius resides. Ancient cultures thought that those hot days of summer were caused by the light and heat from Sirius, adding to the light and heat of the Sun during those few weeks. Canis is the "dog," thus the "dog days of Summer." I can't understand why no one got it!

THE "TWINKIE FACTOR"

I have started working on the Roger Nichols Plug-In for Pro Tools. The program that I was writing took up about 30 pages of source code and needed about half-a-Gazillion bytes of memory to run. It contained some of the same routines that were in Wendel (the sampling drum machine used on *Gaucho*). I thought that the original version took a lot less code and memory to perform the same task. I decided to check it out.

I dug into my storage shed where I have kept everything I ever owned. I still have the 6-inch reflector telescope that I built in the summer of 1956. I found a box of canceled checks from 1964. I guess I went in too deep. I was getting closer, I found a pair of speaker cabinets I built from JBL blueprints in 1972. Nice three-way systems the size of a VW bug. I finally found what I was looking for, my first computer from 1976. A 1.8 MHz 8080-based computer with 64-character by 16-line text-only display. I found the floppy disk with my original program on it, and the printer that whipped along at a blinding 30 characters per second. I also stumbled on the Mother Load! Sitting right there where I left them was a 20-year-old carton of Hostess Twinkies, still sealed in their original wrappers. All of the preservatives had kept them in better condition than my 8-inch floppy disks.

I opened a package and broke open a Twinkie. The creamy white center was bigger than a quarter. It felt much more substantial than the Twinkies of today. I remembered the good old days when Twinkies were the staple of any late-night programmer's diet. Only last week I bit into a Twinkie (it was forced upon me by my daughter) and couldn't even find the cream-filled center.

I think that studio maintenance personnel fit into the same category as computer programmers. A nerd of the nerdiest kind. The obligatory pocket protector, Coke-bottle-bottom glasses, pizza-stained shirt, and overflowing trash can. A long burned-out desk lamp, a glaring com-

World Radio History

puter monitor radiating green text at bloodshot eyes, and a printer ribbon that was so over used that you couldn't read what was printed on the back side of used computer forms. Floppy disks used for drink coasters. A fast food drink cup that had exceeded its half life, refilled with Jolt Cola, and that leaked from the bottom.

Yes, this could be why the quality of computer software and studio maintenance has become sloppier. The Twinkies on which they have depended on for sustenance for a generation, are inferior. Would you expect your dog to fetch the paper or wash your car if you offered him substandard treats? I think not.

By the way, I didn't eat the Twinkie from my storage shed. I did feed it to my dog and have been watching him closely for the last week. He seems OK, but I'm finding him sneaking around my computer. I personally won't eat a Twinkie that is more than ten years old, unless I'm really hungry or unknowingly buy one at the snack bar at the NAMM show.

In My Trunk: Oh, if you remember from last year, I made six green CDs that I put into the CD changer in the trunk of my car. They have been there for over two years now, and playing just fine. I checked the error rate on them just for kicks. I made a seventh CD that has been stored in its CD case in my studio for the same amount of time. CD number seven showed an error rate of 12 to 18 Blers per second. All six of the CDs from my trunk now show errors of 2000 to 3000 Blers per second with interpolations and uncorrectable errors subsequent to de-interleave. In other words, the trunk of your car is probably not the best place for long-term storage of Master CD-Rs.

THE ANSWER MAN
To save space this month I am not going to repeat the questions sent to me, I will just supply the answers. In some cases, it will be easy to figure out what the question is. If not, you can substitute any question you would like.

1. I prefer recording my multitrack at 48 kHz and my final mix machine at 44.1 kHz. First of all because it will be harder to synchronize the two later, and why would I take the easy way out?

...3. I think that would be 6.0221367x10²³ Mole⁻¹. That would be the correct number of Avogadros to make good guacaMole.

...12. Yes, burning your own CD-Rs is just like flying in an airplane. Hours and hours of pure boredom punctuated by moments of sheer terror. I have never heard of a case involving a CD recorder and engineer being hijacked to Cuba, though.

13. That would have to be Al Bielek and his brother involved in the Philadelphia Experiment in Philadelphia harbor on August 12, 1943 (or October 28, or never, de-

pending on who you ask), and the Phoenix Experiment at Montauk Point, Long Island on August 12, 1983.

...15. No, no. Compton scattering has nothing to do with L.A. gangs and confrontations with the LAPD. Compton scattering is similar to the photoelectric effect except that only part of the incident photon energy is absorbed by the electron that is ejected from the atom. The rest is carried off as a lower energy photon. The sum of the kinetic energies of the free electron, the remnant nucleus, and the emitted photon are equal to that of the incident photon.

16. You are correct. Sometimes tuning digital synthesizers to an acoustic instrument can be a drag. The increments are in digital steps, and the exact amount you need is in-between steps. Doppler to the rescue. If you play the synth through a speaker and have the assistant hold the microphone and at the punch-in point, run toward the speaker (if the synth is slightly flat). The speed at which he must run toward the speaker is found by the following equation:

F = the frequency in Hz you desire

E = the frequency in Hz of the actual synth note

V = 343 m/s the speed of sound at sea level with an air temperature of 20 degrees C

R = the speed of the assistant engineer in meters per second

$R = (V(F-E))/F$.

You can probably only get one note per punch, but it will work, I've done it.

...19. Wow, thanks for asking. I haven't been this moved since the 9.5 earthquake in Chile in 1960. Me and the wife are doing just fine. The kids are great, one girl is a high school senior this year and the other is in the seventh grade. I had to move from Nashville because my dog had bruised ribs from kicking her when I came home from the studio in my pickup truck. I moved to Miami because it is the home of the World Headquarters of Burger King. Their address is even 9700 My Way.

Cool, huh?

NON-DISCLOSURE
Warning, the following column contains the following:
A: ADULT LANGUAGE
N: NUDITY

I received bags of e-mail with questions about whether or not I was working on the new Steely Dan album. The problem is, I can't discuss it. Donald and Walter made me sign a non-disclosure agreement. They won't let me tell anyone what goes on behind closed doors, or even admit that the project exists. This is serious stuff! If you don't believe me, I can prove it. Here is a copy of said agreement.

Non-Disclosure Agreement: I, Roger Nichols, being of sound mind and body, do hereby agree to the follow-

> I WON'T ADMIT IT TO ANYONE BUT YOU, BUT I BOUGHT ONE OF THOSE RADIO SHACK CLOCKS THAT WAS SUPPOSED TO IMPROVE THE SOUND OF YOUR STEREO IF YOU PLUGGED IT IN ANYWHERE IN YOUR HOUSE.

ing terms and conditions of my employment as engineer-in-chief of the new Steely Dan album, *Two Against Nature*.

1. I shall neither compose, nor shall I publish, any contemporaneous accounts pertaining to the events taking place during the making of said Steely Dan record, during the lifetimes of the principal artists Donald Fagen and Walter Becker, or until such time as all three of us are dead and gone.
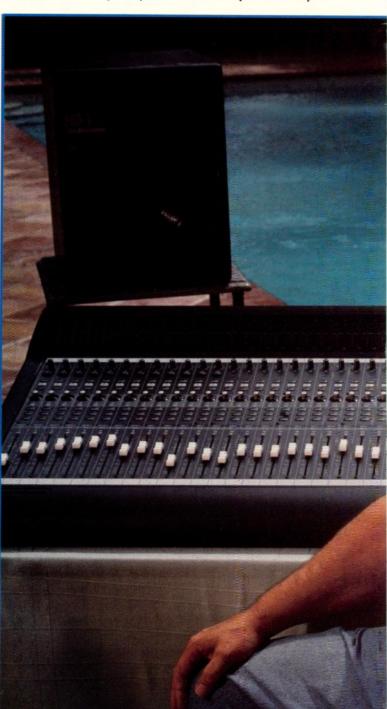
2. For the duration of the recording project I shall refrain from using the following terms of art in any of my regularly published columns in *EQ* or anywhere else:

a) "increments"
b) "milliseconds"
c) "Slow Tools"
d) "punch at the section"
e) "SampleSlut"
f) "furlongs per fortnight"
g) "Why a duck? Why not a chicken?"
h) "adrenal cotex"
i) "I won't take it home, I'll just eat it here"
J) "none of my tricks?"
K) "cantilevered clit-shelf"

3. In deference to the fragile mental health of myself and my employers, I will refrain from referencing, in whole or in part, the following anecdotes: "2nd Arrangement," "your everlasting summer," "punch at the section," "track 25," "Malibu Sheriff," "mother/daughter undercover agents," "the girl with four nipples," and "what do mullets look like?"

4. I will refrain from leaving pamphlets and spec sheets for expensive audio and computer gear on the producer's desk or anyplace else where it will likely be seen by Walter and/or Donald.

5. I shall tithe a small portion of my salary from the project to my family for food and household expenses, rather than spending all of it on expensive audio and computer equipment, as is customary.

6. I will not charge more than $2000/month in 900-number toll calls to the telephone at the studio or in my hotel room.

7. I will not order food from any restaurant whose name ends in the word "junior" or whose name contains the word "fat." Examples: "Fatburger," "Bob's Big Boy Jr.," "Carl's Jr."

8. I will curtail my efforts towards spontaneous human combustion for the duration of the project.

9. Puns — I will do the best I can. I promise.

I, the undersigned, do hereby agree to the above terms for the duration of my employment as engineer-in-chief on the new Steely Dan album and until five years after the release of said album. —Roger Scott Nichols

So, I guess I must talk about something else.

ANOTHER BAD FLASH BACK

I just got back from the 1998 AES show that was held in San Francisco. As I meandered from display to display, ogling over all of the new digital recorders, digital consoles, and digital workstations, I often heard someone whining about not having enough features on their whiz-bang digital system. The audio industry has given them

wings with which to soar, and they want to soar higher. Well, I remember the days when I had to walk 40 miles to school in snow up to my neck, and it was uphill both ways.

For just a minute, I am going to flash back (it's not just Viet Nam vets that have horrible flashbacks) to the days long ago, before digital recording, when the reigning tape machines were the 24-track and 16-track

recorders that used 2-inch-wide analog tape recorded at 15 or 30 inches per second. Performing overdubs on these machines went well, unless you wanted to record multi-

distortion to the signal. This is not acceptable if you are striving for the best possible sound quality.

If you will not accept the generation loss, then you only had two choices. The first choice was to "punch in" on the original track if you wanted to change anything in the vocal. There was a very high "pucker factor" if the performance was very good, but the vocalist insisted on an attempt at a better performance. You would roll the tape, punch in on the vocal line to be replaced, and… whoops, it wasn't as good. Too late! The only thing left to do was try the punch over and over until the vocalist got a take that was up to par with the one you erased. There was no "undo" in analog recording.

To make matters even worse, there was no "auto-punch" on analog machines. You had to perform the punch-in and punch-out over and over each time manually, without making a mistake. Analog punches were not clean, perfect punches, either. Analog machines would erase a little spot where you punched in and leave a little hole where you punched out, so you had to be good at it to replace short vocal phrases, or even words, on the master vocal track. If you screwed up, it was like trimming your sideburns. Each time you missed the punch you would have to punch in a little earlier or punch out a little later to fix the botched attempt. The "punch on the lead vocal track" was the method we used for Steely Dan vocals, and more than once my sideburns met at the top of my head.

The second method was to record multiple passes on the tape and then leave them where they were, without bouncing, and select between the tracks during the mix. This is the way we did guitar and sax solos. Let's say that we just recorded three great guitar solos on three separate tracks. After microscopic appraisal, we would decide on the pieces of each track that would comprise the final solo. We would then erase all of the pieces that were not going to be used, so that during routine playbacks we would hear a complete solo without having to make switches on the console during each playback. There was only one problem with this method — you very quickly ran out of tracks. I must remind you that Murphy's Law applied to analog recording as well, and you always needed one more track than you actually had available.

On one occasion, we had a guitar part that spanned three or four tracks as outlined above, and we needed to overdub a saxophone over the same section of the song. We had no more open tracks. We ended up recording in the holes on the guitar tracks. The first few bars of the sax on one track, the next few bars on another track, and so on. Talk about pucker factor. If the sax player was playing a line that would go past the start of the guitar, you would have to punch out regardless in order to save the master guitar track. I, uh, uh, I can't go on.

ple passes of, let's say, a vocal, and then combine them to make a master track. When you copy a vocal from one track to another on a piece of analog tape, you lose the transients and add additional tape noise and harmonic

MORE SIDE EFFECTS OF TECHNOLOGY

It has been ten years since I wrote my first *EQ* column. It was titled "The Side Effects of Technology." I think it is time to revisit our love/hate relationship with technolo-

gy. I like to think about how easy it was to work on an album (they used to call them record albums) where the musicians sat around in the same room, played together, and recorded all of the parts onto the same piece of tape. Recording, overdubbing, and mixing a pop album would usually take from three weeks to three months, depending on the artist and the budget.

Mixing before automation was easy. You usually mixed two or three tunes per day. If there was a section of the song that was particularly hard, you would mix the rest of the song, and then edit the piece into the master by cutting the tape (and sometimes a finger) with a razor blade. I have often said that automation was invented to keep you in the studio longer, not for better mixes. Console automation enabled you to spend more time fretting over minor little things that probably didn't matter anyway. The reason you spend four hours balancing the vocals in one chorus is because you can, not because you need to.

...A percussionist came into the studio the other day to play some shakers and stuff on one song. At the end of the first shaker pass, the musician said that he thought he could do a better pass. The artist pressed the talkback and said, "That take was fine, we only need one good bar."

How about sampling? Almost every rap record uses a sample from someone else's record. Don't get me wrong, I like most of them. I like listening to them to try to figure out where the loop came from. Piano licks from Lee Michaels and Bruce Hornsby, guitar licks from Steely Dan, keyboards from Isaac Hayes.... There were two different artists who used the same guitar lick from Steely Dan's "Black Cow," and one of them accused the other one of stealing the lick off of his rap record!

...When there were only four tracks on a professional tape machine, the records got done using those four tracks. The strings on track 1, the whole rhythm section on track 2, the vocals and backgrounds on track 3, and the horns on track four. The first time I moved up to eight tracks, I had two tracks left over with nothing on them. All of the Steely Dan records got made using one 24-track analog machine. We tried once locking up two machines, but the brain damage from the lock-up process in the '70s was an excessive price to pay for the extra tracks.

About when the 24-track analog machine was introduced in '72, the desire for even more tracks spread like a disease. One company built a 40-track machine using 2-inch analog tape. MCI built a 36-track machine that used 3-inch wide analog tape. Both 3M and Ampex made tape for it, but the tape was so heavy that the reel motors couldn't handle it. Second engineers had to do weight training before a tracking date just to be able to change tapes.

In the "good old days," the number of tracks was limited, so multiple instruments had to share a single track. If you needed empty tracks for vocals, you would combine instruments to a single track and add the effects during the bounce. After the bounce, you were done. That instrument with that effect was locked in. No changes were possible. In 1999, there are infinite nondestructive processes you can perform on every track you have recorded. Just to listen to all of the possibilities will take you the better part of a lifetime.

For the past few years, Walter Becker and I have started what we call The Preset Factory. When auditioning sounds on a synth, or reverbs in a multi-verb box, or compression in a DSP plug-in, we only listen to the presets. If the sound wasn't good enough for the manufacturer to include it in his list of presets, then it probably wasn't worth hearing. This cuts down the auditioning time considerably. Now there are only a finite number of settings to listen to. It is possible to tweak settings beyond the preset, but we need a majority vote from all involved in the session. Each member of the band is also allowed one "gotta-have-it" for the entire album project. If he invokes the gotta-have-it, it overrides the vote. Sick, huh.

UNDO HARDSHIPS
The other day I was working with an engineer in the studio who accidentally erased a piece of a guitar solo. When I walked into the control room, he was blankly staring at the Sony 48-track remote. I asked what was going on, and he said, "Where is the Undo button on this machine?" Wouldn't it be great if life had little Undo buttons along then way? How about:
Speeding Ticket:
• Undo Ticket (F3) — Court date is the same day as my first paying Pro Tools gig.
• Save (F5) — Register for topless traffic school, may be worth it.
• Print (F6) — Copies of ticket to put under friend's windshield.
DIVORCE
• Undo Alimony (F3) — Or I won't be able to make my Pro Tools payment.
• Save (F5) — So I can remember how lucky I am to be away from her.
• Print (F6) — Fake money on 2400 dpi printer for alimony payment.
GUITAR SOLO
• Undo Erase (F3) — Or I will lose my job and can't make Pro Tools payments.
• Save (F5) — That solo was crap, maybe he will do better now.
• Duplicate (F7) — Make repeat loop of crap solo and burn CDs for friends.

> IN 1999, THERE ARE INFINITE NONDESTRUCTIVE PROCESSES YOU CAN PERFORM ON EVERY TRACK. JUST TO LISTEN TO ALL OF THE POSSIBILITIES WILL TAKE YOU THE BETTER PART OF A LIFETIME.

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.

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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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.

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.

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

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.

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Preamp-Equalizer

BY FRANK FILIPETTI WITH STEVE LA CERRA
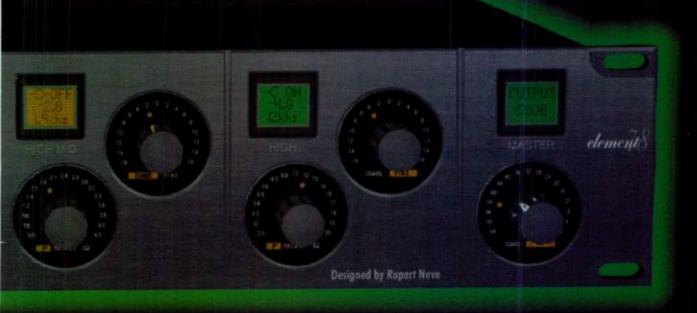
Summit's high-quality preamp/EQ provides a digitally controlled analog audio path and some other unusual features

Summit Audio — well known for producing high-quality outboard audio processors — recently introduced the MPE-200 mic pre-equalizer. The MPE-200 marks the debut of the company's "Element 78" series, which is defined by two unique features: [1] the devices retain an all-analog audio path under digital control and [2] Rupert Neve designed the audio circuitry. Employing a discrete, Class-A, transformer-coupled audio path, the MPE-200 is capable of some unusual functions, including 25 user-preset EQ curves and the ability to copy EQ settings between channels.

Our review unit used software version 1.23, which included the ability to save and recall presets, as well as link multiple MPE units. Version 1.31 — which includes remote control via MIDI — should be available by the time you read this, as should the MPE-200S slave unit.

Since the MPE-200 allows simultaneous, discrete use of the mic pres and EQs, the rear panel contains XLR microphone inputs and mic preamp outs, and a pair of XLR line-level EQ I/Os. You have the option to electronically "patch" the EQ into the mic path or use the mic pres while processing different signals through the EQs (more about this later). Also found on the rear panel are MIDI In/Out/Thru jacks, an IEC power receptacle, a fan vent, and a power switch.

Front-panel controls on the MPE-200 consist of a series of rotary encoders and backlit LCD screen/pushbutton switches — which, along with the unit's beautiful blue faceplate, form a striking appearance. A total of ten encoders and five display switches provide complete control over all parameters for left channel, right channel, or both channels when the unit is linked for stereo.

Summit Audio
MPE-200
Microphone
Preamp-Equalizer

Since these controls are visually grouped into six sections, let's look at their functions per section:

• Input: Pushing the input switch toggles through three screens. The mic gain screen (green) allows gain adjust of either or both channels by pushing the encoder knob. A meter at the bottom of this screen shows signal level from –28 dBu to +9 dBu, with a marker at 0 dBu. If you exceed +18 dBu, the screen turns red to indicate clipping.

Pressing the input display switch turns the backlight amber and brings up the filter screen. Each mic pre has two 12-dB/octave filters, either of which can function as high-pass or low-pass. Additionally, the filters may be "stacked" for use as a single, 24-dB/octave filter. Pushing the encoder selects a filter; turning the knob adjusts frequency. Setting the filter to "off" removes it from the audio path. The filters do not affect the line-level signal path.

Another press on the input switch turns the screen yellow and brings up control over phantom power on/off, phase normal/invert, or mic in/EQ in. Pushing the encoder selects one of these three functions; turning the knob changes the status. The mic-in/EQ-in option is quite important. If you select "mic in," then the output of the mic pre is electronically patched into the EQ. The EQ'd signal appears at the EQ out jack (the un-EQ'd mic signal always appears at the mic pre out jack). If you want to EQ a line-level signal such as a tape track, you physically patch the track to the EQ in connector and set the input to "EQ in."

THE EQUALIZER SECTION

All four EQ bands share a few common functions. Pressing a band's display switch alternates between EQ off (amber) and EQ on (green). The screen shows amount of boost or cut, frequency, and bandwidth or filter type. Pressing any boost/cut encoder toggles between coarse adjust (2.0-dB steps) and fine adjust (0.5-dB steps). All four bands have a frequency select encoder, but pushing this encoder elicits varying results as follows:

• High and Low Frequency Bands: Pushing the frequency encoder lets you choose peak or shelf filter curves. In either case, the curve is fixed at 12 dB/octave.

• Low-Mid and High Mid Bands: These are parametric filters. Pushing the frequency encoder allows selection of filter bandwidth from 1.6 to 0.5 octave.

THE MASTER CHANNEL SELECT

This is probably the most important switch, accessing gain trim, preset, setup, and fader screens. A small padlock graphic indicates which channel is currently selected for adjustment, as well as showing (pre-EQ) gain settings for the channels and channel level. When this green screen is active, pushing the rotary encoder switches front-panel operation from channel A to channel B. When a signal level of 18 dBu is reached, the display flashes red to warn of clipping.

Pressing the master display switch once calls up the amber preset page. The MPE-200 can memorize 25 EQ curves containing all EQ and control settings — except input setup and output fader. Divided into two halves, the screen shows status of each channel. The top row indicates (with a padlock graphic) which channel is active, which preset is loaded, which preset is "in cue," and which preset was last loaded. You select a new preset by turning the encoder, and load it by pushing the knob.

Pressing the switch again accesses the yellow master setup screen, where you may lock or unlock a preset (the padlock reflects the change), set the unit for stereo or dual-mono operation, or turn the master fader on and off. Pushing the rotary encoder selects a "menu" function; the function will blink to indicate it is active. Turning the encoder then changes the status between (for example) preset locked or unlocked. The last choice on this menu is for the fader. NoFade means the fader — and its control screen — is bypassed. This is a true, precision output attenuator, and, as such, should be set to "0" unless you are doing a fade.

OPERATION

Because the MPE-200's front panel has a modest number of controls, the user interface can be tough to master. It takes a bit of practice to rapidly hop around the EQ bands, pushing the boost/cut knob to toggle between coarse and fine adjust, or toggling the frequency selector to bandwidth adjust. We were initially puzzled when trying to set up the unit for dual-mono: it seemed that even when set to "2-Chan," making a change to one channel's EQ affected both channels. We didn't realize we had the same preset loaded into both channels (preset 1), so any time we selected either channel for EQ'ing, we were actually making changes to the same EQ preset. The trick in dual-mono is to make sure that different presets are loaded into channel A and channel B.

On the master setup screen, presets are locked or unlocked by pushing the encoder until the channel's padlock graphic blinks, and then turning the knob to lock or unlock it. When the preset is locked, your changes are not saved — when you plan to save an EQ curve, unlock the preset, make the changes, and then lock the preset. Certainly one of the coolest abilities of the MPE-200 is this screen, where you can "cue" a new preset or alternate between the last two EQ curves for an A/B comparison (finally someone did it!). Exercise some care when changing presets, because a preset retains the gain trim setting. As a result, it's possible for signal level to jump if, say, you're using a preset with a gain trim of –15 dB and then switch to a preset where the gain is set to +5 dB.

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Summit Audio
MPE-200

Microphone Preamp-Equalizer

IN THE STUDIO
We used the Summit MPE-200 in a variety of situations for tracking and mixing, trying the EQ both with and without the mic pre. Unlike a lot of outboard EQs, the mic pre in the MPE-200 is not just a gratuitous extra — it has plenty of headroom and great extension at the frequency extremes. This was evident when tracking a kick drum using an Audix D4 mic, where bottom octave was wonderful. If you're looking for a neutral- or transparent-sounding mic pre, go somewhere else. This unit has a sonic character that adds a little low-mid beef to male vocals and acoustic guitar — making it ballsier than, say, some of the

EQ LAB REPORT

MANUFACTURER: Summit Audio, Inc., P.O. Box 223306, Carmel, CA 93922. Tel: 831-728-1302. Web: www.summitaudio.com. E-mail: sound@summitaudio.com.

APPLICATION: High-quality EQ with mic preamp for audio tracking, mixing, and mastering.

SUMMARY: Two-channel (stereo or dual-mono), 4-band parametric EQ with discrete, Class-A circuitry. Mic preamps can be used separately from EQ paths.

STRENGTHS: Unit can store and recall 25 EQ presets; each EQ band may be individually bypassed; extended high- and low-frequency response on both EQ and mic pre paths; three-year warranty.

WEAKNESSES: User-interface a bit difficult to learn; no global bypass per channel; frequency overlap could be extended (see text).

PRICE: $4495   **EQ FREE LIT #:** 101

Focusrite designs. In light of the low-frequency extension, the low-pass filter on the mic pre is definitely useful for filtering out stand-transmitted vibrations. It'd be nice if you could also use these filters on the EQ input. Circuitry in the mic pre is very quiet, so you'll have no trouble cranking up the gain for low-SPL instruments or vocalists.

We used the MPE-200 as a stereo program EQ on an analog mix for Mariah Carey, and found the top and bottom of the frequency range quite extended. The fact that the frequency selectors

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JOE JACK
@ Joe's Pub

The fine art of
turning a coat check
room into a remote
recording studio

By Steve La Cerra

SON
in NYC

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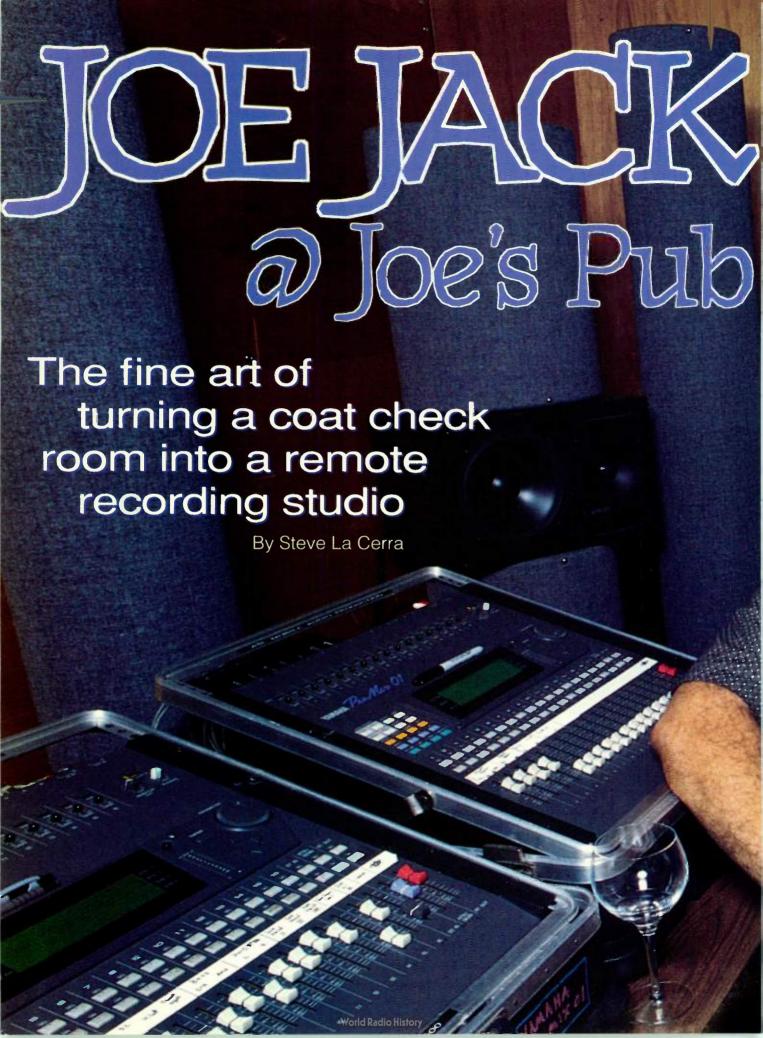
The streets of New York City are sweltering in the summer heat, so it's a refreshing change when I step into a well-air conditioned building located on Lafayette Street. This certainly doesn't look like a place where a major story for EQ Live would be developing, but as I size up the lobby, I begin to hear something that sounds like a piano being tuned. Must be the right place. A quick walk 'round the corner and I'm in the back of Joe's Pub, which — at least for a few days — has literally become Joe's pub. Joe Jackson's pub, that is.

I meet up with Tony Ungaro, Jackson's production manager, who clues me in on the proceedings: "Joe planned to do a few live shows here. Then we started tossing the idea around that since we were going to be in the same place for several nights we could record the shows for a live album release."

Though the concept appeared straightforward, the execution proved a bit more challenging. Apparently there are strange sonic reflections bouncing around the room, some of which are coming back to the stage. Although bassist Graham Maby and drummer Gary Burke have stage wedges for monitoring purposes, Joe is using a pair of Future Sonics In-Ear Monitors. Audio reflections from the house PA are leaking through Joe's ear pieces, diminishing isolation of his monitor mix, and making for some pretty strange sonic artifacts. Ungaro — along with Jackson's piano technician David d'Arcy, and Nick Schwartz-Hall (production manager for Joseph Papp's Public Theater) — are on the case to solve this problem. In order to reduce the amount of high-frequency bleed from the house system back to the stage, they've baffled off the house cluster of EAW KF200's using foam. This cures many of the acoustic anomalies without creating an eyesore. A look at the KF200's reveals the foam "gobo" atop the cluster, which ultimately will disappear into the dark when the house lights go down.

As I make my way onstage past the Baldwin Grand piano, I notice a rather odd sight on Joe's mic stand. There are two mics taped together for Jackson's vocal — a B&K 4011 strictly for recording, and an AKG C535 used for recording, the house feed, and the monitor mix.

Referring to the C535, house engineer Ken Travis mentions that "Joe has been using the C535 for years and he's very comfortable with it. It's a very open-sounding microphone. He gets that mic in his ear pieces along with a bit of kick, snare, and bass guitar, but it's mostly piano and voice. An Avalon VT737 mic pre/compressor/EQ serves as the front end for the C535, and an Aphex Dominator is inline with the Future Sonics unit to protect Jackson's ears

in case a frequency goes off. Other than that, there's just a bit of reverb on the piano and voice from a Lexicon PCM80 to wet the mix a bit." For increased isolation in the vocal mics, a small piece of Plexiglas has been mounted on Joe's mic stand, forming a visually unobtrusive baffle between the mics and the rest of the stage.

I'LL BE IN THE COAT CHECK
Looking around Joe's Pub, I'm wondering what's missing. Oh yeah — the recording gear. Where is that stuff? Ungaro ushers me out the backstage doorway and across the hall where I find engineer extraordinaire Steve Remote of Aura Sonics Ltd. (New York) displaying a mischievous grin. "It's over here in the coat room," proclaims Remote as if he heard the question in my head (perhaps he can, in which case I'm in big trouble). If you want to put your coat in here it'll cost you 100 bucks! I step into the coat room to see it overflowing with audio and computer gear. Steve and producer/Pro Tools whiz Sheldon Steiger have set up shop here and they look quite comfortable with the situation they've created.

Three of the coat room walls are lined with wooden panels. Remote wanted "to keep the listening area symmetrical, but noticed that these wooden panels are only on three of the walls. If all four walls were identical, I'd have set up the long way just for better sound." Although there are quite a few ASC Tube Traps lining the walls of the listening area, Steve wouldn't mind having a few more at his disposal. "Typically this room would be filled with them, but the space actually sounds pretty decent," he declares. A pair of speaker-stand Tube Traps support Genelec 1031A's used to monitor the sessions. "We love the sound in here," continues Steve. "Once they close the metal sliding door behind the stage and we close the door to this room, the isolation is insane. When you turn the volume off you can barely hear the band. We lucked out in a big way."

A PRO'S TOOLS
All of Jackson's shows are being recorded to Pro Tools 24, with backup to TASCAM DA-88's. Signals coming off the stage are going to a split; the direct signals from the split feeds a rack of 24 API 3124 mic pre's, while front of house gets the transformer side of

Steve Remote raised the coat check fee to $100 after turning it into the control room.

JOE JACKSON
@ Joe's Pub in NYC

the split (another split upstairs serves the monitor desk). Outputs from the API 3124's are patched directly into Apogee AD-8000 audio interfaces, which act as the front end for Pro Tools. Since the AD-8000's have TDIF cards installed, they also feed a digital signal to the TASCAM machines for the backup. "What's nifty about the AD-8000's," says Sheldon, "is that there are two TDIF cards in each unit to split, so we have six machines, 24 bits, and 24 tracks." Steiger is running Pro Tools 24

(left to right) David d'Arcy, Sheldon Steiger, Tony Ungaro (front), Nick Schwartz-Hall (back), and Steve Remote.

Sheldon is going to fit five two-and-a-half-hour shows in 36 GB of drive space. "I can't possibly keep all five shows on my hard drives simultaneously," he admits, "so I'm backing up to DDS3-format tape using a really neat program called Mezzo from Grey Matter Response. It's made specifically to work in conjunction with Pro Tools by keeping track of all audio and fade files associated with a given session, no matter what drive they reside on. This is very convenient for anyone working with multiple volumes and

Mix v4.3 on what appears to be an Apple Power Mac 7500 "but really isn't," he points out. "All the guts have been ripped out and replaced with a 260 MHz CPU, SCSI accelerator card, 256 MB RAM, and all the VRAM I could possibly need."

Audio data from Pro Tools is stored to two Glyph rack units, each of which contains one 18 GB Seagate Cheetah drive.

(top to bottom) Apogee converters, ATI Pro mic preamps, and API 3124 mic preamps.

Though the Glyph units have two drive bays each, Sheldon configured the drives this way "to distribute the heat. Like most high-end equipment, these drives like to be cool. Since we had the space in the rack anyway, I put them in separate enclosures. Before each show I clean the drives off completely and we start writing fresh. I reformat every two or three nights but I don't do a low-level reformat — I do a high-level reformat (much faster), which works fine in this type of situation, where I'm

just rolling continuously in record for long periods. I do a low-level reformat about once a month. Continuous recording like this is tough on a drive, but the more serious fragmentation occurs when I'm in the studio doing a lot of editing and fades. Under those circumstances I reformat more regularly."

After a little bit of head scratching and mental multiplication, I come to the conclusion that there's no way

large numbers of audio files. I get the backup going at the end of the show and it runs overnight. If it were continuously backing up and I had someone standing here waiting to switch tapes for me, it would take about six hours to back up the show. In the morning Hugh O'Flaherty (monitor engineer at Joe's Pub and part of Remote's Aura Sonic Ltd.) comes in and switches tapes because a show will fill up two tapes plus a bit more on a third."
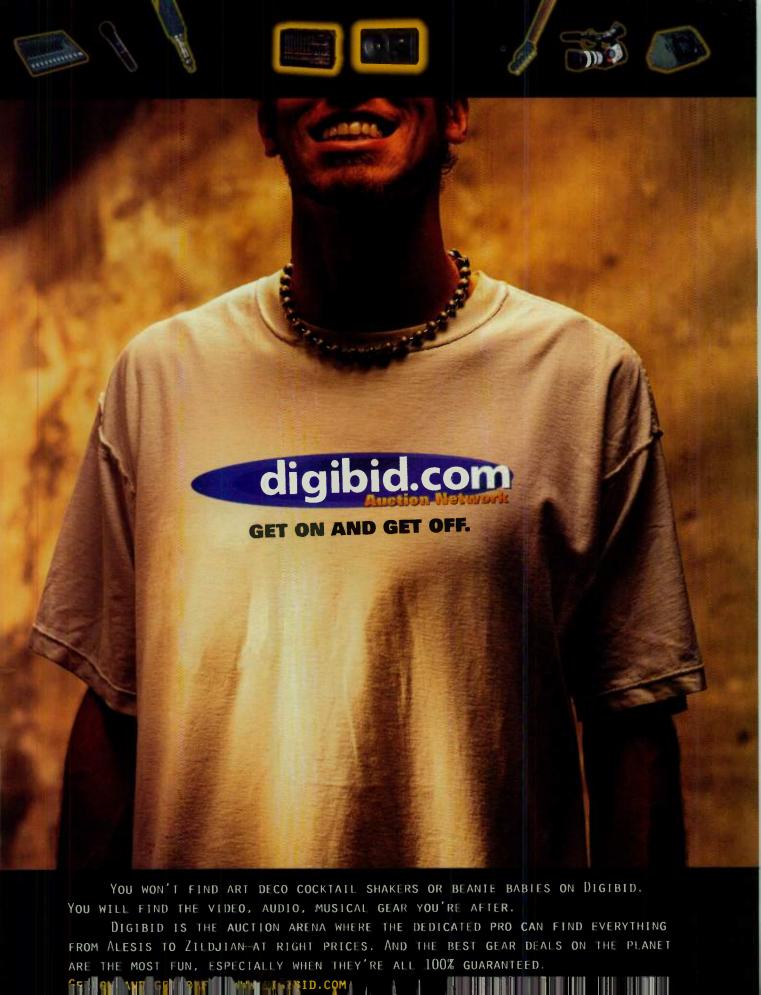
MULTITRACK MADNESS

As mentioned earlier, Jackson's vocal is being recorded with a B&K 4011 taped together with the C535; the 4011 is set back from the front of the C535 so the recording has a bit of air to the vocal sound. Output of the 4011 is split to one of the API 3124's as well as an ATI Pro 6, providing an uncompressed vocal track from the API and a slightly-compressed track using the ATI. "[I don't] really plan to use the processed vocal in the mix," explains Steiger, "but it might come in handy if there's a spot where the uncompressed vocal is distorted. I'll be able to paste a piece of the compressed vocal into the unprocessed track."

PREPARED PIANO

In addition to Jackson's vocal, one of the most important considerations for the recording is the piano. Even at a quick glance it's clear that the Baldwin 7-foot Grand sitting onstage 'ain't no normal piano. David d'Arcy — Joe's piano technician — has made a number of modifications to the instrument.

Provided by the Baldwin Concert & Artists department of New York, the piano was fitted with MIDI by David some time back, and is a favorite workhorse for many artists (in between Joe's gigs, d'Arcy whisked it up to Woodstock '99 for a couple of hours with Bruce Hornsby — a regular user). David says it's both warm and bright, so it sounds good solo and still cuts through a band. "It's a very stable piano. It's been moved around a lot lately, and just did a 1000-mile trip in a truck in 97-

JOE JACKSON
@ Joe's Pub in NYC

degree heat, and it's like it never left the room! Baldwin makes the strongest pianos in the world. I once saw a Baldwin 9-foot fall out of a crane and crush a Volvo. It played a concert one hour later, as scheduled. The Volvo went to the scrapyard! "

David shows me how he's completely lined the piano case interior with 3/8-inch foam rubber because "when the lid is closed, it's parallel with the surface of the sound board, so the mics hear all sorts of reflections and standing waves, dips, and peaks at frequencies with a wavelength close to the 8 inches between the lid and the sound board. The rubber helps reduce that." Since the lid itself tends to bounce around due to vibration from the bass and drums, he's also lined the edge of the piano case with foam, thus dampening vibration from the lid when it is closed for the performance.

For further isolation of the piano mics and pickups, d'Arcy has essentially sealed the piano from below and from the action cavity as well. A peek underneath the instrument shows a series of carefully sculpted foam blocks (lined with the 3/8-inch dense stuff) placed in between the piano's support beams, closing off the sound board from the outside world. "Of course adding this destroys the acoustic sound of the instrument" acknowledges David, "but it maximizes the isolation. In this case it's temporary but if I were to do a permanent installation, I'd attach a board underneath shaped like the piano, with screws you could remove to go acoustic. The sonic isolation also has the added benefit of isolating the atmosphere in the piano, which really helps tuning stability when the piano is touring."

Inside the piano, there's all sorts of interesting stuff mounted for audio pickup, including a Shure VP88 stereo condenser mic, an optical MIDI pickup made by Gulbransen, a Helpinstill magnetic pickup, and a Countryman Isomax Type C microphone mounted on the Capo d'Astro bar to pick up sound from the sympathetic strings. David informs me that the MIDI pickup is "really just being recorded for posterity, and so that Joe will have a score to follow while he's editing the show. We also have a Helpinstill — basically a big magnetic pickup. d'Arcy is in the process of adjusting the Helpinstill, and explains to me that one note in the midrange is noticeably louder than the rest. He fiddles with the pickup, plays some more notes and we observe that the note is now sounding at a volume consistent with its neighbors. While the Helpinstill is part of the house sound and is being recorded, it's a very dry sound which is not being mixed into Joe's ear pieces.

The Isomax Type C cardioid condenser mic has been taped to the plate Capo d'Astro bar and is used to capture sympathetic vibrations from the Capo section, near the tuning pins (it's a 1-

David d'Arcy makes an adjustment inside the Baldwin 7-foot Grand piano.

inch length of string, not actually struck by a hammer, that resonates in sympathy to the notes being struck). David describes this sonic element as "a dirty, mostly out-of-tune element essential to the sound of the high resister of the piano. Though it doesn't travel very far, you can hear the sound change if you mute these strings. This mic has an EQ to wipe out everything (theoretically) under 8 kHz. We want it to start coming in after the last fundamental note, which is about the same frequency as the first sympathetic string. It helps shine up the piano sound a bit, and keeps the air and ambiance of the sound often overlooked in close miking." As a final nod to acoustic isolation, d'Arcy lays a black blanket on the floor beneath the piano to mute any reflections.

A few moments later, Steve ushers me into the coat room for a listen as the band loosens up. Analog output from the Apogee AD8000's (on D-Sub connectors) has been patched by Remote into two Yamaha 01V's for monitoring the recording. When Steve solos the various instruments, it's startling to hear the amount of isolation they've achieved between the drums, bass, piano and monitors/FOH speakers — most engineers would be happy to get this kind of separation in a studio, never mind on a stage.

EDITING SMARTS

One of the most daunting tasks facing Ungaro, Jackson, and Steiger shall be editing the material from the five performances down to a CD's worth. "The band has rehearsed 42 songs, with Joe spontaneously calling them during the set," explains Ungaro. "There are some standards, some pieces from David Bowie, Duke Ellington, some new material from Joe, and some of Joe's older compositions, which he's re-orchestrated for the trio." Given all those possibilities — plus the fact that each show would take a major amount of time just to load into Pro Tools — how do they plan to handle the editing process? There's actually two answers to that. First, the monitor mix from the 01V's is being mixed live to a dual DAT deck; the DATs will be auditioned for prime takes. Second,

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Remote has patched this mix back into two channels of Pro Tools, so in addition to the individual tracks, Pro Tools is also recording a stereo mix of the show. This mix remains on hard disk even though the tracks are off-loaded. If Sheldon needs to test an edit between two different nights, he can do it in Pro Tools on this stereo mix without having to restore the individual track files onto the drives (although ultimately each show will have to be divided into individual song files).

Remote and Steiger agree that they could have brought in a lot of processing gear and toys, but that wasn't the point. "It's really happening over there on the stage," concludes Sheldon. "With Steve's good mic placement, we just let the band do their thing and it goes to disk." As we listen to the band warming up, it's apparent they're well on their way to achieving their goal. Now, which one of these guys took my coat?

Check out Sheldon Steiger's Web site at www.sheldonsteiger.com. Also, check out Steve Remote's Web site at http://members.aol.com/aurasonic

A FEW MINUTES WITH JOE

In addition to the recording project at Joe's Bar discussed in this article, Joe Jackson has several other major projects happening right now, including a new release of his Symphony Number 1 and a book entitled A Cure For Gravity. After soundcheck, I had a brief chat with Joe to discuss his latest endeavors.

Tell us a bit about A Cure For Gravity.

Well, it's basically two books in one — both a memoir and a book about music. It's not really an autobiography, though I do talk about my early years as a musician and my apprenticeship as a musician. It starts when I was a kid and goes up to right around my 24th birthday — which was when I was in the studio recording Look Sharp. That's where I decide to end the book because that was when I stepped out of the wings and into the spotlight. I think the earlier stuff was much more interesting than the pop success, touring, drugs, and hotel rooms. Anyone who has a hit record goes on tour and it's the same bloody story. I think the early years of struggle and failure, disastrous gigs, weird characters, and hilarious, bizarre situations are much more interesting. So that is what I wrote about. But I also wrote about the process of learning and becoming a musician. I use it as an excuse to talk about music. I have a very varied background from the beginning.

Your new release Symphony Number 1 (as well as "Symphony in One Movement" from the Will Power CD) displays strong Classical influences. Were you Classically trained as a musician?

Yes.

How did you wind up in the pop world?
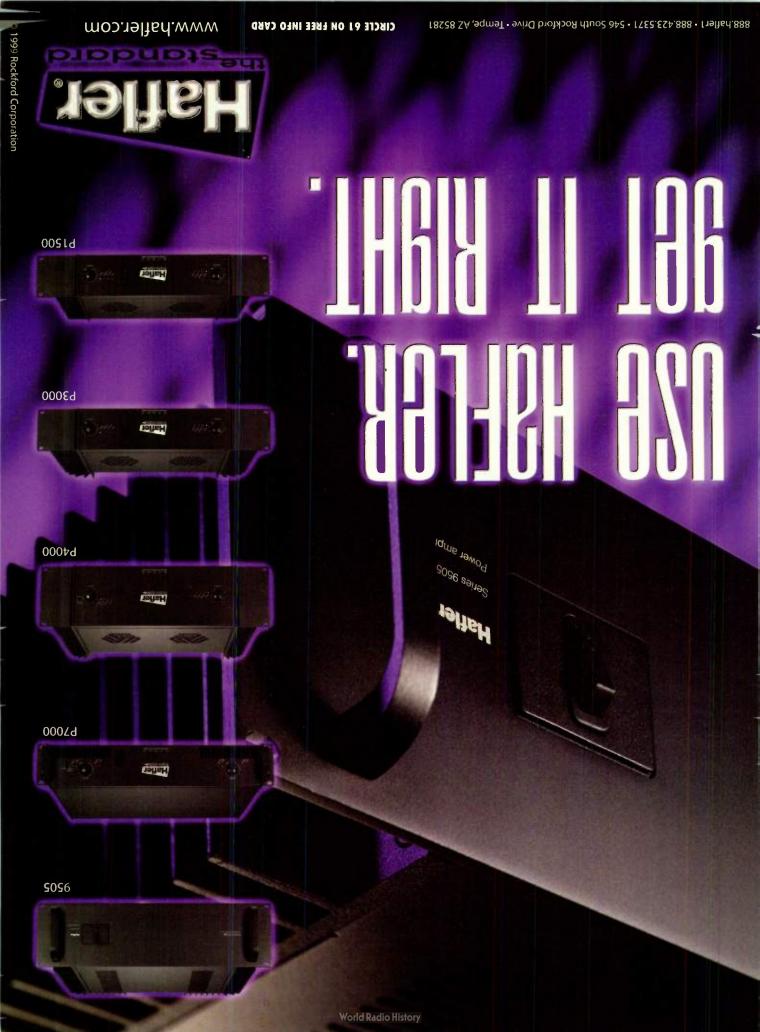
I just liked it. I've always liked all kinds of music. I was a Beatles fan when I was a kid, then I was a Beethoven fan and from there I went to jazz, progressive rock, and back into pop again. But there was a point when I studied composition at the Royal Academy of Music and became very alienated from the classical world, because there wasn't really a place in that world for me as a composer. More and more, I identified with the pop world and that's where I wanted to be. So I started writing pop songs.
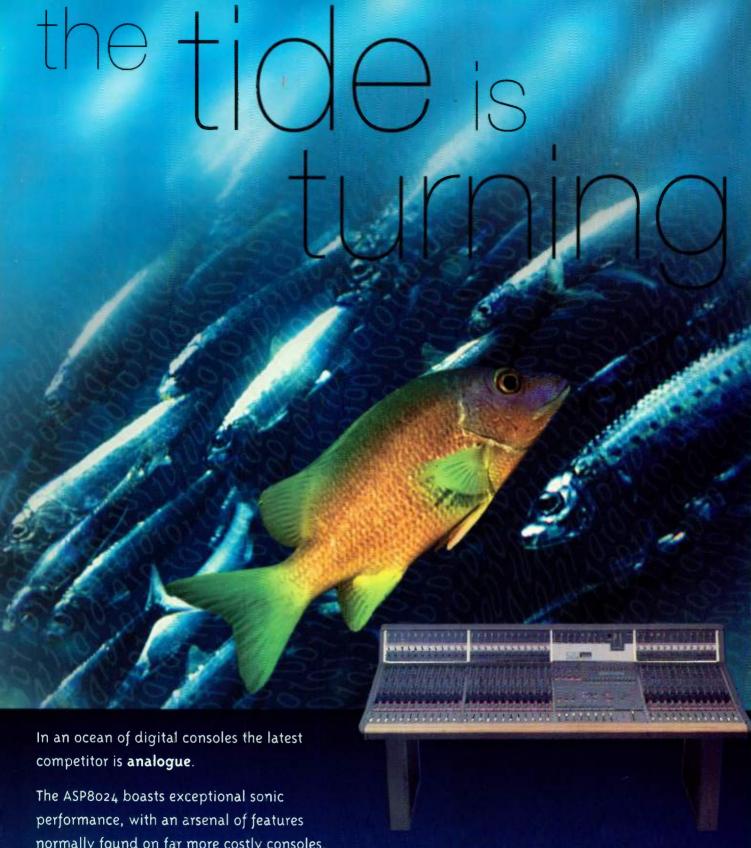
Yet you held a foot in the 'serious music' with your soundtrack work and Symphony In One Movement...

Yes, I never stopped studying scores or learning. About three years ago, I had a teacher at Julliard who taught me counterpoint for about a year. At the Academy, I wasn't particularly good at counterpoint, so I went through the whole thing all over again. I'm much better at it now.

Do you find this feeds your creativity?

Yes, I think everything helps. It seems to me that in the world of music, there are rather artificial barriers everywhere which people seem to be attached to. People ask "How can you write pop songs and then go write a symphony?" as if it's the most bizarre thing they've ever heard. They can't believe it. Well, Mozart and Schubert did it. Bernstein wrote symphonies and also wrote a Broadway musical. Gershwin wrote pop songs and piano concertos. Duke Ellington as well. I'm not comparing myself to them or anyone else for that matter, but those are the people I admire and aspire to be like in some way or another. I don't see why musicians shouldn't be more diverse. I'm sure that the musical abilities I've developed through studying (for example) scores of symphonies and making piano transcriptions of symphonic pieces were used in writing songs as well. It all feeds into the whole picture. We live in a world directed by marketing. Everything has to fit in a demographic and be labeled. But that's just not me, nor anyone trying to be a real artist. I'm doing what seems right to me. I'm going with my instincts. That's what I have always done.

TELLING IT LIKE IT IS: The author (right) goes under-cover after revealing stupidity in sound reinforcement.
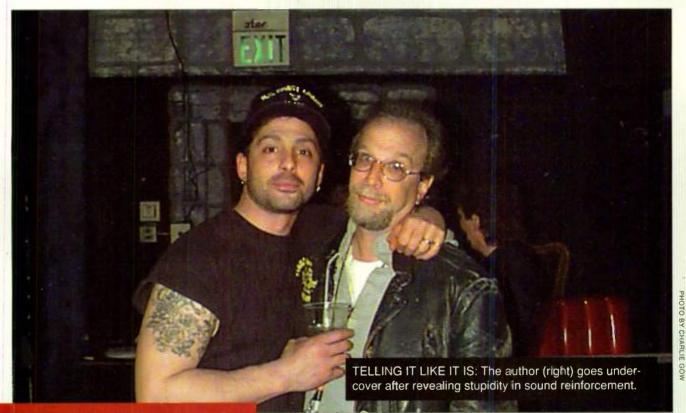
MOMENTS
OF CLARITY

BY ART RING

Ahhhhhhh-right! EQ *swore they'd never allow a squealer like me to get in their pages again, but I'm back! Well, as long as I'm here, I figure I may as well try to do something worthwhile for those of you who need a moment of clarity. This time it's all about advancing the gig. You can ask all the questions you want about the PA system when it's a promoter-provide, but if you don't ask the right questions, you're in big trouble. Sounds simple, right? Yeah. Read on.*

Obvious question: How many channels does the console have?
Always your first one on the list. Follow it up with: *How many of those channels actu-*

ally work? (make that Stupid Question number 1). With a bit of luck, you'll only run into busted channels once in a while. Don't forget to make sure there's enough inputs to accommodate your effects returns and the CD player you need for show cues.

Now here are the *stealth* questions you need answered. Try not to laugh too hard because these are way too silly for me to have made up. SQ number 2: *How many of those channels have mic inputs?* This one comes from that gig in Pocatello, ID where we advanced a 32-channel console. Our first surprise came when we found out that eight of those channels were stereo. line-level-only inputs — leaving us only 24 mic inputs. We realized that our next SQ should have been: *How many snake channels run from the stage to the mix position?* (make that number 3). Imagine my ulcer when we learned the snake had only 16 channels and four returns. After a lot of head-scratching, the PA company

rounded up another 12-channel snake to help make up some of the difference.

Obvious question: How many channels does the monitor desk have?
Sneaky PA companies love this one. Be on the alert when they tell you the monitor desk has the same number of channels as the house console. It could be because the house console *is* the monitor console. What?! No one told you we were mixing monitors from the house position? Good grief. Stupid question number 4: *Is the monitor desk separate from the house console?*

Once you've established that there actually is a separate monitor console, the real information cascades forth to your brain when you ask: *How many channels does the snake split have?* (SQ number 5).

Ten stupid questions you need to ask when advancing a gig

Remember that gig where we advanced a 32-channel console, got 24 channels with only 16 mic inputs and a 16-channel snake? You guessed it — the splitter snake only had 16 channels. Now what are we supposed to do with the other eight channels that need to be in the monitor mixes? I tried to tell the bass player that he's just a frequency and doesn't need to be in the monitors, but he told me... well, *EQ*'ll never print what he told me. Anyway, make sure you're getting a split with the requisite number of channels.

Obvious question: How many compressors and gates will we have?

When you're asking for eight compressors and eight gates, you gotta be *specific*. Everyone out there who has a Behringer Composer, dbx 166X, or Presonus ACP88, raise your left hand. Now put your right hand on the Sound Reinforcement Bible and repeat after me: "I promise not to mislead anyone into thinking that — just because these units have a gate and comp on each channel — it amounts to eight separate comps and eight gates." It doesn't! We're supposed to get eight gates

and eight comps so that we can put them on different channels. You shoulda' seen one house guy's face light up during *that* moment of clarity. SQ number 6: *Are those gates and comps separate?*

Obvious question: How many microphones do you have?

This one is really tricky because a club may have all the mics you want. Ahhhh-right. But they never told you that they only have 25 *cables* (hope they all work...). Another day of sittin' around, waiting for a runner to return from the local music store with an extra 12 cables. It holds up soundcheck for way too long, and I don't wanna be the one to tell the band soundcheck is being held up because there aren't enough cables. It's bad enough when they don't get the cookies. Besides, these band members have really important things to do — like go back to the hotel and watch TV or nap. So SQ number 7 is: *Do you have working cables for those mics?* While you're at it, don't forget the kicker, SQ number 8: *How many microphone stands are available?* ("Well, we have a lot of Z-clamps..." Oh boy. Try miking a rack tom with a Z-

clamp.) Asking this one might save the runner a trip — he can get the extra cables *and* mic stands at the same time. Plus it makes you a hero with the band because they'll get an extra hour to watch TV.
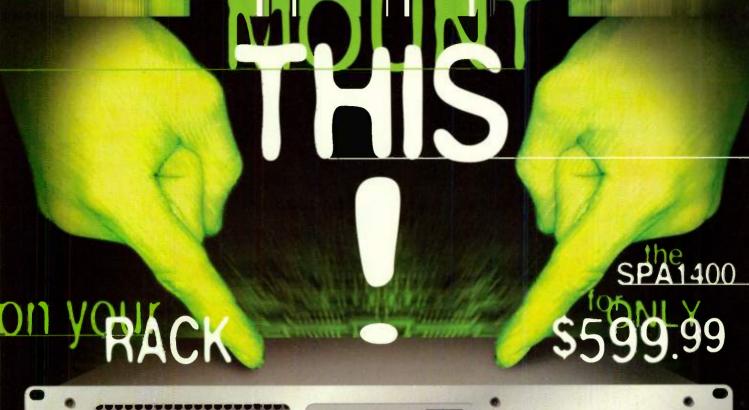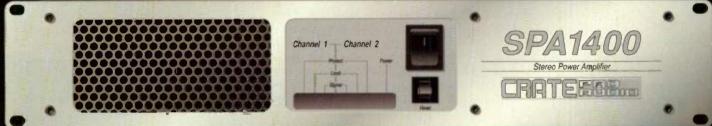
Obvious question: How big is the stage?

"Big." Ahhhhhh-right!. "Bigger than last time." Oh, that really helps. Now we might have room to put the entire backline on the stage. "Thirty-five by forty-five feet." But someone didn't tell you that the Molson girls get 10 x 10 on each side of the stage to sell beer. It's not that we *mind* the Molson girls trying to earn a living... SQ number 9: *How much of that space is actually usable for gear?* If it's a bill with co-headliners, don't neglect SQ number 10: *Are there multiple risers available so the crew can pre-set drums and keyboards while another band is playing?* During set change you can just roll the drum and keyboard risers out on stage and won't have to do all the setup during the 15 minutes your crew is supposed to be able to do it in. "Oh, you wanted *wheels* on those risers?"

Good grief. Gotta go... **EQ**

ROAD GEAR

A HEAD ABOVE THE REST

Nady, which pioneered the first headworn mics on the market with the original Nady Headmic over 15 years ago, introduces the HM-1 Headmic. Designed for high-performance live stage vocals, the HM-1 features a miniature, highly sensitive dynamic element that delivers close-up high gain-before-feedback operation and a 100—15 kHz frequency response for a full transparent sound without irritating background noises or loss or masking of vocals. Ergonomically designed for maximum comfort and durability, the HM-1 is ultra lightweight and is available for use with either hardwired or wireless microphone transmitters. For more information, call Nady Systems at 510-652-2411, fax them at 510-652-5075, or visit www.nadywireless.com. Circle EQ free lit. #126.

KEY PERFORMER

Yamaha introduces the S80 Synthesizer keyboard. Featuring an 88-note, weighted keyboard action — combined with a high-quality, expandable synthesis engine — the S80 is ideal for live performance applications and makes a perfect controller/synthesizer to place at the heart of a complex MIDI system. The synthesis engine at the heart of the S80 provides 64 notes of polyphony divisible among 16 multi-timbral parts. In a significant advantage over the competition's 88-note synths, the S80 has 64 multimode resonant filters, allowing

for the creation of analog-style synth sounds. A massive 20 MB ROM, including sounds from Yamaha's award-winning EX series synthesizers, provides the raw materials used to create the S80's sizable collection of voices. Onboard sounds include stereo-sampled pianos, strings, brass, and choir voices. In addition to the 256 preset sounds, users can create their own sounds using the easy-edit macros. The suggested retail price is $1999. For more information, call Yamaha at 714-522-9011 or visit them at www.yamaha.com. Circle EQ free lit. #127.

COLISEUM CALIBER

Always keeping up with the customers' needs, Yorkville Sound introduces a number of new speaker cabinets to its already successful Coliseum Installation Series. The first of the new cabinets is the CM1260, a 150-watt wedge monitor featuring a 12-inch woofer and a 1-inch high-frequency driver in a side-by-side format. The new C2560 is a 300-watt enclosure with
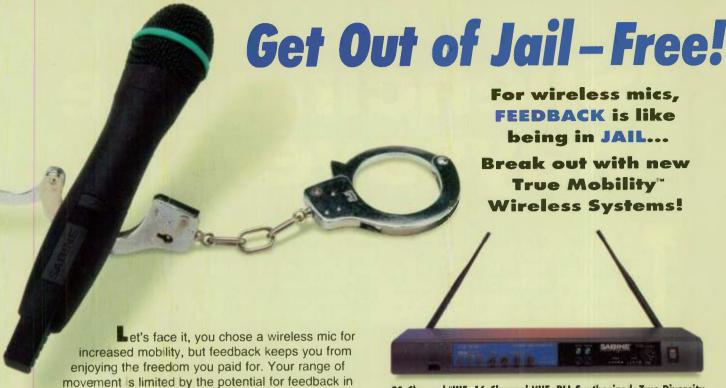
a 15-inch woofer and a 1-inch high-frequency driver mounted on a 60- x 40-degree wave guide horn. The 12-inch-based C2260 and C2290 feature all the characteristics of the company's existing C1260/C1290, but employs a larger woofer for greater output and enhanced low-frequency performance. With custom designed Eminence woofers gracing every Coliseum enclosure, each cabinet features white splatter paint finishes, curved white-painted metal grilles, versatile Flyware points, and binding post inputs. The prices are $439 for the CM1260, $499 for the C2560, and $499 for the C2260/2290. For more information, call Yorkville Sound at 716-297-2920, fax them at 716-297-3689, or visit www.yorkville.com. Circle EQ free lit. #128.

FLEC-TONE

ClearSonic Manufacturing introduces the FLECTOR 8 and FLECTOR 12 "Personal Monitor Discs." FLECTOR 8 and FLECTOR 12 are mic stand-mountable, 8- and 12-inch discs designed to reflect sound back to the musician so that they can hear themselves better. Also, the FLECTOR diffuses the narrow high-energy acoustical beam associated with brass instruments, making life somewhat bearable for anyone directly "in the line of fire." For more information, call ClearSonic at 330-650-1420, fax them at 330-650-1445, or visit www.clearsonic.com. Circle EQ free lit. #129.

EQDrive!

FIGURE 1: Schematic for a Foxx Tone Machine, pulled from the Effectronics site.

FIGURE 2: Four VST plug-ins downloaded from Maxim Digital Audio.

WEB SITE QUICK PICKS

BY CRAIG ANDERTON

Welcome to the fourth Web Site Quick Picks, with ten more sites that are well worth the connect time. Clear out some space on your hard drive for downloads, dial up your favorite service, and get ready to surf.

Gene6 (gene6.idegrif.com/index.html). Go here to download a freeware Windows memory monitor that lets you know just how much RAM those programs and plug-ins are *really* using. Unzip the file, run the program, and you'll see a little on-screen graphic that indicates the total percentage of RAM being used.

Effectronics (www.geocities. com/CollegePark/Library/1355/schematics.htm). There are lots of great guitar box schematics on this site. While you're there, also check out www.geocities.com/CollegePark/Library/1355/sites.htm, which has links to parts sources, other DIY-related sites, and lots more. Besides, where else are you going to find a page dedicated to spring reverb?

Google (google.com). There are search engines, and there are search engines. This cross-checks against other search engines and references, and I always seem to get much more relevant matches using Google than with the alternatives. It's still in beta, but works exceptionally well.

audioMIDI (www.audiomidi. com). This is an e-commerce site for computer-oriented musicians, but their demo section has lots of working demos for commercially available software, and seems to be updated conscientiously. If you want to "try before you buy," check this out.

PARIS Discussion Group (www.greatidea.com/PARIS). This independent discussion group also includes tips, software, links, and lots of information that is vital to users and abusers of the PARIS hard-disk recording system. While not affiliated with either Ensoniq or Intelligent Devices (the developers of the app), personnel from both companies often show up and answer questions.

Maxim Digital Audio l Freeware (www.abel. co.uk/~maxim/ index.htm). If you're looking for free software, this site from Paul Kellett offers VST plug-ins for Windows and Mac, a disk utility to read/write/format Akai S-series disks on your PC, drum synthesizer, audio analysis tools, and more. The stuff's free, it works, and it's good.

Ten more cool audio-related sites for your pointing and clicking pleasure

BITS
AND PIECES

BITHEADZ VOODOO DRUM SYNTHESIZER (MAC)

SUMMARY: Sophisticated virtual drum module with primitive sequencing.

BASICS: Voodoo is a virtual drum module with 14 on-screen pads (fig. 1). Each can load up to four velocity- or controller-switched samples (AIFF, SDII, DS-1, or audio sample CD import), and process them in several ways, including destructive digital audio editing (including flange, trim, delay, reverse, parametric, and shelving EQ), as well as "synth"-type modules (pitch, amplitude envelope, filter+envelope, LFO, and dual FX buses for the two onboard FX). You can trigger the drums via MIDI, VST 2.0's "ReWire," OMS, FreeMIDI, mouse, on-screen virtual keyboard, or sequenced MIDI patterns (either imported or created in Voodoo). Patterns can also be assembled into songs. You'll need a minimum 120 MHz PPC with 64 MB RAM (a Windows version is due out soon). Voodoo uses password copy protection.

More mini-reviews and software update news

PROS & CONS: Voodoo works best for setting up drum kits by loading and modifying samples; you can also control the synth-type parameters, in real time, by up to 16 MIDI continuous controllers (a Peavey PC-1600 or KeyFax PhatBoy is ideal). For sequencing, although you can record easily enough, editing is limited to an event list — there's none of the groove-oriented, real-time drum programming found in programs such as Koblo's Gamma 9000, Steinberg's ReBirth, etc. Nor can you use groove templates, or even quantize with swing, although you can filter notes for editing according to vari-

ous criteria. You can import SMFs, though (and there are enough editing options to tweak them if needed), as well as record a sequence to disk as an AIFF audio file.

BOTTOM LINE: While having a solid drum synth is welcome, the package would be much stronger if as much attention had been paid to creating rhythms. Use Voodoo to assemble your cool drum kits, but generate and edit your sequences elsewhere.

PRICE: $199

CONTACT: BitHeadz, 4400 Capitola Road, Ste. 202, Capitola, CA 95010. Tel: 831-465-9898. Web www.bitheadz.com.

EQ FREE LIT. #: 102

IK MULTIMEDIA GROOVEMAKER 1.1 (WINDOWS/MAC)

SUMMARY: Instant techno/jungle/house tracks for the vinylly-challenged.

BASICS: Priced like an impulse item, it's easy to dismiss Groovemaker as a toy for making instant techno tunes. However, for project studios, it can generate high-quality, genre-authentic, dance-oriented sound tracks that are eminently cus-

tomizable and usable.

On the live performance-oriented Groove screen (fig. 2), you assign loops to tracks from a collection of loops sharing a common tempo and style (a "song"). Eight songs come with the program. You can change track volume and pan, bring individual tracks or groups of tracks in and out, change loops on the fly (sync'd to loop boundaries), and even add real-time synth arpeggiation on top.

If you hit a particularly nice groove, invoking a marker lets you sequence it later using the separate sequencer screen. You assemble the marked grooves in any desired order, and can record the final result to hard disk. A third "virtual DJ" screen creates endless permutations of loops, which is hit-or-miss, but can make some good sounds.

PROS & CONS: The quality of the music depends on the quality and quantity of the loops, and, so far, "official" support — while of good quality — is limited to songs on the program CDs and three additional CDs of material ($99 each). You can import your own loops, although they must all be the same tempo to work properly. This can personalize the results, and you can change

BY CRAIG ANDERTON

FIGURE 1: The Voodoo screen. Note the graphic icons that illustrate which type of drum you've chosen for a particular pad.

FIGURE 2 (top): Groovemaker's futuristic interface may look intimidating, but it's easy to use.
FIGURE 3: The tablature and virtual guitar neck are in the background; the session drummer is about to be invoked. Note the Edit Menu's "deglitch" function, which cleans up MIDI guitar parts.

tempo (the loops follow along if they're the same length).
BOTTOM LINE: The interface is striking, and you can have a ball with GrooveMaker. Sure, the musical results are generic — but when a producer is looking over your shoulder saying, "I need a dance music track in five minutes," Groovemaker is up to the task.
PRICE: $69
CONTACT: Ilio, Box 6211, Malibu, CA 90265. Tel: 818-707-7222, 800-747-4546. Web www.ilio.com.
EQ FREE LIT. #: 103
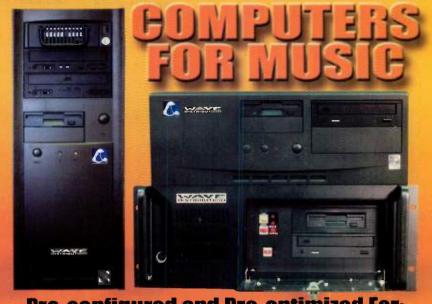
SAMPLEHEADS' DAVE SAMUELS MARIMBA AND VIBES
SUMMARY: Superbly recorded sample CD.
BASICS: If you want to stick a 5-octave rosewood marimba or 3.5-octave set of vibes in your sampler, look here. Both instruments are sampled chromatically, with 300+ MB of material on the audio CD. The marimba has soft, medium, and hard hits, as well as

TAKE A BETTER ACID TRIP: *TOP 20 TIPS*

BY CRAIG ANDERTON

Sonic Foundry's ACID busted all the traditional recording and sequencing paradigms, resulting in a truly original piece of software for the dance/groove musician. Perhaps its coolest aspect is the user interface, as it invites jumping in and getting started without even reading the online documentation.

However, lurking beneath that deceptively simple façade are some powerful and useful features, many of which are not mentioned in any documentation. If you want to enhance the quality of *your* ACID trip, turn on your lava lamp, tune out any distractions, and check out these 20 groovy tips.

How to use Sonic Foundry's software in ways you may never have thought

☺ **Group Therapy:** To group faders, click on a track, and Ctrl-click on any other track or tracks you want to control. Moving one fader will also move the fader(s) on the other track(s).

☺ **Getting Grounded:** Double-click on a track fader's knob to return it to 0.0 (or Center for panning).

☺ **The Zoom Power Trio:** You can zoom in three different ways: time, height, and "marquee" around any area of the track window. To toggle among these, while holding down the left mouse button, click on the right mouse button.

☺ **ACID Needs its Personal Space:** Make sure that the drive where ACID saves its files is not running out of space. ACID goes on a bad trip when it tries to save and there's no place to do it.

☺ **Check Out Those Weird Sounds:** Select a track's Properties tab and double the number of beats in the loop to create slowed-down sounds, or halve the number of beats for sped-up effects. The sound may be lo-fi, but that's part of the fun.

☺ **Living in Harmony:** To do "intelligent harmonization," break a segment into smaller segments and add different amounts of pitch shifting as required. For example, to transpose some sections up a major third and others up a minor third, first isolate each segment that needs a particular harmony. To do this, click on the segment beginning, right-click on the segment, and select Split at Cursor. Similarly, click on the segment end, right-click on the segment, and select Split at Cursor again. Select each segment individually and choose the appropriate harmony (fig. 1). Speaking of which...

☺ **Faster Pitch Shifting:** The quickest way to pitch-shift a segment up or down is to click on it, then use the – or + keys of the keyboard's *numeric* keypad to shift down or up, respectively.

☺ **Wow, Look at All Those Colors:** There's no real advantage to using 16- or 24-bit color, so use 256 colors for speedier operation.

☺ **Make Sure You're Saved:** Always Save As... in the appropriate file format. A standard ACID project saves only the set of directions that tells other files (samples) what to do. These files may be scattered over several drives, or even CD-ROMs. If ACID can't find the files when loading the associated ACD project, it will ask you to locate them. The save as "ACID Project with External Audio" option saves the ACID project file

FIGURE 1

and all associated audio files individually. Typically, when I'm ready to mix, I open up a new folder and Save with External Audio to this folder so everything is in one place. Saving as "ACID Project with Embedded Audio" melds the audio and ACD project together into one, big file that's useful for file transfers and backup.

☺ **File Comparisons:** To compare a newer version of an ACID file with an older version, double-click on the ACID program icon and call up one file, then double-click on the ACID program icon again, and open the file you want to compare (fig. 2). You can't copy and paste between the two sets of files or play them simultaneously, but you can do visual comparisons.

☺ **Adding Speed to ACID:** To speed up adding volume or pan envelopes to multiple segments, group the desired segments by clicking on the first one of the group, then Ctrl-clicking on segments to be added to the group. Now right-click on any of the selected segments, and select Volume or Pan envelope. This adds an envelope to *all* segments in the group, even if they're in different tracks.

☺ **All Together Now:** To mute (or solo) all tracks at once, focus on the track windowpane (click on one of the tracks, for example). Then go to Edit Menu > Select All, or type Ctrl-A. This selects all tracks. Click on mute (or solo). The same basic approach also works for changing output assignments for all tracks. This can be very useful if you have a multichannel digital I/O board.

☺ **Slow Groove:** If you create your own loops for use in ACID, recording them around 100 BPM allows for the widest possible tempo range variation. It's easier for ACID to speed up audio than slow it down.

☺ **Getting Totally Looped:** Loop recording is a little tricky. Set your loop start and end, enable looping, then click on Record. Each take is recorded on the same track, one after the other. For example, if a 16-measure loop starts at measure 1, then the second take will be in measures 17–32, the third in measures 33–48, etc. Isolate the desired take with the loop locators, solo that track, then Mix to New Track. Now you have a file with only that take; drag it into place and you're done.

☺ **Distorted Reality:** When you mix down an ACID project to a WAV or AIFF file, make sure that combining all the tracks together doesn't create distortion. To check this, select the Mix tab and see whether the tops of the meters show a red rectangle, which includes a number that indicates the amount of overload (in dB) above 0. To eliminate the distortion, reduce the associated mixer faders by an equal amount (as indicated by the numbers next to the faders).

☺ **Going With the Flow:** Try to place tempo change markers so that no segment straddles the change. Tempo changes should occur between segments.

☺ **Peace, Love, and Loop Creation:** ACID makes a great digital audio sequencer: create your own loops by snapping several one-shot samples — kick, snare, hi-hat, etc. — to the rhythmic grid. Set loop points around the loop, then use Mix to New Track to save the loop as a WAV file.

☺ **Stretch Your Bass:** With low-frequency loops (like bass), select the Track Properties tab, then the Stretch tab, and, under Stretching Method, choose Pitch Shift Segments. This can create a smoother sound when transposed, but sometimes changes the pitch.

☺ **Going Through Some Changes:** If you use the vector-based ("rubber band") editor for panning, set the main track pan to Center and do *all* your panning with rubber bands. If the track pan setting is off-center, then panning with the rubber bands will also change the level during panning.

☺ **Audio Hallucinations:** To add doubling and chorusing effects with hard-disk tracks, duplicate the track (right-click on the track name and select Duplicate Track), turn off the snap function, and drag the duplicated track forward or backward by the desired amount of delay.

Bad acid jokes notwithstanding, I hope you find these 20 tips useful in your own work. If you want to see more ACID tips, let me know. I have a ton of 'em! **EQ**

FIGURE 2

Where The Next Hot Band Is Coming From.

Right here.

Via *your* PC. Where you can record, edit and mix with MxTrax™ software running a Yamaha® DSP Factory™ card—for an instant professional-quality recording studio.

Really.

Just your PC, Yamaha DSP Factory and *our* MxTrax with signature features like drag-and-drop mixer assembly and tool-free editing.

MxTrax is extremely powerful, yet surprisingly easy to learn and use. And one more surprise. Very inexpensive.

So now that you're ready, just take that next step toward achieving hot band status. Check out MxTrax at your local Yamaha DSP Factory dealer *and ask about our limited-time special offer.* Or, contact us at Minnetonka Audio Software: 612-449-6481 or, at our website: www.minnetonkaaudio.com.

Minnetonka AUDIO SOFTWARE, INC.

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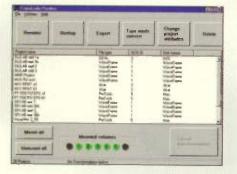
DRIVE! GEAR

WHAT'S YOUR TYPE

TimeLine Vista, Inc. introduces TransAudio PipeLine, a file utility program that supports backup, export, and tape mode conversion of popular digital audio workstation file types. The program is designed to aid postproduction facilities that must deal with archiving

and converting digital audio files created by some of the most popular digital audio workstation systems. TransAudio PipeLine provides the ability to read several different digital audio file types, including Digidesign Pro Tools Sessions, WaveFrame Projects, Akai DD8, TASCAM MMR-8, Zaxcom DEVA, Sonic Solutions, and OMF/Sound Designer II. The program runs on PCs using Windows 95/98/NT. The list price is $795. For more information, call TimeLine Vista at 760-761-4440, fax them at 760-761-4449, or visit www.digaudio.com. Circle EQ free lit. #130.

VIVA VEGAS PRO

Sonic Foundry has initiated the worldwide launch of Vegas Pro, its technologically advanced, nonlinear multitrack media editing system for Windows-based PCs. The Vegas Pro system features a unique, multi-threaded architecture designed to squeeze "over-the-top," real-time performance from a standard Windows PC. With Vegas Pro, users can perform nondestructive edits during playback and run multiple plug-in effects, all in real time. Vegas Pro's complete flexibility and enhanced media handling also allows the mixing of file properties, bit depths, and sampling rates. Minimum system requirements include Microsoft Windows 9X/NT 4.0, a 200 MHz Pentium processor, 32 MB of RAM, 20 MB of available hard-disk space, a VGA display, a CD-ROM drive, and a Windows-compatible sound card. For more information, call 800-57SONIC, fax them at 608-256-7300, or visit www.sonicfoundry.com. Circle EQ free lit. #131.

MEASURE UP

AudioControl Industrial introduces an innovative, two-channel measurement mic preamplifier, the MP-200. The MP-

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SeaSound

Finally, a Computer Recording Interface

Solo
24 bit/96kHz Digital Audio System

Trim • Mon Pan • Rec
Inst • Level
Mic • Preamp 1
Phantom Power • On

Trim • Mon Pan • Rec
Inst • Level
Mic • Preamp 2

Mon Pan • Rec
Level • Line 1 • Rec

- Two custom mic pre-amps with 48v phantom power
- Two high-z instrument pre-amps
- Two line inputs, inserts, direct outputs & aux. inputs
- Built-in mixer with two headphone amps
- Signal activity, clip indicators & 10 segment VU meters
- Optional footswitch controller
- 24-bit/96kHz A/D & D/A converters
- Mac or PC compatible PCI card included
- S/PDIF IN and OUT, all standard sample rates
- MIDI IN, OUT & THRU
- ASIO, Sound Manager & WAVE drivers
- All at the price of a pro sound card!

200 is compact and designed to work with computer soundcard-based measurement systems. It is battery powered for portable use. The MP-200 includes a switchable input gain, variable output pink noise generator, 94 dB acoustic SPL reference light for calibrating computer programs and other level meters, and a calibrated microphone. One of the unit's channels is mic level only, while the second may be switched between mic or line level. The MP-200 is priced at $349. A second mic is available for $145. For more information, call AudioControl at 425-775-8461, fax at 425-778-3166, or visit www.audiocontrol.com. Circle EQ free lit. #132.

A BLANK CANVAS

Edirol has announced the release of the Roland ED Sound Canvas SC-8850, the world's first musical instrument to offer a USB (Universal Serial Bus) connector. The SC-8850 connects directly to the USB port on Windows 98 PCs, Apple iMacs, and the new blue-and-white Macintosh G3 towers. The module can play up to 64 musical parts (MIDI channels) simultaneously. Because the unit's dual MIDI ports remain active when it is connected via USB, desktop composers can also use the sound module as a MIDI interface, accessing an additional 32 MIDI channels on external instruments. The retail price is $1195. For more details, call 800-380-2580 or visit www.edirol.com. Circle EQ free lit. #133.

SAY HI TO PLUGGO

Cycling '74's Pluggo is a collection of 74 VST plug-ins that also allows signal processing patches created with MSP to work as plug-ins. The effects include synchronization, delay effects, filters, filter/delay combinations, pitch effects, distortion, granular synthesis, spectral modification, visual display, meta-plug-ins and audio routing, sampling and synthesis, dynamics and reverb, sound localization/panning, and parameter modulation. Pluggo is available as a stand-alone product for $74, as well as part of Cycling '74's MSP, which is a set of DSP extensions to the MAX graphical programming environment. For more info, visit www.cycling74.com/pluggo. Circle EQ free lit. #134.

You Can Get Your Hands On!

Solo includes Steinberg Cubasis AV, Sonic Foundry Acid Rock & an iSong Power to Play Sampler CD ROM.

EQGroove

FIGURE 1: FreeFilter at work, running as a DirectX plug-in within Steinberg's WaveLab.

LEGALIZE IT!

BY CRAIG ANDERTON

Recently, Cologne superstar remixers Dr. Walker and Joker remixed two of my tunes. I was ecstatic with the results, but three of the loops they used had come from vinyl, and had not been cleared with respect to copyright. I really wanted to use the remix in my upcoming loop library for ACID, so I figured I'd play something myself that captured the feel of the loops, and substitute them for the "illegal" loops.

So far so good — but after finishing the loops, even though the notes worked just fine, the loops didn't. That's when I realized that part of the reason for adding other loops is because they provide timbral variety. Anything I create in my studio is always going to have a certain sound because from one day to the next I use the same basic selection of mics, the same mixer, the same guitars, and so on. Loops created elsewhere are going to have a completely different "vibe." Was there any way I could capture a bit of the vibe in the "legal" loops I wanted to create?

THE FREEFILTER CONNECTION

One of the illegal loops was a simple be-bop-style ride cymbal beat, with a walking acoustic bass in the background — very common stuff. However, the loop sounded as if it had been recorded in a closet using a dynamic mic that had been dropped a few times, which helped provide its deliciously funky charm. The loop I played didn't sound like that at all.

Fortunately, Steinberg's FreeFilter is the perfect tool for impressing one file's timbral characteristics onto another file. It works by "learning" a reference file's spectral response (in this case, the funky loop), then analyzes the "target" file, and notes the difference in spectral response between the two files. Next, it comes up with whatever frequency response curve is required to have the target file's timbre mimic the reference file's timbre, and the process works so well it's scary. After being "freefiltered," my ride cymbal and bass loop sounded uncannily like the original loop, even though several of the notes were different.

Fig. 1 shows FreeFilter at work. The yellow line toward the top of the FreeFilter window shows the curve that was generated to make the target file have the same spectral content as the reference file. The lower green and red waveforms show the pre- and post-processed spectral response of the target file. Because FreeFilter is also a 1/3-octave equalizer, if you want to tweak the timbre further, it's no big deal. For example, one "illegal" loop had a prominent peak at about 6 kHz, which was just too much. Pulling down the slider that correlated to 6 kHz on the target file cleaned that right up. There's also a "morph" slider that can reduce, or in-

When you don't have time to deal with copyrights for a loop, redo it yourself

FIGURE 2 (top): Two files before dynamics tweaking. FIGURE 3: The file on the right has been tweaked so that its dynamics resemble the reference file more closely.

crease, the influence of the reference file on the target file. This is helpful if you want a result that's part original response, part "freefiltered" response.

THE DYNAMICS CONNECTION

Once the timbre was squared away, the sound was almost there. But there's one other element that needs to be considered: dynamics. In fig. 2, the file on the left (down02) is the reference file, while the file on the right (Big Beat 4) is the target file. It's easy to see that the reference file is very tightly compressed and has less dynamic range than the target file. Calling up Sound Forge's compressor and applying it to the target file created dynamics characteristics (fig. 3) that were much more like those of the reference file.

Generally, the most accurate emulation seemed to come from doing the dynamics first, then applying FreeFilter. Even if you played some pretty different notes while creating the "legal" loop (and since you're redoing it, you might as well tailor it to your tune), tweaking the dynamics and spectral response produces result that sound amazingly like the original file.

THE FINAL WORD

Redoing loops saves you the hassle of dealing with copyright clearance, and the above techniques help retain the timbre that may have been one of the main reasons you wanted to use the loop in the first place. Check this approach out sometime; you might be amazed at how effective it can be. **EQ**

ILIO
SAMPLE CDS

Ilio doesn't produce or distribute a huge number of sampling CDs, but they all meet high standards for quality, consistency, and the ability to translate well to different sampler formats. All products reviewed here are available in CD-ROM for Roland, Akai, Kurzweil, and SampleCell samplers (Ensoniq and EMU samplers can usually translate from other formats). WAV file versions are not available. Except for *Symphony of Voices*, the CD-ROMs include an audio CD for quick auditioning — an excellent idea (audio CDs are also available separately).

A sampling of four sample titles from Ilio

Regarding licensing, the basic deal is *only you* are licensed to use the sounds in *only your* compositions. Ilio warrants that their sample CDs are 100 percent "copyright clean." Okay, let's load 'n' listen...

DISTORTED REALITY 2

This two-CD set from Spectrasonics is one of those rare instances where the sequel surpasses the original, and considering how good *D.R. 1* is, that's saying a lot. The "Beats" section contains extremely creative, electronic-sounding loops at tempos from 50 to 200 BPM (with one at 480). They're ideal for layering over more conventional loops, or serving as the foundation for less-conventional compositions. The four "Groove Menus" (available only on CD-ROM versions) each play a relatively lengthy riff every 5 BPM, from 50 to 135 BPM. There are also gorgeous ambiences, beds. electro-bells, drones, weird semi-distorted stuff, hits, alien vocal sounds, mallets, pads, special effects, sweeps, synth bass, and more — even a folder of "bonus" sounds from *D.R. 1*. Spring for the CD-ROM; there's more material, and as this is the kind of disc you will probably use a lot; having everything keymapped and ready to load is a real time-saver. CD-Audio, $99; CD-ROM, $199.

BY CRAIG ANDERTON

TRANCEFUSION

This is the "un-drum loop" CD. It provides drones, swells. transitions, and the other types of sonic "glue" that hold dance music together. The disc starts with both long and short swells (reversed-type effects that rise, then cut off abruptly); next, it moves to long and short "carry-overs" that mostly fade in, and then out, to provide transitions. "Lasers and stabs" are much more evolved than similarly named sounds typically found as synth presets, while "percussive sweeps" resemble a high-speed LFO modulating a pitch sweep (think pitch transposer with predelay on the pitched sound, and tons of feedback). There are also some wigged-out percussive "snare fills" (not quite what you'd expect, but cool), reversed cymbals, lots of arpeggios in various keys and tempos (a gold mine for ACID aficionados), two "delirium" sections (weird FX, semi-human sounds, etc.), and several "textures" (complex sounds that beg to be put into movies — when Indiana Jones opened up the Ark of the Covenant, several of these would have worked just fine). CD-ROM, $199. (SampleCell format is currently not available, but slated for release soon.) CD-Audio, $99.

FINGERSTYLES

You're scoring a promotional video called *Bed and Breakfasts of the Scenic Northwest.* As the camera pans across beautiful Puget Sound, you need a sprightly 45-second acoustic guitar ditty while the announcer introduces the next B&B. Meet *Fingerstyles,* an acoustic guitar CD with 38 compositions (each in a particular key and tempo), and typically deconstructed into loopable patterns, melodies, endings, and strums (some compositions omit one of these categories). The idea is to assemble these elements into a particular arrangement, and, within those constraints, the con-cept works surprisingly well. However, you can't really stretch too far, as you typically have only two to four chord changes to work with; there were often times when I would have been willing to trade-off fewer compositions for more available variations. Overall, *Fingerstyles* isn't particularly versatile, but it does provide quick, canned acoustic guitar parts that are well-played and ready to roll. CD-Audio, $99; CD-ROM, $199.

SYMPHONY OF VOICES (VERSION 1.1)

Spectrasonics's ambitious five-CD set covers choirs (men, women, both, and boys), classical soloists, Gregorian chant, layered pop vocal stacks, Latin phrases, truly spooky choral effects — even whistling, shouting, and laughing. Most of the samples were recorded in acoustic spaces with real reverb; while this theoretically limits your flexibility, in practice, the ambience sounds great. Also, in some cases, chords were recorded as a single sample because of the sonic difference compared to playing multiple sampler notes. The pop stacks are the result of overdubbing singers multiple times, giving that glassy, smooth, pop vocal quality. Although the human voice is extremely difficult to sample, *S.O.V.* is truly awe-inspiring. While it's a "specialist" among sample CDs, if you need realistic-sounding, tastefully recorded voices with variety and accuracy, look no further (besides, after investing the time to audition all five CDs, you won't *want* to look further!). *S.O.V.* seems ideal for the film and video post crowd, although these sounds are equally well-suited for straight-ahead musical applications. In any event, they're exceptionally good. CD-ROM set, $499.

MANUFACTURER: Ilio Entertainments, Box 6211, Malibu, CA 90265. Tel: 818-707-7222; 800-747-4546. Web: www.ilio.com. Circle EQ free lit. #135.

GROOVE GEAR

NEXT! IN LINE

next! introduces the Millennium Series mixers. The PRO-80 and PRO-100 are packed with the latest features. The PRO-100 is for DJs who demand more from their mixer. It features an effects loop to add outboard DJ effects such as a DJ sampler. The effects loop can be assigned to any of the four channels on the PRO-100. The mixer features balanced XLR outputs for better sound quality. Another DJ-friendly feature is the cue pan crossfader, which is assignable to the headphone or zone outputs. The PRO-80 is designed for the more budget-conscious DJ. It includes a 3-band EQ and Trim on each channel; Master, Zone, and REC outputs; Cue section with PFL function; a fully assignable crossfader; and a 12-volt BNC lamp input.

For more information, call Stanton Magnetics at 954-929-8999, fax them at 954-929-0333, or visit www.stantonmagnetics.com. Circle EQ free lit. #136.

GET INTO THE GROOVE

Cakewalk has signed a licensing agreement with Italy's IK Multimedia Production to exclusively distribute GrooveMaker and GrooveMaker MAX in North America. GrooveMaker, the loop-based mixing software for creating electronic dance music, is the world's only real-time, remixable groove generator. GrooveMaker runs on both Mac and Windows PCs, and can be used as a standalone application . The MAX edition includes two additional loop library CD-ROMs for use with GrooveMaker. Real-time mixing allows musical samples to be loaded and mixed "on the spot"

without interrupting the groove. Users can create electronic music in a range of styles, including Techno, Trance, Dub, Jungle, House, Drum 'n' Bass, and Progressive. A wide variety of instrument samples provides unlimited variations and styles. Audio export options allow GrooveMaker projects to be incorporated into other software products, such as Cakewalk Pro Audio, Home Studio, and Metro. The Groove-Maker software is available for $59. The MAX version is $99. For more details, call Cakewalk at 888-CAKEWALK or visit www.cakewalk.com. Circle EQ free lit. #137.

GO WIRELESS

Gemini Sound Products has launched the UX Series of UHF wireless mic systems. Designed for DJs and musicians, the UX Series features a range of three receivers — the single-channel UX-801, the true-diversity UX-810, and the dual-

channel UX-820. All three models offer auto-mute circuitry that tracks the signal to help eliminate popping and noise when out of range. Other features include AF (Auto Frequency) level control, 1/4-inch output jack (two on the UX-820), and adjustable squelch control. The systems are available in eight different factory-preset frequencies. The UX Series is offered with a choice of three beltpack transmitters — the UB-82L with lavalier mic, the UB-82H with headset mic, and the UB82G for electric instruments. All feature quartz lock control to lock in the signal for more stable transmission, adjustable sensitivity control, and a LED battery indicator. Call 800-476-8633, fax 732-969-9090, or visit www.geminidj.com. Circle EQ free lit. #138.

WORK HORSE

Korg USA's TRITON Series Music Workstation/Samplers feature Korg's HI (Hyper Integrated) synthesis system. The series includes the 61-note TRITON, the 76-note TRITONpro, and the 88-note (weighted action) TRITONproX. All TRITON Series instruments include an integrated sampler, enhanced performance capabilities, a sequencer, and a wide selection of user-installable options. The onboard sampler enables users to record their own samples, or load in data from a number of popular formats. The units' 48-kHz/16-bit linear, mono/stereo sampling functionality provides 16 MB of sample memory, allowing approximately three minutes of monaural sampling. The memory is user-expandable to a maximum of 64 MB, for up to 11 minutes and 38 seconds of mono sampling. Stereo or mono AIFF files, Windows WAVE files, Akai S1000/S3000 (floppy or CD-ROM), and Trinity format data can also be loaded into sample memory. For more information, call Korg at 516-333-9100 or fax them at 516-333-9108. Circle EQ free lit. #139. **EQ**

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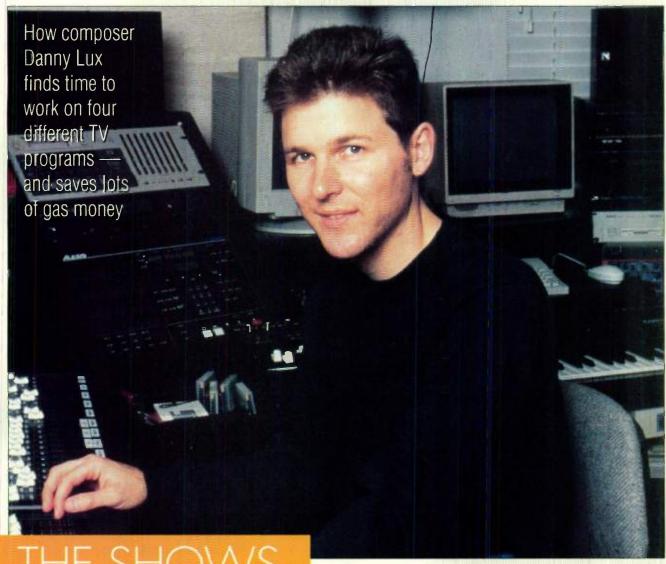
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How composer
Danny Lux
finds time to
work on four
different TV
programs —
and saves lots
of gas money

THE SHOWS
MUST GO ON

*Music composer Danny Lux has been
scoring the hit television series* Ally
McBeal *on a weekly basis since the*

BY STEVE
LA CERRA

*show's first episode.
Because Lux's sched-
ule is also filled with
scoring activities for
the TV shows* Profiler,
Sliders, *and* Sabrina the Teenage
Witch, *time is of the essence — espe-
cially under such deadline-oriented
projects. Lux produces music for these
shows in his project studio, and he's
worked out a clever method of using*

*his room to make the most of both his
time and his equipment. Danny re-
cently took some time to speak to* EQ,
*offering some insight on his compos-
ing/recording method.*

DO YOU SEE SPOTS
BEFORE MY EYES?
Danny first sees any new episode of
Ally McBeal (or *Profiler, Sliders,* etc.)
at a "spotting session," where he, one

of the show's producers, and a music editor watch the episode together while deciding where the music cues will go, and what feel those cues will have. At this point, the show has been "locked" — in other words, the visual part has been edited and (generally) will not change. According to Lux, "We'll get specific down to the frame about which scenes have music, and where the music comes in and goes out. Beyond that, I'm left on my own fruition to write the cues as I feel appropriate. They usually give me between four and seven days to turn it over."

Sounds like plenty of time, right? Well, yeah, if you don't have anything else to do. But last season Lux had five series to score, which, for him, typically meant five spotting sessions per week. Though the spotting sessions themselves may only last an hour or two, Danny estimates that he lives "about 45 minutes to an hour from the studios. I spent a good portion of last summer trying to figure out a way of not having to drive in for all these meetings. I basically came up with a way to do a live video conference. As soon as the show is locked, they messenger me a videotape of the show that I digitize in my computer (an Apple PowerPC 9600) using Adobe Premiere and a MiroMotion DC30 video card. The video file runs about 900 MB for an episode, which is stored on a Jaz® cartridge. When we're ready to spot, we use these Gentner Microtel telephone interfaces that essentially have audio inputs and outputs, and connect to a standard telephone receiver

> When we're ready to spot, we use these Gentner Microtel telephone interfaces that essentially have audio inputs and outputs, and connect to a standard telephone receiver jack.

jack. The producers connect the SMPTE output of their master VCR to an audio input of their Gentner box. They call a phone here at my studio where I receive that SMPTE through another Gentner interface. Then I take the audio out of the interface — which now has SMPTE — and I run the SMPTE code into a channel of my console.

"Sometimes I have to EQ the code so that my system recognizes the SMTPE for what it is, and sometimes I don't need the EQ. It's interesting — *Ally McBeal* and *Profiler* never need the EQ. When I was doing *Sliders* — which came from Universal — it always needed tons of EQ, and it wasn't consistent on a week-to-week basis. There's definitely something going on there in the phone lines! But once I have the setting, I store it on the 02R and it works flawlessly." [*At all times, the SMPTE remains in the analog domain —Ed.*]

With a SMPTE link to the producer's VCR established and the new *Ally McBeal* episode living on a Jaz disc in his Power Mac, Lux is almost ready for the video conference. Ultimately, Danny will compose and record the music cues for *Ally McBeal* in Emagic's Logic Audio Platinum. So he opens a new song in Logic and then uses the "Options > Movie..." command in the arrange window to open the video file of the episode as a QuickTime movie within the new song. Logic's sync source is set to "external," and is now awaiting timecode. Danny calls the producer on another phone line, and "as soon as they hit 'go' on their end, within about 1/2 second, I'm locked and we're looking at the same thing." Generally, Danny shrinks the QuickTime window down as small as possible on his Mac to keep it out of the way of his Logic windows. The MiroMotion card simultaneously spits video out to a full-size monitor so Danny can have the image at full size on a TV while using the Mac monitor for Logic.

Although it sounds simple, Danny reveals that it didn't happen "quite so easily. I had to do some

experimenting with various SMPTE interfaces to see which ones would recognize the code through the phone interface. I tried a few interfaces that didn't work, and then I had an Opcode Studio 3 in a dusty cabinet somewhere. So I tried that, and it worked. Now I'm using a Translator Pro Sync, so I would assume that all the Opcode boxes can read it. Taking it even further, midseason last year I set up the music editor with the exact same system. I'm out in Agoura, the music editor is in Burbank, and *Ally McBeal* is out in Manhattan Beach. To make it more ridiculous, the producer calls the music editor, the music editor three-way calls the SMPTE to me, and when the producer hits 'go' in Manhattan Beach, the computer in Burbank and my computer are locked up within about 1/2 second.

SEE SPOTS RUN

With the show spotted, Danny is ready to begin writing and recording the music for the episode. Logic Audio becomes the master for the process, but, at this point, timecode is not run from any external source — Danny is referencing to SMPTE numbers within Logic. When he's writing a cue, he uses the cue's start time as an offset for the audio/MIDI file so that the audio lines up with the scene. "I usually write an episode from beginning to end," Danny explains "but on a show like *Profiler*, I might have a ridiculously long cue of maybe nine minutes, so I'll attack that first. When tackling really long cues first, you can create the bulk of the palate if there is a thematic sound to a particular episode — so you might be able to get the shorter ones done more easily."

Danny uses a combination of synths, samplers, and acoustic instruments in the music he creates for *McBeal*. "For a lot of cues on *Ally* I'll start with just the piano, but if it's a guitar-prominent cue, I'll lay the guitar down first.

"The piano is sampled, drums are samples of my own kit played from a TrapKat, and percussion is sampled. But the bass, guitars,

woodwinds, and cello are all live instruments. I play just about all the guitars and bass; the only thing that's not me on *Ally* are the woodwinds and the cello."

All of these instruments are recorded at Lux's studio into Logic Audio using three linked Yamaha 02R's serving as the centerpiece of his room. ADAT I/O interface cards are installed in the 02R's, linking the 02R's via Lightpipe to a Korg 1212 I/O card in the computer (the 1212 I/O serves as the "front end" to Logic Audio). The first 02R in the chain serves as the word clock master for the entire studio, feeding the other 02R's, the 1212 I/O, TASCAM DA-88's, and an Aardvark TDIF card that is installed in a second computer (more on this in a second).

GETTIN' GIGY
When asked about whether he has a favorite sampler or synth module, Danny says his most important consideration is "the sample libraries. It almost doesn't make a difference which sampler you have because they're all so good nowadays. I have a couple of (EMU) E4's and E6400's, K2000RS, an Akai S5000, and I used to have a rack full of Roland samplers (only one S760 now). But I've become a big fan of the Gigasampler. Before I had it, I found that I'd fill up my samplers with palates for the show. One E4 might have strings, and another brass or woodwinds. I would often load a whole E4 with 128 megs of sounds from which I might only use a couple of the patches occasionally, but I want these sounds available and ready to go.

"The Gigasampler allows me to buffer all those sounds in its memory,

put those patch names within Logic Audio, have all those sounds instantly available via patch change commands, and I now have an E4 totally free for other sounds. I also found that when I'm writing — and I have to write a lot of material quickly — sometimes I'd think about a great sound that I have in a library, but I didn't have room for in the sampler. It'd be a pain to stop working, find the sample, load it into one of the samplers (if there is any room available in RAM).... With the Gigasampler and a huge hard drive, all I have to do is find the sound on the screen, grab it, drop it, and it's ready to go instantly.

"It's limited by the drive space and RAM in the machine, though it seems RAM is less important. I have 256 MB and what I can have buffered into it at one time seems to be virtually endless." The Gigasampler runs on a PC with an Aardvark TDIF card installed; it interfaces with an 02R via TDIF card in the 02R, keeping the audio in the digital domain from input to output.

Danny admits that every show "has its challenges, and taking on a project where you're writing music you've never written before is always fun. The time saved in working like this is huge. What typically would be a four- or five-hour project for the spotting session now happens in about an hour or hour and a half. Multiply that by three or four shows, and I save 15 to 20 hours per week."

For more information on Danny Lux's unusual techniques, he may be reached via e-mail at Esounds@aol.com.

Hey Danny — that's almost enough

Hot House **PRM 165 Monitors**

Hot House's new compact monitors tell it like it is

BY STEVE LA CERRA

The PRM 165 from Hot House Professional Audio is the newest in the company's line of high-resolution, professional reference monitors. Combining a shielded 6.5-inch, long-excursion woofer and a 1-inch, recessed-dome tweeter in a 13-inch deep cabinet, Hot House intends the PRM 165 for console-top or stand-mounted placement. Hot House engineers have designed this passive speaker to be free of bumps across the frequency range, and response is stated to be within 1 dB from 300 Hz to 15 kHz. Hot House Professional Audio sent *EQ* a pair of PRM 165's, plus their ASB 112 subwoofer and Model Four Hundred and Model Two Thousand power amps.

Although the PRM 165's look like "typical" compact studio monitors, closer examination of the black-painted cabinet reveals a lot of attention to detail. To minimize diffraction and help maintain accurate stereo imaging, the front baffle is free from any sharp angles and does not employ a grille. Depth of the cabinet is noticeably greater than that of most monitors with a similar-sized front panel — a factor that must surely contribute to the PRM 165's extended low-frequency response. We patched our Yamaha 02R console into the input of Hot House's Model Four Hundred amplifier, which we used to drive the 165's.

Even upon a first, casual listen, it's obvious the PRM 165's are dif-ferent from many monitors. The lower-midrange is a bit laid back when compared to some other monitors — not because the PRM 165's are *lacking* in that region, but because they are accurate in this area where many monitors hype the response. Further listening and session work revealed that they sounded as flat as they measured.

We used the PRM 165's on a variety of tracking and mixing sessions, including a drum tracking session where the snare drum was a Noble & Cooley Alloy Classic. This drum has a very distinct, subtle tone, which can easily be masked through other monitors, but when heard through the PRM 165's, all the snap and clang from the drum came through loud and clear. Side stick played on the Alloy Classic sounded crisp and natural. Kick drum was solid and tight, if not capable of moving enough air to rock your world (we wouldn't expect any 6.5-inch woofer to do so). Cymbals were clean and extended with plenty of air, yet without sounding harsh. On floor tom, we noticed something we don't usually hear through different monitors with the same signal chain — the pitch bend of the drum was clearly audible. Although the 165's sounded a bit lean in their bottom octave, this characteristic is the lack of boominess and coloration present in most other monitors of this size.

Using the Model Four Hundred am-plifier, the 165's played plenty loud for a control room roughly 12 x 15 feet, within a nearfield of about 4 to 6 feet. (We'll tell you about our results with the Model Two Thousand in a forthcoming review!) Larger control rooms may need more SPL; a requirement that can be met by the addition of the ASB 112 subwoofer (more on that subject later).

These monitors scored high on what we call the "ruthless scale" — they are revealing enough to let you know what's really going to tape — in our opinion, the most important factor when choosing a reference monitor. The PRM 165's maintained a transparency in the upper mids, which minimized coloration of male and female vocals, guitars, and piano, and they managed to be sonically revealing without sounding harsh in the top end.

When it came to mixing, the PRM 165's were easy to listen to and produced minimal fatigue (we generally monitor at around 87 dB SPL). Off-axis response was very consistent, with a large, useful listening window with the speakers placed about six feet apart. When mixes on the PRM 165's were played on home systems, tonal balance was ever-so-slightly shy in the top end — almost certainly due to the Hot House monitors' extended frequency response and the lack thereof in most home systems. However, there were no sonic surprises, especially in the critical midrange and lower-mids. Vocal balances translated well to other rooms, as did bass tones and balance between kick drum and electric bass or synth. This means the PRM 165 is just about as effective for mixing as it is for tracking.

We did our final tests of the PRM 165 in conjunction with Hot House's ASB 112 active subwoofer. This sub contains a 32-pound, 12-inch driver and a built-in, 450-watt MOSFET power amplifier. An

ASB 112 with a pair of PRM 165's can deliver flat response down to below 30 Hz, and enough SPL to make any club music fan happy. If you decide to add this subwoofer to your control room, be sure to have another person around to help — this thing is heavy!

Connection of the ASB 112 into your system is easy. In our situation, output from the Yamaha 02R was patched to the rear-panel input of the ASB 112, which contains an active crossover (thanks to Hot House for providing both balanced XLR and unbalanced RCA I/O jacks). Rear-panel XLR outs were then patched to the input of the Model Four Hundred. In addition to the I/O jacks, the rear panel is also home to a power switch, IEC receptacle, continuously variable low-pass and high-pass filter frequency select knobs (independently adjustable and fully overlapping), a low-frequency level control, and four switches. The four switches provide control over bass EQ on/off (intended for home theater applications), phase reverse, sensitivity adjustment for balanced or single-ended input sources, and LF bypass.

We set the ASB 112's controls as per the manufacturer's suggestion — high- and low-pass at 80 Hz (approximately 1:00) and
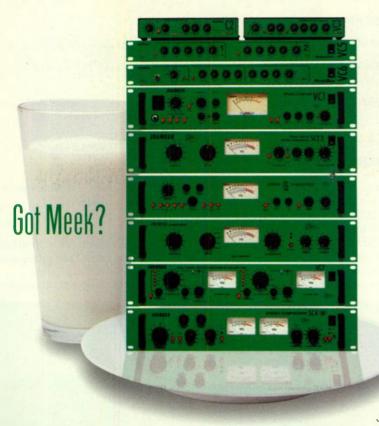
low-frequency level at about 12:30. Though placement of the ASB 112 is less critical than that of the PRM 165's, we experimented a bit with location until we settled on a spot approximately three feet from any wall. The result was a very natural and seamless integration of the subwoofer with the PRM 165's, and a noticeable increase in headroom as well as maximum SPL of the system. With the PRM 165's now relieved of their duties below 80 Hz, imaging and clarity in the midrange was brought up another notch. We thought we heard a subtle increase in the top end of the PRM 165's when we added the ASB 112. Hot House confirmed this perceptual change as a result of the Model Four Hundred amp being high-passed by the 150 kHz-bandwidth crossover — no longer using energy on the lower octave, there's an increase in dynamics and "air" in the top end.

The ASB 112 is very well-matched to the PRM 165's — if you didn't see the subwoofer, you wouldn't know it was there. One thing we didn't like about the ASB 112 was that — because the electronics are active — the subwoofer must be powered on to pass signal. In studios where engineers frequently like to switch the subwoofer in and out of the sys-

tem for comparison purposes, this means having to repatch the console output directly to the power amp for the PRM 165's. Definitely an inconvenience, though, according to Hot House, the signal path would be compromised by addition of bypass switching circuitry. Most importantly, the ASB 112 increases the low-end extension of the PRM 165's without ever sounding sloppy, boomy, or giving you "one note" bass. It certainly ain't cheap, but when you consider it as a major upgrade to your monitoring system, which includes a power amp and active crossover, the price looks more reasonable.

Hot House Professional Audio has certainly done their homework on the PRM 165's, and, at the price, they are extremely stiff competition for similarly priced monitors.

Joe Meek SC-4 Optical Compressor

JOEMEEK's latest addition to their compressor line lets users go wide

BY EDDIE CILETTI

If you are wondering why JOEMEEK has created yet another compressor, keep an open mind. The SC-4, while based on the same optical and support circuitry as their other models, offers a twist. It borrows a concept from the popular Fairchild "stereo" compressor-limiter that allows users to manipulate the "width" of the stereo image. Otherwise, the SC-4 is much like the SC-3, in that both provide 24-bit converters and an optional digital interface, and the SC-2, which has no digital interface.

NEW FEATURE: WIDTH

Did you ever end up with a mix that was vocal-light and effects-heavy? Rotating the SC-4's Width control counter-clockwise will reduce image width all the way to mono (if you'd like), effectively raising the vocal and putting the effects in their place. Conversely, mixes that seem boring and effectively dry can be made ambient and exciting by clockwise rotation, increasing the width. (Some may recall the B.A.S.E. box several yeas ago, which also manipulated width, *sans* dynamics.)

I/O

Two discrete channels enter the SC-4 via analog or digital inputs. Designer Ted Fletcher has addressed limitations found in the SC-3 by adding a consumer/pro switch for the digital ports via optical and AES connectors. Unlike the SC-3 (which allowed the digital input — once converted — to combine with the analog inputs), the SC-4's input selector switch works in a traditional fashion. Unbalanced analog sum and difference signals appear on 1/4-inch TRS-style Insert jacks, also labeled with the respective "M-S" (Mid-Side) designation.

INTERFACE AND METERING

The SC-4 includes Input, Output, Compression (Threshold), Slope (Ra-

portion of the SC-4 is the master or slave.

The VU meters were not labeled on the early production unit I tested. This was an oversight that will be corrected in later production runs. The left meter indicates the "sum" or "volume" information (as the manual refers to it), pre output gain. This can be a problem when large amounts of compression are used, resulting in a low meter reading. Raising the output gain (recovery) control allows the signal to be heard, but not seen. The right meter indicates the amount of compression. There is a Bypass switch, which removes the detector signal from the gain reduction device.

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MANUFACTURER: JOEMEEK c/o PMI AUDIO (distributor), 23773 Madison Street, Torrance, CA 90505. Tel: 310-373-9129/877-563-6335. E-mail: themeekman@joemeek.com. Web: www.pmiaudio.com.

APPLICATION: Stereo dynamics processing.

SUMMARY: Analog, optical compressor with built-in 24-bit digital audio converter.

STRENGTHS: Easy to use compressor with stereo Width control; TRS Insert allows access to Sum and Difference (M-S) sidechain.

WEAKNESSES: No independent control of sum and difference parameters; Width control "only" goes to 200 percent.

PRICE: $2199.99, SC-4; $299.99, DC4 digital board.

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tio on a stepped switch), Width, Attack, and Release controls. The Sum and Difference matrix can be bypassed by pressing the M-S switch, so that the SC-4 can directly process M-S encoded signals. Front-panel selection of Sample Rate and Sync modes determine whether the digital

Long release times will slow the return to Bypass mode.

REAL TEST

I tested the SC-4 with the help of Jeff Roberts, whose Latch Lake Music studio is south of the Twin Cities.

My wish list for the SC-4 includes

COOL FRONT.◆
SERIOUS BACK.◆

HHB CDR850 CD RECORDER
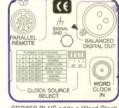
One look at the back panel and it's easy to see why the HHB CDR850 leaves other CD recorders where they belong: in the bedroom! XLR and RCA analogue inputs and outputs, an AES/EBU digital input, optical and coaxial SPDIF digital I/Os and parallel remote control, the CDR850's comprehensive connectivity perfectly complements its class leading sound quality and ease of use. And if you work entirely in the digital domain, the new CDR850 PLUS adds a Word Clock input and a balanced digital output to the CDR850's already very impressive spec.

Before you buy a CD recorder, it pays to check around the back. See both sides of the HHB CDR850 at your nearest dealer.

CDR850 PLUS adds a Word Clock input and balanced digital out.

A full range of HHB high performance CD-R media includes Silver, Gold, Printable, Re-Writable and Bulk Packaged discs.

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an extension of the Width range all the way to full "difference," if only for monitoring purposes (to know what's there). Using the difference signal alone is dangerous unless you purposefully want to generate rear-channel information for matrixed Dolby Pro Logic® surround. The signal will completely disappear if summed (to mono). In addition, I would like access to both sum and difference compressor settings. Right now the only way to do so is to take advantage of the rear-panel inserts.

On the bench, I did find that the Width control's center détente position did not correspond to "normal stereo" (100 percent) mode. No internal calibration was found. True stereo was achieved with the control set to the "one o'clock" position using a test tone in M-S mode.

MANUAL

Like [dearly departed engineer/producer] Joe Meek's philosophy of recording loud, aggressive pop records, the JOEMEEK manual is about twisting knobs until things sound good. The one truly useful bit of information is the explanation that the ear is a nonlinear device and that dynamics processors that can mimic the ear's characteristics are often more desirable than those that may be more technically precise.

RETRO?

The retro mystique of JOEMEEK gear is a bit of a mystery considering that the analog portion is so very '80s. That vintage signal processors are known for their sound has as much to do with the support electronics as it does from the gain-manipulating device(s). With all JOEMEEK compressors, the support circuitry is much like that found in a Trident Series 65 console — not terrible, not amazing. The ICs are TLO72 (dual) and TL074 (quad) opamps. I/O is interfaced by 4559 series dual opamps. The sound in this case really comes from the range of attack and (especially long) release times, which, when combined with the choices of Slope (ratio), manipulate the optics into delivering pumping effects that most VCA compressors can't get (but could).

FINALE

That said, there are reasons to like JOEMEEK gear. First, the large 2U front panel with mechanical VU meters makes it straightforward and easy to use, especially when you consider that compressors that are more complex are squeezed into a single rack space. (The green background can make the silk-screened black text hard to read under low-light conditions.)

Second, if you like the JOEMEEK style of compression, you'll really appreciate the Width control, a feature that makes the SC-4 my favorite in the JOEMEEK line. Few products have the ability to manipulate the stereo image. We preferred the width set to maximum (200 percent) — at least with the mixes we used for the test. **EQ**

DISC STORY

Anyone familiar with the Fairchild compressor-limiter has noticed that the channels are labeled "Lateral" and "Vertical." If you didn't know, Fairchild made disc-mastering equipment starting back in the monophonic days and ending as the loser in the battle of the stereo cutting systems. Theirs: Lateral/Vertical. The Winner: 45/45. Both systems produced a compatible stereo groove, but 45/45 was easier to visualize. Two drivers, mounted at a 45-degree angle to the disc surface, modulate the stylus.

For both systems, mono information makes a lateral-only groove. Mono, but out-of-phase information, makes a vertical-only groove. The advent of stereo turned disc mastering into cruel and unusual punishment for the engineer, the cutter head, its amplifiers, and the playback stylus/cartridge. The Fairchild compressor was designed to minimize the problems by providing separate dynamics control over the lateral and vertical signals.

By combining left plus right into mono to feed the lateral cutter-head and subtracting right from left, and applying that difference information to the vertical cutter-head, mastering engineers could decide how much stereo information made it to the record. Too much out-of-phase information at low frequencies would create such violent vertical excursions as to cause the cutter-head to lift off the disc, creating a break in the groove. (Unacceptable, of course, for R&B records!)

M-S: That's Sum Difference!

The JOEMEEK SC-4 replaces the Sum and Difference — Lateral/Vertical references — with Mid-Side (MS), a term describing the microphone technique that provides greater after-the-fact spatial manipulation of the stereo image than simple pan pots. The Width control, in the center-détente position, delivers a normal stereo image. Turning the control counterclockwise decreases the image width to mono. Clockwise from détente increases the difference signal, making the image up to 200 percent wider, but does not completely eliminate the sum. The raw difference information is comprised of two, identical, ear tickling, out-of-phase signals that will sum to nothing.
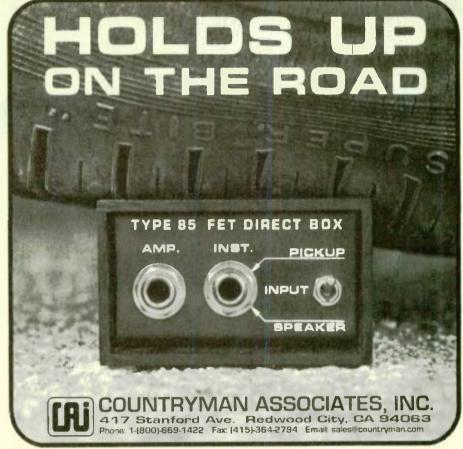
AND THEN THERE WAS LIGHT

By converting amplitude into light, a photosensitive receiver manipulates gain in a gentle way. It's what makes the Teletronics LA-2A and LA-3A the classics we know them to be (no support circuitry other than amplifiers). Neither has Attack/Release controls, because the choice of transmitter and receiver determined the attack time — relatively slow — and the release time, which was nonlinear (initially fast, then slow). Modern opto-electronics are faster all around, allowing designers to include Attack and Release controls, but still it is hard to make any optical compressor sound bad. Interesting, perhaps, but not bad!

A Question of Balance

The LA-2A and the LA-3A are both single-channel devices. They can be linked for stereo, but to match and identically process two channels was a challenge then as now, because the optical time-response characteristics and the "on resistance" are device-dependent. With so many "new" optical limiters — many of which are stereo — manufacturers are spending a good deal of time trying to match optical components. (Optical parts vary widely when new and must be pre-tested, graded, and matched. In addition, the characteristics change as the parts age.)

One solution to minimize the effects of mismatching between any two gain manipulating devices is to convert the stereo signal from two discrete channels (left and right) into a "sum and difference" matrix. At a recent AES convention, I suggested this option to several manufacturers (citing Fairchild's disc-mastering compressor/limiter as an example) as an alternative to the time-consuming task of matching devices for stereo. Ted Fletcher, the principle designer at JOEMEEK, took me up on my idea, thus creating the SC-4. (Ted, however, did not give me credit in the manual for the idea, as he had promised….)

Millennia Media **HV-3B Preamplifier**

Bring out the best in your quality mics with this unusual preamp

BY MIKE SOKOL

Boy did I have fun with this. When I met John La Grou of Millennia Media at a tradeshow earlier this year, little did I realize that I would soon get to review what has to be the greatest preamp/microphone combination I've ever laid my hands (and ears) on.

In a nutshell, the HV-3B preamp from Millennia Media is an unassuming-looking single-rack unit, with the standard controls one would expect on the front panel. Except for an additional switch labeled High Voltage and a nonexistent pad, it could be anyone's preamp. However, a look at the options list quickly puts that thought to rest. First, that mysterious "High Voltage" switch. What does it do? Glad you asked.

When I first got the call from John at Millennia, he nonchalantly asked if I would like to try the B&K 4003 High-Voltage mics with the preamp. He said they were "really nice" for piano. Since I had a piano recording date scheduled

in a few weeks, it seemed like the perfect match. When the box arrived, I found a pair of small diaphragm, omnidirectional B&K microphones, which physically resembled the Earthworks mics I reviewed in these same pages last year. The B&K 4003's have a 4-pin XLR output and require a 130-volt power supply to make 'em go. Therefore, John shipped the HV-3B with the optional B&K connectors and power supply in place, allowing me to try these unique microphones.

I put them up spaced about 18 inches apart, 8 feet off the floor and about 6 feet back from the piano. The result? It was the best-sounding solo piano recording I've ever done. The detail and sense of immediacy was phenomenal. It was hard to believe that a set of omni mics so far back from the piano could get such string detail, while still sounding warm. All this without the use of tubes or transformers in the signal path. It was all effortless. (I did have to be a little careful down the recording food chain so as not to overload the circuitry.) Simply no compression or limiting action took place at the head-end.

The company literature promises +23 dBu of input headroom (31 volts balanced P-P), making an input pad unnecessary. Furthermore, the listed +32 dBu of output headroom (88 volts balanced P-P) could cause meltdown in the

next part of the signal path, so I did have to watch clipping at later stages. While I didn't have any simple way to directly verify these numbers with my test equipment, my ears don't lie. This pre has more headroom than the Grand Canyon.

Back in the studio, I tried the Millennia/B&K combination on acoustic guitar with the same sort of epiphany. It kicked butt. I didn't have a drum session booked, but I'll bet it would be awesome on overheads and percussion in general.

Later, I used the HV-3B with a variety of other large-format condenser mics I was testing for *EQ* (see "Love at First Listen" in the June '99 issue) and it "played nice" with everything to which I hooked it up. This is not a preamp that adds "coloration" via a tube or trick circuit that will make a standard mic sound better. Indeed, using it on a SM58 gave me a ho-hum "this is a '58" signature — as one would expect from a noncoloration preamp, which should act like a wire with gain. However, with Audix CM-111 mics it was stunning. The GT Electronics mics from Alesis took on a transparent quality when teamed up with HV-3B. The ADK mics really liked it as a partner, too.

For something that's not supposed to do anything except increase the gain, it sure allowed me to hear all that these excellent mics were capable of

putting out. I fed the output of the HV-3B directly into my UREI LA-3A compressor, and it resulted in the coolest, quietest vocal chain I've ever hooked up in my studio. There was no end to the headroom, and adding the signature compression of the LA-3A gave me a magic combination on vocals. It provided an "A-Room" quality, and I wasn't doing anything ex-

MANUFACTURER: Millennia Media, Inc., 4200 Day Spring Court, Placerville, CA 95667. Tel: 530-647-0750. Web: www.mil-media.com.

APPLICATION: High-Voltage microphone preamplifier for studio and live recording.

SUMMARY: One RU stereo mic preamp that can be optionally outfitted with B&K high-voltage inputs, 20-/24-bit A/D converters, or 36-position mil-spec gain controls. This is one serious mic pre.

STRENGTHS: Headroom…headroom…headroom…; over-designed power supplies and driver stages means never having to say you're clipping; transformerless design with discrete transistors makes it sonically transparent; did I mention this thing's got headroom?

WEAKNESSES: I had to send it back…Might I 'ave mo', sir?
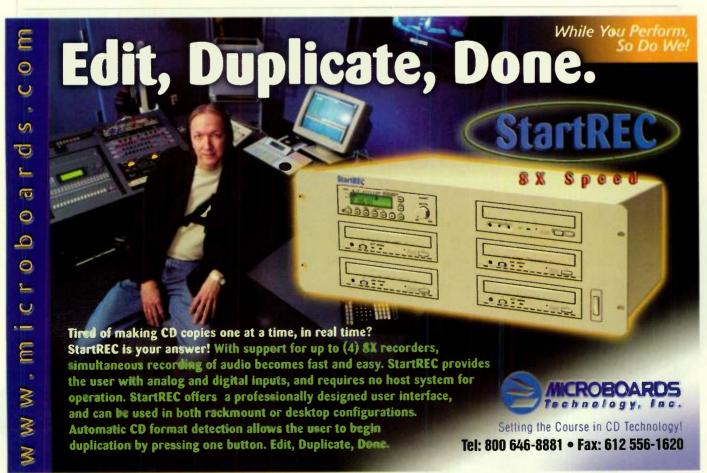
PRICE: $1895. B&K high-voltage option, $400; A/D converter option, 24-bit available 2nd-qtr 2000; stepped gain option, $150 per channel.

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ting about it. Moreover, MM's La Grou is a great resource on stereo mic techniques. The process of close miking every instrument, limiting everything, and "panning" the stereo has degraded many potentially great recordings. If you have a balanced-sounding room with a great performance, you can capture the whole thing with just a pair of mics in the proper position. It's how it was done "in the old days," and

cept turning it on. I like tools that make me look good without requiring some sort of blood sacrifice.

Millennia Media offers an A/D converter option that I bet would be pretty cool for direct-to-DAT or hard-disk recording. It did have the stepped input

controls, which worked flawlessly, as one would expect from a preamp of this class.

Millennia Media's design approach is impressive. Everything's over-designed with mil-grade and industrial parts, and that makes you comfortable with putting it into a 24/7 operation and just forget-

those old-timers knew a thing or three about recording the whole soundfield, not just the close sound of each instrument. I've done it myself on a number of recordings, and it's astonishing how great just a pair of excellent mics with a quality preamp can sound. **EQ**

TASCAM TM-D1000 Digital Console

Get all the control surfaces and basic processing gear that a project studio needs in a single, affordable box

BY MIKE SOKOL

I work with mixing consoles of vastly different sizes on a weekly basis. I've found that, while a really large console is quite impressive looking, many times you: [a] can't afford one; [b] can't put it where it needs to go; or [c] realize it's just plain overkill for the job at hand.

Unlike an analog console where form follows function, with a digital console many things are hidden "under the hood." This makes for a rather steep learning curve, and, indeed, on every digital console I've ever played with there's a certain amount of setup required just to get the thing to pass signal. As most of you know, with an analog desk all you do is assign the channel strip to the stereo bus, bring up the fader trims, then adjust the output faders. If nothing's muted, signal will come out of the outputs. If

you want to go to a subgroup, reassign the strip to the group you're interested in, then assign the output of the group to the stereo bus (or whatever). Aux sends and returns on an analog console work the same way, of course. Furthermore, you can physically patch in a compressor or equalizer with a Y-insert cable, and, as long as the signal levels and connector types match, you're in business.

However, things are not so obvious with a digital console, no matter how small it is physically. A digital console emulates all these functions (and more), but exactly how it's patched is just not available to the eye while you're looking at the board. Indeed, many times while at the helm of a new digital console I've sat there scratching my head simply trying to patch in the dynamics or figure out what's clipping. It's just not obvious to the casual observer (and operator). So how does TASCAM's new TM-D1000 measure up? Well, having used it on a number of live and in-studio recording sessions, here's the long and short of it.

SMALLER IS BETTER

TASCAM has designed a small digital console at an almost unbelievably low price. (Do you remem-

ber when they shocked the world with the introduction of the PortaStudio™?) Housed in a rugged steel chassis, the TM-D1000 is a tiny 16-channel digital console with many features. It's not meant to compete with the big consoles from TASCAM and others. It will find a place either with those stepping up from the PortaStudio™ environment or in an established studio as a main console add-on to get some extra returns and processing.

Digital routing and processing allows the designers to pack a lot of stuff into a small box. Three-band EQ with a parametric mid and sweepable frequencies are included on every channel, and dynamics, digital reverbs, and other effects are available to patch into the various busses at the push of a few buttons. You may not have been able to afford dynamics and processing before for your small studio or production truck, but now you can. Still, the learning curve for any digital console is rather steep, so make sure you put in a few days of playing with it before you try a paying gig with this (or any) new console.

DIGGING DEEP

The input channels 1 thru 4 accept either XLR mic level (with 48-volt phantom power) or TRS signals, these channels also include an insert jack for hooking up an external processor. Channels 5 thru 8 accept either mic or line-level TRS signals. There is an analog trim pot on these channels, but no pad. The upper inputs 9

PAGE	BRAND	INFO#	PHONE#	PAGE	BRAND	INFO#	PHONE#
153	Advance Sonic Concepts	47	609-726-9202	118	Kurzweil	21	800-421-9846
21	AKAI Musical Instrument	80	817-831-9203	39	Lucid Technology	56	425-742-1518
55	AKG/Harman Pro	87	615-399-2199	120	Lynx Studio Technology, Inc.	75	949-515-8265
2, 93	Alesis Corporation	4, 5	800-5-ALESIS	22-23, 85	Mackie Designs	48, 65	206-487-4333
37	Antares Systems	25	831-476-9780	11	Manley Laboratories, Inc.	33	909-627-4256
9	Apogee Electronics	7	310-915-1000	172	Mark of the Unicorn	19	617-576-2760
19	Audio Independence/Turbosound	53	608-831-8700	143	Markertek Video Supply	78	800-522-2025
149	Audix	54	503-682-6933	120	Marshall Electronics, Inc.	73	310-390-6608
154-157	B&H Photo-Video-Pro Audio	8	212-807-7474	60, 104	MediaFORM	96	800-220-1215
90	BBE	92	714-897-6766	36, 137	Microboards Technology, Inc.	81, 100	800-646-8881
153	Bell Electronics	45	903-784-3215	108	MIDIMAN	49	800-969-6434
41	beyerdynamic	11	516-293-3200	115	Minnetonka Software	38	612-449-0187
25	CM Automation/Sentech Electronics	9	818-709-4020	150	Music Yellow Pages	XX	516-489-6514
75	Conservatory of Recording Arts	94	800-562-6383	129	Musician's Friend	85	800-776-5173
135	Countryman Associates	83	650-364-9988	7	NHT Pro/Vergence Technology	52	707-751-0270
105	Crate Pro	31	314-727-4512	49	Opcode	59	650-429-2400
89	CAD Professional Microphone	37	888-702-7075	31, 95	Peavey	24, 26	601-483-5365
150	D.W. Fearn Company	29	610-793-2526	101	Rockford/Hafler	61	800-366-1619
4-5	dbx	69	801-568-7660	16-17, 28-29	Roland	41, 66	323-890-3700
97	Digibid	60	816-300-0311	107	Sabine Musical Mfg. Company	23	904-418-2000
67	Digidesign	36	650-842-7900	59	SAE Institute of Technology	57	212-944-9121
120	Disc Makers	74	800-468-9353	99, 125	Samson	32, 70	516-364-2244
153	Discount Distributors	42	516-563-8326	116-117	Seasound	86	415-331-4978
121	Electrix	50	250-544-4120	171	Sennheiser	20	860-434-9190
3	Emtec Pro Media/BASF	18	800-225-4350	43	Shure Brothers	34	800-25-SHURE
33	Event Electronics	17	805-566-7777	12-13, 63	Sonic Foundry	43, 44	800-577-6642
100	FMR Audio	90	800-343-9976	111	Sony Electronics	XX	800-635-SONY
27	Focusrite	28	516-249-1399	153	Sound Affair	40	714-540-0063
53	Fostex	22	562-921-1112	143	Soundmirror, Inc.	79	617-522-1412
113	Frontier Design Group	67	603-448-6283	126	Surround Professional	XX	516-944-5940
45	Genelec	35	508-652-0900	47	Steinberg North America	58	818-993-4091
61	Generalmusic, Inc.	27	800-323-0280	15, 164-169	Sweetwater Sound	46, 63, 64, 71	219-432-8176
135	Grandma's Music & Sound	82	505-292-0341	39	Symetrix	56	425-787-3222
75	Great River Electronics, Inc.	93	612-455-1846	52	Tannoy North America	97	519-745-1158
147	Guitar Center	76	818-735-8800	18-19, 51	TASCAM/TEAC America, Inc.	16, 88	213-726-0303
79, 133	HHB Communications Ltd.	30, 68	310-319-1111	153	Terrasonde	39	303-545-5848
121	IVL Technologies	50	250-544-4091	145	The John Hardy Company	77	847-864-8060
69	JBL Professional	XX	818-894-8850	100	The Recording Workshop	91	614-663-2544
131	JOEMEEK	84	877-563-6335	64	True Audio Systems	95	860-434-5220
123	KRK	84	714-841-1600	111	Waves Distribution	89	973-746-9417
				57	Yamaha Corporation of America	72	714-522-9011
				44	Yorkville Sound	99	716-289-2920

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thru 16 are available only via the included TDIF-1 jack, which makes hooking up a DA-38 (or other TAS-CAM-format 8-track deck) as simple as plug and play. Outputs include both TRS and RCA jacks for the analog world, and AES/EBU and S/PDIF jacks. A word-clock I/O is also included (bet you didn't know you had to properly synchronize these things to make them work).

Each input strip has a nonmotorized 60-mm fader with a mute, select, and pan control. Other switches are included for bus selection, equalization, and direct output assigns. A two-line display with four rotary pots gets you into all of the internal functions. These include a 3-band equalizer with sweepable frequency centers (and mid-range Q) on each input; 4 effect/bus sends; dynamics (comp/limiter); and a pair of FX processors with reverbs, chorus, phasing, etc., that can be assigned via the Aux 1 and 2 sends. Multiple mixes can be stored in memory and later recalled either manually or via the MIDI port with any external MIDI storage controller.

What TASCAM has attempted with the TM-D1000 is to put all the control surfaces and basic processing gear that a small project studio might need into a single, affordable box — and there are a lot of features. Nonetheless, you can't have it all; some things were left out. For instance, motorized faders. Yes, the onboard faders operate on the digital signal itself, but you can't automatically recall them physically to a previous position. What happens is a sort of "virtual" digital recall, which then allows you to manually reset the faders to their actual positions via a sort of "null" display on one of the screen functions. This does take a little getting used to, since you don't get a true visual indicator of what the board's actually doing, and, if you're not careful, you can make the console jump to the actual fader positions. A digital pad is included on each channel, but its function is strictly an internal one, operating on the digital gain. It's needed if

you're going to be boosting or processing a really hot digital input signal and don't want to run out of digital headroom. This pad, though, doesn't operate on the analog side of things, so I had trouble trying to run a hot line-level signal into the XLR inputs.

There are four auxiliary/group busses, but since they're shared, you can't get both functions on all four busses at once. A Mode control allows you to toggle between four mono aux sends; two aux sends with two groups; four aux sends with 1 and 2 ganged in stereo; and four groups only.

An optional Digital Effects card doubles the internal effects capabilities, and an MA-AD8 digital mic preamp can be used to add eight additional analog inputs. TASCAM makes a variety of interface boxes such as the Lightpipe to TDIF (IF-TAD, $249) and AES/EBU to TDIF (IF-AE8, $599), so you never have to worry about having the wrong format digital connection.

HEY, I'M LISTENING

The TM-D1000 was sonically fine as long as you watched out for the limited analog preamp headroom. The equalization could fix a wide variety of acoustic sins, and having adjustable EQ frequencies was very nice on a small

TASCAM has designed a small digital console at an almost unbelievably low price.

board. The effects sounded about as good as many introductory-level reverbs and compressors I've used recently. I found I could easily navigate around the various menus without too much trouble, but did get myself in a bind at one field recording session when I couldn't figure out how to assign the inputs to the subgroups. And before you say it, yes, shame on me for not having the instructions on hand (you can flog me with a spiral-bound manual for that one). In that case, you have to simultaneously press Shift and Param Sel, which should have been obvious, but I must have left my brain home that day.
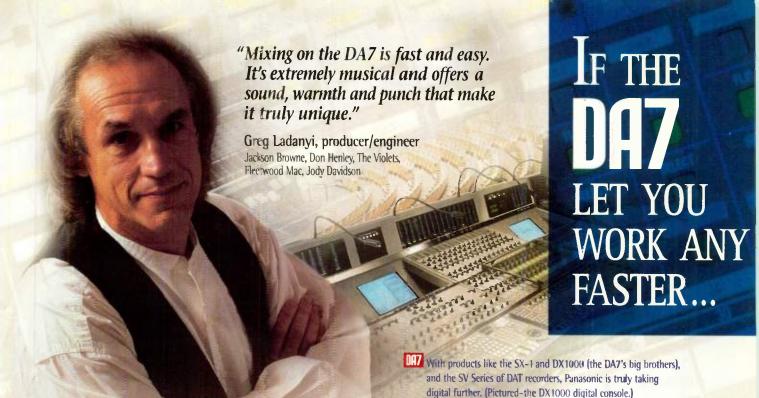
On the subject of manuals, it does come with a very well-written and illustrated Tutorial, which not only includes all the functions of the TM-D1000, but a crash course in digital audio as well. It's well worth the time spent reading, even for those of us who hook up this sort of thing on a daily basis. Speaking of instructions, TASCAM has just made a two-hour video tutorial named (what else?) *The TM-D1000 Video Tutorial*, which is now included with every TM-D1000 shipped. If, however, you've already bought one of these consoles and didn't get the video, contact TASCAM directly at 323-726-0303, ext. 782 and they'll get you a copy.

For those of you wanting to take the leap from a PortaStudio™ environment into the 8-track world, simply add the TM-D1000 to a DA-38 deck and you're in business for a small amount of money. On the other hand, if you've already got a big console loaded to the hilt and just need some more returns for digital decks or processing, the TM-D1000 would make a nice side dish. **EQ**

Lucid AD9624/DA9624 **Converters**

Transparent conversions and ease of use highlight this pair of converters

BY MIKE SOKOL

Over the past year, I've been doing reviews of A/D-D/A converters, and learning what features keep a user happy on a day-to-day basis. Gone are the days when we could accept simply accurate A/D conversion. There now also has to be excellent analog support electronics to allow a converter to even approach the dynamic range available with 20- or 24-bit operation. Furthermore, we also expect to be able to link to both professional and consumer data pipes. If an

A/D-D/A converter is hard to interface or won't play nicely with your other digital children, then it's probably more trouble than it is worth.

That said, Lucid has come up with an excellent pair of converters that combine transparent data conversion with an easy-to-use interface. They went all out and put in all the digital spigots you might ever need for 24/96 stereo data handling. Then they added an excellent LED, headphone driver with separate level control, and both XLR and TRS outputs that make hookup to a workstation or DAT recorder very simple.
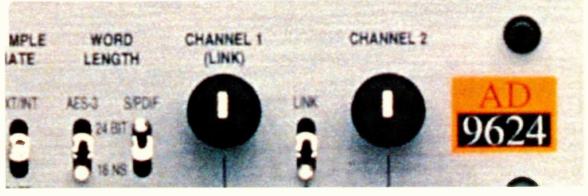
DESCRIPTION

The AD9624 and DA9624 are a pair of dedicated A/D and D/A converters, respectively, each housed in a rugged desktop box and powered by its own external line transformer. An optional rack-mount kit lets the two units live side-by-side in a single RU space. Power is by a line-wart-type transformer. Now, in the past, I've voiced my dis-

pleasure at having to use external power supplies. However, in this case it uses a line-wart, which only eats up one plug space on a power strip. In addition, it terminates in a rugged 7-pin DIN connector, which eliminates the wiggle factor found with those awful pin-barrel power plugs. Therefore, if it must be done, let it be done like this. [The engineering advantages of mounting the power-supply transformer external to the case are that it keeps the 120-volt line-voltages and transformer flux away from the audio electronics — probably one of the factors that contributes to the very respectable S/N ratio of 114 dB (A-weighted).]

I also liked having LED meters to show you what's happening. I hate crawling around to the back of a rack to troubleshoot a bad patch connection — especially when all I need is a meter on the gear to see what's happening. Both the D/A and A/D boxes have large 20-segment LEDs with lots of photon output for easy viewing in all situations but direct sunlight.

INS AND OUTS

The A/D box has a selector switch that allows you to sync to an external reference or use the internal clocks of 96, 88.2, 48, 44.1, or 32 kHz. This should cover just about anything for the foreseeable future. The digital output mutes for a second as the converter resets, but, thankfully, there are no burps or thumps on the data stream. Dual-input attenuators are included, and a link switch allows you to adjust both levels with just the left potentiometer, or set the levels independently as you see fit (after all, it is a free country). The S/PDIF and AES/EBU outputs can be independent-

ly set to 24-bit or 16-bit mode. This allows simultaneous recording to a 16-bit DAT deck and 24-bit workstation without relying on truncation of the lower bits for the DAT.

The D/A box thoughtfully incorporates a headphone output with reasonable output level, which also aids in troubleshooting a signal gone bad and gives you one more tap point. It has its own level adjustment independent of the main output control, thus allowing you to adjust it to a comfortable level without affecting the main XLR/TRS output.

As far as the analog output goes, as shipped, the DA9624 uses a true differential balanced driver for balanced connections. If, though, you need to use it in an un-

State of the *Net 1999*

We've been writing this damn column for three whole years!

BY JON LUINI AND ALLEN WHITMAN

Particulates and gas coalesce in a mad plunge following the Big Bang of Internet audio. Startling amounts of money and resources are hurled in frantic attempts to be the Last Word on the digital music delivery standard. Great strides are made in processing speed, but computer audio interfaces remain buggy, non-intuitive, and downright odd. Pick up a guitar, and it's pretty obvious how to play. Now turn on a computer. It still takes too long to learn how to use the damn thing. We're not asking for developers to dumb-down. We're asking for developers to talk to real-world users of products — the musicians, the people who actually put down instruments and pick up a mouse. The truth is: the focus has shifted from great services and tools to great hype for a great IPO.

You've allowed your FezGuys to begin our fourth year of writing about Internet audio enlightenment. Thank you. On this anniversary we'll talk about the state of Inter-

net audio applications for the desktop, an environment where noncompatibility is accepted and irritating problems are common. The industry is fraught with rapid technological advances, but that's no excuse for user unfriendliness! We celebrate companies improving and upgrading their products, but pledge to you, faithful FezReader, that we'll watch closely to see who settles for merely "acceptable" and who goes the extra mile to common sense usablility.

COURT MP3

After many months of public freaking and backroom speaking, the RIAA (www.riaa.com) finally realized people aren't going to stop listening to music. The mouthpiece of the Big 5 record labels dropped a suit (after several judges hammered them flat) to prevent Diamond Multimedia (www.diamondmm.com) from manufacturing and distributing the wildly popular Rio portable digital music player. A "mutually satisfactory" resolution depending heavily on the SDMI has been reached. The SDMI is the Big 5 consortium (along with several major technology companies) attempting to dictate a standard for secure digital music downloads using the Internet. Hey, if you can't sue 'em, join 'em. Expect bandwagon jumpers to add a half dozen new portable digital music devices to the market by the '99 Christmas shopping convulsion. The upside for consumers and musicians is that tools for digital music manipulation will be streamlined by the cartels looking to control the new marketplace.

EASIER XING

The popular and very useful Xing Audio-Catalyst (www.xingtech.com) encoder Version 2.1 (for PC only) was released in August. V2.1 features a simplified way to add your CD's information into the CDDB (www.cddb.com) database. There's expanded Help documentation, and the upgrade is free to 2.0 users. New users will pay $29.95, and your FezGuys say it's still worth it. Xing also has released a version of the MP3 encoder for Linux.

REALNEWS

Real big, RealNetworks has released the Beta 2 version of RealJukebox and a new RealJukebox Plus. The Beta 2 has multiple GUI "skins" (user interface design options) and features support for portable digital music players (including the above-men-

tioned Rio) and Liquid Audio products. The Plus is $10.99 to owners of the RealPlayer Plus and $29.99 if you are a first-time buyer. You need the RealJukebox Plus if you intend to encode at bitrates above 96 kbps. Real calls anything above 192 kbps "CD quality." The Beta 2 only encodes to 96 kbps. The Plus also features a 10-band EQ.

The new features are nice, but the install of RealNetworks products is problematic in the extreme. The company's apps make file destination decisions without asking. Once installed, RealJukebox software attempts to make itself the default audio application for every audio format, from CD audio to MP3, MP3 playlists, Liquid Audio, WAV, AIFF, and even AU files. It seems impossible to divine what audio types it's configured to play. Once RealNetworks tentacles are latched onto your hard drive, they're stuck! Sound familiar? Isn't this what RealNetworks testified to the Supreme Court that Microsoft does with their stuff?

And another thing(s)! On startup, RealNetworks apps play an irritating four-note jingle. Music listeners don't want to be interrupted! And don't bother using the Beta 2 with a filesystem shared on a network, it still can't find music there. Last but not least, the RealJukebox remains a Windows-only environment. C'mon people now, smile on your brother, everybody get together, etc....

The consensus now: RealNetworks products display user interface problems. Real is not the only audio environment for computers, friends. There are other applications out there.

ONE FOR THE MACS

SoundJam (www.soundjam.com) is the first integrated MP3 encoder/player and CD player strictly for the Mac. It's written by Casady and Greene (who bring you Conflict Catcher), and they know the Mac environment very well. The front end offers a choice of 13 silly/strange skins and an option to import skins from other players, like Winamp. Interestingly, the default skin on SoundJam looks suspiciously like the QuickTime default interface. Some aspects of the interfaces are not very intuitive. Example: to turn on the 10-band EQ, users must click a button that says, "Off." To turn it off, users must click the button that says, "On."

Also, to rename tracks, you must use the Mac Command-I on each individual audio file. This is also how ID3 information is accessed for each track. Music "Genres" (a nec-

essary evil) should be alphabetically sorted, too. This would be a very small fix! By way of comparison, this Mac-only encoder/player is somewhat slower than the Xing Audio-Catalyst encoder. SoundJam has also added a bitchin' bong-hit light show in the form of wacky screen-savers that react to music.

According to documentation, Sound-jam allows MP3 streaming, but on first try (and though the app said it was streaming), we heard nothing. Trying the same set of streams the next day (after a reboot) proved successful. The product is $40 as a download and $50 as shrinkwrap. A limited demo version can be found at the site.

ANOTHER ONE FOR THE MACS
Macamp, the plucky and popular little shareware MP3 player, is now available for free preview. The registered version can be purchased for $25 at www.macamp.net. Version 1.0 (assumed to be their first official "commercial" product release), boasts a new GUI, sports "sexy" (to quote the site text) visual plug-ins, has a 10-band EQ, and supports streaming audio. There is a short-cuts toolbar, an enhanced ID3 tag editor, and even a sleep timer (to quit or shutdown computer). Macamp is a quality product and is recommended.

ONLINE REMIXES
The FezGuys direct your attention to Mixman (www.mixman.com), a distribution system for creating and posting remixes of songs from provided audio templates. Choose from various sample banks, sounds, and grooves. You can post your own and review other's work. It's a real/virtual community and amusing for DJs, nascent remix producers, or folks who don't only play musical instruments.

The shrinkwrap product and demo are Windows-only, the Mac version (available by the end of September) is download-only. The basic version is $50 and the Pro version is $90 at stores and $80 online. Additional sample banks are available. Missy "Misdemeanor" Elliott and Art Of Noise offer sample banks of a song for remixing. Try one yourself and/or see what others have done.
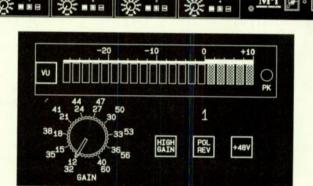
QUICKTIME SOURCE CODE RELEASED
The popularity of MP3 stems, in part, because the source code is freely available. Anyone can purchase the specifications and build their own encoder/player. This fosters speedy technological and interface improvement without the bureaucratic obligation of commerce. Now Apple has made the source code for their QuickTime streaming server (code name: Darwin) available for free. This en-courages the expansion of QT server products beyond on the Mac OS. If you care to download the source code, go to: www.public-source.apple.com/projects/streaming. By creating your own QT server, you won't have to pay for software or streams. People who host their own content can stream QT content without having to run NT (Windows Media) or pay a per-stream fee to RealNetworks. It's available on a number of platforms.

Related item: Check out a resource site about streaming servers (including Darwin) at www.streamingserver.org. Also, Sep-tember will see the final release of the Soren-son Broadcaster product (a streaming QuickTime 4 encoder). Go to the Sorenson Vision, Inc. site: www.s-vision.com/products/SorensonBroadcaster/.

GREEN WITCH PROJECT
Old world mentality in a new world order department: get Internet radio airplay by using the U.S. Postal Service to mail your CDs to Green Witch, an MP3 Internet radio station. It's free, and they seem nice. Check 'em out at www.greenwitch.com.   **EQ**

Love Is **All Around**

The proximity effect, mentoring, Tom Petty, and politics

BY EDDIE CILETTI

Anyone in New York City, whether attending the AES show or not, may be suffering from The Proximity Effect, a technology overdose, a.k.a. "Gear Fatigue!" I dedicate this — my second nontechnical column in a row — to all of you, and promise redemption with two deeply technical articles on tape machines and compressors slated for the next two issues.

A BASF advertisement recently caught my eye because it featured recording engineer Tony Maserati — a client and a friend. Included in the ad were his comments concerning "mentoring," the process of helping others climb the ladder via assistance, encouragement, and direction. In order to improve communication among freelance engineers (like himself) and studios, Tony started "The Allied Pool," as well as heading up one of the early studio-centric bulletin boards known as "Micronic Music Makers Online."

I asked him about the mentoring issue via e-mail. Here is a brief summary of his comments…

"I've really been into the mentoring process because some very talented people put up with me being a bad assistant. They patiently explained the combined art of engineering, politics, peace, love, and understanding technology (all while never letting the client see ya sweat). The Allied Pool encouraged studios to raise the bar for their assistant engineers. "Micronic" survived for about five months, but the amount of money it cost to run — and the software of the day — wasn't really making it a viable business."

That's a quick indication of the kind of individual Tony is.

HELLO PEOPLE

My first studio technology mentor was Greg Geddes, musician, *mime,* and former member of a band called Hello People. Greg taught me the basics of console signal flow and is currently a principle at Novastar, a post house in Hollywood. Hello People, along with guitarist Dick Rosmini, made a demonstration record to show off TEAC's first 4-track open-reel recorder, the 3340S.

Greg also introduced me to Steve Selberg (who went on to work at BGW) and Todd Rundgren. Steve emphasized the importance of having low-impedance output circuitry. I still have a "breadboard" of one of his line amplifier designs intended as a retrofit for a mid-'70s Allen & Heath console, to improve performance. (I had once dreamed of becoming a recording engineer for Todd with the hope of improving the sound of his records. This never happened.)

EATING THE BEAR

At Bearsville Studios, Michael Guthrie — whom I had hoped to work under, instead of replacing — gave me a three-day crash course in studio politics and technology before returning to the Record Plant in New York City. Michael was big on modifications, and was the first to explain to this grasshopper the importance of the grounding within a product. Despite all the informative "dope" that was freely distributed my way, I never thought to run a tape recorder to capture the booty. What I didn't remember had to be learned the hard way, hence the foundation of this column — documentation *and* sharing.

I was eventually lucky enough to work at Record Plant Studio (RPS) in New York City a few years before the studio closed. There, wannabe engineers and producers started in the tape library, got to clean ashtrays, and learn to make coffee all before ever thinking about assisting an engineer. Part of the tour of duty included a month in the shop for a technician's perspective of studio life. All of this experience built character and people skills, the latter quite necessary when working on difficult sessions.

MOM KNOWS BEST

At RPS, Paul Prestopino became my mentor, teaching me to think logically, to treat any system or piece of gear with respect, and avoid the potentially egotistical approach of trying to "improve" everything. Guitar amps best exemplify this lesson because, on the service bench, they look horrible even when functioning according to their design. Like a bulldog, one person's "ugly" is another's "character." (Translation: If it ain't broke, don't fix it.)

MULTI-TALENTED

A recent phone call from Bob Moore at Sound Lab in South Carolina reminded me that anyone who owns an Ampex MM-1200 has heard of "the Record Plant Mods" for that multitrack analog recorder. Like all hands-on studio owners, Bob must balance his engineering duties with light maintenance and heavy business decisions. Clients will basically accept your equipment roster if you are a famous engineer/producer. When in competition for business, however, the heat is on to have the gear on hand that customers request (whether they need it or not).

The New York Record Plant was lucky to have Paul for most its existence. "Presto" was able to create effective and economical solutions, translating client requests into reality, even in lean times when the budget was minimal to none. In addition to his responsibilities as technician, Paul also excels at fine leather, wood, and metal work (including engraving), even *sewing* — plus being called into sessions to play mandolin for everyone from Aerosmith to The Hooters. (More recently, Paul has been

touring with Peter, Paul and Mary. If you see him in a close up, check out those socks!)

I sometimes referred to Paul as "mom," because he reprimanded any "cowboy technicians" who attempted re-engineering without approval. He was always right. No matter what the urgency, Paul often asked, "Why is there never enough time to do it right, but always enough time to do it over?" There had to be a plan, and Paul's logical approach was always sound. Before undertaking a project, always create and organize documentation. Doing so will automatically generate enough "what if's" along the way to ultimately improve the end result.

RE: MOTE
One thousand miles west of New York, at the Target Center Arena in downtown Minneapolis, Record Plant Remote (the truck formerly known as "Fang") was recording Tom Petty and the Heartbreakers for a later radio broadcast. (They did not play the Mary Tyler Moore theme.) I walked in about two songs into the set to a remarkably balanced mix (albeit occasionally interrupted to solo individual channels). The truck's owner, Kooster McAllister, leaned over to tell me that "only one EQ and two filters were being used" by engineer Richard Dodd. The only other processing was compression — a UREI 1176 on Tom's voice, dbx 160's on the other vocals, and a Neve 33609 compressor/limiter on the mix bus.

Of the ten TASCAM DTRS machines onboard — an assortment of DA-98 and DA-88 decks — five were used as the master recorders. The remaining five provided a safety copy, as well as overlap in the event the tape ran out on the masters. (Kooster had purchased the "white" truck from Record Plant in 1989. Originally built by Wally Heider, and for many years outfitted with a Trident Series 80 — from the Minneapolis studio, Sound 80 — the body has been extended, and the truck and console replaced, the latter with an API 96-input console.)

HOST WITH THE MOST
The mix made an especially good impression on Ted Martin — host of my Web site (tangible-technology.com). (Ted's hikarate.com Web site is also home to the Hamer Guitar Fan Club. The entire site comes from a 486 run-

ning LINUX via DSL connection.) I will be recording Ted's band in the fall, and had been emphasizing the importance of preproduction and dynamics. As part of an international information exchange program, Ted and his Web partner, Dan Gardner, will teach me Web hosting secrets.
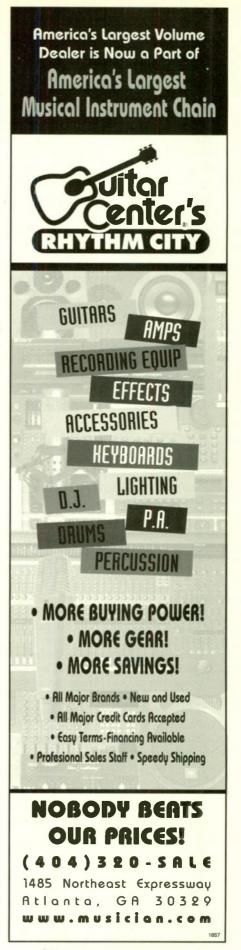
APRES SHOW: REEL FLANGING
After the Heartbreaker's show, both recording engineer Richard Dodd and live sound engineer Robert Scovill credited the skilled group of musicians for making their jobs easier and more fun. (Mr. Scovill successfully managed to convince his boss, Mr. Petty, to try the Neumann KMS-150 on vocals.) I was impressed with the Meyer HM-1 coaxial monitors. Those little devils deliver effortless wide-range sound (assisted by a passive subwoofer) despite their small size (biamped, with a 7-inch woofer and 1-inch soft-dome tweeter).

Richard also told a great story about the politics of being a recording engineer dating back to a time when flanging was done with a tape recorder. Early on, he realized the importance of politics as one of the many "tools" in an engineer's arsenal. On one occasion, he was called in from another session to flange an opening drum fill. (The client knew that Richard was the only engineer in that studio with the skill to quickly patch together what was necessary.) Though treated poorly by the chief engineer, Richard credited him for showing Richard the trick; a little white lie that made everyone look good.

SCORING POINTS
The point of all these stories is to show that audio savvy is about more than the gear. It is about sharing helpful information and being ready when the opportunity presents. On remote trucks, live shows such as Live Aid, Farm Aid, and Woodstock put your gear and your nerves to the test. I learned about the importance of procedure doing remotes for Record Plant, and the importance of remaining calm. On the subject of karma reimbursement, I want to thank Hector La Torre (EQ's executive director, and a fine record producer). He encouraged me to write, persisted until I agreed, and nurtured until I was good enough to get this column. (Thanks also to everyone who helped move me into the new MST facility in West St. Paul, MN.) **EQ**

World Radio History

are stepped is a mixed blessing — sometimes we couldn't quite grab the frequency we were searching for, but, when we did, the Summit's circuitry was able to open it up. Mastering engineers will love the stepped frequencies for precision and repeatability, but we'd like to see smaller steps (and more of them).

Compared to a Pultec, the Summit EQ was much more consistent in its treatment of the frequencies near the one you actually pick to EQ. For example, on a Pultec, when you boost at 100 Hz, you lose a few dB at around 40 Hz. So there might be a tendency to EQ at 60 in the first place — which is not where you wanted to do the EQ anyway. On the other hand, if you boost 10 kHz, 15 kHz might come down a bit (all of which is OK because it's part of the Pultec sound). However, when you EQ 12 kHz on the Summit, you don't lose anything at 15 kHz. Nice.

There are certain things you cannot do on the MPE-200, most of which are related to the limited overlap between the frequency bands. Sometimes we like to start a 1/2-to-1 dB shelf at 5 or 6 kHz, while adding a peak at around 12 kHz. The MPE-200's high mids won't go up high enough for you to do that. Similarly, you cannot set a shelf at 100 Hz while adding a peak in the response at 60 Hz. Perhaps some sort of multiplier (e.g., "x3") could be added to a later revision, or maybe the low mids could be extended at the lower end and the high mids extended at the top end. Furthermore, though we like the ability to by-pass each band, we do wish there were a "global" EQ bypass per channel.

Summit has constructed an EQ/mic pre that definitely attains the upper class of pro audio. This unit's internal build quality is something to behold, offering a beautiful, clean circuit layout and modular construction with separate cards for mic pre input, line amp output, shelving filters, and midrange filters. Almost 1/3 of the chassis is devoted to the power supply, featuring a massive toroidal transformer, which (you'll be happy to hear) appears very well shielded from the audio circuitry. Even the fan was carefully implemented, barely audible under only the quietest of circumstances. All switching of filters and frequency bands and changing of gain settings was completely silent. The MPE-200 is a gentle and subtle — as opposed to a surgical-type — EQ. We see it used not so much for correcting flaws

as for adding flavor to recordings at recording and mastering studios. If only we weren't saving for that new car...

Frank Filipetti was a child prodigy who started recording and mixing at four years old. In 1975, he won first place in the Rock category at the American Song Festival for his composition, "Analysis." Five years and no hits later, he decided to try his hand at engineering. He has since recorded, mixed, and/or produced Foreigner, James Taylor, Carly Simon, Elton John, Mariah Carey, Barbra Streisand, KISS, Courtney Love and Hole, Tracey Chapman, 10,000 Maniacs, Natalie Cole, and a host of others. In 1998, he won two Grammys (Engineering and Production) for his work on James Taylor's Hourglass.

WEB PICKS

Synthesis 99 (www.realtime. com/~fap7/synthesis/index.html). This is a great collection of links to manufacturer sites, newsgroups, extremely cool Mac software, and sites with samples (watch out for copyright, though). Original content includes some useful FAQs (the one on CD-R is excellent), some decent samples (need a Gregorian Choir?), and four real-time utility programs you can run while online.

Random Access — Your Forum For Digital Audio Recording (www.pixelite.com/windaw). If you're up for some discussion, this forum has separated threaded discussion groups for PC and Mac audio hardware, PC and Mac systems, studio gear, and tips and techniques, along with several company-specific forums. There's quite a bit of useful advice here, with fewer flamefests than on some other discussion groups.

Synthzone (www.synthzone.com). If you need support for just about any synth, look here. You'll find FAQs, software, samples, links to other pages, and more. That's not all there is to the site, but the synth section alone is a compelling reason to check in.

Circuit Cookbook (www.ee.ualberta.ca/~charro/cookbook/audio/guitar). This has tons of guitar effects, including commercial boxes, custom designs, and mods. While you're here, click on the parent page to find schematics for a variety of other useful devices. **EQ**

balanced situation (for instance, with a mono 1/4-inch plug or into an XLR input with pin 3 shorted to ground), then Lucid recommends opening the hood and resetting four jumpers. Moving the jumpers reconfigures each output driver for "pseudo-floating, cross-coupled" operation by adding an SSM2142 opamp in series with the differential drivers. What this means to the studio user is that the DA9624 will drive an unbalanced load to +24 dBu — unlike the standard differential output driver that will drop 6 dB when configured for unbalanced operation. There is a small tradeoff, of course: a 1-dB rise in the noise floor and the need to ground pin 3 in an XLR hookup (or the ring in a TRS hookup). (Yes, I promise to write an article in the near future on how unbalanced/balanced hookups work. A mother's work is never done.)
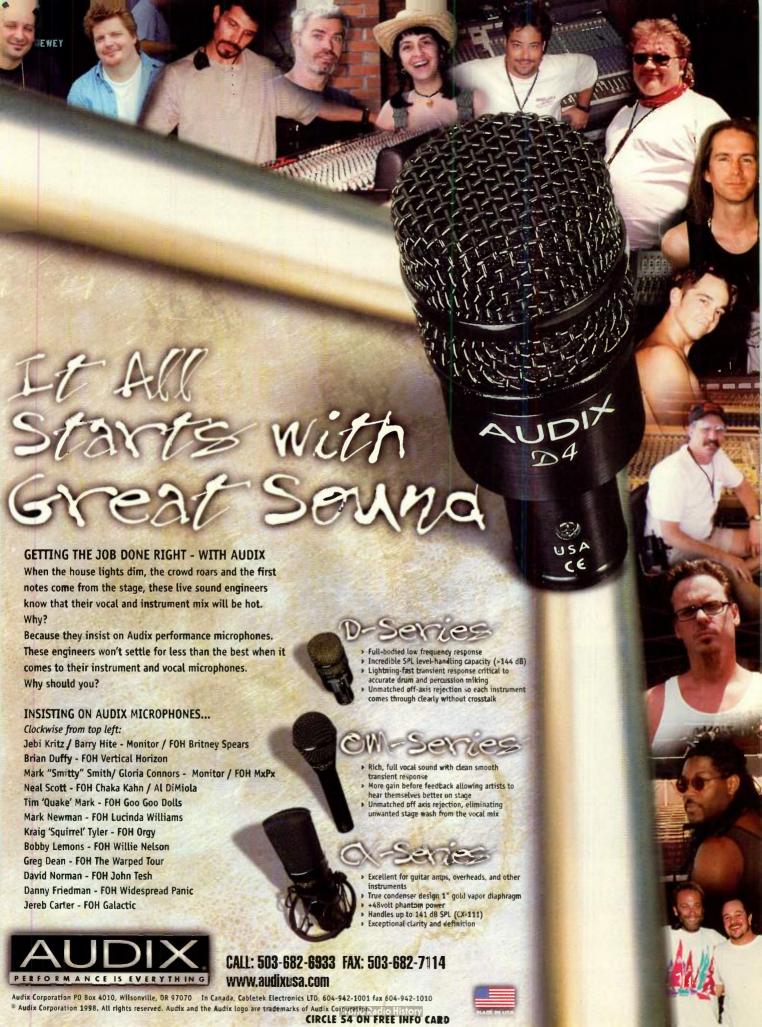
A selector allows you to toggle through the various digital input spigots, which include AES/EBU and S/PDIF in both coaxial and fiber (TOSLINK) connections. A "lock" display tells you when you've got sync.

IN USE
I tried both the AD9624 and DA9624 in a variety of studio and live situations. The A/D was first used to feed a DAT deck directly from a pair of RùDE™ tube mics and an SPL preamp for an on-location, stereo piano recording. Besides giving me some nice meters to watch to stay out of trouble, the sound of the converter was like velvet. Furthermore, the noise floor was too low for me to hear even with the headphones cranked up.

Back in the studio, I tried the D/A box with a variety of digital streams from my console, DAT deck, and CD player. It digested all of them without a complaint, and never popped, burped, or otherwise made obnoxious noises. When I fed an improper digital signal to the box, it simply muted, and when the signal was good, it came on. That's just how it should work.

If you're doing any serious recording, it pays to get your signal started off well with something like this $1700 pair of converters. Remember that your signal will never sound better than for that microsecond when it's converted to little 1s and 0s. Therefore, use the best gear you can possibly afford where it really counts. **EQ**

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Clockwise from top left:
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Brian Duffy - FOH Vertical Horizon
Mark "Smitty" Smith/ Gloria Connors - Monitor / FOH MxPx
Neal Scott - FOH Chaka Kahn / Al DiMiola
Tim 'Quake' Mark - FOH Goo Goo Dolls
Mark Newman - FOH Lucinda Williams
Kraig 'Squirrel' Tyler - FOH Orgy
Bobby Lemons - FOH Willie Nelson
Greg Dean - FOH The Warped Tour
David Norman - FOH John Tesh
Danny Friedman - FOH Widespread Panic
Jereb Carter - FOH Galactic

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MADE IN USA

CIRCLE 54 ON FREE INFO CARD

BITS & PIECES

medium "dead" hits. There are also major 2nd, major 3rd, and minor 3rd rolls sampled over a more limited (2-octave) range, and two mallet scrapes.

Vibes-wise, there are medium, medium loud, soft short, and hard short hits, as well as some riffs (glisses, stick on bar, fast licks, etc. — 16 total). The CD is available in audio, EMU (EOS/E4), Roland, Akai (S1000), and Sample-Cell/Mac formats.

PROS & CONS: As is typical of Sampleheads CDs, the samples are all clean and superbly recorded. While an all-vibes/marimba CD is specialized, this one definitely delivers the goods.

BOTTOM LINE: According to Sampleheads, Dave Samuels recorded these for use with a MIDI mallet controller for those occasions where it wasn't logistically feasible to cart around "the real thing." This CD proves once more that the best products are often the ones that spring from someone's genuine need. Moreover, if you need some good vibes, here they are.

PRICE: Audio $99.95; CD-ROM $149.95.

CONTACT: Sampleheads, 1600 Broadway, Suite 704, New York, NY 10019. Tel: 212-262-3488, 800-807-6056. Web www.sampleheads.com.

EQ FREE LIT. #: 109

CAKEWALK GUITAR STUDIO 2.0 (WINDOWS)

SUMMARY: A major upgrade to a guitarist-friendly sequencer/recorder.

BASICS: Guitar Studio 2 adds significant features to the original Guitar Studio, which was a surprise hit among tech-savvy guitarists. Based on the familiar Cakewalk Pro Audio engine, the main additions include editable tablature display (with the ability to handle alternate tunings and up to 8-string guitars), "vector-based" level and pan automation, twice the digital audio track count (16), up to 16 simultaneous real-time effects, and a "lite" version of the amp simulator plug-in from their Audio FX2 plug-in pack. Finally, audio and MIDI tracks can synchronize to digital video. Some of the new features, like fretboard "mirroring" and tab fingerings showing up in the score window (fig. 3), seem targeted at the educational market as well as recording guitar players.

PROS & CONS: The updates, particularly the digital audio upgrades (tracks and plug-ins), are most welcome. Some will be disappointed the Song Wizard from Version 1 is gone; it's been replaced by the Session Drummer MIDI FX plug-in with 100 presets so you can play to grooves instead of just a metronome, along with the Jammer Songstation. I find these more useful, actually, but I wouldn't mind seeing even more "songwriter-friendly" features, like being able to drag-and-drop chords into a sequence to rapidly build a chord progression. Interestingly, although several StudioWare panels are included, the only guitar-specific ones are for the Line 6 POD, Roland GR-30, and Yamaha G-50.
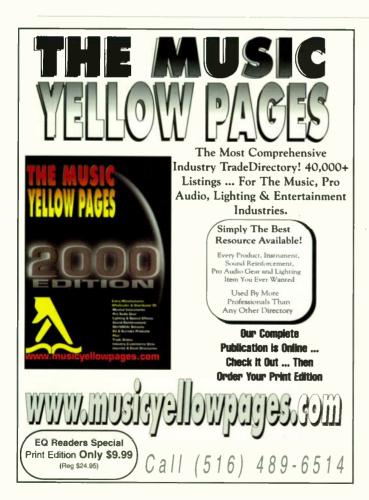
BOTTOM LINE: If you're a guitar player who works with computers, it's really refreshing to work with a program that's designed for you, not keyboard players. Guitar Studio may be the only game in town for guitarists, but this upgrade shows that Cakewalk is not being complacent about improving an already successful product.

PRICE: $249; upgrade from Version 1, $59.

CONTACT: Cakewalk, 5 Cambridge Center, 10th Floor, Cambridge, MA 02142. Tel: 617-441-7870, 888-225-3925. Web: www.cakewalk.com.

EQ FREE LIT. #: 110

THE **EQ** ROAD TOUR IS ROLLING AGAIN...
Surround Sound
for the project studio

Featuring **EQ Magazine** Contributing Editor **Mike Sokol** discussing the real-world creative facts about **5.1 surround sound** including creative mixing, room calibration and set-up, mixing for consumer release, monitor speaker calibration, surround equipment, DVD, bass management, center channel use and much, much more.

The **EQ Surround Sound Road Tour Part II** begins on September 9, 1999 and rolls through October 1999. Making stops at eight top recording programs in the country. These are:

EQ

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September 9:	**Sheffield Institute for the Recording Arts,** Phoenix, MD
September 14:	**Hartt School of Music (U. of Hartford),** Hartford, CT
September 16:	**Ontario Institute of Audio Recording Technology** (+ AES chapter), London, Ontario, CANADA
September 18:	**U. of Ohio** (+ NARAS chapter), Athens, OH
September 21:	**U. of Memphis** (+ NARAS chapter), Memphis, TN
October 6:	**Columbia College of Recording,** Chicago, IL
October 8:	**AES, Upper Midwest Chapter; St. Catherine's College,** Minneapolis, MN
October 20:	**Full Sail** (+ AES chapter), Winter Park, FL

Contact your individual tour sites directly for times and directions.

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AND TO EQ'S WEB SITE **WWW.EQMAG.COM** FOR TOUR UPDATES!

World Radio History

Dealing with *Sticky Fingers*

It may be more than the band's demo tapes and discs that leave your studio

BY MARTIN POLON

No subject brings the rush of blood to the faces of studio owners and operators of all sizes and shapes of facilities than the topic of petty and not-so-petty theft. In fact, as one studio owner described it over several intoxicating brews, it could be even called petti theft:

"I had stayed late one Friday night to catch up on some paperwork. I have all the studio surveillance cameras routed through my office. We had recently installed some sub-miniaturized cameras that are nearly invisible to the casual observer. That may be good in some ways, but the bloody things are so small that much of the inhibitions felt by band members are lifted by these unseen cameras. This was a typical four walls session with only one of my staff members present. He was otherwise occupied in the studio while a band member and his girlfriend were fooling around on the way to a quickie in the control room. This caught my eye needless to say, but the quickie I observed had nothing to do with sex.

"The band member was handing various XLR-style shunts, adapters, and inline transformers from a drawer below the console to his lady — who had raised her long skirt to reveal a slip with a zippered pocket near the hem. The entire store of these rather expensive gadgets were thusly deposited. A foreign travel petticoat my wife called it; used for passports and money where pickpockets are active! Despite having viewed the purloining of our valuable goods, I despaired of any confrontation with a female carrying the XLR spoils at the lingerie level, as it were!"

This anecdote, better than most, illustrates the dimension of the sticky fingers problem in today's project and commercial recording studios. There is no rhyme, reason, or constant rule as to who steals what and why. In general, many, if not most, bands are cash poor. There frequently is resentment as to the large cash outlays for recording studio services measured in the hundreds of dollars per hour! For example, connector inline adapters, attenuators, tone-oscillators, phase-reversal shunts, and transformers cost from $50 to $100 a pop. A pocket full (or a skirt full) goes a long way to equalize the outlay of studio time in the minds of many young musicians. Many of those caught justify their form of shoplifting with phrases such as they can afford it or we've paid for it, or these things cost a dime a dozen. Even with user-owned project studios, along comes a dissatisfied member of the band who begins to slowly and carefully steal cables, adapters, etc., for the day when he opens his own recording facility!

As to constant inventories, a topic that has been much heralded in these pages in the past, studio owners and operators tell us that the smaller the facility, the more difficult inventories are to maintain on a regular basis. In addition, the realities of real-world recording interfere with administrative assumptions. Another studio operator told us: "It's a four-wall rental on Friday and our guy finishes up at 1:10 AM in the morning. He has to be back here at 9 AM for another session and the last train for the night leaves at 1:30AM — after which he cannot get home and must sleep on the old lumpy sofa in Studio D. Do you think he is going to do a complete inventory then or that I want him to do so under these circumstances? Or do you think I am going to bring another tech in on Saturday morning on a four-hour minimum to run an inventory? I think not!"

The solution to the pilferage problem that is beginning to be accepted, at least in discussion, by some if not by many studios is the inclusion of a pilferage fee on all sessions. Some studios have been doing this for years, either by burying the fee in the hourly rate or by labeling it wear and tear or as breakage. Now, we are hearing some owners/operators talking about applying a percentage fee to every hour billed that is calculated by dividing the total amount of equipment believed to be pilfered per year by the number of studio hours billed per year. While it does punish the innocent with the guilty, it is also adjusted each year up or down depending upon the driftage to be compensated for. Additionally, each group that records receives a printed statement about the fee so that the studio's position is strongly communicated.

It is unfortunate that this topic comes up again and again, but many studio operators cite such thefts as one of, if not the most important, issues in preventing studio profitability!

The solution to the pilferage problem that is beginning to be accepted by many studios is the inclusion of a pilferage fee on all sessions.

ANALOG TO DIGITAL CONVERTERS

Rosetta 96k
APOGEE — 24-bit Analog to Digital Converter

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

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

FRONT PANEL:
- Power switch • Sample Rate (44.1, 48, 88.2, 96kHZ)selector • 16-bit (UV22), 20-bit (UV22) and

- 24-bit resolution selector • S/PDIF-ADAT optical selector • Soft Limit on or off • 12-segment metering w/ over indicator & Meter Clear switch • Level trim

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

lucid — AD 9624
24-bit Analog to Digital Converter

Transparent analog to digital conversion designed to bring your music to the next level. XLR balanced inputs feed true 24-bit converters for revealing all the detail of the analog source. 16-bit masters can take advantage of the AD9624's noise shaping function which enhances clarity of low level signals.

FEATURES–
- 24-bit precision A/D conversion • Support for 32, 44.1, 48, 88.2 & 96kHz sample rates • Wordclock sync input • Selectable 16-bit noise shaping

- Simultaneous AES/EBU, coaxial and optical S/PDIF outputs • 20-segment LED meters w/ peak hold & clip indicators • **ALSO AVAILABLE:** DA9624 24-bit D/A converter

DIGITAL MULTI-TRACK RECORDERS

TASCAM

DA-88 Modular Digital Multitrack

The standard digital multitrack for post-production and winner of the Emmy award for technical excellence, the DA-88 delivers the best of Tascam's Hi-8 digital format. Its Shuttle/Jog wheel and track delay function allow for precise cueing and synchronization and the modular design allows for easy servicing and performance enhancements with third-party options.

FEATURES–
- 1:48 minutes record time on a single 120 min tape
- Expandable up to 128 Tracks using 16 machines
- User-definable track delay & crossfade
- Shuttle & Jog capability
- Auto punch with rehearsal

- SMPTE, MIDI and Sony 9-Pin sync capability
- Options include RC-828/898 Remote Controllers, IF-AE8/IF-88SD digital interfaces, MU-Series meter bridge, MMC-88 MIDI machine control interface, SY-88 Sync Card

DA-38 Digital Multitrack for Musicians

Designed especially for musicians, the DA-38 is an 8 track digital recorder that puts performance at an affordable price. It features an extremely fast transport, Hi-8 compatibility, rugged construction, ergonomic design and sync compatibility with DA-88s.

ALESIS

ADAT M20
20-bit Digital Audio Recorder

The M20 represents Alesis commitment to meeting the high standards of world-class audio engineers, producers, studio owners and high-end video and film post production studios. A new professional digital multi-track, the M20 records 20-bit for outstanding sound quality. Combined with a host of production features like SMPTE/EBU, the M20 is a powerful tool.

FEATURES–
- SVHS Recording format - up to 67 minutes recording
- 18-XLR connections (9 in and 9 out) as well as a 56-pin ELCO connection • Digital I/O
- Includes LRC remote and a digital cable.

- 24-bit, 64x oversampling recording, 20-bit, 128X oversampling playback

ADAT XT20

The New ADAT-XT20 provides a new standard in audio quality for affordable professional recorders while remaining completely compatible with over 100,000 ADATs in use worldwide. The XT20 uses the latest ultra-high fidelity 20-bit oversampling digital converters for sonic excellence, it could change the world.

FEATURES–
- 10-point autolocate system
- Dynamic Braking software lets the transport quickly wind to locate points while gently treating the tape.

ADAT OPTIONS–
- **BRC** for all Adat except M20) w/ 460 locate pts, smpte/absolute time & bar and beat timing references, digital editing and transport control for up to 16 ADATs
- **AI3** 20-bit 8 channel analog - optical I/O interface

Digital Audio Recorder

- Remote control
- Servo-balanced 56-pin ELCO connector
- Built-in electronic patchbay
- Copy/paste digital edits between machines.
- **CADI** remote control/autolocator for M20 w/ jog/shuttle & rj-45 connector for long distance cable runs
- **Adat/Edit** integrated PCI digital audio card and software for recording and editing on Mac & Windows computers

CD RECORDERS

HHb — CDR-850 CD Recorder

The new HHB CDR850 is one of the most comprehensive CD-R, CD-RW recorders available today. It delivers the outstanding sound quality that HHB is known at a lower price than previous models. Equipped with a complete range of analog and digital I/O and easy to use one touch recording modes make the CDR850 suitable for any audio environment no matter how sophisticated or demanding.

- CD-R, CD-RW compatible
- All functions accessible from front panel menu
- 4 one touch recording modes: 2 manual, 2 automatic
- Sample rate converter accepts any digital signal from 32kHz to 48kHz including varispeed

- Copies all CD, DAT, MD, DVD and DCC tarck starts
- Complete user control over SCMS
- Balanced XLR analog I/O, Unbalanced (RCA) phono analog I/O, AES/EBU digital input, coaxial & optical S/PDIF digital I/O

marantz — CDR 640 CD Recorder

Marantz' flagship CD recorder benefits from 10 years of CD-R experience. Designed without compromise aided with the help of professional end-users ensuring maximum flexibility and stability in the most rigorous studio environments.

Features–
- Balanced XLR Analog in/out • Analog RCA/Phono in/out • AES/EBU & S/PDIF in/out • Records on CDR and CDRW audio and data disks• High resolution 20 bit Sigma/Delta AD conversion • Full SCMS Copy bit manipulation • 0.5 dB accurate level metering • Variable Audio Delay (0-4sec): Offset your audio to compensate for late track ID's • Preset function stores personal settings

- Optional RC640 Wired remote control

MICROBOARDS

CopyWriter A2D CD Duplication System

The first CD to CD standalone duplicator with built-in Analog to Digital Conversion capability. Easy to use and powerful, the A2D has a 2.1GB internal hard drive and a SCSI port for direct connection to a Mac or PC. A perfect solution for audio, data and video applications.

Features–
- Interface includes Microphone in, Audio line in, Audio line out and external SCSI port
- Supported Formats: CD DA, CD ROM mode 1 & 2, XA, CD Bridge, Photo CD, CD Extra, Multi Session, Mixed Mode, Karaoke, (optional)
- Duplication Speed: 8X Read/ 4X Write

- Windows 95, NT, 3.1, Mac OS and Unix compatible
- Headphone output with level control

STUDIO DAT-RECORDERS

TASCAM

DA-45HR Master DAT Recorder

The new DA-45HR master DAT recorder provides true 24-bit resolution plus standard 16-bit recording capability for backward compatibility-making this the most versatile and great sounding DAT recorder available. With support for both major digital I/O protocols plus the ability to integrate the machine into virtually any analog environment, the DA-45HR is the ideal production tool for the audio professional.

FEATURES–
- Word Clock
- 24-bit A/D and 20-bit D/A with dither
- XLR balanced and RCA unbalanced analog I/O
- AES/EBU and S/PDIF digital I/O

- Word Sync In/Thru
- Alphanumeric data entry for naming programs
- Independent input level adjustment capability
- Output trim for XLR balanced analog output
- Optional RC-D45 Remote Controller

FOSTEX

D-15 Pro Studio DAT Recorder

The new Fostex D-15 features built in 8Mbit of RAM for instant start and scrubbing as well as a host of new features aimed at audio post production and recording studio environments. Optional expansion boards can be added to include SMPTE and RS-422 compatibility, allowing the D-15 to grow as you do.

FEATURES–
- Hold the peak reading on the digital bargraphs with a choice of 5 different settings
- Set cue levels and cue times
- Supports all frame rates including 30df
- Newly designed, 4-motor transport is faster and more efficient (120 minute tape shuttles in about 60 sec.)
- Parallel interface • Front panel trim pots in addition to the level inputs

D-15TC & D-15TCR

The D-15TC comes with the addition of optional chase and sync capability installed. It also includes timecode reading and output. The D-15TCR comes with the further addition of an optional RS-422 port installed, adding timecode and serial control (Sony protocol except vari-speed)

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LIVE PERFORMANCE TOOLS

Why Upgrade to SHURE PSM 700 Stereo Wireless In-Ear Monitors?

- You have the best possible **protection** for your **hearing**.
- Your monitors **sound great every night**, regardless of the venue.
- You have **tremendous freedom of movement** on stage - without losing your monitors.
- You save money as multi-user systems are actually **more economical** than traditional, multi-speaker monitor systems.
- You drastically **reduce the weight and size** of your monitor system.

Why does Shure Dominate the In-Ear Wireless Monitor Category?

- **Sound:** Shure's unique Low Mass/High Energy E5 dual-driver earphones deliver stunning audio quality.
- **Flexibility:** Each transmitter delivers your choice of one stereo mix or two user-selectable mono mixes.

Use any number of receivers with a single transmitter. Everyone on stage can enjoy a clear, safe mix — all for a lot less per band member than most floor monitor rigs! Add up to 16 base transmitters for a total of 16 stereo or 32 mono mixes.

Mark of the Unicorn — the Choice for Powerful Live MIDI & Audio

Live sequencing? It's not just for keyboards and drums anymore! Automate a mix, reset effects and EQs, run your lights, even play complete audio tracks with real-time plug-in DSP effects! **Digital Performer** sequencing software has proven reliability with hundreds of live touring acts and innumerable concert performances. The **MTP AV** patches your live MIDI rig with on-the-fly setup changes —indispensable for keyboards and FOH control of effects processors. The **2408** gives you tremendous audio playback and recording capabilities and the **1224** lets you record your performances in stellar, 24 bit resolution. This combo has quickly become the standard on pro tours, both for audio "sweetening" and live location recording.

Automated Digital Mixing for Live Gigs? The Tascam TM-D1000 Performance Bundle is Here — A Sweetwater Exclusive!

No soundman? No problem! Tascam's amazing TM-D1000 Digital Mixer is perfect for the small ensemble, keyboard player or electronic percussionist that wants great sound and extensive control, without a lot of complicated headaches. Easily create preset mixer "scenes" for each song. Set all mixing functions plus built-in digital effects with a single button push! Or enjoy real-time automation when you control the TM-D1000 from a MIDI sequencer such as Digital Performer.

Sweetwater's Performance Bundle adds Tascam's MA-AD8 8-channel mic preamp/A-to-D converter and FX1000 DSP expander. You get a total of 12 balanced, XLR inputs with 20-Bit D to A conversion, enough for full band. DSP horsepower is dynamically allocatable for up to 8 dynamics processors and 4 channels of digital effects. Save all settings with scenes or automate! Why settle for manual mixing? Call us here at Sweetwater Sound today for our special "ProNet" discount on this great bundle! We'll even **pay you top dollar for your old board when you upgrade** to a Tascam Performance Bundle.

Power and Grace! A Truly Compact PA that Smokes!

What if your club PA had more volume, cleaner sound and less weight? For solo artists and small ensembles, the Community XLT41E two-way cabinet is the perfect choice, balancing top sound quality, pro durability and remarkable portability. Add an XLT51E 15" subwoofer and you've got a full range rig that really kicks, without breaking your back! From the titanium, high-dispersion tweeters to the indestructible construction, Community has taken all of their knowledge and experience with arena and stadium systems and packed it into these little giants!

Enhance your live shows with these advanced tools. What's the best approach for your unique needs? Call us now to talk it over!

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No other system gives you all this

24-bit recording With today's latest converters
Just a few years ago, this level of audio quality would have cost you thousands per channel.

116 dB dynamic range With our new 1224 analog interface
The 1224 gives you stunning audio specs that rival today's most expensive interfaces.

Balanced I/O With the 1224's eight analog inputs and ten outputs
All of the 1224's analog connectors are balanced +4 TRS or XLR for pro-grade I/O.

Tons of ADAT Optical I/O 24 channels of ADAT optical expandable to 72
The 2408 delivers all the ADAT optical you need for today's digital mixers, FX processors and other gear.

Loads of Tascam digital I/O 24 channels of Tascam TDIF expandable to 72
If you're in the Tascam world of digital I/O, no other system even comes close.

S/PDIF and AES/EBU I/O The new 308 gives you AES/EBU and two flavors of S/PDIF
8 channels each of optical "TOSlink" S/PDIF, RCA "coax" S/PDIF and AES/EBU — all in 24-bit glory.

NEW! 308 digital I/O

Expansion With the flexible PCI-324 card — the core of the system
Connect up to three 1224, 2408 and 308 interfaces for as many as 72 inputs/outputs.

Sample-accurate sync With digital transfers between your Mac and MDM's
Say goodbye to worrisome phase issues and other digital audio sync problems.

Broad compatibility With all major audio software for Mac and Windows
Use your favorite audio software with your favorite native plug-ins.

Audio format conversion Up to 24 channels at a time
Own the most flexible format converters out there — without paying extra!

Sample-accurate software with AudioDesk™, the workstation software for Mac OS
Make sample-accurate transfers with ADATS. Edit tracks with sample-accuracy.

Super-easy setup with our step-by-step Setup Wizard
You'll be up and running in no time.

Industry buzz Why is everyone is talking about the 2408?
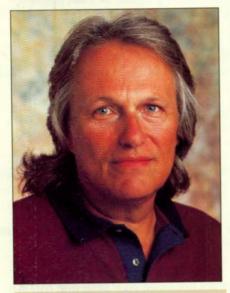Keyboard Magazine says it best: "Is the 2408 the audio interface system we've all been waiting for?...the answer is yes."

Price, price and price Did we say price?
A core 2408 system with 24 channels of input/output is only $995. Add a 1224 24-bit analog expander for only $995 — or a 308 for only $695. Mix and match them any way you like. At these prices, you can own just the right combination.

MOTU

MOTU 2408/1224/308 hard disk recording

The Best of Roger Nichols

Wait! Don't start reading here! This isn't Roger's usual column, but a special 10-year retrospective that starts on page 70.

GIRL PICKED UP IN BAR
• Undo Night of Horror (F3) — I have the shakes so bad I can't use my Pro Tools.
• Save (F5) — She did have nice nipples, *all eight of them!*
• Print (F6) — Girls like that should have Hazardous Warning labels.
And you get the drift.

I'M NO APRIL FOOL

Q: Dear Roger Nichols (you will always be my audio hero). I was wondering if you knew a formula or conversion chart for converting tempos to milliseconds in order to dial up the correct delay setting on an effects processor. I would appreciate any help you can give me. Thank You, Cesar Mejia

A: Hey, this is a smart guy! I may have to get one of those super-hero suits with a red cape. You can make your own chart using Excel or some other spreadsheet program, or you can easily calculate the delay time whenever you need it. The formula is: (60/tempo)*1000 = the 1/4-note delay in milliseconds. The rest of the note values can be figured out from there. An 1/8th note would be half as much. A dotted anything would be 1.5 times the delay of the un-dotted note. A triplet would be 2/3 of the delay value. I always carry around a pocket slide rule to calculate delay times. Since I disconnect the speedometer on my rental cars, I use the slide rule to calculate my highway speed by measuring the time delay between mile markers. Maybe someone should build a metronome that reads out in miles per hour.

IT'S ALIVE!!

I know for a fact that automobiles have a mind of their own. When you are nice to them and keep them clean, they run better. If you stroke the steering wheel just so, you can coax it to start so you don't have to walk home.

I am also a firm believer that audio equipment can turn on you in a heartbeat if you don't treat it nicely. When my digital delay is acting up, I can get it to work fine by plugging it into an outlet of its own on the other side of the room. My Harmonizer™ came back to life after I installed a CPU cooling fan over the digital converter chips.

On the other hand, my G3 laptop computer likes to be slapped around. The LCD screen keeps turning white so I can't see anything. It has never done it when I have taken it in for service, but it happens about 10 times per day when the service tech is not watching. I had to take videos of the screen before the tech would even believe that it was happening. (The guys in the studio thought I was nuts when I would run across the control room, grab my video camera, and start videotaping the screen on my laptop.) So, every time my screen turns white, I have to give it a backhand to the upper left corner of the screen. It then works fine for half an hour or so, then I have to give it another smack.

Then there is the Neve console that I have to whack on the side with my shoe. Kicking it doesn't work, I have to take my shoe off and smack it on the side by module one and the crackling noise goes away.

How about Pro Tools? How many of you have a love-hate relationship with your Pro Tools system? Right now, I bet there are at least a dozen people begging their Pro Tools system to please Save one more time before crashing.

My computer mouse has fun with me all of the time. I click the button to pull down a menu, drag down to the selection I want, and then just as I am letting go of the mouse button, the mouse pointer moves to the menu selection just above or below the one I really wanted. I know the neighbors can hear me when I yell, "*Noooooooo,*" and all of my animals scatter to the far corners of the house. (They have learned to split up so I can't find all of them at once. When I lived in Nashville, I learned to take frustrations out on my dog.)

Feedback is a way that equipment has of communicating its anger with you. After a mic preamp has to endure out of tune vocals for a while, a nice little feedback squeal always works to stop the session for a few minutes. New digital equipment has come up with its own way of slowing down sessions. Some nice subtle digital hair on the tail off of a piano chord, or that crunchy distortion in the DAT copy because the DAT machine didn't like the wire you were using. Hey, all of this new digital equipment is connected. Digital audio contains user bits. Do you use them? I don't think so! This is how digital audio equipment exchanges information. "Uh oh, here comes that same demo this guy's been working on for a year," says the digital console to the CD recorder.

"Maybe if I make a bad CD this guy will take me back to the store where he got me!" The CD recorder answers, "Hey, everything will be OK. You're lucky, you could've been bought by Jeff Baxter." That was a cheap shot. I guess I should apologize to my CD recorder.

Finally, I noticed that all of my equipment is much happier since I started feeding them balanced power. Nothing hums anymore. I guess it's because they finally know all the words.

Well, this last ten years has been a lot of fun. I forgot to tell you about the e-mail I received from a girl who reads my column in the bathroom. She hides the EQ inside a copy of Playboy so her husband won't become suspicious. I love my little show. Thanks for watching!