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AES **REPORT**

BLUE RIBBON **AWARDS**

MASTERING, REMIXING, AND DRUM REPLACEMENT **TECHNIQUES**

marilyn manson

BEHIND THE SCENES OF THE MAKING OF *HOLY WOOD* WITH ENGINEER DAVE SARDY

NOVEMBER • 2000
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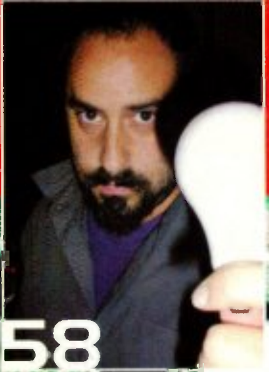
FEATURE REVIEW STEINBERG NUENDO

AES LA 2000 & Plus XXX, Paris



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CONTENTS

FEATURES

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Marilyn Manson and Dave Sardy (left)

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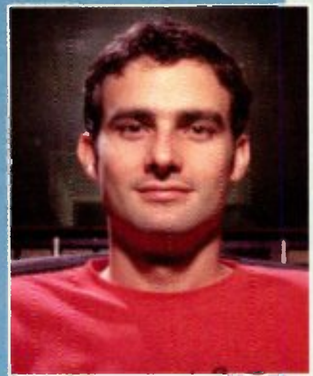

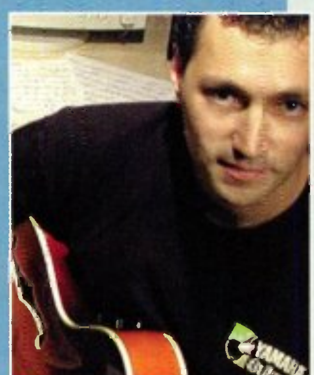

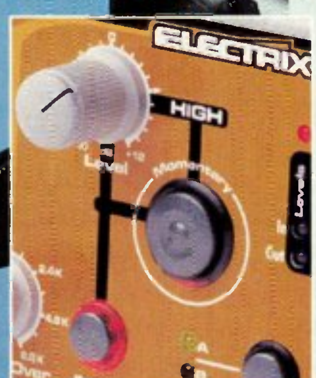

VOLUME 11 • NOVEMBER 2000

CONTENTS

REVIEWS

COLUMNS

DEPARTMENTS

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The Word for the Day

Recently I attended or observed a few events, and noticed an interesting parallel emerging. First, I happened to catch a couple of late-night broadcasts from the summer Olympic Games: some swimming, gymnastics, and weightlifting. Whatever your opinion of the commercialization and hype that surrounds the Games, it's still truly amazing the effort, talent, and commitment exhibited by the competitors — whether it's breaking a record in the pool, executing a new parallel bar move, or lifting three times their body weight, the athletes are simply inspiring. (The effort I expended preparing to watch the games was pretty inspiring, too — it's real work tracking down diet soda and vanilla-fudge ice cream at two a.m.)

Next, I attended the AES show, which was held in Los Angeles in late September. As always, there were tons of cool new products introduced as well as great discussion forums and paper presentations. But what struck me most was the meeting of the minds — the manufacturers with their energy and enthusiasm for the new products they've created and the engineers and producers with their desire to put those products to work to create better productions.

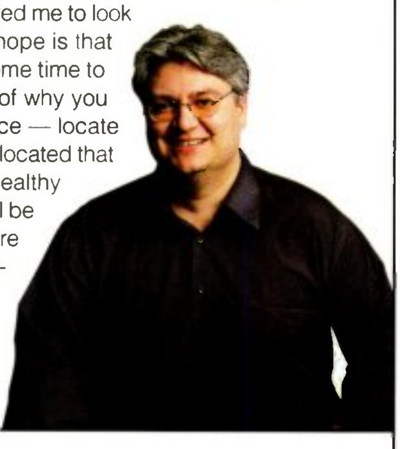
This past weekend I attended the Bass Collective's Bass Day (co-sponsored by our sister magazine, *Bass Player*). Held in a theater in New York, Bass Day consists of a bass-focused mini-tradeshow and a series of performances by respected bassists such as Gary Willis, Billy Sheehan, Will Lee, and others. As with the AES show, there was a meeting of the minds, this time between the gear manufacturers, with their basses and amplifiers, and the players, searching for the ultimate instrument and rig for realizing their musical visions. But, in this case, there was also the performance aspect; each featured bass player assembled a stellar cast of musicians (including Dennis Chambers, Scott Kinsey, Hiram Bullock, and many other outstanding players) and did their best to give the crowd an inspiring performance.

So what do these three seemingly disparate events have in common? Well, there are probably a number of similarities if you want to dig for them (I'm not that ambitious), but the obvious thing is *passion*. In each of the examples above, it was very clear to me just how passionate these people are for what they do; whether it's athletic competition, producing the best possible piece of equipment or musical instrument, or playing an instrument as well as possible.

I'd imagine that most of you are like me; you're involved in the project studio world because you love music and sound — you got into it because of that passion. Unfortunately, with deadlines, bills, budgets, technology problems, cookie-cutter music, and endless concerns over what gear to buy and how to use it most effectively, it can be easy to get jaded and lose that spark, that drive.

For me, the three events mentioned above helped to rekindle my enthusiasm for music and audio; they inspired me to look inside and rediscover my passion. My hope is that you'll find a way to do the same; take some time to find or do something that reminds you of why you got into music and audio in the first place — locate the source of your passion. Once you've located that spark, keep it alive. Make sure it stays healthy and prominent in your life. The reward will be that you'll do your job better, and have more fun doing it — which can only result in better music. And that's something the world always needs.

—Mitch Gallagher
mgallagher@uemedia.com



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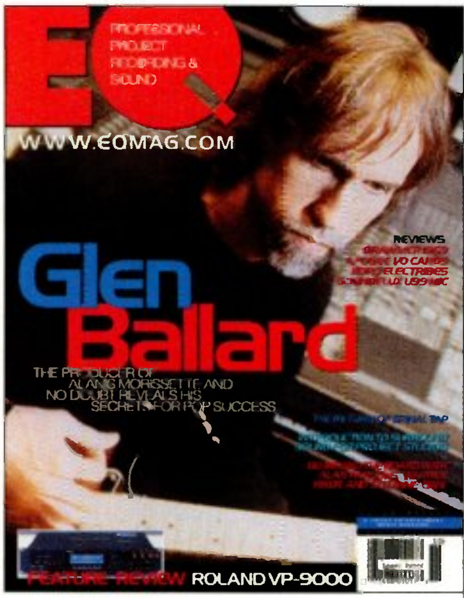

"NICHOLS AND KOOPER SHOULD RUN FOR PRESIDENT AND VICE PRESIDENT. WHAT A ROCKIN' COUNTRY — AN 02R IN EVERY STUDIO!"

—JOHN A. TYLER

FEAR OF WORDS

I have been reading *EQ* magazine for several years. It has always been a primary source for expert information about the music business. In fact, the recent Mr. Bonzai interview with Eddie Kramer [August '00] was a mother lode of information and entertainment. However, I was appalled, annoyed, and disgusted to see you stoop to the lowest grounds of political correctness and censorship by printing the non-existent, euphemistic piece of sh*t word, "sh*tloads."

Please understand that the fear of words is a disease; a neurosis that serves no good purpose, except to cause prejudice, alienation, and a host of other individual and social aberrations. You would have printed "feceloads" or "excrementloads" because they are considered by a certain group of people to be appropriate words to speak or write in mixed company. Well, they still mean "sh*tloads"! And they always will!

Let's use our intelligence to bring people together, rather than discriminate against them because we don't like their choice of words. If you have a fear of the word "sh*tloads," then don't print it. If you are afraid that some fanatical, word-fearing group is going to boycott your magazine, don't print it. Otherwise, exercise your right to freedom of speech, and be the first one on your block to help find a cure for the fear of words.

Michael B. Panasuk
via e-mail

THE FORCE IS WEAK IN THIS ONE

I'm writing to express my concern regarding the irresponsible coverage provided in your July 2000 issue. In Steve La Cerra's "Room With a VU" readers were introduced to Colorado's "The Blasting Room." Although such articles usually provide insight regarding new and unusual studios, they can also serve to provide hints and tips.

I was alarmed by the photo that accompanied the piece. It is clear to even casual observers that the proprietors employ two Watoo drink-cup toppers on each Yamaha nearfield. What arrogance!

Everyone knows that non-powered nearfields need the Anakin Pod Race topper for proper imaging. (At Treelady Studios, we use the Darth Maul toppers for our bi-amped HSRs.) Numerous university studies and tests have concluded that Watoo toppers can cause phase issues, add 2 dB at 2k, and even affect spelling accuracy.

I would expect more from a publication such as *EQ*.

Garrett Haines
Treelady Studios
Pittsburgh, PA

VOTE FOR THE *EQ* PARTY

I just wanted to drop a note and say "thank you" for a consistently excellent magazine. I look forward to my issue of *EQ* every month because I always learn something. I also love the columns. (Nichols and Kooper should run for President and Vice President. What a rockin' country — an 02R in every studio!) *EQ* was a godsend when I was recording my first CD and I've learned a lot of tricks recently for the next one. Keep up the good work.

John A. Tyler
Racine, WI

ONLY HALF-ASSED?

I have been receiving your magazine for at least five years and I enjoy it a great deal. I also count on it for new product information. Roger Nichols's column is always amusing and sometimes very informative. I especially enjoy his writing about the "heavy iron" items that I'll never get to play with myself.

And now the reason for my e-mail (I'll bet you wait all day for stuff like this): Your new graphics and layout eat it. Not that the old graphics and layout didn't eat it. But you seem to have gone from *Popular Mechanics* to half-assed *Wired*. If there was a conscious effort to make the copy harder to differentiate from the ads then that part at least is working.

Steve Etherton
via e-mail

HEY STEVE....

Somebody needs to tell Steve Vai [August '00] that when a Sennheiser 421 is set to "speech" the low-end is rolled off. The "music" setting is what he wants to keep for a full frequency range.

Rob
via e-mail

USE IT DON'T ABUSE IT

In response to the letter from Kurt Foster regarding "Today's records don't sound as good because the talent isn't there" ["Use It Don't Abuse It," August '00], I couldn't agree more. But that's also because we're inundated with the media and record companies shoving "music" down our throats — "commercial" music that people supposedly want to hear. Anything good or having value in my mind is never played on the radio or MTV. Upcoming bands with talent are forced to conform to three power chords and bad vocals or, even worse, rap! The engineers and producers behind these messes are nominated for Grammies, which may be due, but are there not other people doing way more creative things then simply putting a drum machine and vocal rap onto tape?

It's sad that we have some of the most amazing technology around us yet we have to listen to the radio put on another Moby tune and proclaim how great he is with today's technology or how this guy looped this into that. Don't even get me started on rap and DJs! I grew up with formal training studying the great composers. Today we have great composers, but they're not on the radio. If we "shoved" good creative music down people's throats, it would be heard and appreciated.

Let's get to the source: Good music, good musicians, and top-notch creative engineers/producers.

Ian Graham
Small Dog Studio
Kitchener, Ontario
Canada

ABUSE IT, DON'T USE IT

In reference to "Use It, Don't Abuse It" [August, '00], I am amazed by the negativity that seems to creep around the music world. Why don't people leave others' ways of doing things alone? If someone wants to experiment, create, or simply work on something the way they want — *let them!* Don't let it

become some big issue. My digital splicing and editing isn't hurting anyone. In fact, I have fun with it!

If Kurt Foster is so worried about digital editing, then I'm sure he'll tell about how terrible calculators are. Whatever happened to the mathematicians that calculated with their minds, pencils, and paper? Heck, let's just get rid of computers since they make things so easy — ucchh!

How about some positivity? *Relax*...stop messing with people.

Fish
New Orleans, LA

PORN ADS

I have enjoyed your magazine since day one. I am very disappointed in your choice to allow the BASF ad to be a part of your magazine. I find it offensive and am considering canceling my subscription. I promptly ripped the ad out of the new issue I just received. It's seems so unnecessary to put that kind of garbage in your magazine. Please leave the pornography out of your magazine otherwise I will cancel my subscription. If that were your wife or daughter would you want people looking at her like she was a toy to be played with?

Remember she *is* someone's wife and/or daughter. Why must we appeal to sex to sell our products? As for BASF, however good their products may be, I will not support them. Both *EQ* magazine and BASF insult me by publishing such trash.

Ed Bialach
via e-mail

LET'S HEAR IT FOR VINYL

I love Al Kooper, don't get me wrong, but his July 2000 column ["Victory for Vinyl Victims"] proves him to be a three-time loser! First he was a loser for not knowing that LPs are very collectible. Okay, my Aunt Minnie didn't know and sold my dead uncle's collection of priceless Blue Note originals for a buck a piece at a garage sale, but she didn't/doesn't work in the music biz like Al. But then Al, after admitting ignorance and losing once, does it again by making this ridiculous statement: "...but chances are unless you're DJ-ing, they're [your LPs] in a box in the attic...."

Yeah, right Al! File that along with your notion that LPs are worthless! I guarantee you that more people are spinning LPs at home every day than are listening to crappy-sounding MP3s! Al, LPs are still being made, both new and reissued titles.

Millions of new LPs were sold last year. Go to any college campus record store and you'll see LPs and kids buying them. Go to any good used record store and notice the age of the buyers. Youngsters, Al, the smart ones.

According to the *New York Times*, more *new* turntables were sold last year than *guitars*. Granted, many were for scratchin' and DJ-ing, but the makers of turntables for *listening* had a great year last year because millions of us still listen to our LPs. And why? For the reason that Al Kooper is a third-time loser: He actually thinks CDs sound better than LPs! That's because Al has never heard a properly played back LP. As Neil Young once said about 44.1k/16-bit digital, "The mind is fooled, but the heart is sad."

So Al, next time you're in the NY area, you're welcome to come over and listen for yourself. Bring your fave CDs and get ready for another shock to your system. I'll prove to you that 30-year-old LPs, played hundreds of times, once properly cleaned and played, blow the sonic doors off of your CDs. More importantly, they'll remind you of how recorded music once made you *feel*.

Michael Fremer
Senior Contributing Editor
Stereophile

[*Al Kooper replies: "You missed the intention of my column completely. Not only that, but you're preaching to the converted. I own 10,000 LPs and I have no intention of selling them. When I wrote that column, I was merely trying to alert readers that they might be sitting on some extra cash in the form of rare albums they had unknowingly stashed away. And calling me a 'loser' over your comparative opinions of vinyl is a little harsh and immature, wouldn't you agree?"*]

FIREWIRE INTERFACES

In the September 2000 issue, in the "First Look" article titled "Macworld NY 2000," Mikail Graham writes: "The only snag is the lack of analog I/O or PCI slots — you'll need a USB or FireWire audio interface."

He's talking about the new Macs, of course. What I want to know is, who makes a FireWire audio interface? I've been searching ever since I got my iMac DV SE, and I can't find one. Can you help?

Dustin Thompto
via e-mail

[*At the recent AES show in Los Angeles, Apogee Electronics announced FireWire Ambus cards for their line of converters; other FireWire-compatible gear should appear soon.*]

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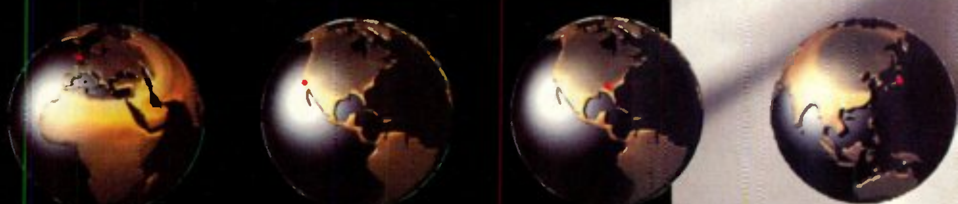

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FINDING YOUR VOICE

How are you supposed to naturally find your voice or most flattering style?

—c.cash

My voice has improved 200 percent since I began singing because of the amount of years I've been singing. It's because of the things I've learned in the last 40-odd years that I have improved my voice. I'd say the two most important items are staying within your range and being able to clearly hear yourself when singing

I wrote many songs in keys where I was conversant with certain chord changes. They weren't the best keys for my voice, but I didn't know any better then. For instance, I recorded "Season Of The Witch" and "Wake Me Shake Me" in F. I've been performing them for the last 15 years in D — much better for my voice.

If I can hear my voice loud and clear in the monitors on a live gig, my pitch has a *much* better chance of being correct. Ditto the studio. If my voice is loud in the 'phones and I can slip one 'phone half-off and hear my actual voice as well, I can sing more in tune. These two little factoids will take you a long way.

—Al Kooper

I've heard that if you monitor too loud it will actually cause your pitch to go sharp. Have you found any truth to this?

—c.cash

I have found that on earlier projects my vocals were lowered in the cans when I sang. This tended to make me over-sing. My vocals ended up coming out very shrill. On my most recent recording I had the engineer boost the vocals in the mix. I was able to sing with less "push" and therefore get the performance without straining.

—PlatEar1

Often enough overly sharp intonation in singers is due to their being overexcited during the recording process — meaning unnecessarily hyped or overly physical — in the same way that underactivation or depression produces a characteristically flat intonation. This pitch phenomenon is a result of the physical position of the palate — high palate, high pitch; low palate, low pitch. Often it seems that an overly loud cue mix (requested to get the singer "up" and feeling energized) can cause a singer to sing louder and with less control over pitch, dynamics, and so on. The real problem is getting a singer, many times the least-trained member of the band and the person most likely to have essentially nonverbal ways of thinking about their instrument, to address issues of technique so that things like pitch problems don't interfere with the energy of the performance. Trying to recreate live-performance energy only by adding volume means the singer needs to get his/her head together about what he/she is trying to record. I've had singers "energize" themselves in various ways, like a little exercise or stretching (obviously not to the point of being out-of-breath), and it's often this that makes all the difference in a singer who wails onstage but doesn't have the extra vibe in the studio — and once the headphone balance is right, it sounds exciting like the stage thing, but is in control enough to be recordable. The pitch problems (assuming a singer with good pitch) are often drastically lessened.

—ana1ana2

I believe I'd get my ass kicked if I asked the singer to do some exercise. Fortunately, there are many ways of energizing the singer. It is also nice to be able to let the singer adjust their own volume; they can get it where they like it better than you can.

—Al Kooper

COMPRESSION AND EQ TO TAPE/HD?

I usually like to print my sound after EQ and compression. Especially when it comes to recording drums. I feel that if I cut things flat, I will "fill up" the tape with frequency-stuff that I know I'm gonna get rid of anyway (in the mix). Would you recommend sticking to this philosophy? If not, what would I gain from cutting things flat?

—hjemmen

How do you know in advance what the drums or other instruments will sound like when they are mixed, and they interact with each other? Everyone EQs or compresses during tracking just by re-positioning a microphone or turning the volume up on an amp. But extreme EQ and compression via console or outboard gear might get you into trouble at the mixdown stage when all the instruments start to blend.

"Flat" gives you more choices come mixdown time. But there is *no set rule*. Some people like to even print reverb during tracking. I guess if you can hear the final mix in your "mind's ear" during tracking, then go ahead and print any EQ, compression, FX, etc., that you like.

—Miroslav

If you record enough projects, you'll start to realize in the mix what you shouldn't have done during tracking. I tend to know what frequencies I'm going to boost and cut before I track, so I leave very few things till the mix. I never compress drums, but I always EQ them. I compress everything else except acoustic and electric guitar.

Now that I track in Pro Tools, I do a lot of stuff on the way in because plug-ins just don't sound as good. Fix it in the mix doesn't work for PT — it forces you to get the sound out of your instrument.

—produceher

COMPUTER DAW VERSUS STANDALONE

I was just wondering if there is much sonic difference between recording with an audio card and computer, as opposed to using something like a RADAR or even a TASCAM DA-88 type system. If I was considering upgrading, which I am, does it pay to look into any standalone type systems? I'm only using synths and samplers, no vocal stuff or acoustic instruments.

—sterlingjerry

In general, the storage medium does not

WEBLINK

Have a question you'd like answered? Visit Roger Nichols, George Massenburg, Ed Cherney, Al Kooper, and David Frangioni online at www.eqmag.com.

AL KOOPER
GUIDE TO THE MUSIC BIZ

ED CHERNEY
BEHIND THE BIG BOARDS

ROGER NICHOLS
DIGITAL RECORDING FORUM

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CIRCLE 05 ON FREE INFO CARD

add or subtract from the sound. Tape can sound better or worse than disk-based recording and vice-versa. A few years ago, high-quality open-reel recorders (both 2-inch analog and 16-bit digital) were the most popular option due to their compatibility and great sound. Now, most pros are going with hard-disk recording because it can also sound great and offers a lot of flexibility previously unavailable with tape-based recorders. This is especially noteworthy because you are using samplers and synths. The ease with which you can fly parts around and combine both MIDI and audio is very powerful with a DAW.

—David Frangioni

LIVE 2-TRACK RECORDING

I'm attempting to improve the qualilty of my live-to-DAT recordings in my rehearsal room. It's a small carpeted room (about 15 x 20) with about a nine-foot ceiling. There is some acoustic treatment on the walls and ceiling; however, I wouldn't classify the room as dead at all. The drums are miked with two overheads (Rode NT1's), kick (D112), and snare (SM57). The bass goes direct, and the guitar amp usually has a '57 on it. If there are vocals, I usually use an Audix OM2, which goes to floor monitors. Everything goes to a Mackie 1402VLZ and then via the tape outs to a TASCAM DA20MKII. Typically everyone faces each other (i.e., it's not a "stage" setup).

Here are my biggest problems: [1] If there are vocals, I can never quite get them loud enough on tape for my liking; [2] I can never get the low end of the kick and the bass guitar present enough on tape. I suspect both of these problems stem from the drum overheads and phasing since the room is small and everything bleeds into them. Should I switch to close miking the toms, or go to small condensers as overheads and reduce the priority of them since they are currently catching the whole kit (and everything else)?

—ajcamlet

It might not be a bad idea to divide and conquer. Try recording only kick and bass on the setup and see if it really is cancellation with the overheads that's getting you. Get them sounding good and then move on from there.

The Mackie should let you boost the lows and I often find I need to do that on live tracked DI bass and kick with a D112, depending on how the kick is miked.

It also may help to EQ everything else; roll off the lows on the vocal mic and drum overheads, roll off the highs on the kick and bass. Roll off both some highs and lows on the guitar and keep the mids. This can really clean up your mix and give things their own space.

Also, try using an aux send for the room mix and mix the vocals as loudly as you need on the master mix feeding tape. The 1402 will let you do this.

—stevepow

You know, I've read and re-read your post, and I don't see where you've changed the phase on the kick mic. If you've checked and corrected the absolute phase of all of your mics (and it's surprising to find out how many deviate from the accepted standard), *then* you'll want to reverse the phase of the kick mic.

Think about it: The snare and overheads may be thought of as being on the *other* side of the kick head from the kick mic. Therefore they'll be out of phase unless something is reversed.

In fact, it wouldn't be a surprise if, when you reversed the phase of the bass, you noticed the level of the bass coming up in the mix with the other mics. There's almost always enough leakage to warrant a quick check to see which phase (positive or negative) results in more actual low-frequency level.

A word of advice: it's not always a clean, clear decision. Often when you switch phase you gain something and lose something. You'll need to make a judgement call.

Oh, the Mackie doesn't have a phase switch? Too bad, you'll have to resort to XLR-M to XLR-F "barrel" adapters, or the equivilent, to do the switch. And without a physical switch to toggle it's infinitely more difficult to get a quick read on which phase you prefer.

—George Massenburg

NEUMANN IMPEDANCE QUESTION

I read somewhere that Neumann designates impedance variations within their microphones with a colored dot located somewhere on the microphone.

I have two U 87's, one has a red dot, one has a light blue dot. I have also seen yellow dots on a U 87. I have a KM 84 with a red dot and one with a blue dot.

Can someone tell me what these dots indicate?

—Mark P

Most Neumann FET mics, as well as some tube mics in the 1960s, used a color dot next to the serial number to indicate whether that particular mic was shipped with a 50- or 200-Ohm output impedance. In the case of tube mics, the presence of a red lacquer dot on the I.D. plate meant that the mic was shipped with 50 ohms, its absence, that the strapping was 200 ohms.

Later, with the introduction of FET mics, the system was modified to drilling a small indentation into the brass body next to the serial number, typically at the bottom of the mic. A bare brass indentation meant the mic was strapped for 200 Ohms, if the brass indentation was filled with red color (changed to baby blue in the 1980s), the mic was shipped with 50 Ohms strapping.

The net-effect of the two different output impedances is that, all else being equal, the 200-Ohm output setting will give you 6 dB more output from your mic if you terminate into 1,000 Ohms input impedance (typical of most mic preamps). The way the two impedance settings are achieved in most Neumann tube and FET mics is by using either the two secondary windings of the output transformer in parallel (50 Ohms) or series (200 Ohms).

The change-over is easily done in most cases by re-soldering the respective color-coded transformer leads on the amp board, next to the transformer (too detailed for this discussion). One mic, the FET 47, has a convenient switch on the bottom to do the change.

Anticipating your next question: Yes, there is a difference in sound whether you connect transformer secondary windings in parallel or series. Most users with relative short cable lengths (under 50 feet) report that there's more beef in the output, particularly in the lower mids, when strapping to 200 Ohms, in addition to the better S/N ratio this strapping provides.

A sidebar: Gotham Audio, Neumann's U.S. importer until the late-1980s, insisted that every mic imported into the country not only be strapped to 50 Ohms (red/blue dot), but also be further reduced in output by another 4–6 dB by audio pads, either on the mic amp board or, for tube mics, on the power supply's audio connector.

The reason for this signal-to-noise-killer? The philosophy of the importer was that Neumann mics would sell better here if they didn't distort the inputs of mixing consoles, which were thought to be too anemic to handle the robust output of condenser mics. (Low-output ribbons and dynamics were the norm when Gotham started to conquer the States in the late-1950s with the high-output U 47's, M 49's, and later U 67's.)

—Klaus H

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Fostex VF-16 Digital Multitracker

By Steve La Cerra

The VF-16 from Fostex combines a 16-track hard-disk recorder with an integral 16-channel digital mixer, forming a completely self-contained digital desktop recording system. Using 20-bit A-to-D conversion and 24-bit D-to-A conversion, the VF-16 stores audio on an internal IDE drive as uncompressed 16-bit/44.1 kHz data.

Each of the sixteen 60 mm input faders on the VF-16 may be used to control a digital audio track. Alternately, the first set of eight faders can be switched to access the eight analog inputs for mixing of digital tracks with external analog inputs. In addition to the 16 "main" tracks, the VF-16 also provides an additional eight "ghost" or virtual tracks, providing space for multiple takes without erasing over previous takes. Since the VF-16's hard-disk recorder offers copy, paste, move, and erase editing functions, audio from the eight ghost tracks may be comped or edited into the main tracks.

At the analog "front end" of the VF-16 are eight inputs. Inputs 1 through 6 are on (unbalanced) 1/4-inch TS jacks (which can accept line- or mic-level signals), while inputs 7 and 8 offer a choice of balanced 1/4-inch TRS jacks or balanced XLR microphone jacks with 48-volt phantom power. Additional 1/4-inch TS jacks are provided for stereo monitor outputs, stereo headphone out, aux send 1 and 2 (pre or post), and a footswitch connection for remote punch in and out. A pair of RCA jacks carries the main stereo bus for mixing to any two-track recorder.

Each of the VF-16's analog inputs has a trim pot for controlling level to a hard-disk track. In the VF-16's "direct" mode, each input is routed via the trim pot directly to a track, bypassing the mixer. This allows simultaneous recording of eight tracks. A second "rec buss" mode allows recording of multiple inputs to a track (or pair of tracks) via the VF-16's mixer. In this mode, the mixer may be used to adjust level, EQ (three-band with parametric mid and high for all 16 channels), and effects from two Fostex ASP (Advanced Signal Processing) engines. The first ASP engine features a variety of hall, room, and plate reverbs, plus chorus, mono-delay-plus-reverb, and panning-delay-plus reverb combinations. Effect 2 features mono, panning, and BPM delays, plus chorus, flange, mono pitchshift, and delay-plus-pitchshift. Compression is available on channels 13/14 or 15/16, plus there's a dedicated compressor for the stereo master bus.

Additional rear-panel features include MIDI in and out jacks (the unit is MMC- and MTC-capable) as well as ADAT optical I/O. By using the ADAT optical input plus the eight analog inputs, a total of 16 tracks may be recorded simultaneously (the ADAT acts as the A-to-D for the second set of eight tracks). The ADAT optical input also offers an easy way to dump tracks into the VF-16 for editing purposes; waveform editing is facilitated by the front-panel 64x128 dot-matrix display. A SCSI interface is provided for backup or transfer of WAV file data into or out of the VF-16.

Other VF-16 functions include a separate EQ for the master bus and a scene sequence mode for automatic recall of mixer parameters for varying sections of a song. Frequency response of the VF-16 is stated as 20 Hz to 20 kHz, with a dynamic range of 92 dB. A removable plate on the bottom panel allows easy access to the IDE drive for project swapping or for expansion to a larger drive. A 5 GB drive provides approximately 960 track minutes of recording at 44.1 kHz/16-bit. ■

WHAT IS IT? An integrated digital hard-disk recorder and digital mixer.

WHO NEEDS IT? Musicians, songwriters, and recordists who require a self-contained recording system.

WHY IS IT A BIG DEAL? Eight simultaneous tracks may be recorded; 16 total when using the ADAT optical input.

SHIPPING: Now.

SUGGESTED RETAIL PRICE: $1,399 (includes 5 GB hard drive).

CONTACT: For more information contact Fostex at 562-921-1112, or visit their Web site at www.fostex.com. Circle *EQ* free lit. #101.

3-Way Active Accuracy!

The first affordable 15" 3-way active system: Mackie's SR1530 with electronic processing and FR Series™ internal tri-amplification.

Meet our active SRM450's big, over-achieving brother… the new **SR1530**. This 3-way triamplified active system delivers flat response at high output with low distortion and extremely wide dispersion in a surprisingly compact, portable enclosure that takes up less than two-and-a-half square feet of stage or floor space.

The new **SR1530** features…

- Electronic equalization, time correction/ phase alignment plus complete electronic and component protection circuitry
- Active circuitry makes the FR Series™ high-current tri-amplification more efficient than passive speakers because passive speakers have inefficient crossovers which cause significant power loss. Because of this, the on-board SR1530 amplifiers, with 500 total watts, can drive the speaker system to 126dB peak SPL.
- RCF Precision components…
- Wide-dispersion, high-output HF/mid-horn design with phase plug
- High-output, low-distortion 6-inch horn-loaded midrange transducer
- High-precision 1-inch exit, high-frequency driver
- 15-inch, high-efficiency, cast-frame woofer with heat dispersing Inside/ Outside voice-coil technology
- Correctly weight-balanced with two comfortable side handles for easy carrying and set-up plus top and bottom handles for easier positioning on stage
- Weather-resistant steel grille

Sound quality that's only possible with a true active system.

Though just 44 inches tall, **SR1530**s generate the sound output of much bigger systems… and are far more accurate to boot. Our active design achieves near-perfect interaction between transducers and internal amplifiers. Together, you get transparent and precise, high-resolution audio performance... only at PA output levels, which is the beauty of properly engineered, high-end 3-way systems

SR1530 rear panel with amp heat exchanger, female XLR input and male XLR full-range through.

Optimized Wavefront™ system "marries" high frequency and midrange.

In typical 3-way designs, mid- and high-frequency horns have symmetrical cross sections which physically force their output to different parallel locations in front of the box, causing uneven frequency response across the audience.

The SR1530's Optimized Wavefront™ horns have asymmetrical shapes with the high-frequency horn firing down into the six-inch midrange's dispersion pattern. Combined with built-in electronic time and phase correction, this creates a unified, wavefront with excellent phase and power response characteristics: Everybody in your audience hears the same thing.

Optimized Wavefront™ horn system

You gotta hear it to believe it.

Yes, you can log onto our web site for more info or call toll-free for an excruciatingly complete brochure. But the real amazement won't kick in until you hear a pair of **SR1530**s at a Mackie dealer.

If you're looking for the smooth frequency response and articulated midrange inherent in a 3-way system, there's only one Active way to go. The new **SR1530** from Mackie. ■

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Made in Emilio Emilia, Italy by Mackies who look almost entirely but not completely unlike those shown above.

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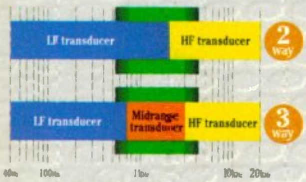
The story behind (and in front of) Mackie's new SR1530 Active 15-inch 3-way speaker system.

The potential of a 3-way system.

A properly-designed 3-way system is capable of reproducing vocals and instrumentals more accurately because it has a separate midrange transducer.



The transition range (crossover point) between transducers in a 2-way system falls in the middle of the critical midrange area (green area above). This part of the sound spectrum is being handled by the extreme high range of the LF transducer and the extreme low range of the HF transducer (where neither is most capable).

In a 3-way system, the HF and LF need only contribute in their more optimal frequency ranges. The key is melding the output of the three transducers.

The **SR1530** realizes the potential of the 3-way design by integrating:

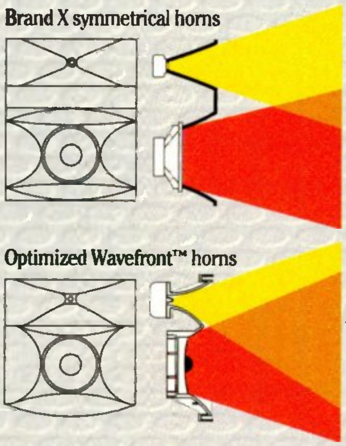
- ***Optimized Wavefront™ horns***
- ***Electronic time compensation, equalization and crossover design***
- ***Tri-amplification***

Optimized Wavefront™ Horns.

To achieve 40° dispersion, conventional systems use a high frequency horn with 20° of angle "on top" and 20° of angle "on the bottom." That beams high frequencies straight ahead, which means mids and high are actually aiming at different spots in the audience.

The **SR1530** is the first speaker in its class to feature a one-piece, 90° x 40° horn that includes both mid and high frequency sections. The 6-inch midrange's basket assembly is designed as part of the horn assembly and is also designed to function as an optimized compression chamber for better efficiency.

Both high and midrange sections of the SR1530's Optimized Wavefront™ horn have asymmetrical shapes with 10° angle "on top" and 30° angle below. High frequencies are directed down into the mid- range's dispersion pattern. This allows midrange and treble reach the audience as a focused, single wavefront with extreme accuracy and pin-point detail.



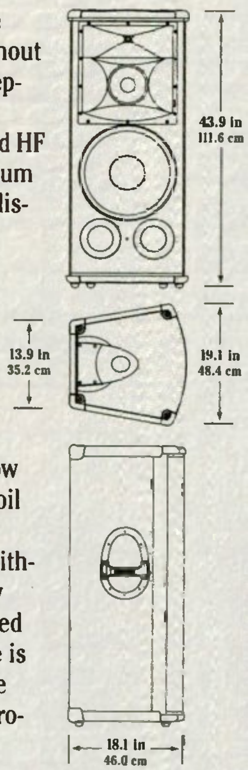
Active electronics.

If we stopped there we'd have a better passive system. But we didn't. Because the **SR1530** is an *active* system, we could employ a sophisticated, phase-accurate electronic crossover (instead of a crude passive one), time correction (to ensure that all three transducers' outputs arrive at the same time) and equalization to optimize each drivers' response to the enclosure as a system.

Individual FR Series™ amps.

The best way to power a transducer is to use a dedicated amplifier that's specially suited to its frequency range and power handling. Until now, that meant three separate amplifiers, a rack full of electronic crossovers and delay units, triple speaker wiring — great for huge touring systems, but very expensive and impractical for smaller applications.

The **SR1530** gives you the benefits of multi-amping without the cost and complication. Separate FR Series modules are directly coupled to LF, Mid and HF transducers, allowing maximum output with protection from distortion and burn-out.



Get the full SR1530 technical story.

These two pages are just a short introduction to the technology we've packed into the **SR1530**. Details like why the inside/outside low frequency transducer voice coil can handle intense amounts of power without damage and how solid and well-balanced the SR1530 enclosure is are covered in our free 80-page SR products brochure.

Call toll-free or log onto our web site and then sound good on stage with affordable 3-way active accuracy:

The new **SR1530** from Mackie Designs. ■

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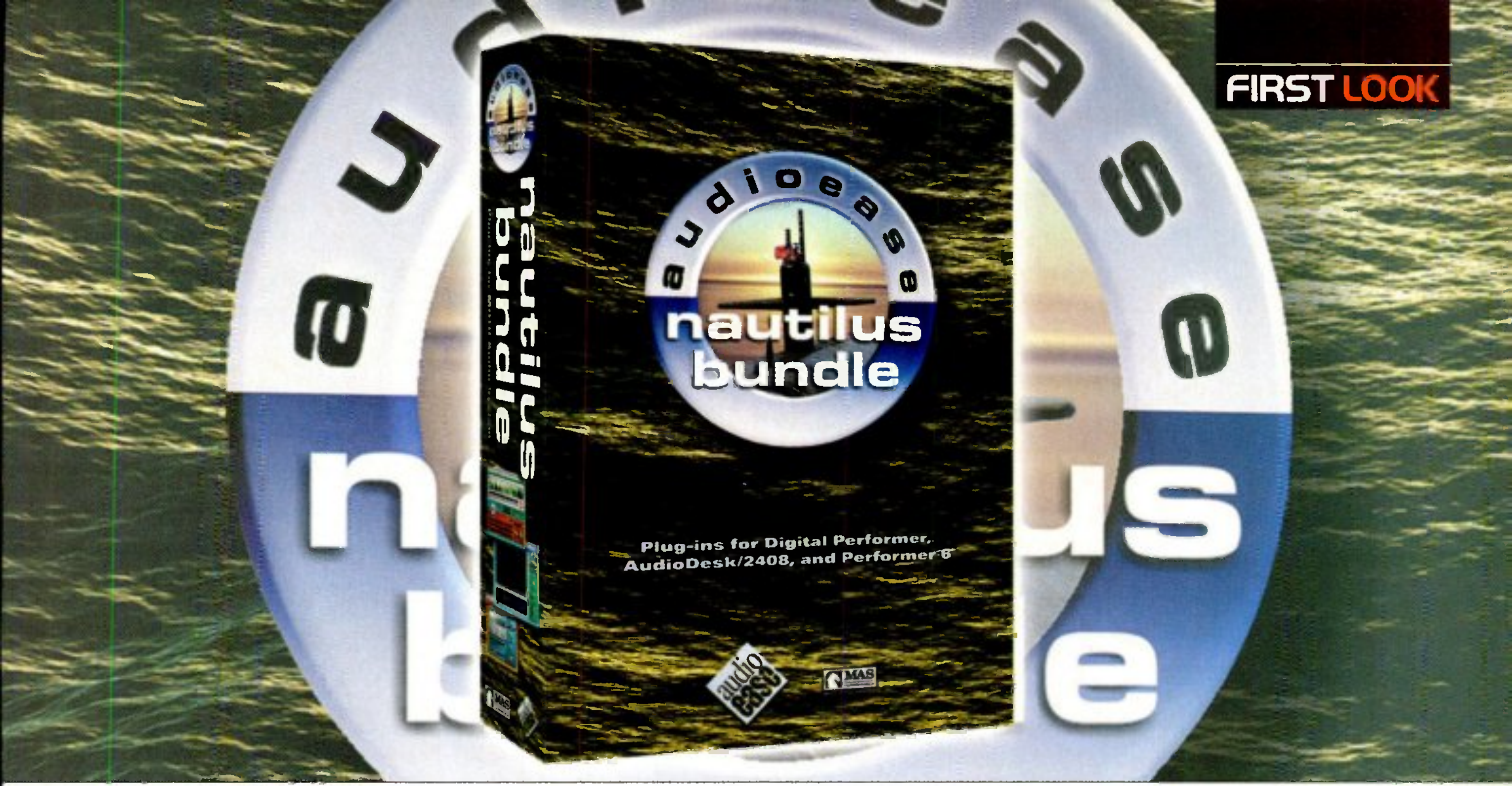

Audio Ease Nautilus Bundle

By Steve La Cerra

Recently introduced by Audio Ease is their second bundle of plug-ins exclusively written for the Mark Of The Unicorn Audio System (MAS): the Nautilus Bundle. Consisting of three new, distinct plug-in effects, the Nautilus Bundle includes RiverRun, Deep Phase Nine, and PeriScope. All three plug-ins are designed to seamlessly integrate with MOTU's Digital Performer in real time, and their parameters may be automated within Digital Performer.

RiverRun is a completely new means of real-time synthesis known as granular synthesis. Audio from a Performer track (or directly from a disk file) can be streamed into and "frozen" in RiverRun. This audio becomes the tone source for RiverRun, functioning much like an oscillator in a traditional synth. RiverRun then breaks the audio down into tiny particles called grains. The grains are routed through an envelope generator, pitched, and mapped across the output channels according to the control settings in RiverRun's window. RiverRun can "read" the grains of audio at variable speeds ranging from a fraction of normal through to faster-than-normal speeds, either in a continuous "flowing" manner or in a more discrete rhythmic manner. Lengths of the grains may be varied, and a Rhythmic mode makes it possible to lock random samples of the audio file in a groove against Digital Performer's tempo — all while simultaneously transposing the samples to preselected chord voicings.

RiverRun also includes unique parameters such as "Grain Glass" and "Walk Slider." Grain Glass lets you choose the portion of the source audio that is being granulated, while the Walk Slider lets you slide the grain glass forwards or backwards over the source audio at any speed you wish.

Deep Phase Nine is a phaser that features up to 24 bands per channel, a variable LFO, and the ability to self-oscillate. LFO wave shapes include sine, triangle, sample-and-hold, one-shot sine, and one-shot triangle. The user may "travel" a path on the LFO ramp by setting an end point and launching the phaser from an arbitrary start point. It will then travel from start to end at the LFO speed and stop. In addition to being able to vary speed of the LFO, the LFO may be synchronized to Digital Performer's tempo. An integral "look-ahead" peak limiter constantly monitors output levels, allowing intentional oscillation of a filter without worry that output levels will become out of control.

Nautilus' PeriScope is a phase-correct, 32-band equalizer providing narrow-bandwidth filters with an adjustment range of –144 dB up to +36 dB(!). Intended for "surgically precise" equalization, PeriScope allows you to zoom in on a specific area of the audio spectrum and apply up to 32 filters to that area. Bandwidth and center frequency of the filters is dependent upon which frequency range is zoomed in upon; minimum bandwidth is 11 Hz. Resolution of the display is user-adjustable, from showing the entire audio bandwidth to a specific frequency range, enabling extremely accurate dissection of audio. As of this writing, PeriScope is the only Altivec- (G4 Velocity Engine) ready plug in; the Altivec performance gain for PeriScope is approximately 80 percent. ■

WHAT IS IT? An MAS-compatible software bundle consisting of three plug-ins: RiverRun, Deep Phase Nine, and PeriScope.

WHO NEEDS IT? Digital Performer users seeking new plug-in capabilities

WHY IS IT A BIG DEAL? RiverRun offers real-time granular synthesis in a plug-in. Deep Phase Nine features extended phaser control. PeriScope provides surgically precise 32-band EQ capability.

SPECIAL NOTES: PeriScope 32-band EQ plug-in takes advantage of the G4 "Altivec" engine for improved performance.

SHIPPING: Now.

SUGGESTED RETAIL PRICE: $299

CONTACT: Audio Ease is distributed in the U.S. by Mark Of The Unicorn. For more information, contact MOTU at 617-576-2760 or visit www.motu.com. Circle *EQ* free lit. #102.

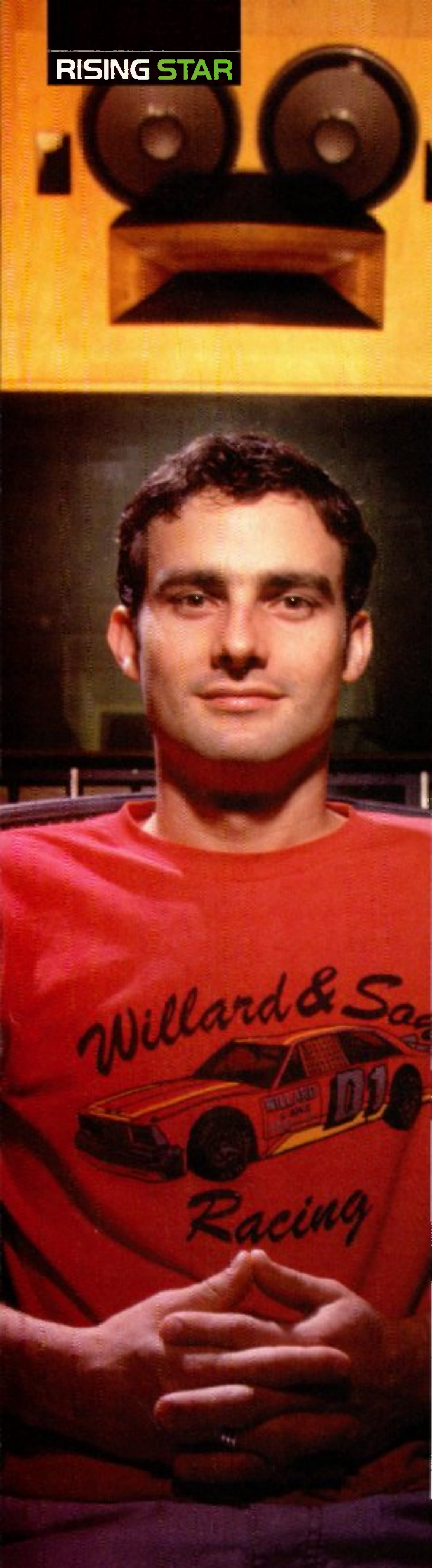

PHOTO BY EDWARD CURTIS COLVER

ELLIOTT BLAKEY

By Howard Massey

EQ poses ten questions to an up-and-coming engineer

Los Angeles's world-famous Record Plant has long been a hotbed of both established and developing talent. We recently dropped in on some mix sessions for an as-yet-unnamed band being produced and engineered by Toby Wright, ably assisted by this month's Rising Star, Elliott Blakey.

As a student in pursuit of a television production degree at the University of Florida, Blakey found himself increasingly gravitating toward the music scene. While playing in bands and recording friends on his four-track cassette recorder, Blakey was introduced to live sound reinforcement by a friend who also happened to have a home studio. Seven years later, with his Bachelor of Science degree in hand, he came to Los Angeles and began cleaning toilets for the Record Plant.

"5,470 food runs later," he says with a laugh, "I was graciously promoted to assistant engineer. No longer occupied with replenishing toilet paper dispensers, I accepted the challenge and responsibility required to work in a multi-million dollar facility." Viewing his role as an extension of the first engineer, Blakey's burgeoning skills have caught the eye of his fellow staff, making the likelihood of a 5,471st food run slim indeed.

EQ: How did you land your current gig?

Elliott Blakey: I just happened to walk into The Record Plant during a "runner drought."

How did you get started in engineering?

I spent my early college summers recording friends and their bands on my four-track cassette recorder.

Where do you see yourself in five years time?

Hopefully, working consistently!

What are your ultimate career goals, and how do you intend to accomplish them?

By keeping my ear to the ground (as I always try to do), I'd love to work with passionate, innovative performers.

Who are your heroes in engineering and record production?

There are so many engineers that I admire: Dave Fridmann, Andy Baker, Bob Weston, David Barbe, and Michael W. Rotolante, to name just a few.

What are your favorite current recordings, and why?

Chavez *Gone Glimmering*, The Flaming Lips *The Soft Bulletin*, Wilco *Being There*. I love any music that stimulates mental imagery as a direct reaction to aural perception.

If you were stranded on a desert island and could only take one piece of studio gear with you, what would it be?

An SSL 9000J would make quite the warm bed on cold island nights.

What's the coolest recording technique you've discovered?

If an idea seems absurd...try it!

What's the single best piece of advice anyone ever gave you?

Don't try to control each cow, just guide the herd! (Think about it.)

What's the single best piece of advice you can give our readers?

If you're working as an assistant engineer, observe and absorb.

WEBLINK

E-mail Elliott Blakey at melyot@earthlink.net.

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DUAL EFFECTS PROCESSOR

DUAL EFFECTS PROCESSOR

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TC Electronic's M-ONE and D-TWO are very powerful yet affordable new multi-effects units from one of the most highly respected names in signal processing. I highly recommend the M-ONE and D-TWO.
Mike Collins - Electronic Musician (US)

M-ONE FEATURES

- 20 incredible TC effects e.g. Reverb, Chorus, Tremolo, Pitch, Delay and Dynamics
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- 1/4" Jacks - Dual I/O

D•TWO
MULTITAP RHYTHM DELAY

RHYTHM TAP FEATURE

TC introduces the truly musical RhythmTap feature: Not only tempo, but actual rhythm patterns can be tapped directly - or quantized according to a specific tempo and subdivision. The delays and rhythm patterns can be up to 10 seconds each. Control the exact number of repeats with Absolute Repeat Control.

D-TWO MAIN FEATURES

- Multi-tap Rhythm Delay
- Absolute Repeat Control
- Up to 10 seconds of Delay
- 50 Factory/100 User presets
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- 1/4" Jacks - Dual I/O

A studio effects processor dedicated solely to providing digital delay is a rare thing in this age of all-singing, all-dancing multi-effects units. In the D-TWO TC have produced a very attractive delay unit that sounds exemplary and offers some unique features as well as all of the expected ones.
Paul White - Sound On Sound (UK)

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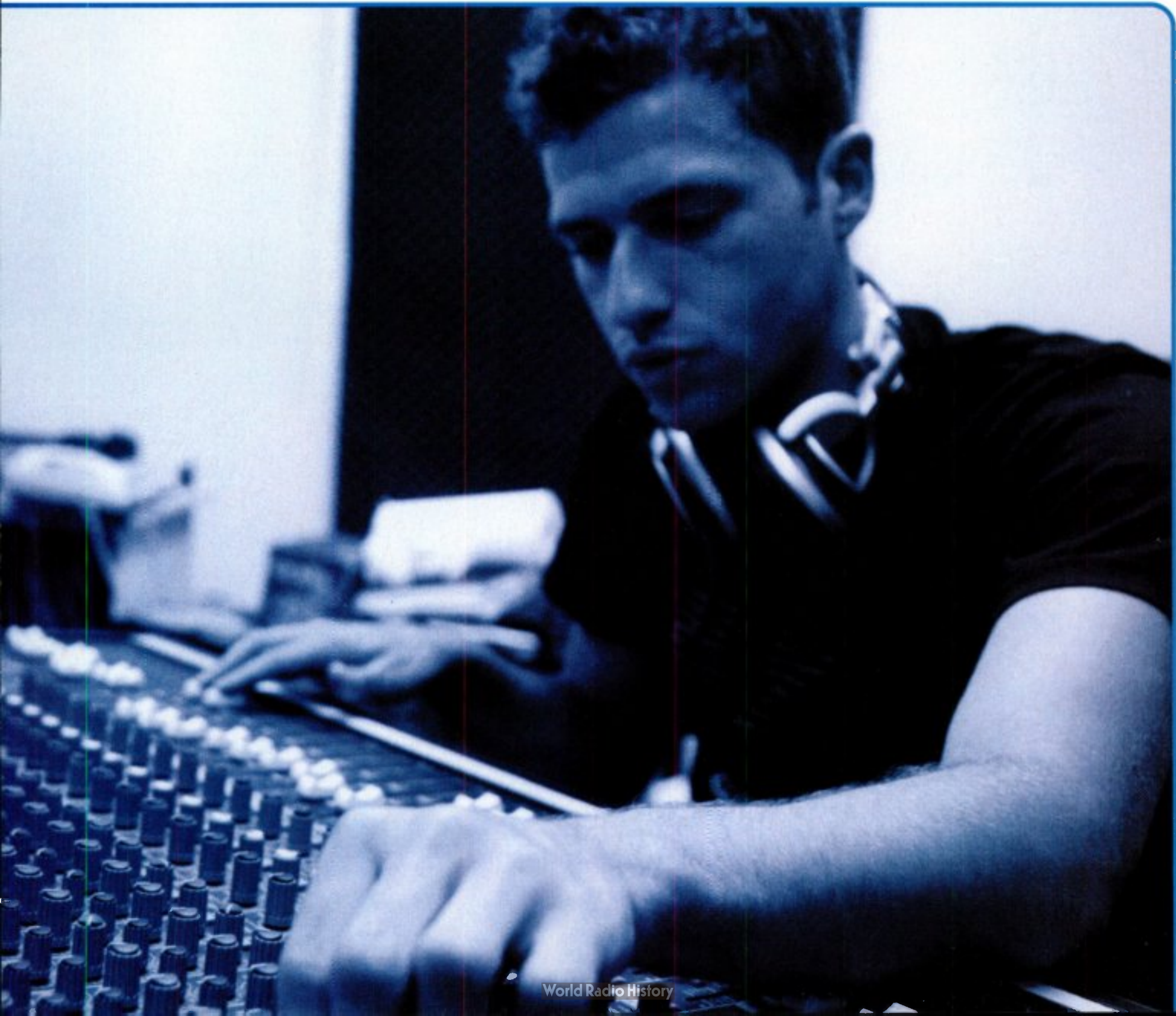

BWA STUDIO

WEBLINK

Visit the Vinny Valentino Web site at www.vinny.com.

BWA Studio

Jazz artist Vinny Valentino's recording home

By Steve La Cerra

STUDIO NAME: BWA Studio
LOCATION: Teaneck, NJ
KEY CREW: Vinny Valentino, John Benitez
CREDITS: In addition to releasing five solo albums, *Now and Again, Distance Between Two Lines, Rudolph Valentino, Live at the Cafe Japone,* and *Center Place* (recorded completely in his studio), Vinny Valentino has shared the stage or studio with jazz greats that include Tom Scott, Gary Bartz, Bob Moses, George Benson, John Pattitucci, Steve Gadd, Patrice Rushen, Charlie Byrd, Buck Hill, and Jimmy McGriff. Valentino has also appeared as both a guest and guest host on BET's *BET on Jazz.* One of the first musicians to perform live on the Web, Vinny continues to pioneer on the Internet. He can be seen at **www.vinny.com** giving online performances, lessons, and master classes.
MIXING CONSOLE: Yamaha 01V
MONITORS: Yamaha MSP10 (powered)
RECORDERS: Yamaha DSP Factory and MT44; Sony DATman
OUTBOARD: Valentino notes that he's "sold all the outboard stuff, and I use plug-ins for compression and gates."
EFFECTS: Yamaha REV500, DSP Factory
MICS: Sennheiser MD441, MD421; Shure SM57, SM58, SM53; AKG C414 [2]
SAMPLERS/KEYBOARDS/MIDI
MODULES: Yamaha MU80, MU100
COMPUTERS: PIII 500 MHz with Asus motherboard, 128 MB RAM, Graphics Blaster video card with 16 MB RAM; 4.5 GB SCSI hard drive, 8.5 GB, 4.3 GB, 20 GB IDE hard drives; Yamaha SCSI CD burner. Second computer is a PII 350 MHz with 128 MB RAM and 20 GB hard drive. Both computers share a KDS 15-inch monitor and are networked via ethernet.
SOFTWARE: Cakewalk Pro Audio 9.1, Guitar Studio, Musician's Tool Box II and III, Q Tools, Cakewalk Effects I, II, III; DSP Effects, Steinberg Wavelab 3.0
STUDIO NOTES: Valentino used "a Yamaha MT44 four-track recorder when I first started," he explains. "The studio has since graduated to a Yamaha 01V digital mixer and DSP Factory, which are the heart and soul of my setup. I also use the latest version of Cakewalk Pro Audio 9 for recording and sequencing."
EQUIPMENT NOTES: "A good headphone mix is one of the most difficult yet sought-after aspects of studio recording for jazz musicians. In order to get any separation, you need to monitor through the headphones. The 01V and DSP Factory allow for good separation. For every mixing and recording project, I'll record a scene memory for each song, including settings for the headphone mix as well as EQ, panning, output, and effects. You can automate with the 01V in many ways, but I do it via MIDI for program changes, fader movements, and effects."
PRODUCTION NOTES: Although Valentino's guitar parts on *Center Place* were recorded directly into the DSP Factory, sessions at his studio "generally revolve around the miking and tracking of acoustic instruments. The *Center Place* CDs acoustic bass and hand percussion were run directly to the 01V. I took the 01V's digital sub outs for the parts I wanted in stereo, and those went to the DSP Factory's stereo digital input. I then recorded into the DSP Factory (on a number of different hard drives) to Cakewalk. Within Cakewalk, there is the Audio X console, which is a DSP Factory console inside the software for easy control over DSP Factory parameters." ■

By Steve La Cerra

B.O.M.B. Studios

Danger Danger drops the B.O.M.B. on pro baseball

STUDIO NAME: B.O.M.B. Studios
LOCATION: West Hempstead, NY
KEY CREW: Bruno Ravel and Steve West
CREDITS: As founding members of the band Danger Danger, Steve West (drums) and Bruno Ravel (bass, guitar, and vocals), have released five albums, including their current CD, *The Return Of The Great Gildersleeves*. Their forthcoming CD, *Cockroach*, is slated for release in November 2000 on Low Dice Records (JVC in Japan). Steve and Bruno have written and recorded "Team Themes" for the New York Yankees, Baltimore Orioles, Boston Red Sox, and New York Mets (released under the BMG Special Projects label). Their recent activities include producing the band LMNT ("Element"), on a soon-to-be-released CD, recording tracks for the band West World, and producing and recording music for a track on the upcoming CD for DJ Hurricane (of The Beastie Boys).
MIXING CONSOLE: Mackie D8B
MONITORS: Mackie HR824, Audix Studio 3A, Yamaha NS10M, Sony MDR-7506 headphones [3]
AMPS: Hafler P4000, Samson Q5 Headphone Amp
RECORDERS: Alesis ADAT XT [3], TASCAM DA-30mkII
OUTBOARD: Empirical Labs EL8 Distressor [2], Avalon VT-737SP, Rocktron Chameleon preamp, Line 6 Pod
EFFECTS: Lexicon PCM90, MPX500, MPX100; Yamaha SPX900, Eventide H3000 SE, Alesis Wedge
MICS: AKG C414, Rode NT2, Lawson L47, Shure SM57 [2]
MIC PREAMPS: Avalon VT-737SP, Demeter VTDB-2b Tube Direct Box
SAMPLERS/KEYBOARDS/MIDI MODULES: Digidesign SampleCell II, Kurzweil K2000, Nord Lead II, Roland JV-2080 ("with a bunch of expansion cards"), JX-8P; Korg Trinity TR Rack, M1; Alesis Nano Piano
COMPUTERS: Apple Mac G4/450 MHz with 320 MB RAM, Maxtor Diamond Max Pro 40 GB drive [2], Quantum Fireball 10 GB drive and Cheetah 18 GB drive; both connected to an Adaptec 2940U2W SCSI accelerator; Mark Of The Unicorn 308 and 2408mkII
SOFTWARE: MOTU Digital Performer 2.72, Bias Peak 2.5, Digidesign Sample Cell II, Rebirth, BitHeadz Unity DS-1. Plug ins: Waves Gold Bundle, Metric Halo Labs Channel Strip, Kind Of Loud Real Verb, Antares Auto-Tune and Mic Modeler, plus "lots of VST plug ins."
STUDIO NOTES: Steve West explains that "Bruno and I are making records basically in our home, and they're comparable to records with a quarter-million dollar budget. There are some things we do at other rooms, but with what Bruno brings to the table as an engineer, we're able to do great productions. And, of course, we're not rushed, so we can take our time to get it right. It's so comfortable, and it's the best thing that's ever happened to us."
EQUIPMENT NOTES: Bruno has a few pieces of gear he relies on, including the Lawson L47 microphone, the Avalon VT-737SP, and the Distressors. "I love the L47 — to me, it's like: why go any further? Using the L47 through the '737 sounds gorgeous for vocals — sweet and warm. I use the Distressors on just about everything, sometimes even the two-track mix. Digital Performer is the main recording system, though I do use the ADATs often. I might work on analog tape for tracking, then transfer to ADAT and bring it into Digital Performer."

According to Steve, "The D8B has made life so much easier because of the recall — switching between projects is much quicker now. We just had a few different projects going at once: a Christmas song for Danger Danger, a mix for LMNT, then the sports teams projects, and it's seamless going from one to the next." ■

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CIRCLE 37 ON FREE INFO CARD

PHOTO BY WES BENDER

RCA BK-10B

MICROPHONE NAME: RCA BK-10B (MI-11018-B)
FROM THE COLLECTION OF: Bob Paquette, The Microphone Museum, Milwaukee, WI
PRICE WHEN NEW: Unknown
YEAR OF MANUFACTURE: circa 1960 (see notes)
TYPE OF MIC: Ribbon
FREQUENCY RANGE: 50 Hz to 15,000 Hz
DIRECTIONAL CHARACTERISTICS: Unidirectional
EFFECTIVE OUTPUT LEVEL: –55 dBm for a sound pressure of 10 dynes per square centimeter
HUM PICKUP LEVEL: –128 dB, referred to a hum field of 1 x 10^{-3} Gauss
OUTPUT IMPEDANCE: 200 Ohms (can be changed to 40 Ohms)
EXTERIOR FINISH: TV gray and black
DIMENSIONS: 10.875 long x 1.75 diameter (inches)
WEIGHT: 2.75 pounds (less cable)
MOUNTING: 7/16-inch hole provided in the mounting arm for boom attachment
MIC NOTES: Also known in the industry as the "Perry Como" microphone, RCA's BK-10B was a highly directional microphone produced by RCA in very limited quantities (many of those produced went to the Canadian Broadcasting Company). Developed specifically for NBC Television Studios, the idea behind the BK-10B was to produce a microphone that could be boom-mounted for television applications, had the sound of RCA's BK-5A uniaxial microphone, and yet would be able to reject unwanted background sound in favor of Como's soft voice. To accomplish this, RCA engineers used two closely matched BK-5A elements in the BK-10B, mechanically arrayed in a single tube with vents for each element. Output of the two elements is electrically summed through a passive network contained in the microphone body. The combined output of the two elements yields a tight pickup pattern akin to today's hypercardioid pattern, and capable of "reaching" almost twice the distance of a cardioid mic.

According to company literature regarding the BK-10B, "an RCA-developed blast filter is incorporated in the unit to reduce the possibility of damage from gun blasts."(!) Although the microphone appeared in few (if any) RCA catalogs, technical approval of the BK-10B was signed off May 12, 1960.

For additional information, see the RCA BK-5A MicroPhile in the April 1996 *EQ*.

Technical information furnished through the courtesy of Arthur Garcia and Bob Paquette.

The "kid from Iceland" records Sheryl Crow, the Wallflowers, and Los Lobos

Husky Höskulds

SUSPECT:	S. "Husky" Hoskulds
ANCESTRY:	Born and raised in Iceland
HEIGHT:	6 ft. 4 in.
OCCUPATION:	Recording engineer
RESIDENCE:	Los Angeles, California
VEHICLE:	1991 Suburban
DIET:	Smoked trout
CREDITS:	Suspect has tracked and overdubbed the Wallflowers, Michael Penn, Sheryl Crow, Los Lobos, Vonda Shepard, etc. Currently recording Joe Henry.
NOTES:	Höskulds studied audio engineering at UCLA. First job was as a runner at One On One Studios (now Extasy Recording), followed by an assistant engineer gig at Grand Master, then Hollywood Sound, and finally the Sound Factory, where for a number of years he assisted the producer/engineer team of Mitchell Froom and Tchad Blake.

Mr. Bonzai: Why did you become a recording engineer?

Husky Höskulds: I was just a fan of music and interested in the gear, so this seemed like the only job where you could meld the two.

Did you have any teachers?

As far as making records, it would have to be Tchad and Mitchell. On the strictly engineering side, Tchad definitely had a lot to do with the way I do things now. Working with those guys was an eye opener. Being a part of making records with artists like Los Lobos — it feels like you're working on the Beatles. You gotta make sure you don't do anything stupid while taking it all in.

What is Tchad's greatest quality?

He's incredibly generous — that's hard to find in this business. He's fun to work with and fun to hang out with.

What gear do you have in your personal arsenal?

I have a lot of percussion instruments, like funky old tambourines, bongo drums, and a bunch of old snares. I try to bring things that are different from the norm, something odd that has a different kind of sound from what most drummers would bring to the studio. The most logical place to start — if you want to have an influence on the sound — is to go to the source, as opposed to doing it all from the other side of the glass. I also have some keyboards — an old S-4 tube Hammond organ, a Clavinet, a pretty decent pump organ — instruments with a different texture.

What about outboard gear and processors?

I've got a couple of ratty compressors and a pretty unimpressive collection of gear compared with the "good" stuff. I don't have any LA-2A's or 1176's, but all the studios do. The ones I have are the dbx 163's, Orban stereo compressors, and things that have a drastic color — a big effect on the sound. I like gear that has a strong character to augment what is in the studios. I also have a couple of nice M-1 preamps made by John Hardy that are great for ribbon mics and for vocal work. A lot of gain, but they are very clean.

What about the major gear — any preference on consoles?

I don't have to work on an API or a Neve, but they happen to sound great. I find it easier to work on the APIs at Sound Factory, because they are well laid out. You don't have to get up and walk over to the other end of the room to push up the tambourine on channel 74.

Analog or digital, which do you prefer?

I'm fine with both. I usually do my tracking on analog two-inch, and then I have a Pro Tools rig, the 001 system, that I use in tandem — more as a piece of outboard gear and a sampler, a device to loop things and do a couple of quick edits. I use it as a back up, for archival purposes, and when I need to twist things around a bit.

Quick impressions of people you've worked with — was it a thrill to work with Bob Dylan's son?

Absolutely, but it was also great fun to work with such fine musicians, with great senses of humor. I'm just a kid from Iceland, you know, and there was no pop music when I was growing up, so the fact that he is Bob's son wasn't really such a big deal to me.

How was it working with Sheryl Crow?

Pretty cool. She had done a considerable part

HUSKY HöSKULDS PHOTOS BY MR. BONZAI

Tchad Blake (standing) and Husky Höskulds

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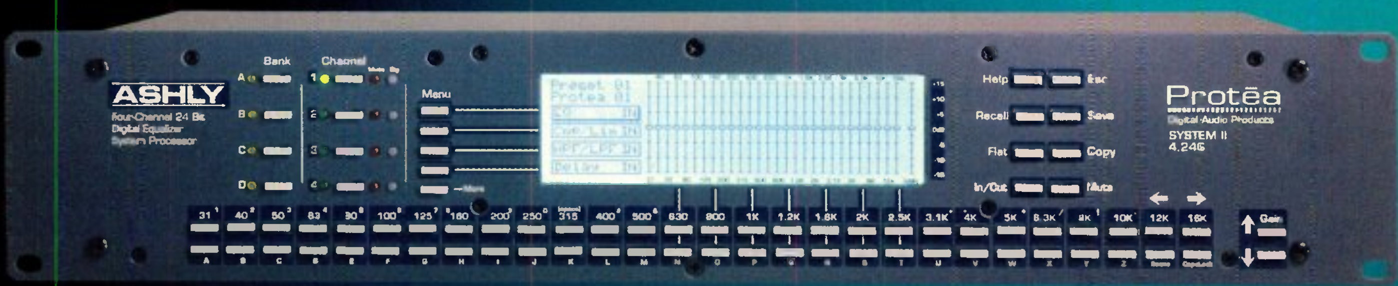

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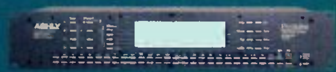

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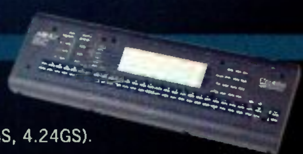

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CIRCLE 06 ON FREE INFO CARD

of the *Globe Sessions* by the time she got to L.A. and I got involved with the project. We did three or four tracks, and it was good fun. The pressure was off, because she had already done most of the record. She'd be working on a song with her guitar player Jeff Trott, and we recorded pretty fast — just blew through the material. It came out really well, and she was excited.

How did you get the gig?

I had worked as an assistant for Tchad on her previous record.

What have you been up to lately?

Last night was one of the best sessions I've ever worked on. I'm recording Joe Henry now, and we had Ornette Coleman come down and play. Incredible people involved, the players are great, songs are amazing. Ornette came in, got his horn ready, and just started blowing. We put down three or four tracks and after each pass I'd turn around and look at Joe, our producer Craig Street, and our arranger Steven Barber — their jaws just dropped. The first take was great, then it just got better and better. The last take really blew the roof off and, after I stopped the tape, he said, "Yeah. That was it. On those other takes I was just playing the sax, but on the last one I was playing the music."

What kind of mic did you use?

I had a Coles ribbon mic.

Did you ever screw up in the studio?

Of course not — that's not good press.

WEBLINK

Questions or comments for Mr. Bonzai? E-mail him at mrbonzai@mrbon-


SHURE
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If you were a microphone, which would you be?

I'd be a skinny long one like me, maybe a C12 — y'know, very bright.

What's wrong with the music industry?

I'd say there are too many people getting record deals who can't really play.

What do you listen to while you're driving?

Lately I've been listing to Ennio Morricone, Raymond Scott, Esquivel, and Mr. Bungle.

If you could go back in time, what would you like to record?

I wish I could have been around to work on those Esquivel sessions.

What is the first music you remember hearing?

My grandfather, the choral director, rehearsing the town church choir in the living room every week. Grandma would make coffee and cookies. It was like going to church, except you could go to the bathroom whenever you wanted to.

If you could pick anyone, is there anyone in the world you would like to record?

Tom Waits.

Do you know any interesting music business tricks?

No, not really. You turn in your invoices like anywhere else — it just takes longer to get paid.

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

I hate to single out just one, but there is no way you can't mention David Hidalgo. But I would like to also mention the people on this Joe Henry record: Marc Ribot on guitar, Brad Mehldau on piano, Abe Laboreal, Jr. and Brian Blade on drums, and Dave Piltch and Me'shell Ndegéocello on bass. Craig Street really put a great band together.

What was your most ridiculous experience in a recording studio?

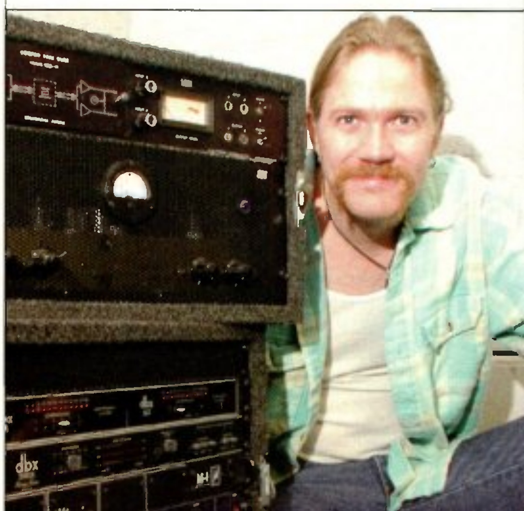
I was working on a hip-hop record at Hollywood Sound as an assistant. The producer brought a little bag with his own master fader so he could run the signal through it and it would "sound amazing." It went in one end, came out the other, and had absolutely no effect on the sound.

Any advice for getting a good start in the engineering field?

You have to really mean it. Many engineers, assistants, and runners are in it for the wrong reasons. You really have to be sure this is the life you want. It's pointless to stick around if this isn't what you really want to do. It's a very time consuming job, and you get back what you put in. You can't just show up and then go home — you gotta put in the extra effort. You can't trick me into thinking you know what you're doing. We've all been runners and assistants and helpers. It's easy to spot the guys who are good and the ones who aren't.

What would you like Santa to bring you this year?

A week off with my family at a bed and breakfast up near Santa Cruz. Nothing fancy — I don't want him to feel taken advantage of. Even a three-day weekend would be good. ■

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CIRCLE 10 ON FREE INFO CARD

Producer Craig Leon explains how he uses similar approaches for recording New York punk and modern classical music

Craig Leon

By Howard Massey

One of the maxims of the music business is that it can make for strange bedfellows. Who would have thought that a classically trained musician with finely honed composition, harmony, and arranging skills would be the primary architect of the second-wave New York punk movement of the mid-1970s?

The man we're talking about is Craig Leon. Not only did he discover the original thrash-rock band — the Ramones — and go on to produce their groundbreaking first album, but he proceeded to bring many other denizens of CBGBs and Max's Kansas City to the listening public — bands such as Blondie, Richard Hell and the Voidoids, and Suicide. Eventually abandoning the Big Apple for the lure of L.A., Leon's eclectic career took a decidedly country twist — working with singers Rodney Crowell and Dwight Tilley — before he relocated to London, his present base of operations. In 1999, Leon reunited with Blondie for their critically acclaimed comeback album No Exit, *and today he finds himself returning to his classical roots as he explores new ways of assimilating modern recording technology with conventional orchestra performance. To say that he has a singular perspective on the past, present, and future of record production would be an understatement.*

EQ: How do you feel your classical background has enhanced your ability to make pop records?

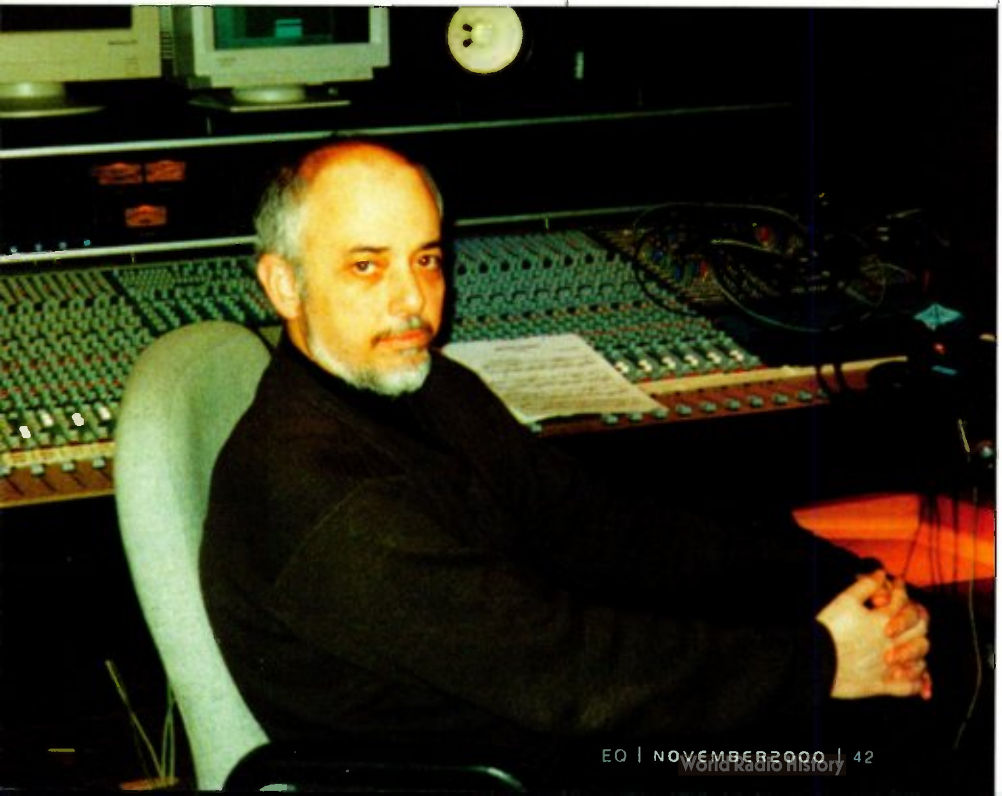
Craig Leon: I taught myself how to engineer in the '70s because I was flying by the seat of my pants and nobody would engineer the way I wanted to. I originally came from an arranging background — orchestration, composition, and piano — and a lot of that ended up in my method of production. I would actually attack each recording as you would a composition — lay it out in blocks, like you would with normal orchestration. So I would do a lot of things in the arrangement that ultimately affected how the record sounded. Not on something like the Ramones, where everything was just a big wall of sound, but that approach is more evident on something like Blondie — music that has a lot of countermelodies in it. Even something which sounds radical, like the Suicide records, were really thought out in terms of sonics — which frequencies people were going to be playing — rather than trying to EQ it. It's about, "Let's change the register where that bass is," going through that type of preproduction to create a hole for the vocal. So, consequently, on a lot of those old punk records — and even up to the new Blondie record — I don't use a lot of EQ and I don't use a lot of effects; the arrangement is what shapes the sound of the record. Which is not a million miles away from what we were doing with classical records. Doing things with harmonies and different chord changes and not necessarily playing the root all the time in the bass changes the way a record sounds more radically than anything you can do electronically. The sound of a record has much more to do with what the musicians are playing and singing than it does mic placement.

Conversely, how has your experience as a pop record producer influenced the work you're doing now in the classical genre?

Actually, what I'm trying to do is to *not* do core classical recording. I'm doing more modern things — something that's very early music, but modernized. I'm trying to incorporate some of the techniques of pop in classical recording, like using multitrack — even though they are live orchestral performances — and doing things with sequences. Sometimes, there are live instruments sequenced [through digital editing] and then overdubbed with more live instruments. So it's actually making very radical recordings in the classical world — I guess I can't ever quit making radical recordings of one kind or another! [*Laughs.*] There's not a lot radical you can do in pop right now; it's pretty much established.

In 1900, they said that everything that was ever going to be invented had already been invented, so that's a dangerous point of view.

Well, I'm not quite saying that. Someone could come out in the next year and make the killer record of all time that nobody's ever heard before — please, somebody do it! But I'd like to do a



straight core classical record in a modern way, where you do something multi-tracked and use different EQs and different echoes to change the shape of the music a little bit. That's one thing I'm really interested in right now.

You said earlier that you learned engineering because nobody was around who could engineer the way you wanted.

Well, you've got to realize that it wasn't that far away from the '60s when we were starting to do Blondie and the Ramones. When you'd go [into the studio], you'd have people that were still unaccustomed to live, loud music. They were still trying to dampen things down, even in 1973 and 1974, when I started doing some of the preliminary work on the things that came out in '75, '76, and '77. So you'd get very conventional engineering, very flat-sounding compared to what I wanted to hear. I wanted to hear the impact of a Ronettes record without having 300 things playing. [*Laughs.*] That was the goal; I wanted to hear the band sound like that. But wanting to get that larger-than-life sound involved a lot of things that were different in those days.

The way we did the new Blondie album is pretty much the same way we did the old Blondie stuff, which for then was really radical but now is the standard way to make a record. It was using a lot of room mics with leakage. On the early Ramones and Blondie stuff, there's hardly any processing, and there's very little reverb. Even though it sounds really roomy, that's actually the sound of a huge studio. Then, that was really radical — you didn't do that. Instead, you put an amplifier in a little cage and you put the drums in a little booth with foam all around it. So I'd have a lot of trouble with engineers wanting to do that.

A couple of guys really got it. I worked with Shelly Yakus on a couple of things back then and he really had it down, but because of the low-budget nature of a lot of the records I was making, I couldn't get him all the time. So I'd have to go and do it with whoever was in the studio. I'd just put the mic wherever my ear was, and hopefully it would sound right. Nobody taught me that — it was just a matter of moving things around until they sounded right. You know, "What's this? A limiter? Well, let's put it across everything and see what happens." In those days, nobody was limiting the mix before it would go to mastering, or putting the whole drum kit through a stereo limiter. Everybody was saying, "This is technically incorrect," especially when it's a pair of API limiters that crunch the hell out of it! [*Laughs.*] So we were trying a lot of things like that, and I had to do it myself because there was no method for doing it. We were still having people that were very used to recording close-miked jazz sessions and stuff, which was really cool also, but it just wasn't what I was after, not with those bands. Later on, when I did Rodney Crowell and people like that, we did a mixture of both — very conventional close miking combined with a bunch of room sounds and having the drummer play in the bathroom and things like that.

> "THE SOUND OF A RECORD HAS MUCH MORE TO DO WITH WHAT THE MUSICIANS ARE PLAYING AND SINGING THAN IT DOES MIC PLACEMENT."

So you are definitely from the "leakage is good" school of thought?

Oh, yeah. That, and getting as much of the live performance as you can possibly get on the record. Even today, when you're hearing things sequenced, get as much live as you can get. I never really separated things a lot — not even to this day.

But, with the exception of the Ramones recordings, there is good separation. You clearly carved out spaces for the individual instruments within this broader brush of the big sound.

That's because there are a lot of overdubs, but the basic core of it is one big sound. We did a lot of the new Blondie record in the basement of Chris Stein's house, in his rehearsal room — a home environment, albeit with a lot of the equipment from my sophisticated studio. But, still, the basic feel of the album was the bass, drums, two guitars, and keyboards recorded live at Electric Lady for a couple of days. We then took that away and stripped things down and built on top of that, so you still have the live core of what the band was sounding like, even though there later is a bunch of separation from the overdubs.

Generally, you can get a very good picture of things — even within the mass — by panning instruments hard left and right. That comes from me listening to old Beatles

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records, where, by necessity, the whole band's on one side and then a guitar solo and a tambourine is coming out of the other side. [*Laughs.*] I thought that was really the way they were supposed to be! Because I grew up in America, I didn't know that the band and George Martin disowned all of that stuff.

Do you tend to use fairly standard drum miking in addition to the extensive room miking?

It's pretty much a close-miked kit plus a stereo overhead of one kind or another. Sometimes there's added emphasis on a cymbal or two if it's not being picked up, but it depends on the drummer. How you mic something and how you EQ it — in fact, everything that you do — is actually driven by the way the instrument is being played. If it's a guy that has a really heavy foot, you would use a different mic on the bass drum than the guy who has a very light foot. You would judge where the room mics and where your overheads are placed more by that than anything else. How the drummer's playing determines your mic setup, so there isn't a standard setup that works; no one thing works all the time. I'll even change mics within a session. [Blondie drummer] Clem [Burke] is pretty consistent in the style that he plays, but within different styles of music, we use different setups; sometimes the room mics and the overheads will be way overhead and sometimes they will be really close, depending on how hard he's playing the cymbals.

"THE MORE I'VE BEEN WORKING IN DIGITAL, THE MORE ANALOG'S SOUNDING OLDER AND OLDER TO ME."

Do you experiment with placing the drums in different areas of the room?

Yes, or even in different rooms. If you're going to throw a whole bunch of digital reverb on it later, it wouldn't really matter, but if you're trying to get an organic sound, the room means so much more than which specific mic you're using.

For example, we did some overdubs on that first Ramones album in a huge room — again, there's that connection between punk and classical music, because it was Arturo Toscanini's old rehearsal hall with the NBC symphony above Radio City Music Hall. So we were in this room that was about 65 by 100 and 30 feet tall, and we had room mics all over the place. We wanted the sound of an explosion on one song so I ran around and had people climbing up ladders and putting room mics all over the place while Tommy [Ramone] hit a tom-tom. For all that, it just didn't sound roomy enough. No matter what we did, it sounded like someone hitting a piece of paper — even if it was 50 feet away and cranked up and run through API limiters. So we just took a pair of grand pianos and put them around Tommy's kit and put bricks on their sustain pedals. Then we put up a pair of [Neumann U] 87's in a normal pattern over the drums, cranked them up, limited them to death, had him do one hit, and it sounds like cannons exploding.

That's not just an isolated incident, either; something like that seems to happen on

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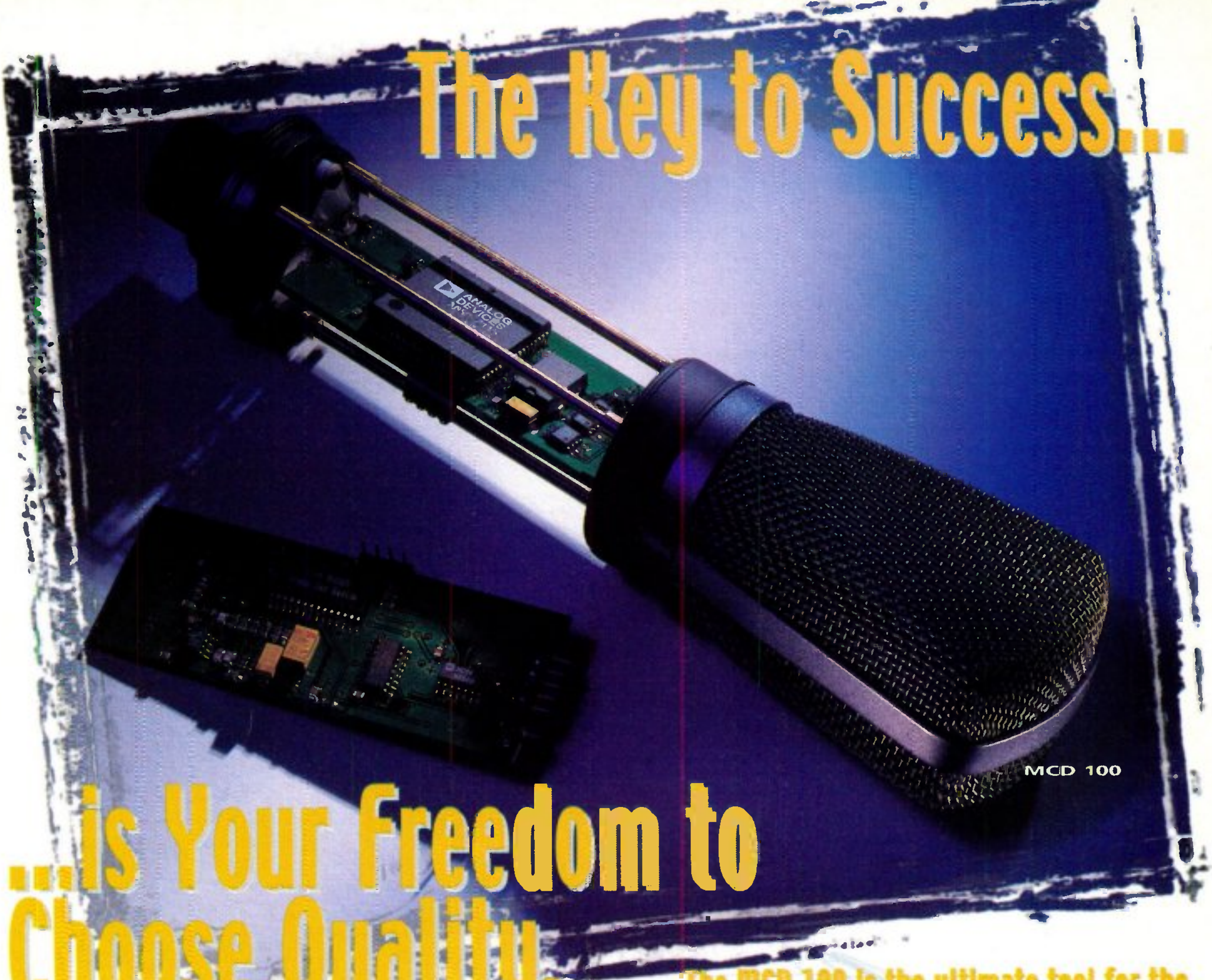

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almost every session. For example, on the latest Blondie album, Debbie [Harry] didn't want to have as resonant a sound as she usually did, so we got different pieces of foam to put up above her head and we'd lower them or raise them to get more or less resonance on her voice. There were even a couple of songs where she wanted to do something *really* deadpan, so she'd actually put the foam on top of her head and wear it like a little hat! [*Laughs.*] Maybe that sounds like 1920s recording techniques — you know, move farther away from the horn — but it works. We'd have people, within the course of a track, moving on and off mic on things. We'd move mics — do half of a guitar solo with close mics and then do another take with different miking, maybe putting the two mics out of phase — and then combine the two in the middle of the solo. As much as I want a live base, I do stuff like that a lot.

Every incident that I'm describing is not an over-the-top trick; it's something that worked because the guy really thought about the way the piece of music was meant to sound, about what the composer intended or what the artist wanted out of his guitar solo. You want to do what emphasizes that and not just do it for the hell of it. And when it's done right, you really get that bit of magic that makes the record something special. You want to use whatever devices you have at your disposal to any extreme in order to make the record sound right.

How will you decide to record a particular instrument on analog versus digital?

I'll use analog if I want to hear tape saturation, and that wouldn't necessarily be on the instruments you would think you'd want saturation on. I've done acoustic cello on a classical recording with tape saturation and heavy, heavy compression. Why not? It's almost like a little homage to George Martin.

So the only reason you'd go to analog would be for tape saturation?

Well, there's a theoretical "warmth" factor, which really isn't the right word. It's just that, because we grew up listening to analog, it sounds right to us. And the more I've been working in digital, the more analog's sounding older and older to me. That's more in my past than my present, but when the transition to digital was happening in the '80s, I'd always have in the back of my mind that it doesn't sound right until it's analog. Now, because we're ending up in a different medium — because we're not going to vinyl anymore — I don't know how valid that is. But there are certain things where I'll go for analog, maybe more from worshipping some sound that I heard in my youth than any other reason. Maybe I'm wrong, but I have a feeling that, as the years go by, I'm going to be less dependent on hearing that old "warm" sound.

That's kind of the tube versus solid state argument, also.

It's the same thing. When I started doing some of these classical things, I had access to the mic cabinet of both Decca records — which is unbelievable — and the mic cabinet of Abbey Road, which has everything, you name it. I would be able to set up 50 [Neumann] M 49's if I wanted to, but what ended up sounding right to me was a modified transformerless M 50. On the other hand, I'll use valve [tube] mics on certain instruments because they sound a bit more mid-rangy

> "HOW YOU MIC SOMETHING AND HOW YOU EQ IT – IN FACT, EVERYTHING THAT YOU DO – IS ACTUALLY DRIVEN BY THE WAY THE INSTRUMENT IS BEING PLAYED."

and brighter to me in an odd kind of way. You'd think the opposite — that they'd sound round and warm and velvety, but they don't. Nowadays, especially when you're recording to digital, the criteria for not using a valve mic is generally noise. When you're doing very, very low-level recording of something, they just won't work. But I tend to use valve mics more in pop stuff than in classical, because it's masked and it gives you a different kind of a punch — kind of a mid-range, lower octave punch. I also like to use valve mics as overheads on drums — a [AKG] C24 instead of a pair of 414's or something like that.

The flavor of tube equipment on tape is also different than tube to digital.

Which is why I'll do things simultaneously. On the Blondie album, there are a lot of things that I ran to analog and to digital simultaneously — bounce some bits over from the analog, edit some bits on the digital. There are times when I've actually used the analog drum take for the verses and the digital drum take for the choruses. It's all subtle stuff, but it builds up over the course of many instruments on a record. I wouldn't say that that's going to miraculously change anything, except in your own mind when you're doing it. But if you apply that philosophy to many different elements of a recording, then it can all of a sudden open up in the choruses as opposed to being really intimate in the verses.

Do you generally mix to both domains as well?

Not anymore, because we're in the CD medium and you might as well get it sounding good in that medium as soon as possible. I'll use outboard analog, but I usually mix to digital.

As an American producer living and working in England, do you think there's an "English sound" versus an "American sound"? If so, what accounts for it?

I do. There's a different approach and a different mentality. I think it's because the control rooms in England are drawn up more by the seat of their pants. You're working on speakers that are extraordinary, but they're not positioned using any kind of scientific method; they're placed according to what sounds good to the people that are working in the room, so it's done by ear. Whereas in America, I think it's done a little bit more technically by the book, and that creates a different sound.

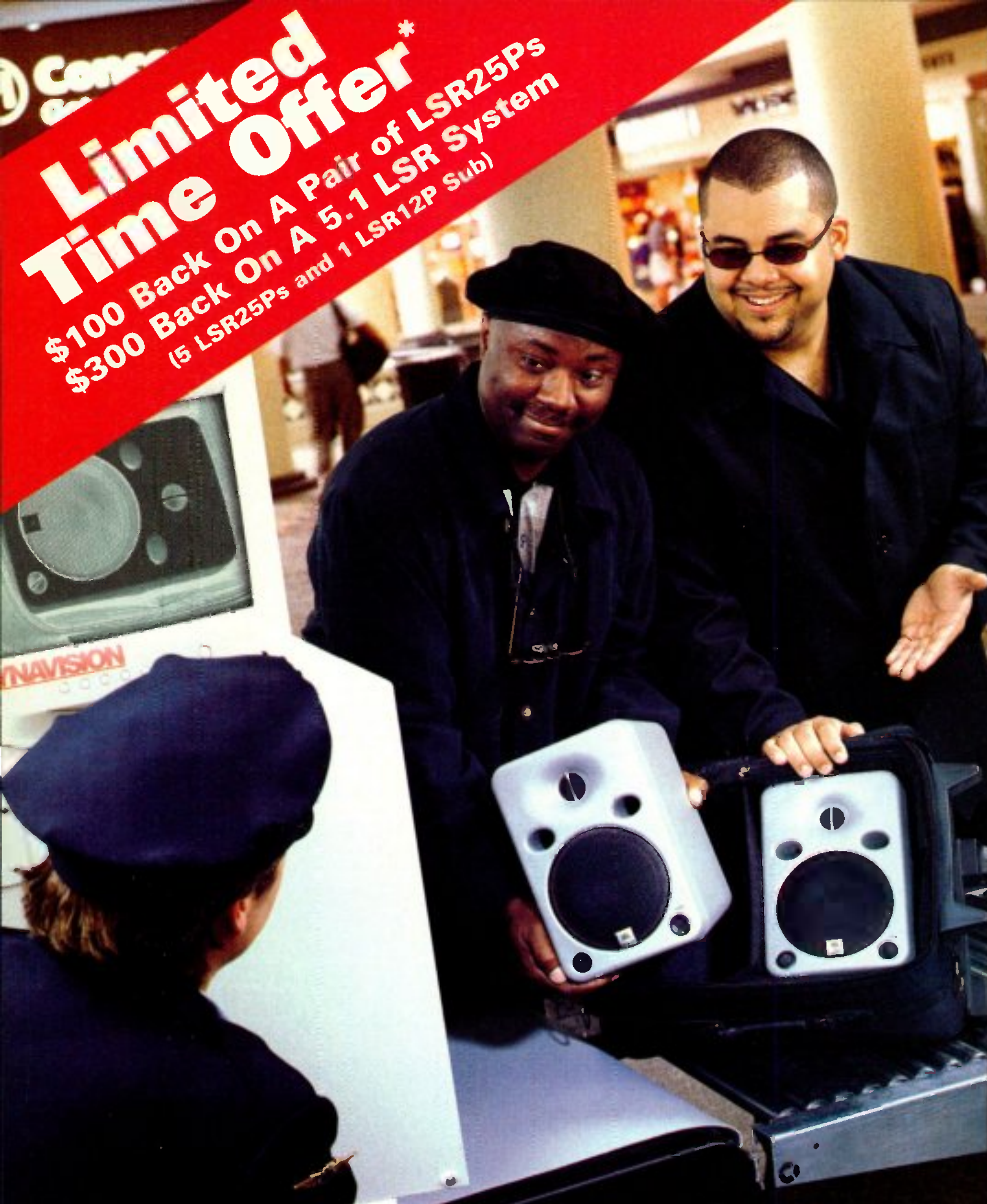
I find that I actually push myself a little more in English studios, because the environments are a bit unnatural. One of my favorite rooms to work in here was the old AIR studios. The control room had big floor-to-ceiling windows overlooking Oxford Street. That was how the room was designed, yet George Martin would sit there and get the most incredible sounding things out of there, and I got really good results as well.

All of which proves that the sound doesn't come from the equipment, but from the way you hear things in the room. The most important criteria, more than anything outboard, desk [mixer] choice or anything, is how you hear something coming through the speakers.

But will a number one British pop record and a number one American pop record have qualitatively different sounds?

These days, it's really odd because chances are both of them were recorded all over the place; the vocals were done on an ISDN line and they were mixed by some guy in Sweden. That's another thing that's becoming great: the ability to work on things in many places. As a result, the sound of records has become more universal now.

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



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Jonathan Peters builds a remix from just a vocal track

Starting From Scratch

by Steve La Cerra

Over the past few years, Jonathan Peters has been at the heart of some incredibly successful remixes by artists such as Whitney Houston and Deborah Cox ("Same Script Different Cast"), Janet Jackson ("Doesn't Really Matter"), Paula Cole ("I Believe In Love"), Sisqo ("Thong Song"), Jon Secada ("Stop"), Vanessa Williams ("First Thing On Your Mind"), Montell Jordan ("Get It On Tonight"), and Brian McKnight ("6, 8, 12"). Jonathan's busy remix schedule — plus his launch of Deeper Rekords approximately five years ago — prompted him to build a studio as a home base. That studio, Deeper Studios (New York), was designed by Fran Manzella, and includes two control rooms containing almost identical equipment, although the B room uses MOTU's 2408 mkII for audio I/O where the A room uses Digidesign 888/24 I/O's. All of these interfaces are connected via a Z-Systems Digital Detangler, allowing digital access to audio files from either control room.

Starting From Scratch

The foundation for a Jonathan Peters remix is the vocal track from the original recording. Jonathan typically receives only the original vocal track, but in some cases — such as Paula Cole's "I Believe In Love" — the record label delivers the entire song as a Pro Tools session on CD. Sometimes the label provides Peters with the vocal track on a DAT. And though he always requests the lead and background vocals separately (preferably dry), lately labels have been sending him "the whole vocal performance, including backgrounds and effects on a two-track — which can make the [time] stretch horrible (see below). If we get a two-track of the lead and background vocals we can usually start working on the arrangement while we wait for the individual tracks."

Jonathan has recently been working with programmer and sequencing wiz Tony Coluccio, but before any sequencing or recording takes place, Peters decides on a tempo for his new version of the song. The original vocal is then loaded into Pro Tools. While the vocal is playing, Peters and Coluccio manually run Digital Performer at the new tempo and watch the clock to see when the vocal falls out of time with the sequencer. Serato Pitch'n Time is then used in Pro Tools to stretch the vocal to the new tempo; when they get it close to the tempo that's running in Digital Performer, they put a click up and check the vocal against the click. According to Peters, "Hiromi Abe, one of our engineers here at Deeper, handles a lot of the time-stretching duties, and she's really good at it. Sometimes I'll walk in and she'll have a vocal in time with a beat, ready for us to start the arrangement." Hiromi also cuts the vocal up into samples, mapping them across the keyboard so that vocal phrases can easily be flown into the new version.

From Pro Tools, the stretched vocal is imported into Digital Performer. Peters rarely uses the songs' original rhythm tracks, preferring to create an entirely new arrangement. "If the song has some sort of a signature sound that helps identify the song — like the percussion sound in the Paula Cole remix — we might use it. But we usually recreate the entire rhythm section."

For working out the basic parts, Tony plays a Roland JV-2080 loaded with the Session expansion card. "There's a Concert Piano in there that's real good, so I'll use that or some kind of pad sound to start. I love the Nord Lead for the dance sounds, and we have the Studio Electronics SE-1, which is basically a rack-mount programmable Moog. I use that for bass or sometimes for a synth-y 'Emerson, Lake, and Palmer'-type of a Moog sound. I'll layer the SE-1 with a harder sound from another keyboard just to give it some punch in the mid-bass, not necessarily the bottom. I'm happy with the bottom on the SE-1, but sometimes the definition is missing, so I layer it with another tone, maybe from the '2080."

The MIDI parts are sequenced in Digital Performer running on an Apple G3/300 at the programming station in control room A at Deeper. In addition to the aforementioned sound modules, Jonathan and Tony also use Emu Vintage Keys, Planet Phat, and Proteus 2000; Waldorf Microwave 2XT and Virus B; Novation Super Bass Station and Drum Station; and Korg Trinity Rack modules, as well as a Roland XP-80 keyboard for sounds. The "dual desk" design

WEBLINK
Jonathan Peters is the founder of Deeper Rekords, www.deeperrekords.com.

THE ACID TEST

One of the most unusual aspects of Jonathan Peters's Deeper Studios is the "lounge" outside Control Room A. Yeah, every studio has a lounge — but this one can make a Jekyll-and-Hyde transformation into a dance club! Jonathan says he's "constantly working on ways to make the studio more slamming. There's an all-EAW club system out here with four Avalon Series DC-4's, one Avalon Series DCT-1, and two SB250P powered subs. When we're rockin' in the studio on a project, I can come out here to play records. I have the studio patched into the DJ mixer [Rane MP2016] and I can play the mix from the control room in the 'club' to get a different perspective. You hear it in here with a completely different array: four speakers, a tweeter array, bass bottoms, and a bigger room…we turn the lights down and get the whole room going like a nightclub! This system is consistent. If we're vibing in the control room and really think we're the sh*t, we come out here and compare our mix to other records. It's a great reality check, and it really, *really* takes our mixes up to the next level."

of the control room (see photo) allows Peters to work in Digital Performer while chief engineer Ryan West works at the mixing desk in the same room. The main console in both rooms is a Digidesign ProControl; the ProControl in the A room comprises four ProControl fader packs for a 32-fader work surface. Pro Tools runs on another G3/300 connected to a 13-slot expansion chassis, which houses a plethora of PCI cards. Storage is provided by Glyph hot-swap, dual-rackmount 9 GB Cheetah drives.

Setting up the control room in this manner accomplishes two things for Peters. First, it allows West to "premix" the synths while Peters works out the arrangement, thus saving time.

The second thing Deeper's control room setup provides is easy access to Pro Tools from the programming station. At the programming station there's a second keyboard and mouse dedicated to the Pro Tools system — allowing the sequencing work to take place without the need for Peters or Coluccio to run back and forth between desks. At the ProControl desk, a Gefen Systems KVM switcher allows a single keyboard and mouse to toggle between the Pro Tools system and the Digital Performer system, so Digital Performer can be run from the front of the room.

There are times when audio is recorded directly into Digital Performer, such as when Peters needs to record a percussionist before building the arrangement. "In that case," Jonathan notes, "the percussion will eventually be transferred to Pro Tools, and that's where our Z-Systems Z16 Digital Detangler comes in: We can print the percussion from Digital Performer to Pro Tools through the Detangler, so it's a digital transfer."

When the MIDI sequence is finished, the sound modules and stretched vocals are recorded into Pro Tools. Peters then proceeds with creating an automated mix via ProControl. Perhaps just as important as the automation is the ability for complete recall. "If an A&R guy wants us to make a change to a mix," maintains Peters, "all we have to do is open the Pro Tools session and *bam* — there it is. It's accurate, we don't have to sit there tweaking, and it comes up fast." ■

Using the Alesis DM Pro drum module as a studio drum replacement tool

Drum Replacement

By Peter Chaikin

If there's one sonic element that most contemporary recordings share, it's powerful drums. Whether it's dance, reggae, hard rock, or country, a slamming drum sound is paramount to a strong mix. But what do you do with a song that has (let's be polite here...) less-than-ideal drum sounds? Replace them! No, not the drummer or their performance — just the *sounds*. Drum replacement can be a very powerful technique in the modern recording studio arsenal. It wasn't so long ago that this technique was limited to creative producers and engineers who had racks full of triggering and sampling devices. But if you had, for example, an Alesis DM Pro percussion module (okay, there's the big plug-ola), you could perform drum replacement just as easily in a project studio. This guide explains how you can use drum replacement to improve the sonic quality and versatility of your productions.

What is Drum Replacement?

Drum replacement involves taking pre-recorded percussion sounds and replacing them or blending them with different, sampled sounds for a beefy, combined tone. Typically, audio from a recorded drum track is fed into a trigger-equipped sampling device. The sampling device responds to the incoming audio signal by instantly playing a sound from its built-in selection of drum samples. The sampler's audio output is then fed into a mixing console or recorder. When done properly, these new drums sounds will be neatly synced to the original recording, and can be mixed with all the other instruments in the song. Sometimes only a single drum is replaced — most often a kick or snare drum. In other cases an entirely new drum kit can be substituted for the original recording (but don't tell the drummer).

Why It's So Cool

Using drum replacement techniques, you can choose exactly the drum sounds you want at any time in the production process. You can experiment with a variety of sounds and listen to them in context of fully arranged mixes. You can even create variations of a song, with each version containing a unique drum kit — and then simply try different drum sounds during mixdown. Your flexibility is limited only by the percussion module you choose to use. Our example — the DM Pro — contains over 1,600 sounds. Because the DM Pro (as well as Alesis' older D4 percussion module) can translate trigger information to MIDI, you can use the MIDI out of the Alesis module to access any MIDI-compatible sound module or sampler.

Drum replacement can also help fix problematic tracks. Many technical problems with mic placement, phase cancellation, tuning/ringing, and ambience from the original recording can be resolved by getting rid of the problematic sounds, while keeping the actual performance. You may even find you can update and strengthen some great songs hidden away in the vaults, even if they were recorded many years earlier.

Drum Replacement Basics

Drum replacement is pretty straightforward. Once you've learned the procedure, it's easy to expand by using the technique on a number of simultaneous tracks. An important consideration: make sure your trigger/sound module has plenty of capacity for inputting, processing, and outputting multiple drum tracks. Though we're referring to the DM Pro, we've also had success with other drum modules. You can use any percussion module that includes trigger inputs.

Step 1 — Select the recorded drum track(s) you want to change. You can access the drum's audio signal from your mixing console using a bus output, aux send, channel direct out, or directly from the output of the recorder.

Step 2 — Send audio to the sound module's trigger inputs. Each audio track needs to be connected to its own trigger input. Do this by patching the bus out or aux out of the console to a trigger input. If you're replacing more than one drum simultaneously, make sure that only one drum is routed to each trigger input. For example, put the original kick drum on bus 1, and then

Using a percussion module to perform drum replacement techniques, you can choose exactly the drum sounds you want at any time in the production process and experiment with a variety of sounds.

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BitHeadz Inc. 4400 Capitola Road, Suite 202 Capitola, CA 95010 USA (888) 870-0070 www.bitheadz.com patch bus 1 out to trigger input 1. Put the snare drum on bus 2 and patch bus 2 out to trigger input 2.

Step 3 — Route the sound module's output back to your mixing board. Patch each of the sound module's audio outputs to inputs on your mixing board. On the DM Pro there are six audio outputs, so you can simultaneously record at least six sounds to six tracks (more if you don't mind grouping the toms to a stereo pair of outputs). At this point you can add effects from the module or at your mixing console. If you need to replace more sounds than the number of outputs, you can always record the first set of replacements to tape and make another pass for the remaining sounds.

Step 4 — Adjust audio input signal and trigger settings for optimal response. This is the key step for successful drum replacement, involving two stages. First, evaluate your audio input — it may need to be cleaned up a bit. The goal is a relatively tight, noise-free signal as a triggering "spike" for the sound module. A compressor can help you clarify attack and decay. Noise gating can dramatically control crosstalk and bleed. The DM Pro includes built-in gates for this purpose, but, if you're using a module without gates, you can patch one in between the output of the tape track and the trigger input of the module.

Second, adjust the trigger settings on your sound module, if needed. With a little tweaking and experimentation, you'll get crisp response, good dynamics, and no false triggering. The DM Pro's default trigger settings (gain, velocity curve, crosstalk, threshold, and retrigger) provide a good starting point, and work well with a wide variety of audio inputs. When these elements are properly fine-tuned, you'll get well-defined, responsive sounds from your drum module. The most important of these parameters is the gain adjustment: set it too high and multiple drums may trigger the replacement; too low and you'll miss the soft notes.

[*Tip: When using certain drum modules you might notice a change in the feel of the drum tracks due to a miniscule delay between the time the original drum hits and the triggered drum hits. You can adjust this by advancing or delaying the tracks slightly from a digital recorder. If you're using an analog recorder, you may need to experiment with triggering from the sync head while playing the rest of the tracks from the repro head.*]

Step 5 — Choose your sounds. This is the fun part, and the more samples you have at your fingertips, the more fun it is. You can replace the original drum sound entirely, or blend it with other sounds for distinctive stacked percussion tones. If you're layering, listen carefully for "flamming" of the two sounds. Each sampled sound has a unique "start time" relative to the trigger input; some will sit in the song better than others. Experiment as much as you like — you're free to shape your rhythm sound and try out new ideas. While the song is playing, try variations of the sounds (*e.g.*, different kick drums), and keep in mind that you're not locked in to one drum sound for the whole song.

> "MANY TECHNICAL PROBLEMS WITH MIC PLACEMENT, PHASE CANCELLATION, TUNING/RINGING, AND AMBIENCE FROM THE ORIGINAL RECORDING CAN BE RESOLVED BY GETTING RID OF THE PROBLEMATIC SOUNDS, WHILE KEEPING THE ACTUAL PERFORMANCE."

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Peter Chaikin is the director of product marketing at Alesis. (www.alesis.com).


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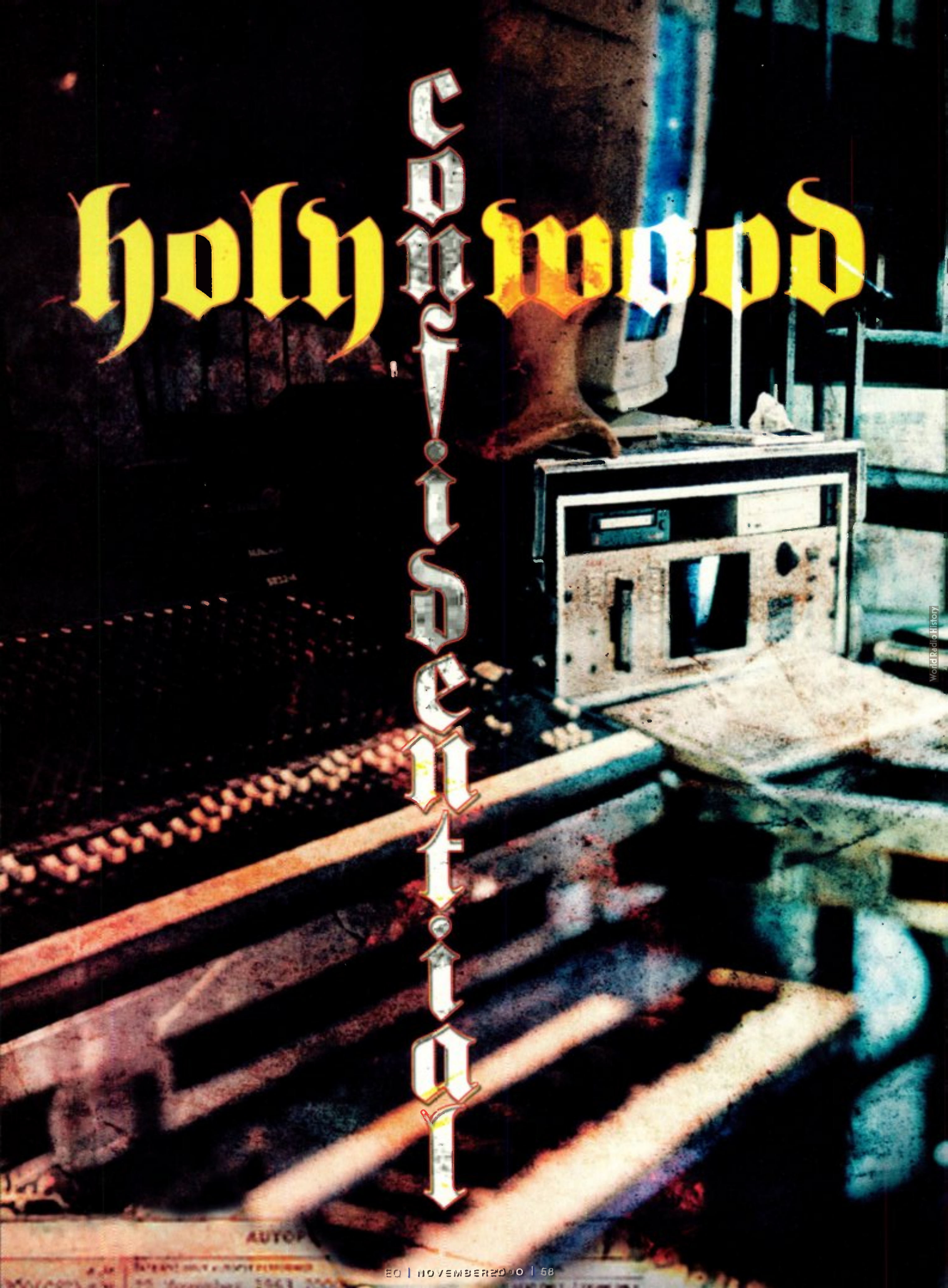
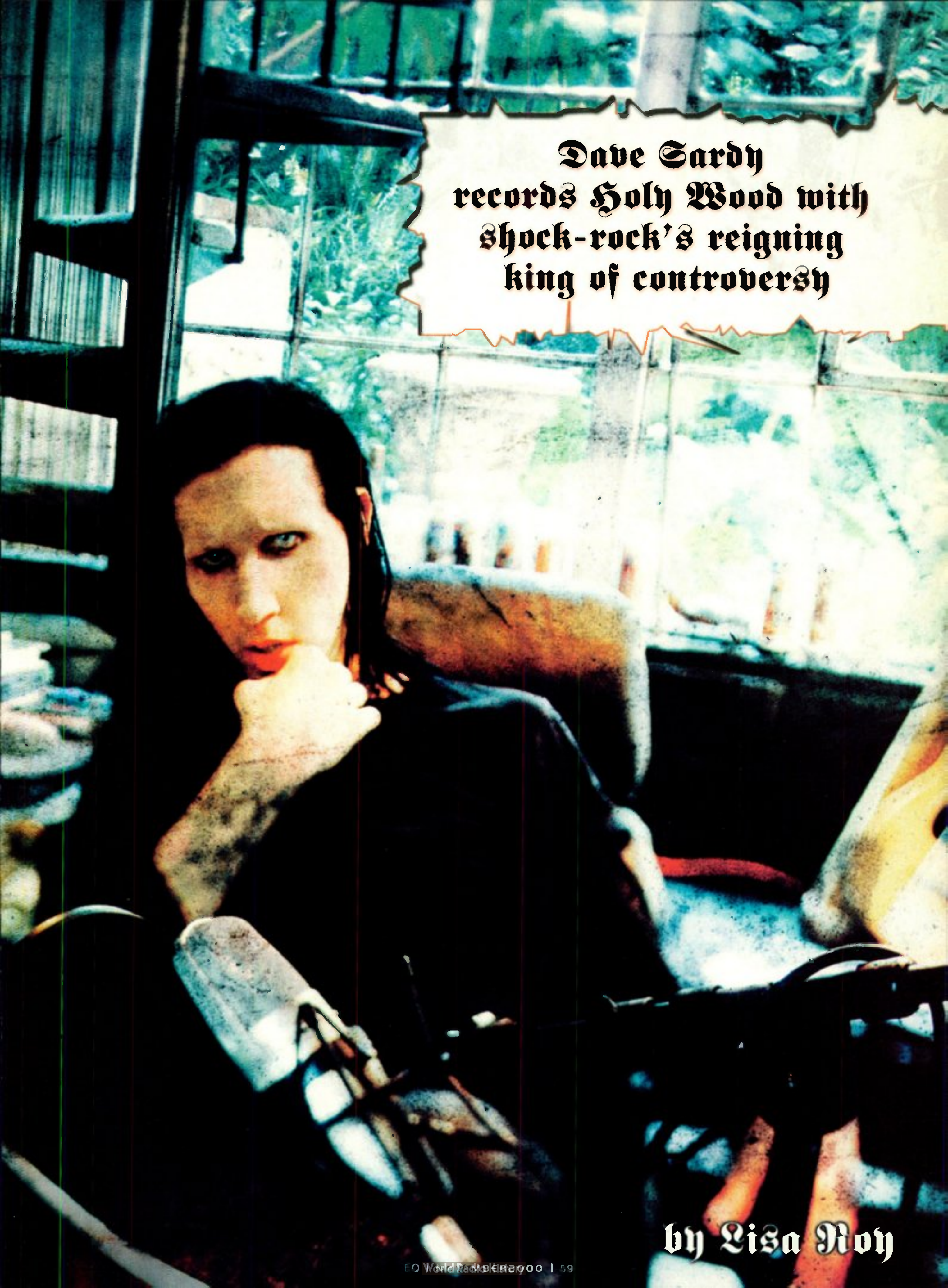




Dave Sardy records Holy Wood with shock-rock's reigning king of controversy

by Lisa Roy

holy wood confidential

On November 14th, Marilyn Manson and the death metal band bearing his name release their third studio album to an anxiously awaiting public. Let's face it: love him or hate him, this guy — a man who *thrives* on controversy — is anything but normal. Therefore, it should surprise no one that the recording process was less than traditional. For a look inside the making of Manson's newest offering, *Holy Wood, EQ* talked with the man behind the production of this metal masterpiece, Dave Sardy, as well as with Manson himself (see sidebar on page 63).

"Making this record was like any kid's wet dream," is the way Sardy describes a nine-month-long recording experience that involved more than 20 drum kits, 50 guitars, 300 pedals, 15 classic amplifiers, and a variety of speaker cabinets. At the core of this project was a very connected producer/artist relationship that, in Sardy's opinion, was instrumental in the success of the finished project. "I think, sonically, Manson and I are coming from a similar place and have a similar history, so it was really easy, real obvious. That's why they kept asking me to come back and work on more stuff with them."

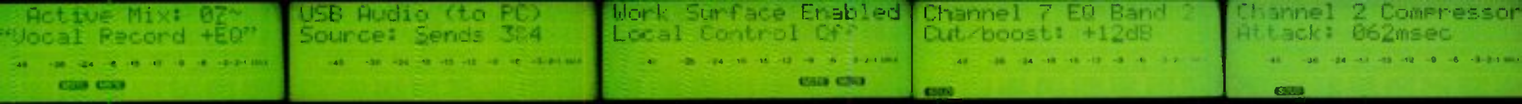
Sardy's relationship with Marilyn Manson began in 1998, when he was asked to mix "Highway to Hell" for the movie *Detroit Rock City*. Manson and the band were drawn to Sardy based on work he had done in the past, including production for Interscope act Cop Shoot Cop, Slayer, and his own band, Barkmarket. As is often the case when artists are courting producers, Sardy found himself getting to know Manson and, more importantly, understanding the very defined vision the artist has for his music. After mixing a song called "Astonishing Panorama of End Times" for MTV's *Celebrity Deathmatch,* Sardy checked into L.A.'s Village Recorder, where he was given the opportunity to mix a live album for the band. "They've got a phenomenal old [Neve] 8038 with flying faders at the Village. That's just a great sounding mix room. That's my preference: old consoles with new equipment," he confides.

One piece of "new equipment" that Sardy and Manson relied heavily on was [Digidesign] Pro Tools|24 MixPlus, especially when it came time to go into preproduction for *Holy Wood*. "There is no reason not to use Pro Tools," he states. "It puts 64 tracks at your disposal, as opposed to having to have a gigantic studio. It's pretty phenomenal, especially for doing demos." Sardy points out that the band has a Pro Tools rig that they take on tour with them. "Those guys are constantly working; there's not a moment that they're not writing and recording and archiving. They already had 20 or 30 songs together by the time I got involved, and then we did another 10 or 15 that they wrote after that point. It all got narrowed down to about 20 songs, which we recorded. So it's a pretty big process."

The process began over a year ago in Marilyn Manson's home studio, which features a Mackie 32x8 console, the aforementioned Pro Tools rig, a TC Electronic Finalizer, and a slew of compressors and EQs, along with a large number of samplers, a huge catalog of drum samples, and Manson's favorite Shure KSM32 mic.

Upon completion of preproduction, Sardy, the band, and a pack of dedicated assistants and engineers transformed the Houdini Mansion in Laurel Canyon into a massive recording complex. "The Houdini Mansion, which has more than 30 rooms, is set on top of a mountain overlooking an orchard," Sardy recalls. "It was pretty spooky up there — there are no other houses around." No doubt that was a good thing, as the shenanigans that were to follow would stir even the wildest of souls. They included "setting lots of things on fire and recording them," recording outside in the rain during a lightning storm, throwing the guitars in the swimming pool for a week, and throwing bass guitars down the mountain to see how it affected the sound. "Let's just say I walk with a limp now," Sardy jokes. "It was so insane."

The mansion's 2,000-square-foot ballroom, with its 14-foot ceilings and hardwood floors, was designated as the main control room, with Sardy's favorite console — a Neve 8038 — as the centerpiece. In the small wooden library connected to the ballroom there were two small marble rooms that were also converted into control rooms, one for programming engineer Ron Harris, who Sardy praises for his work on *Holy Wood*. "We hung deflection [material] all over the place, but it was definitely commando style. After we deadened the place down with carpets, we found this gigantic American flag and used it to cover the whole ceiling, which really dampened down the sound. We wound up with a



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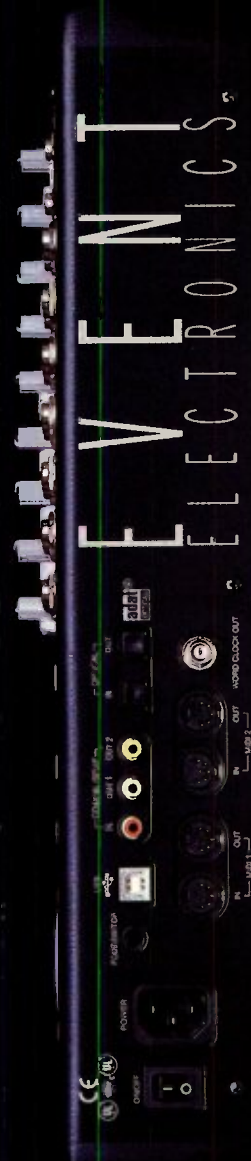

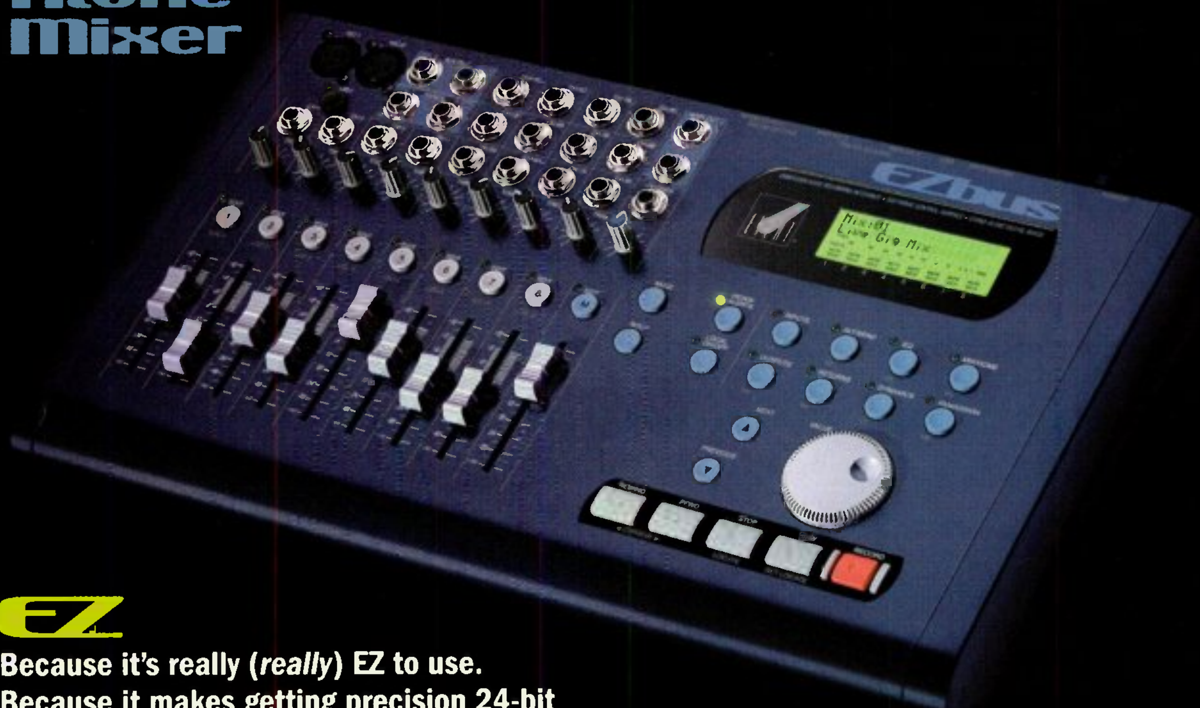

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huge space that was kind of dead, so you actually had all this really great low end floating around the room. The drum sounds we got were pretty amazing."

Most of the mansion's massive space was utilized during recording. In addition to using the hallways for demo rooms, Sardy recalls that Manson's keyboard player, Pogo, took the upstairs and filled it completely with keyboards, MIDI gear, and yet another Pro Tools system. One of the many bizarre "rituals" that accompanied the recording process took place in this portion of the Houdini House, which was usually "off limits" to anyone else. "Pogo had a room set up completely with weird rare synthesizers and old programming equipment in which he would run these insane experiments. He'd set sequencers in motion and leave them running for weeks until they start to just disintegrate, information starts to fall apart, so it was like every single room had these weird experiments going on between synthesizers just running by themselves. Plus, there was this rule that nobody was allowed to touch anything, because sometimes we'd wait for weeks on a sound and see what happened to it over the course of time, then we'd use it."

Pogo certainly didn't corner the market on eccentricity in recording. That honor should probably go to Manson himself, who co-produced the album. Astute listeners can find examples of his unique flavor on two songs, "The Fall Of Adam" and "Count To Six And Die." For these tracks, the guitar sounds were achieved in a most unorthodox way. Sardy explains, "The song breaks down and there's a protracted acoustic guitar moment. There was a gigantic thunderstorm and [guitarist] John 5 was sitting outside, in the rain, with lightning hitting all around him, playing an acoustic guitar miked with two [Neumann] U 67's covered in plastic bags. It was incredible! Okay, so we destroyed a couple of microphones, but the sound of it was amazing!"

Not everything on the

▶ continued on page 64

Manson on Manson

EQ: Tell us about your home studio.

Marilyn Manson: It's called Holy Studios. I write on [Emagic] Logic and record with [Digidesign] Pro Tools|24 MixPlus. I have a small Mackie board (32x8) and a lot of outboard gear, but I mostly use my studio for writing, and if anything became more complicated — like the drums and any type of tracking — that would require more space. That's why we moved into the Houdini Mansion, but I do a lot of vocals at my house. It's kind of like a small, intimate setup but with enough equipment so that the quality is there. I moved in a year ago, and I set it up right when I moved in.

I understand there's an interesting musical history in this house.

The house has a good history because the Stones lived there when they wrote *Let It Bleed*. If you watch [the movie] *Cocksucker Blues*, there's a moment at the beginning when they're in this house and they're writing. They're actually working on "Gimme Shelter," and that's my living room.

How did you get Dave Sardy to capture your vision on tape?

I think it was just a matter of having him understand it. On our last studio record, my intention was to make the seven songs that were supposed to be sarcastic jabs at pop music sound a lot different than the seven songs that were supposed to be the internal emotions being felt. But I don't think that I really got that across to [producer] Michael Bienhorn. The songwriting is so drastic that I think the point is still made, but sonically it could have been done a little bit differently, and the same with *Anti-Christ Superstar*. But in both cases there was somebody else who was kind of leading the production on that side more than I was. When we were writing it, we captured a lot of the intention of how it was supposed to sound. Sardy's job was to make it that much better. I think we have a lot in common, so it wasn't hard for him to understand what I wanted.

How did you come up with some of the recording techniques used on your vocals?

Well, there's a bit of ritualistic element to everything I do. Some of it's just to have some sort of ritual in your life. When it came to singing, we would just create different things that would either put us in the mindset or just take us out of the regular world. The more you become into the world of what you're doing, the more you take yourself out of the normal world and you're not thinking about what everyone else is doing. You're only thinking about what you're doing; you become very focused.

Sardy had a good understanding for how I like to work. It took awhile to figure that out because I'm really used to working alone as much as I can. I often have people start recording something and then leave me alone until I finish it, and then come back. Part of it might be because I'm experimenting or because I don't feel comfortable around other people, but I felt really comfortable in front of him — even considering the fact that he's a singer, which could be intimidating in a way. But it worked, and he was good at pushing me to do better. I would do something I thought might be as far as I could go, and he would make me do it again and again and again until it became really strong.

Tell us about the electrical storm, when you were recording guitars outdoors.

That's the track, "The Fall Of Adam." That was very interesting because everything on the record is very symptomatic of how I'm relating it in my head to parts of the story that I wrote. I wanted it to represent that musically, either with dynamics, or just with how things are recorded and presented. That song represents a part in the story where, if it were a film, it would have been the first time in the movie when it rains. In cinema, whenever it rains, that signifies a change.

Ironically, when we began recording it, it wasn't raining. There's a line in there about cars arriving for this event, so I suggested that we record outside so we could hear the sound of the cars going by in the street. I would have loved to have incorporated rain and things like that into it, but at the time I wasn't really thinking about that. I was thinking about the sound of cars, and the minute we started recording outside, this huge thunderstorm just erupted and we captured it on tape. It was just something that was magical and I admit it was kind of strange, meant to be or something.

What would you say was the strangest recording technique you used on this album?

I'm sure the people that rented us the mics don't appreciate that technique [U 67's in a rainstorm]! You know, there were so many different things. I don't like to do vocals normally. Sometimes I'll lay down or sometimes I'll put myself in a corner somewhere where I can't see anything, just whatever it takes, whatever rituals I need to do. There were also times that we made little expeditions out to the desert. We recorded different things, but not a lot were used on the album because we could only use a portable DAT recorder. Sometimes when you do things without all of the studio benefits and without the technology, you're forced to be more creative. That was probably the best thing. There's a song [on the album] called "Burning Flag," and nearly all the percussion sounds are made up of us hitting each other — sounds most people would associate with industrial music of the early '80s. I would attribute it back to the stuff that Lennon would do when he would just be real experimental. A lot of times when you're making a record, you're being real safe and you do what you know works, but we really did things differently on this one.

You've worked with some great producers. What makes you a great producer?

I think that there's something that people take for granted when an artist produces themselves, but what I did on this record is what I always do. I was just more in control than ever. It remains to be seen what I would do with someone else, producing someone else's music. I think that's a better test as far as being a producer goes. What I'm trying to say is, I don't think I did anything that I wouldn't expect any other artist to do when they're making their own album. In a sense it's producing, I guess, and it comes natural to me when I'm doing my own thing. But I suppose it's a matter of more than your technical knowledge; it's knowing how to accomplish what the artist wants. For me, that was picturing in my head how I wanted the record to sound and actually managing to accomplish that. That was the greatest goal. Because a lot of times things would change — sometimes for the better, sometimes for the worse — but in this case it only became more perfect as we went along, as far as what I thought perfect was. ■

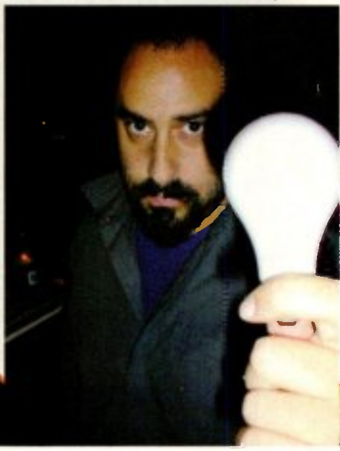
holy wood confidential

▶ continued from page 62

Manson recording session was done in a strange manner. Like many producers, Sardy relied on trial and error when selecting the mic to record Manson's signature vocals on. "His voice is just so unusual. He's got unbelievable low end, and lots of midrange and high end. I've never recorded a voice like his. Initially there was a lot of testing to find which mic sounded the best with his voice, and a lot of those mics that you would assume would sound phenomenal didn't."

The verdict of the mic shootout was an Elam UM251 run through an LA2A limiter, but there was one "incident" while recording vocals that involved Sardy's own (now deceased) U 47. "At one point we balanced the U 47 on these huge daggers that Manson had bought. The daggers were on the floor and basically he's singing over them because he goes through all of these movements while he's singing. We had to poke little tiny holes into the grills of the mic; I'm not even really sure why we were doing it. There was a lot of rigmarole that we had to do for this particular vocal. It actually sounds really bizarre because you get these weird metal reflections, and it's pretty subtle, but in a lot of stuff that we did, sometimes the most simple thing in the world sounded insane, and the most complicated thing in the world sounded simple."

Producer Dave Sardy



Sardy admits that much of the recording described above was the result of what he calls "happy accidents,"

▶ continued on page 139

AES2000 REPORT

THE 109TH AES CONVENTION SURROUNDS L.A. IN SOUND

Late September found audio professionals from around the world heading to the Los Angeles Convention Center for the 109th AES (Audio Engineering Society) show — a pilgrimage fraught with opportunity for networking, news gathering, new gear lusting, and socializing. A dual-purpose gathering, the AES show comprises a convention with numerous research paper presentations, discussion panels, and other educational opportunities, and also includes a tradeshow where pro audio manufacturers exhibit their newest wares to engineers and studio owners.

The theme of this year's show, "Surrounded By Sound," put the spotlight on 5.1 surround sound, but there were a multitude of new releases shown in virtually all product categories. As is always the case with a tradeshow as large as AES, it's impossible for us to cover every single new product in a report like this; we've kept our focus on new offerings that we feel will be the most significant to the professional project studio owner/operator. Look for more AES new gear coverage over the next few months in our regular Product Views department. A final caveat: the information you are about to read is subject to change; pin down the manufacturer or your favorite retailer for the final word on specs and prices. *—Mitch Gallagher*

MICROPHONES

As we expected, there were many new microphones (and even some old ones!) introduced at this year's AES Convention. AKG introduced a transformerless, large-diaphragm condenser mic intended for broadcast purposes: the **C4500B-BC**. With its internal, multi-layer pop filter, we suspect that the C4500B-BC will sneak its way into many a kick drum.

Audio-Technica showed the **AT815ST** and **AT835ST** stereo shotgun mics. Both feature mid-side pickup patterns with an output that may be switched from internally matrixed MS to non-matrixed signals. Announced at the Audix booth was the **ADX-90**, a miniature condenser mic with a low profile for easy placement on drums and percussion. The ADX-90 is also appropriate for use on brass instruments. Beyerdynamic introduced an entire line of new transducers called the **Opus** series that included new dynamic, condenser, and wireless models. The CAD booth was brimming with debuts, highlights of which include the **M177** cardioid condenser as well as the **M179** omni condenser. Both models feature a single-pattern version of the Equitek E-300 capsule and a low-noise, high-speed head amp. At the Earthworks display we eyeballed the **M30BX** measurement microphone. Powered for 1,500 hours on a single AA cell, the M30BX delivers enough output to drive a single-ended input such as a computer sound card.

New microphones from Neumann always spark a lot of interest; they introduced the **M150,** an omni, tube condenser drawing its influences from the company's classic M50, yet with a much lower self-noise spec (15 dBA). Two retro-Telefunken ELAM 251's appeared at the show, though they had strikingly different appearances: Lawson's **L251** sports a beautiful gold and black finish, while the **ELUX 251** from Soundelux looks very much like the original. Royer's line of ribbon mics grows to three with the debut of the **SF-1**, an ultra-compact, monaural ribbon microphone with a 1.8-micron (!) transducer.

Sennheiser added an electret condenser mic to their popular Evolution series with the **e865**, a handheld cardioid mic capable of handling a maximum SPL of 150 dB. Also on display at Sennheiser was the **MKH800** studio microphone, with a self-noise of only 10 dB (A-weighted) and a frequency response extending out to 50 kHz. Joining Shure's line of microphones is the **KSM44**. Running on a transformerless, Class-A preamp circuit, the multi-pattern KSM44 uses a pair of large diaphragms, and is optimized for vocal recording. *—Steve La Cerra*

MIC PREAMPS

You can never have enough mic preamps for all those wonderful microphones, and a variety were on display. Aardvark kicked off their Direct Pro Series with the **Q32**, an eight-channel pre with phantom power, ADAT optical and TDIF I/O, and 24-bit converters. Its little brother — **the Direct Pro 24/96** — features four discrete mic inputs with phantom power and (as the name implies) 24-bit, 96 kHz audio.

Making a big splash was the GT Electronics **VIPRE** (Variable Impedance Preamplifier). This single-channel, all-tube unit features four different input impedances as well as a variable rise time characteristic. Taking a different approach to microphone amplification was the Avalon **AD2022**, a dual-channel, Class A discrete design. The AD2022 affirms the importance of variable input impedance while providing

Studio OBSERVER

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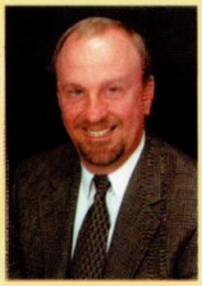
Volume 5 · Number 2 · November/December 2000

BARRY GOLDBERG

ENGINEERING IN THE LAND OF SUNSHINE

Engineer Barry Goldberg has been on a knowledge quest from the beginning. This LA born and bred engineer/producer never attended any of the nations heralded audio engineering programs, he learned the hard way, starting at the bottom of the career ladder. "I started as a runner at The Village Recorder knowing that I loved music and just wanted to be around it. I knew I wanted to be an engineer before I even knew what that meant or what was involved. I enjoyed the technical aspects and working with the artist on the creative side," he recalls. Goldberg, who has had a passion for music most of his life, finds that his rich musical background has come in handy throughout his career in the studio. "I grew up playing different instruments so reading and theoretically knowing music helps me with the communication factor in the studio." Goldberg's first acclaimed studio record was in 1992 with Snoop Doggy Dogg (Doggystyle), a record that he did at The Village Recorder and Larrabee. Through the years he has worked at many of LA's hot spots for recording but now considers Conway and Sony his studio homes on the West Coast. "I love Conway. My relationship with them started when I was an assistant engineer. Then I would turn around and engineer a record there. If I had time between gigs I always went back to seconding there. Its one of the best ways to observe how to and how not to make a record" he adds, "besides it's the most beautiful studio around." While Goldberg has been evolving as an engineer, his discography has been evolving into a who's who list of some of the finest names in pop and rock history. In recent years Goldberg's talent can be found on projects by Fleetwood Mac, Smashing Pumpkins, Crosby, Stills, Nash & Young, and Hole." My career has allowed me to work with some of the

"I prefer analog but it is also a natural and common progression to move right into the digital domain for overdubbing or editing purposes,"



FROM THE TOP

CORPORATE NAME TRANSITION

We have embarked upon a brand transitioning period to change over to **EMTEC** as our new brand name for all of our pro media products. The legalities of this matter are simple, we bought the use of the name "**BASF**" for five years and that time is quickly coming to an end.

This transition period has been underway for months now and I am proud to report that the industry as a whole is very receptive to the change and consumers are beginning to connect **EMTEC** to the **BASF** line of products.

Throughout this transition we want to continue to emphasize that it is essential that our customers understand that ***the only thing changing is the name.*** The brand name change to EMTEC in no way reflects any changes in the factories that manufacture the EMTEC products or the people that bring them to you. The quality that you have found for years in BASF brand tape and the service you receive from our staff will remain consistent and what you expect.

Remember, change is good and at **EMTEC** we welcome it.

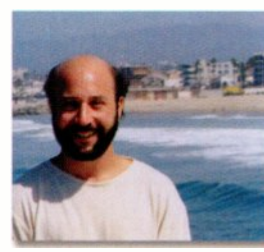
Joe Ryan, President • EMTEC Pro Media, Inc.

greatest record makers, covering the full spectrum of music and personalities. This has enabled me to feel limitless in my creativity," says the engineer who recently finished a project with Japanese New Age artist, Kitaro. "It has been a great

"Getting it on tape properly first creates a freedom with the track and clears direction for the future production."

experience to help such quality people create their art. "Some might call it art...some may not, but regardless, Goldberg was the man behind the board for Marilyn Manson's record, Mechanical Animals. He confides that Manson is a consummate professional in the studio. "Now, I'm not going to say he is a normal Joe, but he is a true artist with a definite idea of what he wants. "For that record Goldberg used BASF SM 900 2" analog and BASF SM 931 1/2" digital tape."I prefer analog but it is also a natural and common progression to move right into the digital domain for overdubbing or editing purposes," he explains. "I appreciate the feel and dimension created with the reel to reel recording, which can't be achieved with hard disc recording. However, I have learned to combine the two formats so that they compliment each other. But analog has the greatest sonic nuances." Goldberg credits his eclectic variety of projects with allowing him to see how truly versatile his favorite tape is. "Being that I work on projects from the completely distorted out type Manson records to the silky lush strings of Kitaro or organic sounds of Ricky Lee Jones record, BASF tape has shown me that the tape really excels in any situation." Like most engineers, Goldberg has a few tricks to his trade that let him consistently get good sounding tracks. For example, he confides that when using BASF tape he never uses the standard over bias. "Depending on the style of recording it will vary. To date, it has always been less over bias than recommended." Also, when it comes to micing techniques Goldberg is quick to admit that this is one area in which he takes a lot of pride and precision. "I use a variety of mics; and I like to experiment. One of the things I do is I actually put my ear where the mic is going to live, "he shares. "When I have a guitar rig set up, I put my ear plugs in and actually listen to each speaker and I find the spot that works best for the recording. The same principle applies for all miced elements." Micing is what Goldberg calls "a lost art." One of his concerns for the future of his profession is that the up-and-coming engineers are not being taught the importance of, nor do they have the opportunity to observe, proper mic placement in the studio. "For me, I'd rather move the mic around to capture the sound right rather than to just start throwing a bunch of processing on it. Don't get me wrong, I love a bunch of processing, but it has to start with a source. Getting it on tape properly the first time creates a freedom with the track and clears direction for the future production."

Goldberg always seems to go that extra step for his recordings and as a result he is in demand, not only as an engineer, but also as a producer. He is finishing up producing, engineering and mixing legendary Black Sabbath drummer Bill Ward's solo project this Fall. Through his production company, Bare Bones Productions (with his partners Kevin Keller of Empire 28 and composer/arranger Joan Barton) Goldberg has been doing artist development for the past few years. Although he scouts talent throughout the country Goldberg didn't have to go far to find the band that he is currently in the studio with–Superfine. "The singer/songwriter, guitar player, Rob Grad is my next door neighbor. He knew that I was a producer/engineer and I had no idea that he was even in a band. He let me hear his demos and away we went. We are putting the final touches on it right now and we are very pleased with the outcome. We'll see...There is a buzz building about them. "Don't look now Barry, but there is a buzz building about you too!"



IN SESSION WITH S. HUSKY HOSKULDS

It didn't take long for S. Husky Hoskulds, who moved to the US in 1991 from Iceland to warm up to the idea of an engineering career in Los Angeles. A brief education in engineering at UCLA resulted in Husky landing a job at One-On-One Studios where he worked as an entry level staff person. This led to even more work at a string of smaller studios in the "grunt work" days of his career. Husky eventually got a staff gig at Sunset Sound Factory but says that it was those early days of working in smaller studios where he learned some of his most valuable lessons as an engineer. Mainly, how to be on your guard while relying on yourself and how to keep a session going. Today this new daddy and budding engineer has his pick of studios and has worked on projects with Vanessa Paradis, Michael Penn, Sheryl Crow and the Wallflowers to name a few.

While first working at Sunset Sound Factory you met and began to work with Tchad Blake. What is it like working with him when he has his producer's hat on and its you instead of him in the engineer's chair?
It's great. The first time I "officially" engineered for him, it was very natural. We've spent so much time in the studio together that we're on the same wavelength. I found myself reaching for the same knobs he was about to. Y'know, things like checking the phase on this mic, too much high-end here, not enough bass there. It was a cool thing. I really owe him a lot for what I've learned from him.

What project in your career are you most proud of and why?
I'm really happy with the way Michael Penn's record (MP4) turned out. I recorded and mixed that one, except for the first song, and it has a sound to it that I'm really pleased with. Being able to follow the songs through to mastering is key. You kind of have a sound or approach in mind for each song, having put it up a number of times throughout the overdubbing stages. You know how things work together, so it's nice to get the opportunity to complete the thought. Also, I think the songs on Vanessa's record came out really well. Tchad is mixing that one now.

July will be a big month for you with the release of the Wallflowers and the Vanessa Paradis albums. Let's discuss the Wallflowers project specifically and your approach to the recording process for this record.
Well, I didn't really think too much about it before hand. I like it when things come to me spontaneously, as the songs are being tracked. I'll hear an approach - sonically - that I think might work for the song, and if I'm lucky, the producer(s) and band will agree.

What vocal chain did you use on Sheryl Crow and were there any memorable moments while working on that award winning record? For that one there were a couple of different ones. Most of it was the 251, while for maybe one song or two we used the C37. The mic pre in both cases was a Neve. She's great though. With a singer like that, you've got a lot of leeway. She sounds good on a lot of mics. To me at least. As far as memorable moments, there were absolutely none. Very boring. No no, of course I'm kidding!! It was a blast! Getting to work with that kind of talent is great. She'd have people come by to play, like Benmont Tench, who is always a pleasure to work with (played a lot on Vanessa's record too), and Dan McCarrol and others, so it was a lot of fun.

When approaching a project, do you usually go analog or digital? Do you find yourself switching formats?
I always go analog. 15-SR if I can. If not, I'll do 30 non Dolby. I also have a Pro Tools rig. It's called "My First Pro Tools." Actually it's the 001. I use it in tandem with the tape, for loops, sound design, fx and things like that - along with the Nord Modular and the Meta Synth. But no Auto-Tune or drum editing on that rig.

And your tape of choice is? Why?
I like the BASF SM 900. I just like the fact that it's consistent and very durable. I also like the fact that it can take some heat. If the drummer gets a bit excited, and starts to really smack the drums, I don't have to worry about it sounding shitty coming back.

Are there any studios that you call "home"?
I'm most comfortable at Sunset Sound Recorders or Sound Factory. Most of the rooms there are pretty great. Not too big, not too small. They just feel right.

Can you give us insight into your micing techniques?
A lot of ribbon mics. Not afraid of those guys... I also like to use cheap mic's to add some color. When you blend them in with the nice ones (on drums for example) and play with the phase, you can get some cool sounds. I built a box that varies the phase on a balanced signal, fully sweepable, so you can get signals that are anywhere between zero and 180 degrees out. I won't say it affects record sales, but I like the way it sounds. The only other thing to mention would be that I had the opportunity to try Monster Cable when I was doing the Wallflowers record and have used it ever since. Great stuff makes a big difference!

Can you give us a peek in to your future. What projects do you have coming up?
Now my main gig is being a father to my 8 month old son. And I might do a jazz record, all Louis Armstrong stuff. Something different. I think it'll be fun. That kind of music I think lends itself to my style of recording. Mono drums, mono piano, lo-fi everything. Also, Craig Street is producing, he makes pretty cool records. And Abe Laboreal jr., Greg Leisz and Smokey Hormel (among others) are playing. I can't really say no to that one.

TAPE TALK

by Jean Tardibuono

Vice President of Sales and Marketing
Studio and Broadcast Products

EMTEC Pro Media, Inc.

One of my favorite aspects of working in this industry are the great friendships and business relationships you develop along the way. As EMTEC continues to grow within the marketplace, and as our commitment to the audio market intensifies, we have also expanded our broad base of industry partners.

I am thrilled to report that we recently embarked on a great new relationship with Studer North America. Studer North America shares our passion for providing customers with the very best in products. This quarter we entered into a pack out arrangement with Studer. They will now be including BASF brand tape in all future shipments of Studer's popular tape machine line including, the Studer A807, Studer A827, Studer D827 Mk II MCH, and Studer V-Eight. We feel that the natural marriage of the best in multi-track tape, and the best in recorders will result in positive things for both companies and our valued customers.

During Summer NAMM, EMTEC was also able to partner with industry giants Solid State Logic, Lexicon and Audio Media Magazine for the grand opening of engineer Chuck Ainlay's new studio, BackStage. BASF tape will no doubt be the tape of choice at BackStage as Chuck is a long-time user of our products. We congratulate him on his new venture.

To all our industry partners, we look forward to growing our businesses together in the coming year. Thanks for letting EMTEC be a part of your successes.

a high-impedance instrument input and sealed silver relays for an uncompromised signal path.

Designed by David Blackmer, the **1024** from Earthworks is a four-channel unit boasting a distortion spec of under one part per million (0.0001%). High-frequency response of the 1024 extends to a stratospheric 100,000 Hz. Grace Design has entered the project market with the **Model 101**. This compact, chrome-faced beauty houses a single channel identical to the preamp used in their model 801. Great River Electronics announced their two-channel **MP2-NV**, a discrete design in the tradition of the Neve 1073 preamp.

Expanding their line of mic pres was PreSonus. They introduced the **Blue Tube** two-channel tube-based unit and **Digimax**, an eight-channel mic/instrument pre offering built-in 24-bit A/D conversion and "dual domain" limiting. Sonic Sense, Inc. and Benchmark Media Systems announced the release of **Sonic ADK2+** mic preamplifier, a compact unit featuring two channels, 12-volt DC powering and 24-bit, 96 kHz compatibility. Following on the heels of the Precision 8, TRUE Systems rolled out the **P2** digital. This two-channel mic pre includes analog and 24-bit, 96 kHz digital outs, plus multichannel optical output routing, mid-side stereo decoding, front-panel phase meter, and FET DI inputs. *—Steve La Cerra*

AUDIO-TECHNICA AT815ST AND AT835ST

CHANNEL STRIPS

Channel strips, which combine mic preamps with EQ, and other processing have become increasingly popular recently. dbx unveiled the **376** Tube Channel Strip, a combination mic pre, compressor, EQ, and de-esser in a one-space rack unit with 44.1/48/96 kHz and 16/20/24-bit digital output. The Oram "Al Schmitt" **Pro-channel** includes an optical compressor and step-switch six-band EQ, along with the ability to connect three different microphones and switch between them for A/B/C comparison. Announced by GML is the **2020** high-resolution discrete input channel, which combines a line/mic preamp with EQ and dynamics control.

If you missed out on the '70s, two different manufacturers are giving you the opportunity to recapture the famed Trident sound. Trident Audio Ltd. (keepers of the flame) unveiled their **S 80 Producer Box**, which is essentially two channels of a Series 80 console, including dual faders, transformer coupled mic preamps, and 3-band EQ. If you prefer the sound of the A Series, the JoeMeek **Trident-MTA** features classic inductive EQ and provides a set of three high-pass and three low-pass filters — all discrete Class A amplifier second order filters, said to be exactly the same as on the original. *—Howard Massey*

RECORDERS

DVD-Audio has missed yet another Christmas selling season, and Sony's big push of DSD (Direct Stream Digital) at the show proves that they're more than willing to step into the breach. In fact, SACD (the dual-layer disc that incorporates both DSD and Red Book CD audio) was big news at AES, with the debut of a number of new players, including Sony's **DVP-S9000ES** (which plays both DVD-Video and SACD discs) and **SCD-C333ES** (which boasts a five-disc carousel). Philips also announced the imminent release of the **SACD-1000**, the first player capable of reading multichannel discs, as well as a giving attendees a sneak preview of an eight-channel DSD recording/editing workstation code-named **Sonoma** (it's expected to go into production in the fall of 2001). Partnering with Sony, TASCAM announced that a new two-channel version of their DA-98 — the **DA-98HR** — will record DSD audio onto standard DTRS tapes. Need more tracks? No problem — up to sixteen of the machines can be synchronized together for 32-channel recording.

Speaking of tracks, the iZ Technology **RADAR 24** is the successor to Otari's RADAR I and RADAR II standalone hard disk recorders, providing 24 tracks (instead of 12) and support for sample rates as high as 192 kHz.

If you don't already own a CD burner or three, you might want to check out the newest offering from Sony: the **CDR-W66**, a "pro" version of the CDR-W33 Super Bit Mapping recorder unveiled at Summer NAMM (see our October 2000 issue). New features include support for 96 kHz audio (which is automatically down-sampled to 44.1), a double-speed duplication link, balanced analog and AES/EBU I/O as well as a word clock input, plus the ability to apply the built-in DSP functions to incoming digital as well as analog signal.

Last but not least, the HHB **Portadisc** proves that there's still some life in the MD format. Optimized for broadcast and field applications, this rugged portable stereo recorder boasts a heavy-duty professional drive mounted in an all-steel chassis, yet only weighs 2 kg (4 lbs 7 oz) with eight NiMH AA batteries installed. Battery life is said to exceed three hours; other features include a large, illuminated display, one-touch recording, lockable record level, time/date stamping, a built-in monitor speaker and backup microphone, and a six-second pre-record buffer so you'll never miss the start of a take. *—Howard Massey*

MEDIA

BASF presented their formatted **DTRS** blanks for the TASCAM/Sony digital recording system format. HHB announced an upgrade of their **2.6 GB** and **5.2 GB Magneto Optical discs**; a rigorous testing procedure ensures that the initial number of bad sectors on a disc is below the maximum percentage allowed within the MO format spec. In the "this is very cool" category, Medéa introduced the **AudioRaid SCSI,** a plug-and-play SCSI array that supports up to 64 tracks of 24-bit, 96 kHz audio recording. AudioRaid looks like a single

SCSI disc to your computer, negating the need for fiddling with RAID software and hardware. *—Steve La Cerra*

RECORDING CONSOLES

It seems that every AES show features a product that lowers the price bar so much that even jaded journalists like ourselves have to shake our heads in amazement and say, "How do they do that?" Expected to ship next spring, the TASCAM **DM-24** eight-bus digital console falls firmly in that category. Featuring motorized touch-sensitive faders, 16 analog inputs, 24 TDIF plus eight ADAT digital inputs, comprehensive EQ and dynamics on each channel, full mix automation, dual built-in effects processors, dual stereo AES/EBU and S/PDIF outputs, full transport control, and the ability to pass 24-bit, 96 kHz audio (albeit with a reduced number of channels), the DM-24 has an astounding projected retail price of $2,999.

Despite all the emphasis on digital consoles, analog mixers are far from becoming an endangered species. In fact, AMS Neve used AES to announce their first all-new analog console in almost 20 years, the **88R**. Incorporating audio circuitry design that's linear from start to finish (no VCAs, even in the monitor section), the 88R comes equipped with full automation, with moving mini and large faders plus automated panning. Also included are comprehensive surround mixing capabilities.

Mr. Neve himself (who now is associated with Amek) was on hand to unveil the Amek **Media 51** entry-level analog console, which utilizes a new mic preamp design, mix automation, and extensive surround features, including downmixing capability, easy insertion of encode/decode processors, and multiple solo, cut, and metering modes. Completing the triumvirate of classic British console manufacturers, Oram unveiled the **BEQ Pro24**, a 24 x 8 analog console that incorporates the classic warm sound of "Oram Sonics" EQ, along with a dedicated sub-bass output with built-in crossover and stereo high-definition EQ and compressor for mastering functions. Last but not least, the upgraded **Status 2** from Otari (a digitally controlled analog console) now offers 24 busses (instead of 12) as well as a five-way panner on each input module and an automated joystick panner for those sweeping surround moves.

If you already own a non-surround-ready analog console and you'd like to get into multichannel mixing without replacing it, you might want to look into Scarab Technology's **Trimension-1**. The system adds a control surface with panning joysticks and motorized faders on every channel, a 24-bit, 96 kHz A/D converter box, and digital I/O router (which supports up to 80 channels) to your existing console. Their Maitre-D' system adds mastering capabilities by integrating a Dolby DP570 processor and providing remote control of Dolby encoders.

Finally, owners of the Roland VM-3100Pro V-Mixing Station will want to check out the company's new **RPC-1 R-BUS Interface Card**, which allows the console to act as a control surface for any ASIO-compliant PC recording software. *—Howard Massey*

SIGNAL PROCESSORS

There was lots of news in this category, ranging from state-of-the-art digital converters to sampling reverbs. Empirical Labs (makers of the popular Distressor) unveiled **FATSO**, which allows you to revisit the days of analog tape by simulating the harmonic generation, soft clipping, and saturation of your favorite strip of oxide-coated material — plus classic knee compression.

If you're looking for something completely different in the world of equalizers, you'll want to check out the dual-channel CLM **Expounder**. It features four overlapping bands of EQ and hi-cut and low-cut filters, with adaptive circuitry that responds to the dynamics of the program material. This kind of unique amplitude-domain processing allows for precise "microsurgery" such as dynamic noise filtering and frequency-limited expansion.

The new TC Electronic **Triple•C** compressor also pushes the boundaries of traditional compressors by offering three kinds of dynamics processing: standard full-band compression, three-band spectral compression, or a unique Envelope Compression, which allows you to alter the envelope of the sound by changing attack and release gain. Other features include look-ahead compression, spectral balancing, and a "digital radiance generator" to add tube-like overtones to the signal.

TC's recent alliance with IVL — TC Helicon — is already bearing fruit with the release of their first product, **VoicePrism**. It provides a full complement of tools for vocal processing, including an onboard mic preamp, compression,

TRENDS: NETWORKING

Handshaking isn't just for friends anymore; one notable trend at this year's AES was the proliferation of networking options and devices that allow interconnection of a wide range of equipment — even gateways to the Internet. Yamaha were showing a range of wicked-fast (200 Mbps) mLAN IEEE 1394-to-audio converters, including a standalone processor, an expansion board for their new line of synths and samplers, and a YGDAI card that can be installed in their 02R or 03D consoles. Otari announced that they will be partnering with Yamaha to jointly develop an mLAN chip that will up the ante by routing 32 channels of 24-bit 48 kHz digital audio at 400 Mbps. Otari also unveiled a new line of Echelon Series connectivity products, including the ND-20, which provides up to 32 channels of A/D, D/A, and sample rate conversion at up to 96 kHz sampling rates. Four I/O card slots are available for remotely controlled mic inputs, line inputs, line outputs, and AES I/O; MADI, TDIF, and mLAN formats will all soon be available as well. Another new Echelon product is the FS-96 — the successor to the popular UFC-24 format and sample rate converter. The FS-96 can handle up to 24 channels of 24-bit audio at sample rates up to 96 kHz, in all the common formats, including AES/EBU, TDIF-1, ADAT (optical), and SDIF-2, with optional MADI and IEEE 1394. Also jumping on the 1394 bandwagon is TC Electronic, who announced that they have licensed the rights to Digital Harmony's Studio portfolio of technologies, including DHIVA 1394 interface modules — a move that will undoubtedly lead to high-speed networking capabilities in the next generation of TC products.

Taking a slightly different tack was Euphonix, who unveiled their own suite of "InterNetworking" products and technologies, including the FC727 format converter, which not only does bidirectional conversion between TDIF, AES/EBU, SDIF-2, ADAT, ProDigi, and MADI formats, but also links directly to Pro Tools. The company also showcased their eDeck software, a free downloadable Rocket-powered Windows player (with a Mac version to come) that enables the playback, conversion, and transfer of 24/96 multichannel files over the Internet.

—Howard Massey

gating, dual parametric EQ, voice harmony generation, ADT, and effects such as reverb, delay, chorus, and flange.

Sampling reverbs — where the acoustic signatures of true acoustic environments are stored in memory and can be applied to incoming signal — are becoming all the rage, and now Yamaha joins the party with the release of the **SREV1**. Able to operate in either two- or four-channel mode, its features include two mini YGDAI card slots, dual AES/EBU I/O, two serial ports, and external word clock input. Storage options include a CD-ROM drive for loading data and upgrades, plus a PCMCIA card slot. With the addition of a DSP expansion board, reverb times can be up to 10.92 seconds. In other high-end reverb news, Lexicon announced a digital I/O-only version of their **960L** multichannel processor.

CEDAR showed their **DNS1000 Dynamic Noise Suppresser**, a 40-bit, multi-band processor designed to remove rumble, hiss, whistles, and other noises that might interfere with recording on location.

There were also lots of new converters on display at the show, including the eight-channel Genex **GXA8** and **GXD8**, each of which supports 24-bit audio with sample rates up to 192 kHz; optional expansion cards allow conversion to and from DSD format. Weiss released the **DAC1**, a stereo 24/96 D/A converter with four digital inputs (three XLR and one TOSlink) and "virtually zero" output impedance. On the other side of the coin, the MindPrint **AN/DI PRO** includes dual mic preamps and line inputs plus balanced inserts, delivering 24/96 audio via AES/EBU and coax and optical S/PDIF outputs. —*Howard Massey*


DIGIDESIGN PRO TOOLS 5.1

STUDIO MONITORS

Garnering buzz on the show floor was newcomer Truth Audio's **TA-1P** passive monitor, which features two 5-inch woofers and a single tweeter for a frequency response of 48 Hz to 20 kHz. Power handling is said to be 165 watts into four ohms. Alesis announced an update to their venerable Monitor One, the **Monitor One Mk 2**. Improvements include new drivers, improved frequency response (45 Hz to 20 kHz), radiused cabinet edges, and enhanced porting.

Holding down the bottom were Genelec with their new

1093A active subwoofer. The 1093A features frequency response from 18 to 85 Hz from a 10-inch driver powered. Loop through jacks allow multiple 1093A's to be daisy-chained when higher SPLs are necessary.

Dynaudio Acoustics was showing their **BM5.1A** surround monitoring system, comprising five BM6A powered monitors (seven-inch woofer, 100-watt amplifier) and an active BX30 subwoofer with 12-inch driver and 130-watt amplifier.

Too much bass is never enough bass; Tannoy introduced their **PS350B** active subwoofer, which includes a 15-inch driver, built-in power amp, XLR inputs, variable low-pass filter, defeatable 80 Hz high-pass filter, and ground loop isolation. *—Mitch Gallagher*


AMEK MEDIA 51

SOFTWARE

Everyone on the show floor was talking about the new **Pro Tools 5.1** software. As its version number implies, there's full support for surround mixing, with integrated automatable panning on each track and a "jellyfish" display that shows the relative levels of all main channels, but there's a whole lot more here than just surround features. Pro Tools owners everywhere will rejoice at the fact that version 5.1 allows you to create stereo tracks — even multichannel tracks — not to mention multiple levels of undo (hooray!) and the ability to have multiple plug-in windows open simultaneously. MIDI users will also appreciate the addition of event list editing. The new mixer architecture allows you to designate more than one output or send for each channel, so you can mix to multiple formats simultaneously, and a new I/O Setup window allows you to create custom routings for each session. Plus, there's increased DSP processing power — TDM systems can run both TDM and RTAS (Real-Time AudioSuite) plug-ins, utilizing both the chips on your DSP card and your CPU's own horsepower. And if you're into tactile control, the new **Edit Pack** option for ProControl adds a color-coded QWERTY keyboard, 20 dedicated edit function buttons, dual motorized touch-sensitive panning joysticks, a trackball, and eight high-resolution meters.

Steinberg unveiled version 1.5 of their acclaimed **Nuendo Media Production System**, now available for both Macintosh and Windows platforms. Support is provided for up to 200 tracks of 24/96 digital audio, as well as a video track and virtually unlimited MIDI tracks. Extensive surround mixing features are standard, and the company announced that a future version will include integrated Dolby Digital encoding. If you're getting tired of mousing around, you might want to consider adding **Houston**, Steinberg's new MIDI/USB remote controller for VST audio engines such as Nuendo and Cubase. It provides nine touch-sensitive motorized faders, eight rotary encoders with LED position indicators, and a matrix of buttons to bring all aspects of the VST mixer within easy reach. In addition, there's a large LCD, a numeric keypad, transport controls, and a jog/scrub wheel.

TC Works **Spark 2.0** for Macintosh features a new file engine for faster operation and an enhanced FXmachine plug-in matrix that supports up to 40 real-time plug-ins.

Internet broadcasters everywhere will want to check out Waves's new **MaxxStream**. Designed to turn your Windows NT PC into a streaming workstation, the MaxxStream software simultaneously encodes audio to multiple formats at multiple bitrates, while the accompanying PCI card provides stereo analog and S/PDIF digital I/O, with 20-bit conversion and support for sample rates up to 96 kHz. Various configurations are available, including the **M1600**, which can condition up to 16 stereo or 32 mono audio streams simultaneously. *—Howard Massey*

PLUG-INS

The world of software-based signal processing continues to expand; Pultec, purveyors of TDM- and MAS-compatible vintage processor recreations introduced a pile of new products, including the **Pultec EQP-1A** equalizer, **Fairchild 660** compressor, **JoeMeek SC2** photo-optical compressor and **VC5 Meequalizer**, **Tel-Ray Variable Delay** (remember that thing?), and **Moogerfooger 12-stage Phaser** and **Analog Delay**. Also introduced was **BF Essentials**, a bundle comprising a plug-in guitar tuner, a meter bridge, and a phase correlation meter.

Waves announced **version 3.0**, which increases processing resolution to 48-bit (dithered) for some TDM plug-ins, adds 88.2/96 kHz support for native plug-ins, as well as graphical enhancements and completely redesigned user interfaces. Also new for Waves is the **Renaissance Reverberator**, and the **Renaissance Collection**, which comprises the Reverberator as well as the Renaissance EQ and Compressor. But the big news from Waves was the elimination of

TRENDS: VINTAGE REVIVAL

Vintage audio gear is enjoying tremendous popularity, trickling down from commercial studios to the project world. Though we've noticed this trend for the past few years, it was more evident than ever at the 2000 AES Convention. Some companies such as Universal Audio are actually re-producing faithful versions of past gear. Others, such as Great River Electronics, introduced new products with a "tip of the hat" to designers-past.

At least one manufacturer showed an all-tube recording console, while several mic pres were introduced with "retro" features such as adjustable input impedance or Class-A operation. There were two newly introduced re-productions of the famed Telefunken ELAM 251 microphone, and ribbon microphones finally seem to be getting the attention they deserve. VU meters are getting bigger, and even products with built-in analog-to-digital conversion are using tubes in the pre stages. This trend is also spreading to software — we saw emulations of Pultec EQs as well as Prophet and Moog synthesizers. If only we could get our hands on a vacuum tube multitrack....

—Steve La Cerra

dongle-based copy protection; the company's products will now use a challenge/response system with online authorization.

As is always the case, the Digidesign Partners booth was brimming with plug-in manufacturers; in addition to the aforementioned Bomb Factory, Line 6 was showing their **Echo Farm**, a vintage delay box emulation plug-in, and Digidesign was showing their **Reverb One**, which the company says "...compares favorably to any software or hardware reverb processor on the market today."

Speaking of reverb plug-ins, TC Works announced the first third-party surround plug-in support for Steinberg's Nuendo platform with their **Surroundverb**, which provides 5.1 reverb with a unique surround metering window.

Native Instruments, best known for their line of software synthesizers, announced the first in a line of FFT-based effects, **Spektral Delay**, which can delay each of 512 frequency bands independently; each band also has feedback and modulation control. Available in PC and Mac VST formats, Spektral Delay also offers MIDI control over all parameters.

Finally, Sounds Logical's **WaveWarp** real-time audio effects processor for Windows can now act as a DirectX plug-in with up to 32-bit resolution. Modules consist of equalizers, dynamic range controllers, time delays, distortion, choruses, pitch shifters, reverbs, signal generators, and more. *—Mitch Gallagher*

SYNTHS/MIDI

The AES show focuses mainly on pro audio, but synth, sampler, and instrument manufacturers also put in appearances. Announced at the show was the Emu **E4 Platinum Sampler**, a fully loaded incarnation of the E4 Ultra featuring 24-bit A/D converters. Also included are the new **RFX-32 Effects Processor**, EOS 4.5, a 20 GB internal hard drive, 128 MB of RAM, 128-voice polyphony, six analog inputs and 16 analog outputs, eight inputs/16 outputs on ADAT lightpipe, dual SCSI ports, and 20 soundware CD-ROMs.

Incorporating the features of Kurzweil PC-series keyboards, the **PC2R** 1U rackmount synth offers triple-strike pianos, stereo strings, classic keyboards, and more, including the company's award-winning KB-3 organ models. The 64-voice module offers dual effects processors, expandable ROM, and 24-bit digital outputs.

New from software synth specialists Koblo is **Tonemill**, a polyphonic, multitimbral, virtual modular software synthesizer for Mac or Windows. Tonemill supports Pro Tools, Cubase, Logic, Cakewalk, and Digital Performer, and its features include a sample player, waveform editor, and "built-in" effects (chorus, echo, reverb, etc.).

Akai was showing their limited edition **MPC2000XL-SE1**, a groove box that combines a sampling drum machine and 64-track sequencer, and sports a whole new visual look. The **Version 2.0** operating system for S5000/6000 samplers includes USB network capabilities (the new

WEBLINKS

- Learn more about the Audio Engineering Society and the 109th AES convention at www.aes.org.
- Surfers can point their browsers to www.musicplayer.com for even more AES show coverage.

IB-S56USB board and **ak.Sys** sampler network software). Thirty-two samplers can now be controlled from a computer, and you can drag and drop directly between samplers without transferring programs and samples to disk or computer. Also new is support for Roland CD-ROMs and Emu E-IV SCSI disks. —*Mitch Gallagher*

SOUND REINFORCEMENT

Plenty of noise was made in live sound circles with a variety of gear. AKG displayed the WMS 40, an affordable UHF wireless system that operates up to 30 hours on a pair of AA batteries. Audio-Technica revealed the **ATW-7373**, a multichannel UHF wireless handheld system, which uses the same element as in the AT4033 studio mic.

Allen & Heath announced a new live console, the **ML4000**, a VCA-controlled, 12-aux desk equally suited for either front-of-house or monitor mixing. ATI pulled the wraps off the **Paragon II**, a sound reinforcement desk with 88 assignable inputs as standard, plus Distributed Intelligence — a feature that allows the console to immediately reset in event of a power failure without the delay of a "reboot" common to other computer-controlled desks. On display at the Peavey booth was a series of compact mixers including the **RQ2310**, **RQ2314**, and **RQ2318**.

Speaker systems were very well represented this year in both powered and non-powered versions. Rather surprisingly, Crest Audio debuted the **CT** Series of loudspeakers, a new line for applications requiring high SPLs and a wide dynamic range. The CT Series consists of four trapezoidal cabinets and two floor wedges. EAW unwrapped their new stage monitor, the **SM84**, a split-baffle design with an angled-front horn and four low-frequency drivers. EAW also debuted their **E-Powered** series of cabinets, designed with locking power connectors, circuit breakers, and automatic voltage selection. Electro-Voice unveiled the **X-Line Array**, a system designed to provide wide horizontal dispersion from a vertical array. The X-Line system consists of the **XVLS** full-range main cabinet, the **X-fil** downfill cabinet, and the **X-sub**.

JBL uncovered the **VerTec Line Array**, a three-way enclosure with two 15's, four 8's, and three compression drivers coupled to JBL's high-frequency WaveFormer. JBL also rolled out the EON Generation 2 line featuring the **EON15 G2**, a bi-amped, 300-watt enclosure, and the **EVO Intelligent Sound Reinforcement System**, a complete, ready-to-use sound system with automatic digital control over room equalization, feedback suppression, delay settings, and amplifier dynamics. The EVO system includes intelligent powered speakers, mixer, wireless and wired microphones, system controller, and all necessary cables and stands.

Mackie Designs showed several powered sound reinforcement speakers including the **Fussion 3000** (active, three-way, full-range), **Fussion 1800S** subwoofer, and the **SR1530**, a triamped, three-way, horn- loaded system. Meyer Sound's **MTS-4A** is four-way powered loudspeaker designed for ground-stack or flown sound reinforcement use. Peavey displayed four new **Impulse 1000** loudspeakers, including the **1012** full-range and **1012 SUB** subwoofer, and the **1015** full-range and **1015 SUB** subwoofer. Both full-range cabinets provide for biamp capability and use a ferrofluid-cooled, titanium compression driver. Radian Audio Engineering presented their latest wedge monitor, the **RMW-1122**, a passive two-way monitor with coaxial 12-inch and two-inch drivers.


MACKIE FUSSION 3000

Alesis extended their RA line of power amps with the **RA150**, **RA300**, and **RA500**. Each amp features DC-coupled, discrete topology, balanced 1/4-inch and unbalanced RCA input jacks, and front-panel level controls. Crown introduced the **CE 4000**, a high-output amp based on their Balanced Current Amplifier (BCA) technology for increased efficiency and reliability. QSC's **Powerlight A Series** adds onboard, analog signal processing capability, switchable gain sensitivity, and adjustable power output to the established line of power amps. Hafler unveiled two amps designed for touring purposes, the **SR2300** and the **SR2600**. —*Steve La Cerra*

FURNITURE

Middle Atlantic showed the **Edit Center Multimedia** studio furniture featuring acoustically damped enclosures for reduced computer noise. Taytrix displayed a full line of studio furniture, including the **Oval Workstation**, a two-tiered work desk mounted in a steel frame. Features include an adjustable meter bridge shelf and speaker wings and decorative black trim on wood surfaces. —*Steve La Cerra*

COOL STUFF

Audio Engineering Associates (AEA) showed their **SPL2020** stereo phase meter. Contained in a lunch box-style chassis, the SPL2020 features a single-point visual display for phase-checking your mixes. Neutrik's **Minilyzer** is a handheld, battery-operated measurement tool; functions include analysis of THD, oscilloscope, polarity check , PPM or VU level metering, and third-octave frequency analysis. RDL (Radio Design Labs) showed their **RU-AED4** digital audio distributor for AES/EBU protocol as well as the **RU-SPX4** S/PDIF signal selector switch. The **Piano Barre** from Slider is an acoustic piano microphone support system, allowing easy placement of mics above the strings and sound board. —*Steve La Cerra*

COOLEST (AND HOTTEST) NEW TECHNOLOGY

Over in a corner of the show floor, a new exhibitor called Noren Products were showing their NoiseLock isolation box, designed to house noisy equipment such as PCs and hard drives. Normally, a functional but unexciting product like this wouldn't rate more than a glance, but the hip thing about NoiseLock is that it uses proprietary "heat-pipes" instead of cooling fans to absorb and dispel the heat generated by the gear inside the box. Their demonstration of the technology couldn't have been simpler: two heat pipes, a bowl of ice water, and a bowl of hot water. Within seconds of dipping the pipe in a bowl, it became incredibly cold (or hot) to the touch. At last, total silence from noisy gear!

—*Howard Massey*

THE BLUE RIBBON AWARDS

THE ANNUAL *EQ* EDITOR'S CHOICE AWARDS

DIGIDESIGN PRO TOOLS 5

NEUMANN M150

AUDIO-TECHNICA ATW-7373

ALLEN & HEATH ML4000

EVENT EZBUS

TASCAM DM-24

GT ELECTRONICS VIPRE

JBL EVO

By
Mitch Gallagher

Each year, myriad new products are introduced at the AES show. To be honest, they're all pretty darn exciting in their own way. But there are always those that emerge as truly outstanding. To find those bright lights, *EQ*'s editors combed the AES floor in search of the best of the new products on display, and rewarded them with our coveted Blue Ribbon Award.

Here's the process: Each editor was allowed to nominate products in each category; products had to be newly introduced at this show. When the nominations were in, voting ballots were sent back to the editors and contributing editors who had attended the show. The ballots were collected, the votes were counted, and the results were tabulated.

And the winners are...

SOFTWARE

Digidesign Pro Tools 5.1 — The new version of Pro Tools, with its multitude of enhancements was the buzz of the AES show, and the overwhelming choice of our editors.

PLUG-INS

This category was hotly contested, with many worthy new products announced at AES. In the end, there was no choice but to award a tie — a four-way tie — to the ***Waves Renaissance Collection***, ***Bomb Factory Pultec EQ-P1***, ***Line 6 Echo Farm***, and ***Native Instruments Spektral Delay***. Each of these plug-ins is a unique offering, and each brings something new to the world of software-based processing.

COMPUTER AUDIO INTERFACES

Our editors chose the ***Event EZbus*** interface for its flexibility and powerful combination of features and price.

SIGNAL PROCESSOR

Empirical Labs FATSO took this category, despite the introduction of a variety of cool new processors. Who can resist the best aspects of analog tape combined with great compression?

MICROPHONE PREAMP

The ***GT Electronics VIPRE*** took the Ribbon this year. As one of our editors commented, "Looking at the insides convinced me...anyone who shock-mounts a tube PCB is all right in my book."

CHANNEL STRIP

Channel strips combine a preamp with EQ, compression, and/or other processing; they've become increasingly popular of late. Our editor's pick is the ***George Massenburg Labs 2020*** high-resolution discrete input channel, which combines a line/mic preamp with EQ and dynamics control.

SYNTHS/MIDI

Raising the bar for "sampling power per square inch," the ***Emu E4 Platinum*** sampler is fully loaded with useful options, and sports 24-bit I/O.

LIVE SOUND SPEAKERS

Our editor's choice for best live sound speakers goes to the ***JBL EVO***; a complete sound-reinforcement package with smart digital control.

POSTPRODUCTION

A number of powerful new post products were introduced at AES this year, but the prize goes to the ***Digidesign EditPack***. As one of our editors said, "Motorized joystick panners, dedicated edit buttons, a color-coded QWERTY keyboard, and a trackball — what more could a Pro Tools operator ask for?"

STUDIO MONITORS

In studio monitors, we've got a tie. The ***Truth Audio TA-1P*** gets a nod for catching the eyes and ears of our editors. Maybe it was the go-against-the-flow approach; it's passive instead of active. Also in the winner's circle is the ***Tannoy PS350B*** active subwoofer with its 15-inch speaker, built-in amp, and LCR/LFE/XLR balanced inputs.

RECORDING CONSOLES

Several important new consoles appeared at AES this year, but none shattered the price/performance barrier as handily as the ***TASCAM DM-24***. Eight busses, analog and digital I/O, EQ and dynamics on each channel, two effects processors, and 24/96 support for $2,995? Scary....

STUDIO MICROPHONES

The prize went to the ***Neumann M 150*** omnidirectional microphone; but the voting was very close. The Shure KSM44, Lawson L251, and Soundelux ELUX 251 tied for runner-up, only a step behind the Neumann.

LIVE SOUND MICROPHONES

As is to be expected at AES, a number of new microphones were introduced for live sound applications. Our pick for the award? The ***Audio-Technica ATW-7373*** wireless system, which uses the same element as the popular AT4033 studio mic.

LIVE SOUND CONSOLE

With its ability to serve as either a front-of-house or monitor console, the ***Allen & Heath ML4000*** took the Blue Ribbon this year.

SPECIAL AWARDS

Not every deserving product can win an *EQ* Blue Ribbon Award. These special award categories honor products that our editors feel deserve extra recognition.

FUNNEST PRODUCT

Without a doubt, this award goes to the *Alesis AirFX*. Powerful processing under unique gestural control; we're betting AirFX is going to be very popular in studios, on stage, and in DJ booths.

COOLEST PRODUCT

Silent temperature control for noisy studio gear? That's what the *Noren NoiseLock* promises.

UTILITY PRODUCT

Our favorite practical utility product was the *LittleLabs PCP Distro* — everyone needs to split a guitar signal at one point or another.

MOST PROMISING

Steinberg Nuendo 1.5 came up repeatedly in our discussions of the Blue Ribbon Awards. With its broad support for media production, addition of VST instrument and ReWire support — to say nothing of Mac support — version 1.5 stands poised to have a big impact.

MOST PROMINENT NEW SUPPORT OF PREVIOUSLY UNSUPPORTED EXISTING TECHNOLOGY

The introduction of DSD-compatible products to assist in bringing SACD to the masses.

SOON-TO-BE-EVERYWHERE

Analog Devices new AD1896 sample rate converter chip — greater than 130 dB signal to noise, up to 192 kHz sampling rates.

Tony Visconti

on crafting the ultimate rock sounds

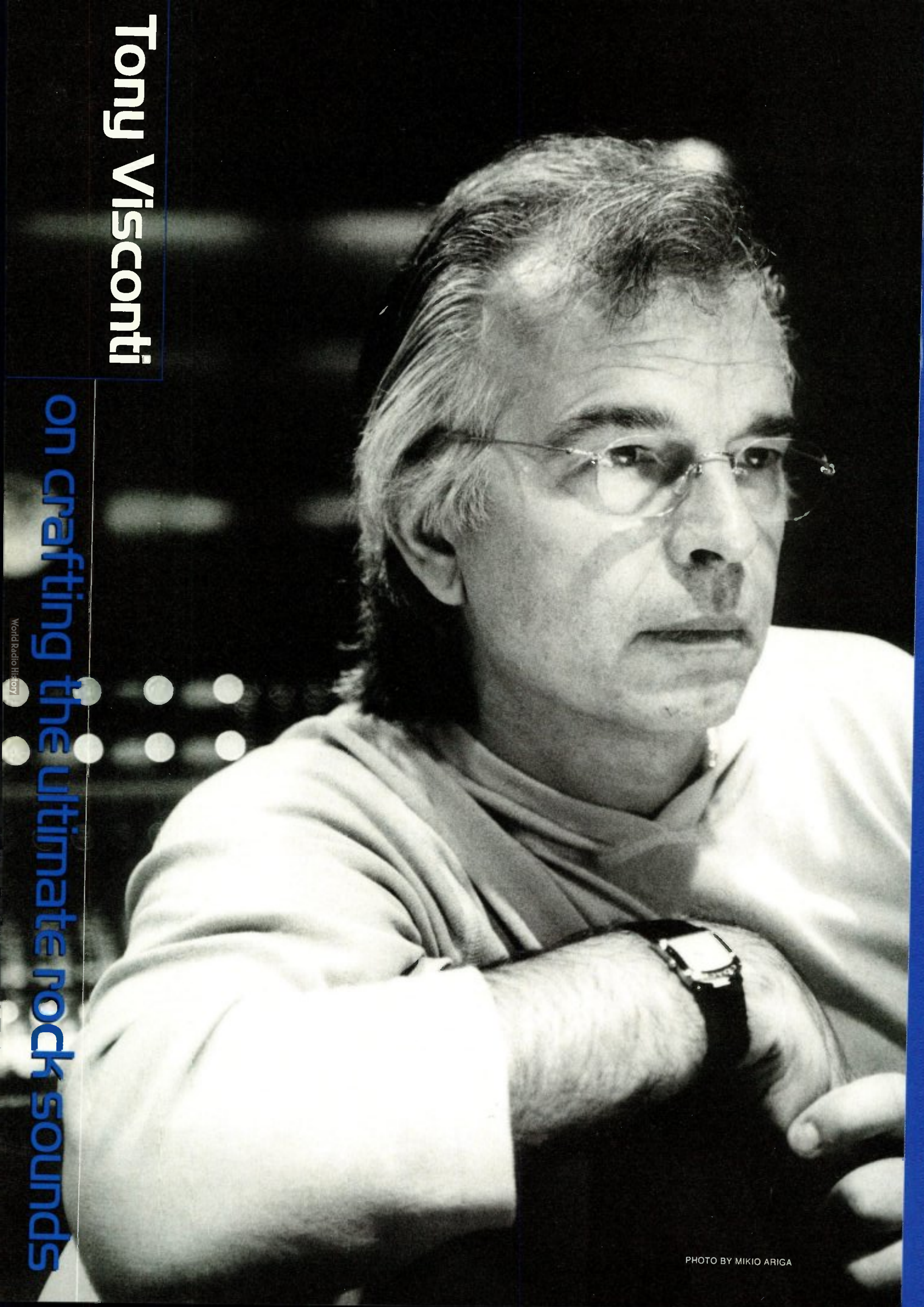


PHOTO BY MIKIO ARIGA

BY HOWARD **MASSEY**

THE SOUND OF ROCK

They say lightning never strikes twice. But they're wrong, as Tony Visconti can attest. Imagine discovering and developing the career of an artist with the stature of a David Bowie, then finding another gifted (if enigmatic) young singer by the name of Marc Bolan, and going on to produce a string of hit albums for him, too.

*Born in Brooklyn, New York, Visconti made his fateful move to London in 1968, initially apprenticing with legendary producer Denny Cordell. His musical and arranging talents soon made Visconti one of the most in-demand engineer/producers on that side of the Atlantic. In the nearly two decades he spent in England, he built his own studio (Good Earth) and helped craft dozens of important records, not only with Bowie and Bolan & T. Rex, but also with Procul Harum, the Moody Blues, and Joe Cocker, as well as numerous post-punk artists such as Iggy Pop, Adam Ant, Hazel O'Connor, Thin Lizzy, the Boomtown Rats, and Sparks. In 1989, Visconti returned to his native New York, where he maintains close contact with the listening public through his Web site (***www.tonyvisconti.com***) and continues to nurture new talent in his state-of-the-art project studio.*

EQ: You're one of a small group of producers who were active in the '60s and '70s who have kept current with technology — for example, you've got your own Pro Tools rig. Do you feel that this is really a better way to make records than in the old days?

Tony Visconti: It's just another tool in my arsenal. I used analog tape last year on a project, and we flew some tracks into Pro Tools for digital editing. When we flew it back to analog, we found it actually improved the sound. Maybe, because of phase shifting, there was more depth to the recording; there was kind of a more three-dimensional sound to it. It's undeniably the future.

You're a bass player, so I'm particularly interested in the approach you take toward recording bass.

I'll split the bass and record both DI and amp. If I'm in a big studio and I've got lots of tracks, I'll record them on separate tracks. Again, I come from the old school where, if the sound is great, you can blend the two — just check the phasing, always check the phasing. I've got a nice Manley DI that I carry with me; it's got tube circuitry and it's one of my high-end boxes. I've also got an ART preamp; I use that sometimes as a bass DI. If I'm playing bass, I don't usually like to play through an amp in the studio, so I'll just use the DI — I have enough gizmos here to make it sound fat.

When you're recording both amp and DI bass, do you tend to use one signal for low end and the other for attack?

Yeah, that would be the case. The DI's going to have more attack than the amp, so you might as well go for a fat amp sound — that's all it's really good

THE SOUND OF ROCK

for. You're not going to get clarity out of a bass amp.

Do you typically compress both signals as you are recording them?

Oh, yes.

Do you compress them again when you mix?

I'm a compressor freak. Compression is the sound of rock. At some stage, you have to compress a lot, especially low end. I don't really slam it when I go in, so it's actually quite nice in the mix. Sometimes I won't compress the bass on a mix, but I'll put it high in level and I'll put compression over the stereo bus so the bass just kind of does its thing and interacts with the kick drum. Pumping is acceptable.

One problem that many people have when recording bass is getting a fat sound without it being woofy and flabby. How do you tighten up the low end?

Well, whenever I compress a bass, I do it through a stereo compressor and put the link in. Your DI might not hit the compressor as hard as the amp would, but the amp might hit it really hard and it'll bring the attack down appropriately. I'll do that on the mix as well.

So you're sending the DI through one side of the compressor and the amp through the other side, but linked.

Yes, so it's like treating it as a stereo source, which it really isn't, but the levels will always ride each other. So on those notes where the DI might be much louder than the amp, the compressor's controlling it. Sometimes after I do that, I might gang the two signals and send it to a third compressor to squash it even further.

How do you EQ bass?

I don't do anything radical except that I look up the frequency of the key that the song is in. Say if it's in the key of F, I know that the low F on the bass string is something like 48 cycles. If that note's not loud enough, I'll pinpoint that frequency and boost it. In fact, you can often get more clarity — you can make the low end seem more apparent — if you boost it again an octave higher. So if I'm boosting it at 60 cycles, I'll also do a slight tweak at 120, and I might even go to 240. Even a bass sound has got harmonic overtones, and sometimes that's where the definition lies — it's not just only in the low end.

When I discovered this principle, I found that, if you go a little higher up, you can actually hear the note itself — not just the warmth of the note, but the clarity of the note. This way, in the whole mix, you can actually hear the notes that the bass player's playing.

Tony Visconti (right) and David Bowie at New York's Power Station in 1979 during the recording of *Scary Monsters.*

Do you have any favorite equalizers, or do you tend to use the board EQ?

Well, for these precise things I have to use a graphic. Any graphic that divides the octave in three or four will do, and parametrics will also do the job.

What's your general mic technique for acoustic piano?

I'll open the top and do a crisscross kind of stereo thing with the two mics pointing toward each other like a pyramid — either two [Neumann U] 87's, or [KM] 84's. Neumann 84's are my favorite for piano.

Positioned over the sound holes or over the strings?

Over the sound holes. I don't go right up to the hammers — I like to raise the mics and let the sound wave form a bit. And occasionally I'll use a third mic if I'm not getting really good low end. I have this philosophy that a stereo piano can sometimes sound smaller than a mono piano. You can't record a piano that sounds as wide as a 20-foot stage — it's a psychoacoustic thing — so sometimes I'll record a piano in stereo and then on the mix I'll just mono it. But I'll record in stereo for insurance, so we've got that sound.

If it's a great room, you've got to put room mics up and blend them in. So with piano, I might use up to five mics at once. Again, I use stereo compressors; I can't emphasize how important that is. If you put a mono compressor on each mic, you're going to have the worst sound balance in the world; you'll have a very lopsided piano.

Are you creating a stereo image out of the five mics and then feeding the stereo send to a stereo compressor?

No. It's usually a two-mic technique — I'm using that 80% of the time — so my stereo mics will go through a stereo compressor. If I'm using stereo room mics, then I'll have a stereo compressor on those stereo room mics so at least both sides will go up and down. I might not use a compressor on the fifth mic at all, and maybe I'll stick it on another track. I don't usually use the low-end mic; in my whole life, I've done that maybe 5% of the time.

What sort of compression ratios do you use? Heavy compression or gentle?

I never use a quick attack on anything — that's ridiculous. It'll be around a 50-millisecond attack so the initial attack gets through, and then it kind of tapers off. The compression [ratio] will be maybe 3:1; usually no more than that. If it's really going to be rock 'n' roll, maybe 4:1.

Are those the same kind of compression settings you'll use for bass or other instruments?

You know, I'll use the same compression settings for any instrument. I kind of have this philosophy where I know how much compression I like to hear, and it almost applies to any instrument. On the mix, however, if it's a very uneven bass sound, I'll really hit it. I might hit it 10:1 on the mix. But basically the same thing applies — 3:1 on the bass, 4:1 on recording. I don't think 2:1 is compression; you might as well turn it off.

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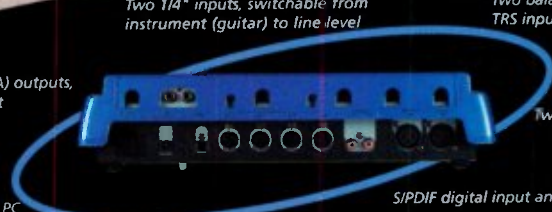

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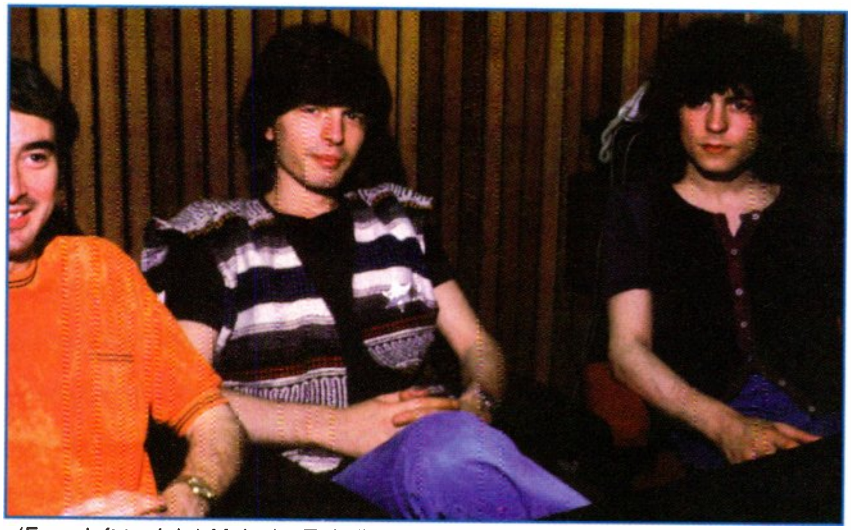
THE SOUND OF ROCK



(From left to right) Malcolm Toft, Tony Visconti, and Marc Bolan in 1969 during the recording of Tyrannosaurus Rex's *Prophets, Seers and Sages.*

In vocals, the same thing: I'll do gentle 3:1 compression on a vocal on the way in, but if it's a vocalist who's really whispering and screaming the next minute, I'll have to hit that maybe 5:1 or 6:1 on the mix. If there is a whisper part, in Pro Tools I'll just slice it out and give it its own track and treat it differently from the scream part. It's ridiculous [to think] that one compression setting will cure all evils. Sometimes you just have to mult something out for different parts of the song.

What's your generic approach to miking drums?

First of all, when I get a good drum sound, I take photographs of the mic placement. I have a series of photos from over the years — they're like photographs of my drum sounds.

Generally, I try to use the minimum number of drum mics until I find a problem — then I'll start sneaking mics in. So I'll use one mic for the kick, one mic over the snare. For hi-hat, I like to get as close to the stick as possible. I have been known to tape an AKG 451 to the actual pole of the hi-hat, or even clip a lavalier mic to the pole, so as it goes up and down, so does the microphone — and it's always following the same signal.

And it's not picking up all kinds of rumble from the stand?

No, because when I record a hi-hat, I'll filter out all the low end so you won't get any air. There's nothing on a hi-hat that is below 200 [Hz] — it's just junk, so I take it out. If I'm mixing someone else's track and they haven't done that, I'll do that during the mix.

Sometimes I'll take a second microphone stand and hang the cable on it so the mic dangles over the hi-hat like a marionette. Another way I like to record hi-hats is to take a [AKG] 451 that has that screw-in elbow so the capsule can be bent at an angle, like the Concorde jet. Then you can sneak the microphone right over the hi-hat and bend it downward. Again, I love to get as close to that stick hitting the metal as possible. For toms, I just like to sneak the mics over the rims, aimed at where the stick will hit.

What's your mic of choice for toms?

I like [Shure SM] 57's on toms; they sound really good. But in the kick drum, I'm not a big fan of the [AKG] D12, and to this

> "EVERY INSTRUMENT HAS GOT ITS RANGE THAT'S BEAUTIFUL, AND THAT'S WHAT YOU WANT TO FIND."

day I'm still putting different mics in there. If I use a D12 or the D112, I find it's hard to get the clarity, so sometimes I'll put in a second mic just for the high end; sometimes that might be an Electro-Voice RE20.

With both mics inside and up close?

Yeah, both inside and up close. I'll put the D12 right at the outside of a hole cut in the front skin. Just at the perimeter to get the maximum low end, to get the waveform a bit. But I'll sneak the RE20

WEBLINK
For the latest on Tony Visconti, visit his Web site, www.tonyvisconti.com.

right in there, a few inches away from the beater. I'll blend the two and I'll send them both to one track. If I've got the sound really great, then I'll commit that to one track.

How about snare miking?

On the top I'll have my latest — a mutated '57 that I took apart — I've got to show it to you! [*Brings out an SM57 that has been cut open and taped together so the capsule is at a 90-degree angle to the body.*] You could sneak this over anything; it sounds really great.

Do you also under-mic the snare?

If I'm not getting brightness, if it's a very dull snare, or something's wrong that day, or it's just not cracking — then I will put an under-mic in. But I will always gate it; I don't like it rattling around all the time.

You gate the under-mic as you're recording?

As I record. I want to clean that bugger up before it gets on tape. It's quite easy to set the threshold, because the only thing that's going on in there is the rattle. Toms don't seem to affect it, kick drum doesn't affect it, so it's quite easy to gate it.

Will the gated under-mic be recorded on a separate track?

I always put it on a separate track. And invariably it's out of phase with the top one. I have never, ever had the good luck of having both mics in phase naturally — so if you do that trick, you must check it.

> “COMPRESSION IS THE SOUND OF ROCK.”

Even though it's being gated, you'll hear a big difference if you just play with the phase button, and you'll find that the low end will disappear if it's out of phase.

What mic will you stick under there?

Whatever I have on top. It doesn't matter.

What do you use for drum overheads?

For overheads I like to use something really hi-fi like an AKG 414. And lately I've been using the [Audio-Technica] 3041 — they're just amazing. They're too fat on the low end, but put in the low-end cut and they work great.

Where do you position them?

I'll just position them until I find a good sound and I'm picking up the ride, with the crashes much louder than the ride cymbal. I'll put the overheads fairly high up, at least a foot over the drummer's head. Sometimes I'll blend the overhead mics with the room mics and record them on the same track. In a small room, the room mics and the overhead mics could very well be one and the same.

Will you filter out the lowest frequencies in the kick drum?

It depends. Again, I'll find the octave of that sound and boost that to make it clear. Often, the bass drum might have a lot of 50 Hz, which you're not going to hear; in fact, it'll wreck most speakers — it'll rip a car speaker apart if you play with that frequency. So I'll find the octave of it — which will be 100 or 200, and then the kick drum is fat and it's clear. Then I'll

▸ continued on page 144

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FINAL CONFERENCE PROGRAM*

Friday

10:00 a.m. Keynote

TRACK A — WHITTIER ROOM

10:30 a.m. – 11:30 a.m.
DVD-Audio: Where Have All The Titles Gone?
Record company executives share their experiences on developing titles and discuss what's in the future for the format.
Moderator: Christopher Walsh, Billboard magazine
Panel: Nick Sahakian DVD-Audio Project Supervisor, Rhino/Warners; Brian Ackley, Director, Production, American Gramophone Records; Mike Hobson, President, Classic Records

11:30 a.m. – 12:30 p.m.
How Big Is The Surround Market?
Panel discussion with representatives from consumer electronics and car audio and retail stores. Plans for marketing surround to consumers will be revealed.
Moderator: Kerry Moyer, CEA

12:30 p.m. – 1:30 p.m.
Networking Lunch

1:30 p.m. – 2:30 p.m.
Residential Surround Environments
Following surround sound from professional production through to design and installation. How does surround content translate as it makes its way from the professional who creates the product through the system until it reaches the consumer? This CEDIA-sponsored panel addresses the pitfalls and challenges in bringing a surround project to the ears of consumers at home.
Moderator: Frederick Ampel, Technology Vision

TRACK B — RODEO ROOM

11:00 a.m. – 12:00 p.m.
Multichannel Miking
Bobby Owsinski gives a hands-on demo of some previously recorded surround sound sessions to demonstrate how different mics and miking techniques provide different results with surround sound in mind.
Presenter: Bobby Owsinski, Surround Associates

12:30 p.m. – 1:30 p.m.
Networking Lunch

1:30 p.m. – 2:30 p.m.
Watermarking
What it is, why you should use it and how to use it.
Moderator: Ed Outwater, Industry Consultant

2:30 p.m. – 3:30 p.m.
Technical Tools
Are the tools ready for professionals to go to work on new surround titles? This session looks at the state of readiness of mastering and authoring equipment, here and now.
Presenter: Paul West, Universal Music Group

3:30 p.m. – 4:30 p.m.
Surround Workstation Tutorial
An insider's view on the technologies, including innovative hardware and software products being made available for professionals looking for surround sound tools.
Presenter: David Frangioni, Audio One

Conference Chairman


Tim Wetmore
United Entertainment Media, Inc.


Ed Outwater
Industry Consultant


Garth Hemphill
American Conservatory Theater


David Navone
Car Sound & Performance


John Kellogg
Dolby


Vincent van Haaff
Waterland Design


Rich Tozzoli
Surround Professional


Al Schmitt
Producer

(Track A Continued)

2:30 p.m. – 3:30 p.m.
The Game Experience
The age of surround gaming is quickly approaching as Playstation 2 continues its success and Microsoft readies the X-Box for release. This panel of game developer experts reveals the strengths of individual gaming platforms for the coming year.
Moderator: Rob Hubbard, Electronic Arts
Panel: Brian Schmidt, Audio, Microsoft XBox Team; Craig Duman, Sound Designer, Interplay

3:30 p.m. – 4:30 p.m.
Just What Is MLP Anyway?
DVD-Audio features new technologies and benefits for music lovers. One of the key technologies which makes the benefits of high resolution 96/24 multichannel PCM streams possible is MLP (Meridian Lossless Packing). Dolby Laboratories' G.M. of Multichannel Audio & Music Production, John Kellogg, will explain the history, functions, features and benefits of MLP, its role in the DVD-Audio format and how Meridian UK, the inventor of MLP, and Dolby Laboratories, work together to integrate MLP encoding and decoding into DVD-Audio.
Presenter: John Kellogg, Dolby Laboratories

4:30 p.m. – 5:30 p.m.
What Do I Do With The Center?
Perhaps more than any other single element of surround, the pioneers of this technology have argued over how to handle the center channel of a mix. This stellar panel of award winning mixers and producers reveals the secret to handling the center.
Moderator: Murray Allen, Electronic Arts Vice President, Post Production, Testing & Technical Support
Panel: Al Schmitt, Producer/Engineer; John Loose, Dolby; Dennis Sands, Film Maker; Ed Cherney, Producer/Engineer; Shawn Murphy, Film Sound Mixer

Saturday

10:00 a.m. Keynote

TRACK A — WHITTIER ROOM

10:30 a.m. – 11:30 a.m.
Film Surround
Mixing for 6.1, 7.1 etc. The subtleties of mixing a seven or eight channel surround program for film is distinctly different from any other endeavor, not to mention these guys have been doing surround mixing longer than anybody else. Some of the world's great film sound mixers reveal their secrets.

11:30 a.m. – 12:30 p.m.
Detroit Driving Surround
Major auto manufacturers are expected to drive growth in surround sound. This distinguished panel of Detroit's major players, as well as the world's most important aftermarket heavies, discuss how they plan to roll out surround to American car owners.
Moderator: David Navone, Technical Director, Car Sound & Performance Magazine
Panel: Jim Fosgate, Automotive Sound Designer

12:30 p.m. – 1:30 p.m.
Networking Lunch

1:30 p.m. – 2:30 p.m.
Technical Challenges For Residential Surround
As the home theater market continues its growth curve, what are the technical challenges that are cropping up for the application of surround and how should they be addressed? CEDIA-sponsored panel.
Moderator: Michael Heiss, M. Heiss Consulting

2:30 p.m. – 3:30 p.m.
Bass Management
Technology visionary Tomlinson Holman peels away the layers of mystery surrounding this arcane art and uncovers the nuances of bass management. It can determine the success or failure of a music project.
Presenter: Tomlinson Holman, TMH Labs

3:30 p.m. – 4:30 p.m.
Surround In Sports
One of the biggest engines driving the entertainment dollar, from television to movies to games, is the world of sports. It has also been at the forefront of experimentation in high definition and surround broadcasting. This session looks at how surround is gaining ground in this arena.

4:30 p.m. – 5:30 p.m.
The Recording Academy Sessions
Creators of the Grammys, the Recording Academy's Producers and Engineers Wing, will offer expert sessions on actually recording and mixing music in surround. This hands-on session is a must see event!
Presenter: Joe Chiccarelli, Recording Academy, Producers & Engineers Wing

*program and times are subject to change.

TRACK B — RODEO ROOM

11:00 a.m. – 12:00 p.m.
Recording Studio Design For The Future
Surround recording is triggering a renaissance in recording studio design and even a rethinking of the recording space itself. As the live studio environment experiences an overhaul, this panel discussion will redefine the demand put upon the modern recording space.
Moderator: Vincent van Haaff, Waterland Design
Panel: Chris Pelonis, Designer; John Storyk, Walters-Storyk Design

12:30 p.m. – 1:30 p.m.
Networking Lunch

1:30 p.m. – 2:30 p.m.
Speaker Setup And Calibration
The final word on this topic. After much confusion and disagreement over standards, this workshop reveals the do's and don'ts of setting up a surround speaker configuration.
Presenter: Bobby Owsinski, Surround Associates

2:30 p.m. – 3:30 p.m.
The Live Experience
Live Sound Goes Surround. Surround sound is not just for the movies and music studios. Live shows are embracing the technology as well. This case study unveils how the experts who set up live sound surround systems apply their knowledge.
Moderator: Richard Zvonar, Level Control Systems
Panel: Garth Hemphill, Resident Sound Designer of American Conservatory Theater (ACT) San Francisco; Naut Humon, Director of Sound Traffic Control, San Francisco; Christian Hugener, Thomas Gregor Associates; Peter Otto, Software Developer, U.C. San Diego

FEATURING A WHO'S WHO LIST OF INDUSTRY EXPERTS

Brian Ackley, *American Gramophone Records*
Murray Allen, *Electronic Arts*
Frederick Ampel, *Technology Vision*
Ed Cherney, *Producer/Engineer*
Joe Chiccarelli, *Recording Academy, Producers & Engineers Wing*
Craig Duman, *Interplay*
Jim Fosgate, *Automotive Sound Designer*
David Frangioni, *Audio One*
Michael Heiss, *M. Heiss Consulting*
Garth Hemphill, *American Conservatory Theater (ACT) San Francisco*
Mike Hobson, *Classic Records*
Tomlinson Holman, *TMH Labs*
Rob Hubbard, *Electronic Arts*
Christian Hugener, *Thomas Gregor Associates*
Naut Humon, *Sound Traffic Control, San Francisco*
John Kellogg, *Dolby Laboratories*
John Loose, *Dolby Laboratories*
Kerry Moyer, *CEA*
Shawn Murphy, *Film Sound Mixer*
David Navone, *Car Sound & Performance Magazine*
Peter Otto, *UC San Diego*
Ed Outwater, *Industry Consultant*
Bobby Owsinski, *Surround Associates*
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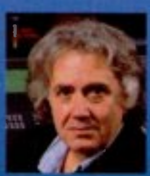
Dennis Sands, *Film Mixer*
Brian Schmidt, *Microsoft XBox Team*
Al Schmitt, *Producer/Engineer*
John Storyk, *Walters-Storyk Design*
Rich Tozzoli, *Producer/Engineer*
Vincent van Haaff, *Waterland Design*
Paul West, *Universal Music Group*
Richard Zvonar, *Level Control Systems*
and more to be announced!


Michael Heiss
Heiss Consulting


Robert Margouleff
Record Producer


Ed Cherney
Producer/Engineer


David Frangioni
Audio One


John Loose
Dolby


John Storyk
Walters-Storyk Design


Tomlinson Holman
TMH Labs


Richard Zvonar
Level Control Systems

Surround 2001 Featured Tutorials

Studer Professional Audio, Brighton Room

Friday and Saturday 10:00 a.m. – 6:00 p.m.

Studer will have a full working 5.1 mix environment featuring their new D950 M2 Digital Mixing System. Attendees are invited to join the pros from Studer for presentations, as well as hands-on time. Studer will features its exclusive Virtual Surround Panning (VSP) which provides full room simulations directly from within the console, including early reflections, as well as reverberation. Multiple acoustic scenes can be set up, allowing creative options, and speed of operation unavailable in any other large format console.

Digidesign, Wilshire Room

Friday and Saturday, 1:30 p.m.

Pro Tools 5.1 Surround Mixing!

Check out the powerful new surround mixing, panning and processing features of Pro Tools 5.1 software. Witness the flexibility of Pro Tools signal routing. See a mixdown to multiple surround formats simultaneously. Watch integrated surround panning and multichannel plug-ins in action.

Friday, 2:30 p.m., Saturday, 3:30 p.m.

Surround Techniques in Pro Tools

Instructor: Rich Tozzoli, R. Austin Productions, New York Rich Tozzoli, composer, sound designer and surround engineer (Al DiMeola, Joni Mitchell, Emerson Lake and Palmer, Foghat, Vernon Reid) demonstrates Pro Tools surround mixing techniques. Topics include surround recording techniques, repurposing recordings for surround, bass management, mixdown, software encoding.

Friday and Saturday 10:00 a.m. – 6:00 p.m.

Pro Tools 5.1 and Plug-Ins from Kind of Loud

In addition to the above demos, you'll be able to see Pro Tools 5.1 throughout the show. We are also showing three surround plug-ins from Kind Of Loud Technologies: RealVerb 5.1, the first multichannel reverb plug-in for Pro Tools, Tweetie, a monitoring and calibration plug-in, and Woofie, a bass management plug-in.

DENON, Roxbury Room

Friday and Saturday 10:00 a.m. – 6:00 p.m.

Presenters: David Birch-Jones, Marketing Manager, DENON; Jeff Talmadge, National Training Manager, DENON

DENON, the leader in digital home theater audio, introduces the AVR-5800 — the world's first A/V component that features THX Surround EX, DTS Extended Surround Matrix 6.1, and DTS Extended Surround Discrete 6.1 decoding. Continuous demonstrations (at least 5 each day) include consumer surround decoding format explanations, along with demonstrations using the latest music and movie surround clips, including DTS ES Discrete 6.1 surround sound clips.

DTS, Santa Monica Room

Friday and Saturday 10:00 a.m. – 6:00 p.m.

Some of the world's leading surround sound engineers/producers will be demonstrating their favorite 5.1 surround mixes in the DTS demo room. Mixes will be demonstrated by Chuck Ainlay, David Tickle, Alan Parsons, Robert Margouleff and Tommy Tallarico. Additional presenters will also be on hand.

Dolby, Palm Room

Friday, 11:00 a.m. and 2:00 p.m.

Dolby E and Dolby Digital in Broadcast Production

Instructors: Steve Venezia/Gary Epstein, Dolby Laboratories

Dolby E is Dolby's new professional multichannel distribution coding system which is essential for 5.1 broadcast production for Digital TV and HDTV. This seminar will show the functions and benefits of Dolby E for multichannel soundtracks for television production and show how Dolby E works with Dolby Digital to create a complete multichannel solution for broadcasters. Also included is a demonstration of the Dolby DP571 encoder, DP572 decoder and the new DP570 multichannel audio tool.

Saturday, 11:00 a.m. and 2:00 p.m.

MLP and Dolby Digital Mastering for DVD

Instructors: John Kellogg/Gene Radzik, Dolby Laboratories

DVD-Audio production is ramping up as more music titles get into the pipeline. MLP (Meridian Lossless Packing) and Dolby Digital are two essential technologies used in the creation of a DVD-Audio music title. This seminar will demonstrate MLP encoding and show the benefits MLP offers for high resolution multichannel music PCM streams of any kind. Learn first hand how to create both MLP and Dolby Digital audio streams ready for authoring. This session is essential for anyone involved in producing or mastering DVD-Audio titles.

AMEK, El Camino Room

Friday and Saturday 10:00 a.m. – 6:00 p.m.

AMEK will be demonstrating the new Media 51 Recording and Post Production Console throughout the show. Stop by their demo room for a hands-on demonstration. Hear for yourself the power and versatility of an analog console featuring Mr. Rupert Neve's pre-amps and EQ's within a fully equipped surround mixing desk. Personally scheduled appointments are available.

Sony, Oakhurst Room

Friday and Saturday 10:00 a.m. – 6:00 p.m.

Super Audio CD — Multichannel Sessions

Sony and Philips will provide an update on the status of Super Audio CD (SACD) as it enters its second year in the market, including ongoing demonstrations of the latest multichannel hardware and software. In addition, key producers and recording engineers will share their experiences in working with this exciting new format.

Conference Registration Form

surround 2001

(Copy this form for multiple registrants)

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- Register online: www.surroundpro.com
- Register by phone: 516-944-5940
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Registration Fees:

- ☐ Two-day conference pass: $495
- ☐ 3-5 registrants: $450 each; 5-10 registrants: $425 each
- ☐ 10+ registrants: $375 each; Limited one-day passes: $300 each
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HOTEL INFORMATION: A limited number of discount rooms are available for attendees. Contact the Beverly Hilton Hotel directly at 310-274-7777 and advise them that you will be attending the Surround 2001 Conference.

PLEASE NOTE: Payment must be received before conference date to qualify for discounted rate. All on-site registrants will pay $550. Surround 2001 must receive notification of any cancellations. Cancellations received before Nov. 22, 2000 are subject to a 50% cancellation fee. (No cancellations after that date.) Substitutions can be made at any time.

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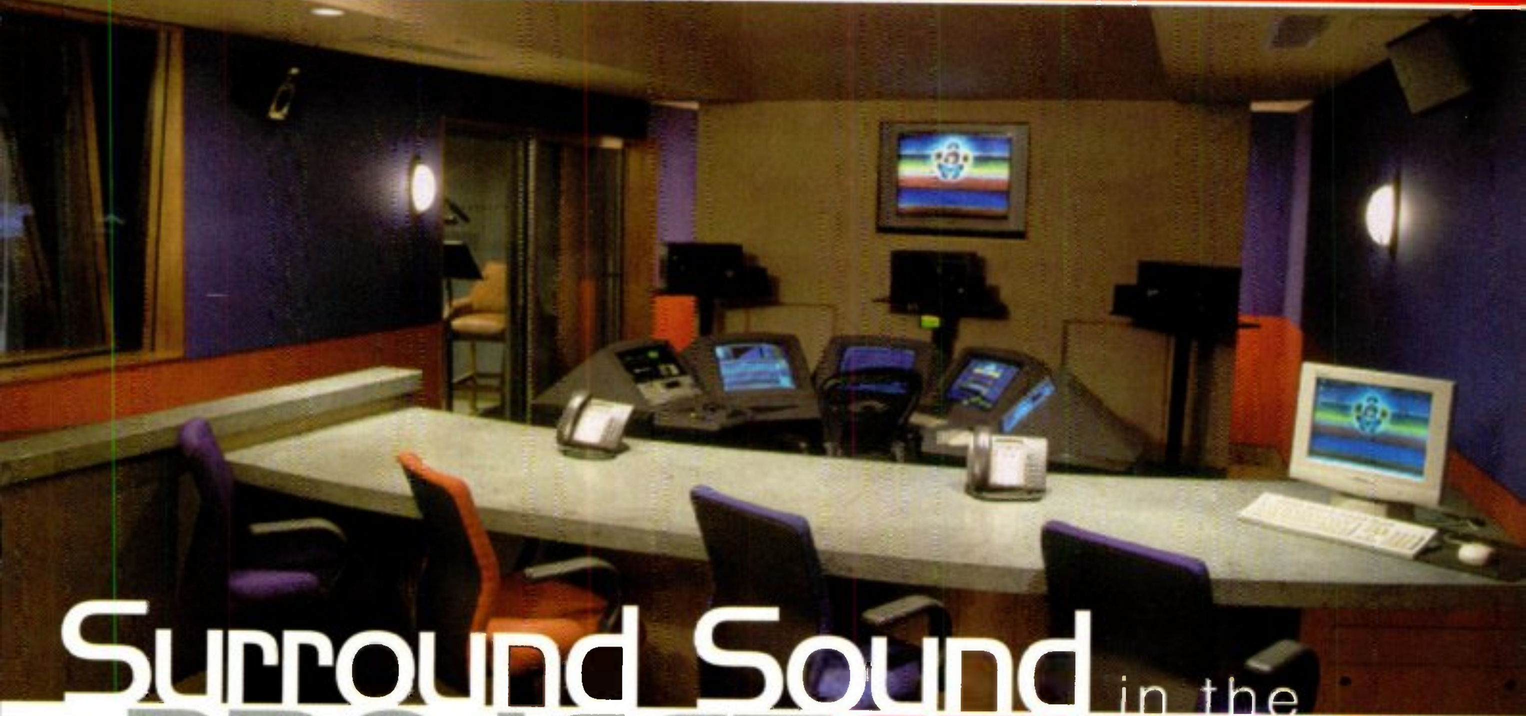
MORE ON HOW TO BRING YOUR PROJECT STUDIO INTO THE SURROUND SOUND ERA



Surround Sound in the PROJECT STUDIO

Part 2

BY HOWARD MASSEY

In last month's installment of this surround sound primer, we discussed basic gear requirements. This month, we get technical, as we cover the topics of equipment interconnection, bass management, speaker placement and calibration, downmixing, and encode monitoring. Be sure to stay tuned to this space for next month's wrap-up, as we dissect the makings of a great surround sound mix.

THE SETUP

The interconnections for a surround sound rig aren't particularly complicated if you keep in mind the main goal — to route signal freely to any or all of the six channels and to route each of the six channels to its own dedicated track and loudspeaker. There is no set bussing scheme, but the Dolby Digital standard (also the SMPTE standard) is:

Bus 1 = left front
Bus 2 = right front
Bus 3 = center
Bus 4 = subwoofer (in film surround mixing, this is called the "LFE" channel, for "Low Frequency Effects")
Bus 5 = left rear
Bus 6 = right rear

Just to make things complicated, the DTS bussing scheme (also used by the Yamaha 02R) is different, as follows:

Bus 1 = left front
Bus 2 = right front
Bus 3 = left rear
Bus 4 = right rear
Bus 5 = center
Bus 6 = subwoofer

It really doesn't matter which of these two bus assignments you use — in fact, you can even make up one of your own — as long as you stick to it. Just be consistent and carefully label the resultant mixes so that the mastering engineer knows which track is meant to be routed to which channel.

Once you've decided on your layout, the idea is to send line-level signal from each of the six busses to both the associated track on your mixdown recorder and to the power amplifier driving the equivalent speaker (or powered speaker). For example, the Bus 1 Out of your mixer needs to be connected to both the Track 1 In of your MDM or hard-disk recorder (whatever you're using to hold the final surround mix) and to the left front speaker.

There are several ways to accomplish this dual

routing. One way is to use Y-cables, signal splitters, or patchbay mults to send the signal from each bus to both the recorder and power amplifier. Alternatively, you can connect the bus solely to the recorder and then route the recorder's outputs to the appropriate speakers (*i.e.*, Bus 1 Out to Track 1 In and Track 1 Out to the input of the first power amplifier). With the latter approach, you'll have to set the recorder to monitor the input signal arriving at each of the six tracks while you're mixing.

An even better solution is provided by digital consoles that have a provision for expansion cards. These cards allow a single bus out signal to be routed to multiple destinations. In my rig, the digital multitrack routing is as follows: I made two-way lightpipe connections between the ADAT cards in my 02R and my ADATs. This allows signal from the 24 tracks to be input to the 02R, and at the same time allows my surround mix to be sent back to the first six tracks of each machine. I generally use two 16-bit ADAT-XTs for source material and a newer ADAT-XT20 to store a 20-bit surround mix, being careful to set the word length for busses 1–6 to 20 bits at mixdown time.

BASS MANAGEMENT

The line-level analog routing is slightly more complicated, because of an important principle called *bass management* (sometimes called "bass redirection"). Yes, I could simply connect the bus 1–6 outs from the D/A card in my 02R to the five main nearfield monitors and a subwoofer, but that would mean that the subwoofer was only receiving signal from the LFE channel, and that's not a great solution for two reasons: one, since nearfield monitors typically only have limited bass response, you won't be able to hear any subsonic anomalies that may exist in those five tracks; and, two, that's not the way most end users will listen to surround sound (most consumer receivers take all low-end signal and send it to the subwoofer). Your job as engineer is to make sure that what you're delivering is technically correct, and that means that you need to be able to hear the full frequency range of all channels. The solution is to route low-frequency signal from all five channels as well as the LFE channel signal to your subwoofer(s), and that's what the term "bass management" means.

A bass management system uses filters to ensure that frequencies below a crossover point (typically 120 Hz or 80 Hz — the Dolby Digital and DTS standards, respectively) are combined with the LFE channel input and routed to the subwoofer output, while frequencies above the crossover point are routed to the "full-range" speakers. There are standalone "bass manager" boxes that do this, but it can also sometimes be accomplished within the mixer itself by circuitry inside the subwoofer, or by an outboard signal processor or software plug-in (*i.e.*, Kind Of Loud's Woofie). Both the Genelec subwoofers in my rig have this capability. I connect the front left, front right, center, and LFE bus output channels to a 1092 subwoofer (which is physically positioned behind and below the

> **"ONE OF THE MOST IMPORTANT FACTORS IN CALIBRATING THE SUBWOOFER LEVEL IS ITS ACOUSTIC PHASE ALIGNMENT."**

center speaker), and then connect the 1092's front left, front right, and center channel outputs to the corresponding 1029A's. The bus outputs from the two rear channels are connected to a 1091 sub (which is positioned between the rear speakers) and its left and right outputs are then connected to the two rear 1029A's. With these connections made, signal from the five main channels goes to a subwoofer (along with the LFE channel) while signal routed to bus 1 is sent to both track 1 of the ADATs (in the digital domain) as well as to the left front speaker (in the analog domain); signal routed to bus 2 simultaneously goes to track 2 and to the right front speaker, and so on. Trust me, it's much harder to describe it than it is to do it!

THE SURROUND MONITORING BUGABOO

This makes, however, for an interesting dilemma: how to set the recording levels and monitoring levels independently. Most mixers — even those touted as supporting surround mixing — don't have surround *monitoring* capability. To do so requires a dedicated six-way potentiometer, so that the level of all six channels can be adjusted simultaneously and separately from the bus send levels. There are a number of

Part 2

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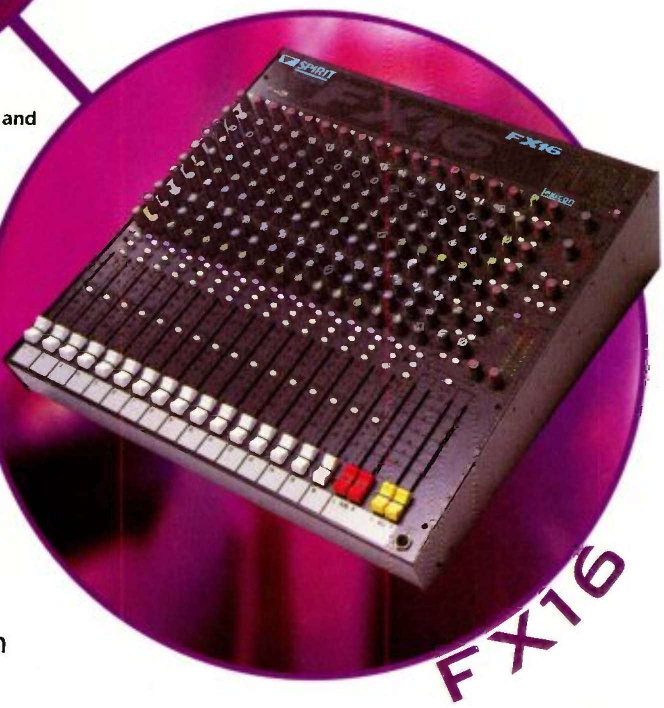

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third-party surround monitoring controllers available from manufacturers such as MartinSound, Audient, Otari, AudioTechnologies, and Baldwin Products, but these are aimed primarily at the high-end market and carry a hefty price tag.

There are workaround solutions to this problem, although they may not be particularly elegant. On an analog eight-bus console, for example, you could use the eight subgroups to feed the monitors — but you'll have to be very careful to balance each output's level accurately. If you're using a computer-based DAW with an audio interface that features multiple outputs that you can access independently, you could use its outputs to feed the speakers directly; you'll have to control volume from within your software as opposed to with a physical knob.

On a digital console such as an 02R, you could use your external aux sends. To set independent monitoring levels assign the D/A card outputs 1–6 to aux sends 1–6 (instead of the default assignment of busses 1–6). Next, assign the tape returns from the target recorder (in my case, an ADAT-XT20) to the aux sends *instead* of the stereo or bus outs — track 1 to aux 1, track 2 to aux 2, etc. Finally, group all six faders together for master volume control, and you're in business.

Other systems may provide different routing possibilities; take a creative look at your rig. If all else fails, you can simply use the volume controls on your power amps or self-powered speakers to set monitoring levels.

SPEAKER PLACEMENT AND CALIBRATION

Needless to say, speaker placement for surround monitoring is important, though perhaps not quite as critical as some audiophiles would have you believe. Let's deal with the five "main" (*i.e.*, full-range or non-subwoofer) speakers first. The front three speakers should be placed in an arc, with the left and right speakers toed in at a 30–35° angle (finally, a use for that $1.98 plastic protractor you were forced to buy in high school!). If possible, the rear speakers should be angled in as well. The accepted practice is to set them at about 110° relative to the front center speaker, but some surround mixers prefer them at a steeper angle for improved phantom imaging. Nashville mix engineer Chuck Ainlay, for example, favors an angle of 145°, stating that "even at 120°, there's no continuity if you're trying to create a phantom image between the left rear and the right rear."

> **WEBLINK**
> Web-surfers can find extensive information on surround sound at www.dolby.com/digital/, www.dtsonline.com, www.tmhlabs.com, www.surroundassociates.com, and www.RobertMargouleff.com.

Ideally, you want all five speakers to be at the same height (ear level is best), though, again, this is largely a matter of taste. "I like to have the tweeter blowing over my head a little bit," Ainlay comments, "and I don't point the speakers dead center, either. If I'm setting up speakers for stereo, they're actually aimed behind and above my head somewhat, to open up the listening position and not make it quite so pinpointed, and also to account for my ears not being dead center in my head. So I do the same thing for the rear speakers — I point them

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somewhat in front of and above my head."

Each of the five speakers should be at an equal distance from your normal mixing position, though, if this isn't possible, you can compensate by making distant speakers slightly louder (by turning up their power amp) and/or by slightly delaying the signal arriving at closer speakers. The basic rule of thumb for this delay is about one millisecond — that's one thousandth of a second — per foot. This means that if your rear speakers are six feet away and your front speakers are four feet away, you should delay the signal to the three front speakers by about two milliseconds. This is easily accomplished in most digital mixers or MDMs, where individual channels or tracks can be delayed precisely.

In practice, I've found that slight distance variations like this are virtually imperceptible. In general, as long as all five main speakers are close to equidistant from the mix position, you're not going to have any problems. The vagaries of my project studio dictate that my rear speakers are about three feet further away than my front speakers and slightly off-center as well. By simply making them a little bit louder than the front speakers, I'm able to achieve a good listening balance without having to delay the front speaker signals at all.

You can set the relative levels of the five main speakers with a reasonable degree of accuracy with the use of an inexpensive Radio Shack SPL meter. Simply position it at the mix position, set it to C-weighting (slow response), and feed a noise source into one speaker at a time, setting its power amp so that the SPL meter reads the same level (85 dB SPL is the film standard listening level) for each.

FUN AND GAMES WITH SUBWOOFERS

Subwoofer placement and level calibration is a bit trickier. As Bob Margouleff observes, "The low-frequency information needs to fill the room. It's like liquid; it fills the space and oozes everywhere, so you want to make sure the subwoofer is placed where the room itself helps support the resonant quality of the low end." Margouleff has devised a unique technique for doing so. "I start by identifying the sweetest spot in the room. Then I take the subwoofer and I put it right in the middle of the sweet spot. I put pink noise or an 80-cycle sine wave through it, and then I walk around the edge of the room until I hear where the subwoofer sounds loudest. That's the best spot for the sub."

If you're using two subwoofers, Margouleff cautions against facing them toward each other (a "sure way" to induce phase cancellation problems, he tells us). "In a modest space, you can place two subs right next to each other; if you're in a bigger space, they can be on different walls at the edges of the room. You want to get the bass energy as dispersed as possible, but I don't believe in firing them into the wall or floor, since, in pop music, they're providing considerable low-frequency musical information and not just rumble."

One of the most important factors in calibrating the subwoofer level is its acoustic phase alignment at the crossover point; incorrect alignment can cause a drop in the frequency response of the whole system at the crossover point. Some subwoofers offer a built-in phase matching control that allows adjustment in precise steps. If yours does, feed in a sine wave at the crossover frequency and adjust it to the point where the sound is loudest at the mix position. If your subwoofer does not have such a control, try repositioning the speaker until the signal is loudest at the mix position. While there is no industry standard as yet, experts such as Tomlinson Holman recommend that the subwoofer provide 10 dB more headroom than the main speakers. This can be accomplished by setting its level so that it delivers approximately 4 dB SPL above the chosen reference level when playing back band-limited pink noise (low-pass filtered at 120 Hz). If you calibrate your full-range speakers to deliver 85 dB SPL at the mix position, the subwoofer should be set to deliver 89 dB SPL.

A number of surround sound speaker calibration tools are starting to find their way to market, including TMH Corporation's DTRS-format multichannel test tapes and Kind Of Loud's Tweetie Pro Tools plug-in.

DOWNMIXING AND ENCODE MONITORING

Downmixing is one of the most dreaded words to the discriminating surround sound engineer. It describes a process by which the 5.1 mix is automatically reduced ("folded down") to stereo or even mono under the control of unseeing, unfeeling circuitry that has no sense of aesthetics or taste. Truth be told, no professional really trusts downmixing. Despite the fact that the DVD-Audio spec provides for something called SMART (System Managed Audio Resource Technique) Content,

▶ continued on page 142

WHAT IS HOT?

THE "WHAT IS HOT" PROJECT SEEKS TO DETERMINE HOW LOUD IS TOO LOUD

"THERE IS A MYTH IN THE RECORD INDUSTRY THAT APPLYING 'RADIO-STYLE' PROCESSING TO CDS IN MASTERING WILL CAUSE THEM TO BE LOUDER OR WILL REDUCE THE AUDIBLE EFFECTS OF ON-AIR PROCESSING. IN FACT, THE OPPOSITE IS TRUE: THESE CDS WILL NOT BE LOUDER ON AIR, BUT THEY WILL BE AUDIBLY DISTORTED AND UNPLEASANT TO LISTEN TO, LACKING PUNCH AND CLARITY. WE HOPE THAT THE RECORD INDUSTRY WILL COME TO ITS SENSES WHEN IT HEARS THE CONSEQUENCES OF THESE PRACTICES ON THE AIR."

—FROM THE MANUAL OF THE ORBAN OPTIMOD-FM 8400 BROADCAST AUDIO PROCESSOR.

BY ALLEN WHITMAN

In June of 2000, a San Francisco-based engineer and member of the online Mastering Web Board (MWB) community seized upon the idea of comparing various masterings of the same mix as a way of answering the question: "How loud is too loud?" The result is the "What Is Hot?" CD. Tardon Feathered (he swears it's his real name), owner of Mr. Toad's Recording, had been mastering since 1995. He felt he had the tools to master a CD to just about any volume, but wanted to know, empirically, at what level he *should* master. Whenever the question of overall level was discussed online, it became clear that it was impossible to directly translate the instructions to the music. Since many members of the MWB gave him slightly different answers, Feathered decided it was time to apply some practical science to the opinions.

Mr. Feathered suggested to members of the MWB that they approach a definitive answer through a comparison: Have each engineer master the same song and then examine the results. It was such a simple, elegant idea that you could almost hear a collective online "Doh!" Twenty mastering engineers from around the globe (four continents and three islands) agreed to participate anonymously in the wholly independent, non-corporate-allied "What Is Hot?" project. Feathered chose a 24-bit mix of modern rock music, burned it to twenty CDs, and mailed them to the participants. When the masters were returned to him in San Francisco, they were laid out in order of volume, quietest to loudest, compiled onto more CDs, and mailed back to the original participants (and any other interested MWB members) for comparison and analysis.

By the end of summer, the discs had been analyzed and discussed, and the conclusions turned out to be remarkably consistent. Engineers who had submitted hotter masters were able to appreciate the dynamic and tonal qualities of versions that didn't compromise dynamics for the sake of volume. Feathered admits he fell into this category. Like so many other independent studio owners looking to keep his clients in this age of DIY boxes, he had felt pressure to make masters with an "extra bit" of gain, thus assuring clients wouldn't feel that volume was an issue when comparing to current mega-compressed major-label releases. Armed with proof drawn from this project, Tardon says he feels "considerably more at ease explaining to a client why a master shouldn't be any louder than necessary to translate well to all mediums, especially radio."

Digital Domain's Bob Katz (**www.digido.com**) describes his take on the results: "I learned that my mastering approach falls in the middle between the hottest/most squashed and the lowest 'unpunchy' CDs — exactly where I want to be. It strongly reinforces my point that standardized monitoring levels produce good-sounding, uniform results."

Tardon Feathered: The man behind the mastering comparison compilation CD.

Each master on the "What Is Hot?" project CD received constructive comments. Mr. Katz mentions he received "valuable feedback from other mastering engineers on the tonal balance of my master. Every time I get feedback from professionals on my work, it helps me to do even better. I look for criticism, I thrive on it."

ONE THREAD LEADS TO ANOTHER...

Typical of research projects with a simple stated goal but a large number of variables, the tightly focused CD volume issue evolved into a much wider discussion of subtle aesthetic techniques. Professional mastering engineers with decades of experience working in high-profile, well-equipped mastering houses shared thoughts and discussion with students using simple DAW-based setups. Enthusiastic conversations about proprietary processors, new techniques, and some rather radical sonic approaches (one contributor nearly made the mix monophonic) grew in many different directions. Yves Zimelman, a Brazilian mastering engineer, said: "This single experiment will advance me several years in the art of premastering."

The "What Is Hot?" project was taken one step further by Robert Orban (designer of the industry-leading Orban FM broadcast audio processors; **www.orban.com**). Mr. Orban, upon receiving a copy of the project, ran the CD through his company's Optimod-FM 8400 processor to illustrate how each master would sound when broadcast over the airwaves. The results were clear: Only the moderately volumed masters retained any sense of clarity, punch, and definition. The louder masters all suffered serious dynamic inversion (where severely limited, emphasized, and compressed audio is inverted so that loud parts are soft and soft parts are loud) as well as offensive clipping artifacts.

TOO HOT?

Because of the subjective nature of sound, it's impossible to draw absolutes from any piece of audio — everyone has

WHAT IS HOT?

WEBLINK

If you're interested in evaluating "What Is Hot?" for yourself, go to http://webbd.nls.net/~mastering, enter as a guest, and select the conference called "Red Hot/Chilly Papas." Look for a topic entitled "Welcome *EQ* Readers." It will provide an overview of the MWB with links to various discussions that grew from the project. It will also detail information on obtaining your own copy of the "What Is Hot?" project CD.

slightly different ears Even seemingly obvious factors such as clarity and definition can be difficult to demonstrate. Still, the "What Is Hot?" project achieved its stated goal and now seeks to offer the results to an industry that could obviously benefit from them. Says Tardon Feathered: "It was good to see this project be of interest to so many people. Discussions of many different subjects related to the masters on the disc…were beyond my original conception of the project. I got into this business for the love of music, and this information will help me continue to work with the best interests of the music in mind."

The quest to understand exactly how audio should be mastered has led to many impassioned conversations on the MWB. As far as volume goes, based on the results of the "What Is Hot?" project, the answer seems obvious: Just because the tools are available to make a CD loud doesn't mean that a CD should be made that loud. The industry-borne myth that "hotter is better" leads to a domino effect of other problems, including those resulting from the application of FM processing at the radio station and listeners' experiences with the resulting compression, limiting, and inevitable clipping. It's clear that, as far as mastering goes, too loud is just that: too loud. ■

AN ONLINE COMMUNITY

The "What Is Hot" project evolved in the online community known as the Mastering Web Board (**http://webbd.nls.net/~mastering**), an informal gathering of mastering engineers discussing issues from technical to aesthetic.

The Mastering Web Board (MWB) began its existence as the e-mail-based Mastering Mail List hosted by Clete Baker's Studio B in Omaha, Nebraska. Now hosted on the Internet by Glenn Meadows of Masterfonics (**www.masterfonics.com**) in Nashville, TN, the MWB's contributors share useful and relevant information ranging from classic techniques to this morning's software bug. MWB contributor Bob Katz says: "The dialog between professional mastering engineers, musicians, and recording engineers on the Web board is absolutely essential to anyone wanting to keep up with events and technology."

The only professed goal of the group is "Education, Education, Education," and with hundreds of individual discussion threads, there's hardly a subject that isn't addressed. One such thread was about the effects of FM radio processing on current mega-loud CDs. "We recently have had legendary design engineer Bob Orban become an active participant," points out Mr. Meadows. "His posts have been very enlightening in regard to how sophisticated FM radio processing has become. I've used that information to start 'enlightening' my clients as to why they don't need to overly compress/limit/EQ their masters."

Now Orville Has A Family

Introducing Eve/Net -

The Eve/Net™ Network Remote Control System provides flexible, total remote control for the Orville™ Harmonizer® processor family - without the overcomplications and high costs of other multi-channel effects processor controllers. And perhaps best of all, there's no steep learning curve. Any Eventide Orville or DSP-series user will be instantly comfortable with the Eve/Net remote. The Eve/Net system links one or more Eve/Net remote controllers with multiple Orville or DSP7000/7500 processors in any combination.

Introducing Orville/R -

Face it. If there's an Eve/Net remote sitting conveniently on your console, or if perhaps you've installed multiple Orville processors in a central machine room, you may never use the Orville's own front panel controls. So why pay for them? Eventide's new "no faceplate controls" Orville/R model has all the capabilities of the standard Orville, but is designed for use exclusively with Eve/Net. Remember, Eve/Net can control up multiple Orville units, so you can mix and match standard Orville and Orville/R processors. You can pay many $1000s more for a competitor's multi-channel processor system and still not equal the versatility, quality and ease-of-use of Eventide's Orville system.

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Eventide hasn't forgotten that when it comes to music, it's still very much a stereo world. We're proud to introduce the DSP7000 Ultra-Harmonizer® Stereo Effects Processor. The DSP7000 is a major upgrade of Eventide's legendary DSP4000-series, featuring up to four times the processing power for higher effects density and superior audio performance. 24-bit conversion at 96kHz of course. The new DSP7000 offers extensive programmability and remote capabilities. It features hundreds of exciting preset programs including your favorites from the DSP4000 line, and it's even easier to use than the DSP4000. The DSP7000 line is fully compatible with the Eve/Net remote system.

Introducing the DSP7500 -

A long-standing Eventide tradition is our "500" model series. The "500" models offer all the standard Ultra-Harmonizer processor features and then some, and are priced to be exceptional values. The new DSP7500 Stereo Ultra-Harmonizer® Effects Processor upholds that tradition beautifully. It's a DSP7000 to-the-max, featuring hundreds of additional presets especially useful in post-production and broadcast applications, plus a 174 second (mono) / 87 second (stereo) sampler with special preset programs which make it one of the most versatile samplers you've ever used. From superb time compression and expansion to auto-replacement of snare and kick drum tracks, you'll never run out of ways to use the extra capabilities of the DSP7500 processor.

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CIRCLE 79 ON FREE INFO CARD

by Mitch Gallagher • mgallagher@uemedia.com

STEINBERG NUE

STEINBERG NUENDO

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

SUMMARY: A multi-faceted powerhouse capable of serving a variety of audio production applications: stereo, surround, video, multimedia, and more.

STRENGTHS: Many productivity-enhancing features. Undo/Redo history. Most edits and processing can be done while audio plays back. Plenty of tracks. Object-oriented design. Surround mixing support. Built-in four-band EQ per mixer channel (with presets and reset). Group channels. Multiple plug-in windows can be opened simultaneously. Excellent automation features. CPU-efficient VST and DirectX plug-in support. Configurable, comprehensive key commands. Configurable VST Master Setups. Zero Latency Monitoring (with compatible hardware). Video track with thumbnails. Easy learning curve. Cubase song import. Good manuals/documentation.

WEAKNESSES: No MIDI notation or drum editing. No surround-capable plug-ins included (except Panner and Pro Logic matrix encode/decode). User interface can get cluttered. Dongle-based copy protection. Can't overlay automation display on waveforms. Cubase song import doesn't include mixer, effects, or automation settings. Undo/Redo history doesn't include mixer moves (except automation edits).

SYSTEM REQUIREMENTS: 266 MHz Pentium II or faster, Windows 98/2000/NT 4, 128 MB or more RAM, compatible audio hardware, "wheel" mouse recommended.

COPY PROTECTION: Hardware dongle (parallel port)

VERSION REVIEWED: 1.02

TEST SYSTEM: 733 MHz Pentium III, 256 MB RAM, internal 40 GB IDE/ATA hard drive, Windows 2000, Nuendo 96/52 PCI audio card, Nuendo 8•I/O eight-channel A/D/A converter, Emagic Unitor 8 mkII MIDI interface, Yamaha 01V as control surface, various audio and MIDI gear.

PRICE: $1,299

EQ FREE LIT. #: 110

STEINBERG'S NUENDO CHALLENGES THE FRONT-RUNNERS IN AUDIO PRODUCTION SOFTWARE

Engineers, producers, and musicians tend to be fiercely loyal to their audio production software — just visit an online audio forum, post a question about the relative merits of two programs, and sit back and watch the flames fly. For even more fun, post a question asking for comparisons of proprietary hardware-based packages versus native programs.

Into this arena bravely steps Steinberg, with a new offering aimed at audio production professionals — a market some would say is already dominated by strong competitors such as Digidesign and few others. The new package, dubbed Nuendo, is native; that is; it runs on a host PC and doesn't require proprietary dedicated DSP or audio I/O hardware.

Steinberg comes to Nuendo with a strong pedigree in audio software; their Cubase VST line of digital audio sequencers has a tremendous following worldwide. Based on new code that incorporates the company's ASIO and VST technologies, Nuendo offers an impressive complement of features: 200 audio tracks, unlimited MIDI tracks, a full-featured mixer, DirectX and VST plug-in support, comprehensive automation, endless editing and processing options, and built-in surround mixing and video support.

But is it good enough to attract users already invested in a competitor's product? Can it pull in new users? Let's take a tour of the program and find out....

NDO

Media Production System

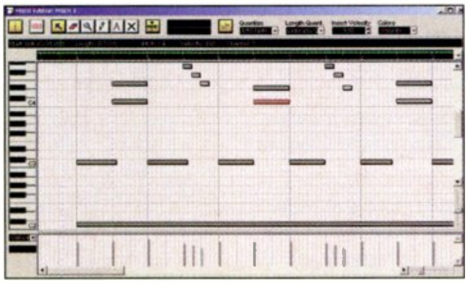


FIGURE 1: DOUBLE-CLICKING A MIDI TRACK OPENS UP NUENDO'S MIDI EDITOR WINDOW. WHILE THE PROGRAM DOESN'T OFFER ANYWHERE NEAR THE MIDI CAPABILITY THAT ITS SIBLING CUBASE DOES, THERE'S ENOUGH POWER FOR MANY PRODUCTION TASKS. BASIC QUANTIZATION IS PROVIDED, WITH SWING. THE STRIP CHART AT THE BOTTOM OF THE WINDOW CAN BE SWITCHED TO DISPLAY MIDI CONTROLLERS, VELOCITY, AND SO ON. USING THE PENCIL OR LINE TOOLS, YOU CAN DRAW IN AND/OR EDIT PITCHBEND, MIDI VOLUME, OR OTHER CONTINUOUS CONTROLLER DATA.

IF YOU HAVE THE INFO LINE TURNED ON (AS IT IS HERE) AND SELECT A NOTE, A NUMBER OF PARAMETERS ARE DISPLAYED AND CAN BE TEXT-EDITED, INCLUDING START TIME, NOTE LENGTH, PITCH, VELOCITY, AND MIDI CHANNEL.

Installation and Set Up

Nuendo installed on my PC without a hitch. The program uses a parallel port dongle for copy protection, which can be a hassle if you work on multiple computers (such as a laptop and a desktop).

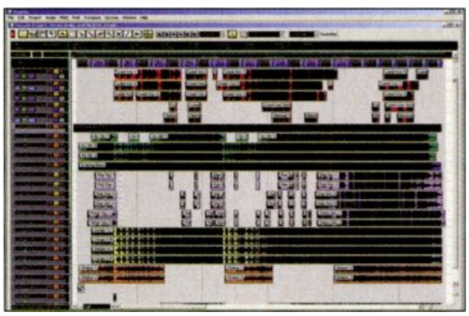
In most cases, you'll be able to dive right in and start working; the program defaults to a usable state. However, you're given plenty of opportunity to customize Nuendo's operation if you so desire. Especially nice are the flexible Master Setups that tell Nuendo about your studio monitors and their placement. These allow for easy switching between stereo and up to eight-channel surround production environments with a simple click of the mouse.

One final setup note: Steinberg recommends a "wheeled" mouse as a nice accessory. I heartily concur; the wheel can be used to quickly edit parameters and to scroll and move about the Project window.

User Interface

If you're familiar with Cubase VST, then Nuendo will be a piece of cake for you to operate. Regardless, Nuendo is very easy to learn and to get around in. I rarely needed the manual, but when I did, the information I needed was there.

But please read the manual: There are just about always several ways to accomplish any given task — including an easy way you can usually find without cracking the manual. But if you *do* dig into the manual, you'll likely find some alternatives (or easier ways to do things), each of which may or may not have specific cool extras associated with it. One example: As you'd expect, you can use the mouse to select a range on a track. But if you position the cursor over the start or end of the range, it turns to a double-headed arrow, which you can use to extend or shorten the selection. Very nice, and more flexible than the old standard "hold Shift, and click to extend the range" standard (which also works, by the way).



THE MAIN WORKSPACE IN NUENDO IS THE PROJECT WINDOW. MUCH OF YOUR TIME WILL BE SPENT HERE EDITING TRACKS AND MAKING CHANGES TO THE SONG ARRANGEMENT. THE PROJECT WINDOW CAN CONTAIN AUDIO, MIDI, GROUP, VIDEO, AND MIXER AND PLUG-IN AUTOMATION TRACKS. EACH ITEM ON A TRACK EXISTS AS AN INDEPENDENT "OBJECT," AND CAN BE EDITED AND PROCESSED SEPARATELY FROM THE OTHER ITEMS IN THE TRACK. YOU CAN ALSO GROUP TOGETHER VARIOUS OBJECTS IN THE TRACK (OR ACROSS MULTIPLE TRACKS) AND TREAT THEM AS A SINGLE OBJECT. MULTIPLE PROJECTS CAN BE OPEN AT ONCE, AND YOU CAN DRAG AND DROP BETWEEN THEM. YOU CAN EVEN HAVE PROJECTS WITH DIFFERENT RESOLUTIONS AND SAMPLE RATES OPEN AT THE SAME TIME.

As with Cubase, Nuendo is very "window-centric" — broken up into lots of windows. You can store and recall screen sets, and the main windows can be opened and closed via key commands, but open windows and clutter can still pile up quickly. In any case, you'll definitely want a large monitor.

Right-mouse click brings up menu of cursor tools (split, erase, select, etc.), as well as a menu of commonly used editing commands that's context-sensitive to whatever window you happen to be working in.

Another nice feature: You can mute individual objects without muting the rest of a track; this is great for trying out different arrangement ideas and for comping tracks.

FIGURE 2: NUENDO'S VST MIXER PROVIDES A CONDENSED VIEW OF ALL THE MIXER CHANNELS, AS WELL AS BUTTONS FOR OPENING THE CHANNEL SETTINGS WINDOW. THE NINE CHANNELS ON THE LEFT ARE SET UP FOR SURROUND PANNING, THE SEVEN ON THE RIGHT ARE CONFIGURED FOR REGULAR STEREO PANNING AND FEED JUST THE FRONT LEFT/RIGHT SPEAKERS IN A 5.1 SURROUND SETUP. ON THE FAR RIGHT ARE THE SIX OUTPUT METERS FOR THE 5.1 SYSTEM, AS WELL AS THE MASTER FADER, WHICH IN THIS CASE CONTROLS ALL SIX OUTPUTS. THE OUTPUT METERS AND MASTER FADER CAN BE HIDDEN IF YOU DESIRE.

NOTE THAT, IN THE MIXER, THE PAN AND LEVEL READOUTS SHARE THE SAME DISPLAY FIELD, WHICH SWITCHES TO THE ACTIVE PARAMETER WHEN YOU MOVE ONE SLIDER OR THE OTHER.

Nicest of all is that you'll rarely need to stop playback. Almost everything in Nuendo can take place in real time, while audio is playing.

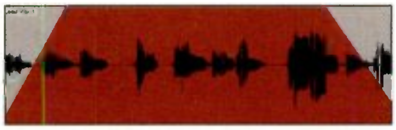
Tracking And Audio

You can get audio into Nuendo in one of two ways. By importing it, in which case a number of file formats are supported, or by recording it through your audio interface.

Recording audio is very easy; enable the appropriate VST input, arm the track, and hit record. Several audio monitoring modes are provided (such as "tape machine style") or you can hit a button to "force" monitoring. If your hardware supports it, Nuendo's ASIO engine provides Zero Latency Hardware Monitoring, which works very well for eliminating latency issues while tracking.

Recorded audio exists in a multi-layered hierarchy: Audio Files are the actual disk data. Audio Clips reference the Audio File, while Audio Events play back the audio clip. This seemingly complex arrangement (it's transparent to the user in practice) allows for a variety of benefits. Because Nuendo is "object-oriented," each clip exists independently and can be edited and processed; similarly, audio events allow for flexible playback and arrangement manipulations.

Audio Events can be combined into Parts, or Events can be grouped — even if they're on separate tracks.



THE BASIC AUDIO BLOCK IS THE CLIP; SIMILAR TO THE "REGIONS" FOUND IN SOME OTHER PROGRAMS. DUE TO NUENDO'S OBJECT-ORIENTED APPROACH, YOU CAN DO A VARIETY OF THINGS TO A CLIP, SUCH AS APPLY A PLUG-IN OR OTHER AUDIO PROCESS DIRECTLY TO IT WITHOUT AFFECTING ANYTHING ELSE ON THE TRACK.

ONE OF THE NICEST THINGS ABOUT NUENDO IS HOW IT HANDLES TRIMMING CLIP START AND END POINTS AND FADES. SIMPLY SELECT THE CLIP; FOUR HANDLES APPEAR IN THE CORNERS OF THE CLIP. THE LOWER TWO CONTROL THE START AND END POINTS; THE UPPER LEFT HANDLE IS FADE IN, UPPER RIGHT IS FADE OUT. EDITING IS AS SIMPLE AS GRABBING A HANDLE WITH THE MOUSE. THIS CAN BE DONE IN REAL TIME, EVEN WHILE AUDIO IS PLAYING.

In this case, changing something on one grouped event changes the rest the same way.

MIDI

Nuendo is primarily an audio production solution, however, there is basic MIDI support. You can record, do basic quantizing, delete doubled notes, manipulate controllers, transpose, change velocity, etc. There's a "piano-roll" style MIDI editing window (see fig. 1), but no drum editor or notation. There is MIDI event list-style editing available in the Project Browser window.

The MIDI features in Nuendo will be enough for many applications, but for serious composition and arranging, you'll want to use a power sequencer — such as Nuendo's sibling Cubase — then import the tracks into your Nuendo project. If you do choose to use Cubase, you can import song files directly into Nuendo. This worked well for me, however, the import process ignores mixer and effects settings as well as any mixer automation data. It would be nice if this data was also included. However, all of the audio and MIDI data in the song will transfer over just fine, which is the main concern.

Mixer

The VST mixer in Nuendo is very similar to the one found in Cubase (see fig. 2). A condensed mixer view contains all the tracks with their faders and pan controls (as in Cubase, I found the pan control hard to grab with the mouse). Clicking a track's edit buttons opens its Channel Settings window (see fig. 3.), which provides more comprehensive control over its four inserts, eight aux sends, and four-band EQ.

FIGURE 3: THERE'S A LOT MORE TO THE CHANNELS IN NUENDO'S MIXER THAN JUST VOLUME AND PAN. ON THE LEFT IS THE "COMMON" PANEL, WHICH PROVIDES ACCESS TO PARAMETERS RELATING TO ALL THE CHANNELS (SUCH AS AUTOMATION, METERING SETUP, ETC.) ALL OF THE SETTINGS FOR A MIXER CHANNEL CAN BE COPIED AND PASTED TO ANOTHER CHANNEL USING THIS SECTION.

THE NEXT PANEL DUPLICATES WHAT IS DISPLAYED WHEN YOU LOOK AT THE VST MIXER. NOTE THAT IN THIS CASE, THE CHANNEL IS CONFIGURED FOR SURROUND PANNING. AS IN THE MIXER VIEW, THE PAN AND LEVEL SETTINGS SHARE THE SAME DISPLAY FIELD, WHICH SWITCHES TO THE ACTIVE PARAMETER WHEN YOU MODIFY ONE SLIDER OR THE OTHER. I'D PREFER A SEPARATE DISPLAY FOR EACH OF THESE PARAMETER VALUES.

EACH CHANNEL HAS FOUR PLUG-IN INSERTS, WHICH CAN BE INDIVIDUALLY OR GLOBALLY BYPASSED. EIGHT AUX SENDS ARE AVAILABLE, WHICH CAN FEED DIRECTLY TO PLUG-INS OR CAN FEED BUSSES, OUTPUTS, OR GROUP CHANNELS. THESE, TOO, CAN BE INDIVIDUALLY OR GLOBALLY BYPASSED.

ON THE RIGHT IS THE CHANNEL'S EQ SECTION; FOUR PARAMETRIC BANDS ARE PROVIDED. YOU CAN ADJUST THE SETTINGS BY TURNING THE KNOBS, INPUTTING TEXT VALUES IN THE FIELDS, OR BY GRABBING POINTS IN THE GRAPH WITH THE MOUSE. EQUALIZER PRESETS CAN BE STORED AND RECALLED.

In addition to audio track channels, Nuendo offers "Group Channels," which function as subgroups to which other channels can be routed (Group Channels can even route to other Group Channels). Since Group Channels offer the same Channel Settings options as regular audio track channels, you can use them to feed multiple plug-ins in series — just insert the desired effects into the Group Channel and route through it.

Channels can be linked, which groups their volume faders. Pan and other controls are not linked. There's no indication which channels are linked together; it would be nice if each group could be colorized, or a "G" appeared on them or *something*. While channels are linked, you can hold the Alt key and make changes to individual faders without affecting the rest in the group.

Nuendo offers full automation support for most parameters in the mixer (as well as in plug-ins that support automation). You can record automation in from a hardware control surface, or access automation subtracks

NUENDO 8 I/O

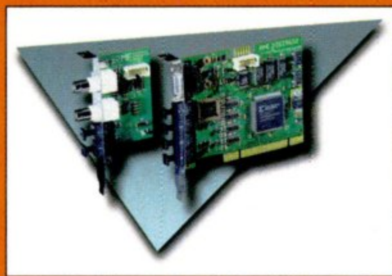
Steinberg's Nuendo 8 I/O is an eight-channel single-rackspace A/D and D/A converter offering 24-bit resolution at 44.1 or 48 kHz sample rates. ADAT lightpipe and TASCAM TDIF (on a 25-pin D-sub connector — you'll need a breakout cable) digital I/O is provided. On the analog side, the Nuendo 8 I/O has eight balanced 1/4-inch inputs and outputs, as well as two 25-pin D-sub connectors. The analog ins can be set to reference +4 dBu, –10 dBV, or "Lo Gain" (no reference spec is given, except for "+19 dBu headroom.") Likewise, the analog outs can reference +4 dBu, –10 dBV, or "Hi Gain" (again, no reference spec is given, except for "+19 dBu headroom"). Word clock in and out (on BNC's) completes the "main" I/O complement.

Beyond this, the Nuendo 8 I/O's back panel includes three more connectors comprising ADAT and TDIF Aux in and out. These connectors are used for the unit's three special functions: Bit Split, which allows 24-bit audio data to be sent to two 16-bit recorders, Combine, which is the reverse of Bit Split, and Copy, which turns the unit into a format converter, digital patchbay, and digital signal distribution box. In Copy mode, the A/D converters are disabled, but incoming digital signal is routed to all four digital outputs (ADAT Main, ADAT Aux, TDIF Main, and TDIF Aux). You can also choose to Bit Split while in Copy mode.

The front panel includes signal present and overload LEDs for each input channel, and a signal present LED for each output channel. Switches are provided for turning on Bit Split, Combine, and Copy modes, as well as for selecting the unit's clock source, sample rate, and digital input.

I used the Nuendo 8 I/O in conjunction with the Nuendo 96/52 card for this review. I had zero problems with it, and found it very easy to use. The sound quality was excellent, with good detail and dynamic range, and solid imaging. At $1,999, it doesn't come in as the least expensive A/D/A converter package on the market, but its sound quality and special functions make it an excellent value compared to much of the competition.

NUENDO 96/52 AUDIO CARD



In addition to the Nuendo software package, Steinberg is offering a selection of "Nuendo" hardware. This hardware isn't proprietary to the Nuendo program; in fact, it can be used with just about any audio software package.

One example is the $799 Nuendo 96/52 audio card; a PCI-based card that offers two ADAT lightpipe I/O and a nine-pin connector, which supports sample-accurate ADAT sync and S/PDIF digital I/O. Included with the main PCI card is a second expander card that mounts in a computer slot opening, but doesn't use the slot itself. The two cards connect with a ribbon cable. This expander card adds a third ADAT I/O and word clock I/O. If you need more ins and outs, multiple 96/52 cards can live in the same computer. No analog I/O is provided; you'll need external A/D and D/A converters, ADATs, or a lightpipe-compatible digital mixer in order to get analog audio in and out of the card.

A handy software "tray" app gives control over the card's I/O, resolution, and sync features, and reports sync status and so on. Drivers include ASIO 2.0 and Windows MME. Windows 98, NT, and 2000 are supported.

The card installed easily in my computer, and worked flawlessly throughout the Nuendo review. Especially nice is its support of ASIO 2.0 Zero Latency Monitoring, which uses hardware routing to minimize latency issues.

that can each contain a single type of automation (see fig. 4). You can either edit automation graphically with the mouse or you can access the automation events in a list and edit them as text values.

Also included with Nuendo is the Surround Panner, an integrated plug-in that provides for graphic panning with surround output setups (see fig. 5). The Surround Panner works very well, and it can be automated. However, I found it a bit touchy; it's easy to accidentally change the pan position of a track while opening the panner or adjusting other parameters. On the plus side, the Panner allows you to turn individual speakers on and off for each track, and you can use it to feed signal to the LFE channel.

Plug-Ins

Nuendo supports both DirectX and VST plug-ins, in addition to a variety of non-real-time offline processes, such as normalizing, phase reverse, time stretch, and so on.

The program comes bundled with 19 plug-ins, plus a Dolby Pro Logic encode/decode matrix. The

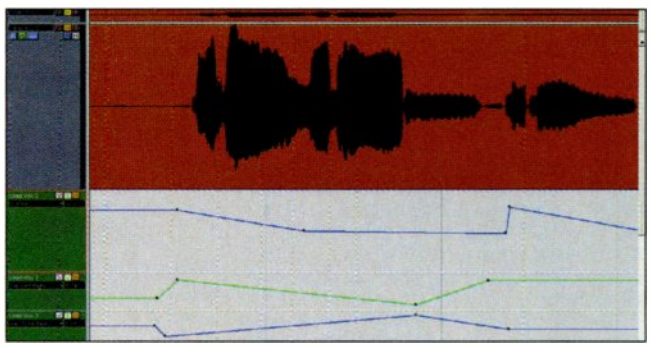


FIGURE 4: EACH TRACK IN THE NUENDO MAIN PROJECT WINDOW CAN BE WHATEVER HEIGHT YOU LIKE; SIMPLY GRAB IT AND PULL, AND IT STRETCHES OUT. THE LITTLE SLIDER ON THE UPPER RIGHT OF THE WINDOW CONTROLS THE WAVEFORM HEIGHT WITHIN THE TRACK; THIS ALLOWS YOU TO OPTIMIZE THE WAVEFORM DISPLAY NO MATTER HOW TALL THE TRACK IS.

AN AUDIO TRACK CAN HAVE A NUMBER OF ASSOCIATED SUBTRACKS, EACH CONTAINING A DIFFERENT TYPE OF AUTOMATION DATA. ANY COMBINATION OF THESE CAN BE OPENED OR HIDDEN. WHILE IT'S NICE TO BE ABLE TO SEE MULTIPLE TRACKS OF AUTOMATION DATA AT ONCE, I MISSED BEING ABLE TO VIEW AUTOMATION OVERLAID ON THE WAVEFORM FOR CERTAIN EDITING TASKS.

FIGURE 5: NUENDO'S SURROUND PANNER (TECHNICALLY IT'S A PLUG-IN, BUT IT FUNCTIONS TRANSPARENTLY AS PART OF THE MIXER) CAN OPERATE IN EITHER OF TWO MODES: POSITION (ABOVE LEFT) OR ANGLE (ABOVE RIGHT). BOTH ARE USEFUL, DEPENDING ON WHAT YOU'RE TRYING TO DO; "POSITION" IS BETTER FOR PLACING A SOUND IN THE SOUNDFIELD. IN THE PANNER ON THE LEFT, I'VE BROUGHT THE LEAD VOCAL OUT INTO THE FIELD A BIT, WHICH HELPS ITS PRESENCE IN THE MIX.

"ANGLE" WORKS WELL WHEN YOU WANT THE SOUND TO COME FROM AN "AREA" OF THE FIELD, RATHER THAN A "PLACE." IN THE PANNER ON THE RIGHT, I'VE TURNED OFF THE CENTER SPEAKER (BY ALT-CLICKING IT), AND SPREAD THE BASS TRACK ACROSS THE LEFT AND RIGHT SPEAKERS. THIS SIMULATES THE EFFECT OF A CENTER-PANNED TRACK IN A NORMAL STEREO MIX.

IT CAN BE HARD TO OPEN THE SURROUND PANNER WINDOW WITHOUT UNINTENTIONALLY CHANGING THE PAN POSITION. YOU CAN ALSO INADVERTENTLY CHANGE THE SURROUND PAN POSITION WHILE THE SURROUND PANNER IS OPEN. THIS PROBLEM IS WORST WHEN IN ANGLE MODE; IF YOU ACCIDENTALLY CLICK OUTSIDE THE CIRCULAR PAN DISPLAY IN THE REST OF THE PLUG-IN WINDOW, THE PAN POSITION TRIES TO MOVE TO WHEREVER YOU CLICKED.

NOTE THAT THE CURRENT VERSION OF THE SURROUND PANNER (SHOWN HERE, DOWNLOADABLE FROM NUENDO.COM) HAS AN ADDED LFE SLIDER FOR SIMULTANEOUSLY FEEDING THE TRACK INTO A SUBWOOFER WITHOUT USING A MIXER AUX SEND.

selection of VST plug-ins includes chorus, FuzzBox, AutoPan, Phaser, and so on. Also supplied are two reverbs, NuendoReverb and NuendoVerb3, both of which I found to be of marginal quality. Plan to use hardware reverbs or to invest in third-party reverbs.

Surprising by their absence are surround-capable plug-ins; given the surround support in Nuendo, I expected at least a few basic multichannel plug-ins. You can insert stereo plug-ins on Group Channels fed to the output busses, and there is an Output Routing Patchbay that lets you change which combination of two output channels feeds through various plug-ins, but neither of these is the same as true multichannel processing. (Steinberg does offer an optional Surround Edition suite of six multichannel plug-ins.)

The current version of Nuendo doesn't support ReWire or VST instruments; this should be addressed in Version 1.5.

Cool Stuff

As I was going through the manual, learning Nuendo, I found myself repeatedly stumbling across nice "bonuses" — the designers behind Nuendo clearly thought their way through how most engineers and producers do things, and provided a large list of extras and convenience features.

Among the more outstanding extras are unlimited undo and redo, along with an undo history that lets you jump back a number of steps. Unfortunately, undo/redo can't undo pan or other mixer moves (it does cover automation edits). Hopefully this can be addressed in an update.

Nuendo supports video playback from an integral video track, which can display thumbnails of the video frames. This track makes spotting and conforming audio to video very easy.

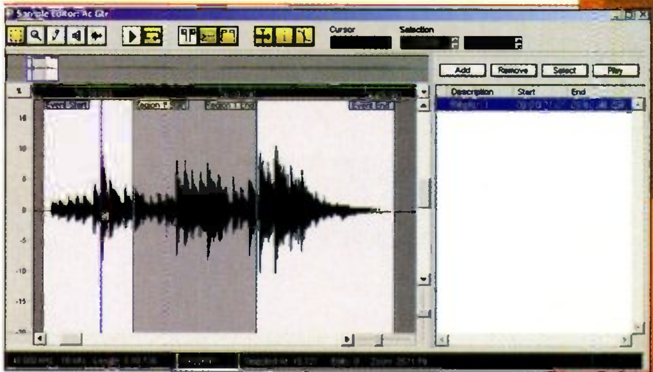
Other convenience features include fully assignable key commands and support for screen sets (Nuendo calls them Window Layouts), which can be recalled from a menu or using key commands.

In addition, Nuendo contains layouts for a number of hardware controllers, such as those from JL Cooper, as well as the CM Automation MotorMix, and Yamaha 01V digital mixer. I was able to use my 01V as a control surface with no difficulty at all. With an 01V, you get control over 28 mixer channels including volume fader, pan, four-band EQ, and six of the eight aux sends. The 01V's stereo master fader defaults to control Nuendo's master fader, which, if you're in a surround master setup, controls the level of all six outputs for a 5.1 system — this is a godsend for those who don't want to invest in a surround monitor controller box.

The Real World

How well does Nuendo hold up under real-world circumstances? As I began this review, I was booked to do a fairly complex 5.1 surround mix consisting of 26 simultaneously playing audio tracks (16-bit/48 kHz resolution). The tracks were imported into Nuendo from a single ADAT-XT, using the nine-pin ADAT sync and lightpipe connections on a Nuendo 96/52 card (see sidebar). I ended up using 24 bands of EQ (spread over 14 mixer channels), 11 insert compressors, two chorus and two NuendoReverb plug-ins (on aux sends), and just for fun imported a video track. I had eight VST inputs and six bus outputs turned on. On my computer (a 733 MHz Pentium III — see Box for more test system specs), Nuendo's CPU usage meter hovered between 50 and 60%; the hard-disk

▶ continued on page 144



DOUBLE-CLICKING AN AUDIO TRACK OPENS THE SAMPLE EDITOR, WHICH ALLOWS FOR FINE EDITING AND PROCESSING OF AUDIO CLIPS. IN ADDITION TO THE EXPECTED CUT/COPY/PASTE, WAVEFORM DRAWING WITH THE PENCIL TOOL, AND SO ON, YOU CAN DEFINE REGIONS WITHIN THE CLIP (IN THIS CASE "REGION 1"), WHICH CAN BE EDITED AND PROCESSED INDEPENDENTLY FROM THE REST OF THE CLIP. YOU CAN EDIT THE START AND END POINT FOR THE CLIP, APPLY PLUG-INS OR OTHER OFFLINE PROCESSING, AND SET THE "SNAP" POINT (REPRESENTED BY THE BLUE LINE WITH THE "S"), WHICH IS USED WHEN LOCKING (QUANTIZING) AUDIO TO A GRID IN THE PROJECT WINDOW. NORMALLY, THE SNAP POINT DEFAULTS TO THE BEGINNING OF THE CLIP, BUT IN SOME CASES, IT'S BETTER TO HAVE IT ELSEWHERE. IN THIS CASE, A STRUMMED ACOUSTIC GUITAR, IT WORKED BETTER TO SNAP TO THIS PARTICULAR POINT.

REGION LOOPING, AUDIO SCRUB, AND ZOOM IN/OUT ARE ALL PROVIDED, AS IS AN OVERVIEW BAR (ABOVE THE WAVEFORM). CLICK-DRAG IN THIS TO MOVE ABOUT WITHIN THE OVERALL AUDIO TRACK. ONE COOL FEATURE: YOU CAN SET AUDIO EDITS TO SNAP TO THE NEAREST ZERO-CROSSING, HELPING TO PREVENT CLICKS AND POPS WHEN DOING WAVEFORM EDITING.

InREVIEW

Yamaha MSP10 Powered Nearfield Monitors

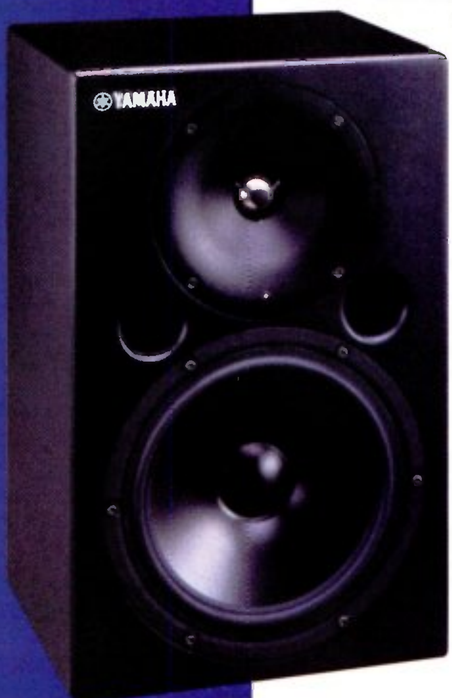
Yamaha's active monitors and companion subwoofer score a solid hit

Yamaha's MSP10 is a two-way, biamped studio monitor intended for nearfield listening applications. Housed in a compact, dual-ported enclosure, the MSP10 combines a plastic-cone woofer and a titanium-dome tweeter with a high-frequency response extending out to 40,000 Hz. Each MSP10 cabinet contains two amplifiers: one capable of delivering 120 watts maximum to the woofer, and the other 60 watts maximum to the tweeter. The MSP10 is available in a magnificent maple sunburst wood finish or a more industrial-looking semi-gloss black lacquer.

Rear-panel features of the MSP10 include a balanced XLR input jack, power on/off switch, low- and high-frequency trim controls, a sensitivity adjustment pot, and an 80 Hz low-frequency rolloff switch. In addition to filtering out potentially damaging low-frequency sounds, the rolloff switch is also helpful when integrating the MSP10 with its companion subwoofer, the SW10 (see sidebar).

The MSP10 is intended for console-top or stand-mounted nearfield use. Since the cabinets are rather heavy for their size (approximately 45 pounds), make sure that the desk or stands you intend to use for support are up to the task. Recessed into the tweeter's conical baffle is an LED that lights green when power is applied and turns red in the event of clipping.

The MSP10 has a lot of intelligent design features, best exemplified by the "tweak" controls on the rear panel. The low- and high-frequency trim controls are switched, with positions for –2, –1, and 0, and –1, 0, and +1, respectively. Although some people might prefer continuously variable trim pots, the switched design makes it easy to obtain consistent settings between the cabinets as well as easy repeatability. Ditto for the sensitivity, which has a detent at the +4 dB setting.



A bit of care is required in the setup process to get the best results out of the MSP10's (unfortunately the manual doesn't provide much more than specs). Initially, I ran the sensitivity pot(s) at maximum, trim controls at 0, and the 80 Hz rolloff switched on. These settings resulted in reproduction that was somewhat on the bright side for my taste. Through experimentation, I found that turning the LF

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cut off and rolling the HF trim back to –1 made for well-balanced reproduction. It then dawned on me that the LF cut was probably intended to be switched on only when the subwoofer was in use.

MSP10 Solo

My first sessions with the MSP10's were without the SW10 subwoofer. Immediately I noticed that the speakers produce great stereo imaging and excellent response to dynamics. They were also very revealing, clearly showing up punch-ins I've never before noticed in mixes I'm very familiar with. This clarity is the big strength of the MSP10's; these speakers really let you hear detail in your audio. For tracking, this analytical quality is invaluable. It also allows the MSP10's to "dissect" a mix — which some people may not like. If you want a monitor that's going to give you the "warm fuzzies," look elsewhere — the MSP10 is more about accuracy than pleasant coloration.

Off-axis high-frequency response of the MSP10's is very good, so even when you're listening outside of the sweet spot you won't feel like someone put gauze in your ears. This can be a big strength for rooms where space behind the console is tight. As you'd expect from a monitor of this size, bass response didn't extend to the subterranean regions, but what the MSP10 did produce on its own was tight and solid.

With The SW10

Adding the SW10 to the MSP10's extends the low-frequency response and adds headroom to the system. Through proper use of the low cut switch, it also frees up the MSP10 from dealing with the bottom octaves, thus improving clarity in the mid-bass and midrange.

Combining the SW10 with the MSP10's took a bit of experimentation, mostly due to lack of clues in the manuals. The MSP10's had been set to maximum sensitivity. With the SW10's volume pot set to minimum (which is the factory default), I was wondering if the SW10 was functioning at all because the trio was so top-heavy. Moving the MSP10 sensitivity control down to the "+4" (notched) position and moving the SW10's "volume" pot up to the notched position parted the seas for me, bringing the entire frequency range into proper balance. I decided to run the MSP10's with their rolloff "on." Since that rolloff is at 80 Hz, it seemed logical to set the cutoff of the SW10 also at 80 Hz. Results at these settings were great, with effortless bottom-end extension, increased headroom, and seamless transition between the SW10 and MSP10's. The SW10 added plenty of "whump" to the system, easily driving the SPL in my 12- x 15-foot control room beyond a comfortable level before I witnessed an occasional flash of the red LEDs.

A unique feature of the SW10 is the three I/O channels, which I'm assuming applies to a 5.1 situation where the third I/O channel would be used for a center channel. Very clever. Another operational aspect of the SW10 I liked is that the output or "thru" jacks are passive. So when you turn off the subwoofer, audio to the MSP10's is unaffected — and you won't hear any kind of "pop." This comes in handy when you're doing a mix and want to turn off the subwoofer for referencing purposes. I've run into some subs that won't pass audio when you turn them off, making it a hassle to quickly switch the sub in and out of the chain.

The Bottom Line

With or without the SW10 subwoofer, Yamaha's MSP10 powered monitors scored big points in my book. They're accurate, play loud without breaking up, and cost about as much as comparable monitors *without* built-in amplification. While they might not be capable of filling huge control rooms with ear-splitting SPLs, the MSP10's will be quite at home in a project or small commercial studio. ■

▶YAMAHA MSP10

MANUFACTURER: Yamaha Corporation of America, P.O. Box 6600, Buena Park, CA, 90622. Tel: 714-522-9000. Web: **www.yamaha.com**.

SUMMARY: Active nearfield monitor for studio applications.

STRENGTHS: Plenty of power. Extended high-frequency response. Excellent dynamic response.

WEAKNESSES: Manual lacks setup information. Inputs are XLR-only.

EQ FREE LIT. #: 111

YAMAHA SW10 POWERED SUBWOOFER

Intended as a companion to the MSP10, Yamaha's SW10 is a powered subwoofer with a 10-inch woofer loaded in a bass reflex cabinet. The SW10's internal amplifier is capable of delivering 180 watts maximum output at 100 Hz with a 1% THD figure. Six XLR input and output jacks are provided on the rear panel for three channels of I/O. A continuously variable low-frequency cutoff ranging from 40 to 120 Hz and a volume pot are provided for matching the SW10 to any "satellite" speakers. A power switch and captive three-prong AC cable are provided at the bottom of the rear panel. A front-panel LED indicates green for power on and red for clipping. Suggested retail price for the SW10 is $849.

SW10 SPECIFICATIONS

Frequency range	25 to 150 Hz (–10 dB)
Sensitivity	–12 dB (at "vol. max") for 100 dB SPL at 1 meter
Maximum output	111 dB at 1 meter
Dimensions	13 wide x 18.25 high x 18.5 deep (inches)
Weight	57.2 pounds

SPECIFICATIONS

Frequency range	40 to 40,000 Hz
Sensitivity	–10 dB at "–6" position for 100 dB SPL at 1 meter
Maximum output	110 dB at 1 meter on axis
Woofer	8-inch, plastic cone
Tweeter	1-inch, titanium dome
Dimensions	10.5 wide x 16.6 x 12.5 deep (inches)
Weight	44 pounds

DSound Stomp'n FX, Vol. I VST/DirectX Plug-Ins

Stomp box-style plug-in effects for DirectX or VST

"Stomp'n FX, Vol. 1" is a set of six plug-ins for Windows (DirectX, VST 2.0) or Mac (VST 2.0) from Czechoslovakian company DSound. These plug-ins emulate guitar stomp boxes; the roster includes the AW1 Auto Wah, CH1 Chorus, CM1 Compressor, FL1 Flanger, GE1 7-Band Graphic Equalizer, and NG1 Noise Gate. The look is appropriately retro (Fig. 1), being somewhat reminiscent of DOD or Boss boxes from the '80s. But these aren't a joke; the sound quality is actually very good, and, besides, they're a lot of fun...so let's Czech into this a bit further.

Getting Started

System requirements are relatively modest; if your system can handle a program that supports plug-ins, it can likely do the job. The plug-ins were tested on an Apple PowerBook G3 running under OS 9.0.4 with 192 MB of RAM and a Q Performance Systems 850 MHz Celeron (overclocked) running Windows 98SE with 256 MB of RAM.

On both the Mac and Windows, installation was trouble-free. Especially nice: With Windows, both DirectX *and* VST versions were installed automatically. Copy protection consists of entering a serial number; from time to time, the program asks for this serial number to verify ownership.

There are a few special considerations: When running as VST effects (PC or Mac), the parameters support automation and all effects can work as track effects within Sonic Foundry's Vegas. The effects are relatively processor-friendly; on the Mac, I opened up all six DSound plug-ins within TC Works' Spark 1.5, and the processor usage was around 32%. Granted, these aren't as complex as something like a high-end reverb, but it's nice to know you can use a batch of them if you're so inclined.

FIGURE 1: The DSound plug-ins sport a suitably retro look, but sound vastly better than their hardware ancestors.

DSOUND STOMP'N FX

MANUFACTURER: DSound, 407 Stony Point Road, Santa Rosa, CA 95401. Tel: 800-449-3213. Web: www.dsound1.com.

SUMMARY: Mac/Windows plug-ins emulate vintage-style stomp boxes.

STRENGTHS: All plug-ins sound and look good. High-resolution, automatable parameters under VST. Simple, obvious operation. Reasonably low cost and low processor loading. For Windows, installs as DirectX and VST.

WEAKNESSES: Doesn't accept external MIDI controllers under VST. Flanger lacks negative flanging. Auto Wah lacks envelope tracking by filter frequency. Graphic EQ could use an "air" high-frequency band.

MINIMUM SYSTEM REQUIREMENTS: PC — Windows 9X/NT/2000, 120 MHz Pentium. Mac — OS 8.1 or higher, 180 MHz 604e.

PRICE: $149

EQ FREE LIT. #: 104

Commonalities

All of these effects have certain common aspects. There's an on/off "footswitch" and a button you can click for options. These include interface size (small and large; choose small unless you're really in love with the stomp box look) and preset management.

Each FX has at least six "factory" presets (some have a few more), but you can create your own presets and save these in a bank (unlimited numbers of banks are allowed, which facilitates saving groups of presets for specific projects). Up to four presets can be associated with four preset buttons for quick selection. As expected, it's possible to delete individual presets or all presets, mix factory and user presets, and compare the modified version of a preset with the original version. You can also reload a just-deleted preset if you change your mind.

A single display line shows the value of the parameter being modified; this is also where you enter preset names.

Parameters are represented on-screen as knobs. To change values, you click on the knob and rotate the mouse in a knob-like fashion. I prefer changing values by moving the mouse linearly (*e.g.*, up for more, down for less); the main advantage to the "knob" approach is that, as you move the mouse further from the knob's center, the resolution increases for a given amount of mouse movement. This makes it easy to change values in very fine increments. However, if you have a "scrolling" mouse with a wheel, you can use the wheel to fine tune values, or shift+wheel for coarser tuning.

With the Mac, help files are HTML documents that can be opened with standard Internet browsers, although there's also a version that works with the Mac Help Viewer. With Windows, calling up help activates a standard Internet Explorer-type help file.

Effects Parameters

Auto Wah: This is an LFO-based wah effect with Rate (0.05 to 10 Hz), Depth, and Center Frequency. A Sensitivity control causes the LFO depth to depend on the dynamics of your playing. An associated Mode switch determines whether depth increases as your playing gets softer or louder. This is a cool effect (there's a hint of overdrive that sounds pretty tasty), but it doesn't allow the traditional envelope follower effect where the filter frequency tracks the signal envelope.

Chorus: Controls include Mix (Dry to Wet), Rate (0.1 to 10 Hz), Depth, and Stereo (sets a different sweep width between the channels). The sound is very sweet (I like using it with synths, too), and it does great vibrato effects.

Compressor: This is standard issue stuff — Threshold (–60 to 0 dB), Attack (0 to 200 ms), Release (20 to 4,000 ms), and Ratio (1:1 to 50:1, then infinity:1). An additional switch chooses between automatic "make up" gain or just letting the output fall where it may. Nice touch: Attack and release are variable in 1 ms increments, and ratio in 0.1 dB increments.

Flanger: Initial Delay (called "FX Tune"), Rate (0.05 to 10 Hz), Depth, and Regeneration. It does positive flanging only, not negative.

WEBLINK

Download MP3 audio examples of the various effects, as well as demo of the DSound Stomp'n FX, Vol. I, at www.dsoundl.com

Graphic Equalizer: All sliders are ±18 dB, with center frequencies at 0.1, 0.2, 0.4, 0.8, 1.6, 3.2, and 6.4 kHz. An additional slider controls output level.

Noise Gate: Threshold (–100 to 0 dB), Attack (0 to 200 ms), and Release (1 to 2,000 ms). As with the compressor, Attack and Decay can change in 1 ms increments.

FIGURE 2: Several plug-ins loaded in Spark's FX Machine. The flanger is highlighted; it's in "small" graphics mode, as opposed to the "normal" mode evidenced in fig. 1. Note how two choruses are placed in parallel at the output. The CPU indicator shows 34.1% — pretty good for seven plug-ins.

The Real World

In addition to testing the plug-ins using Spark on the Mac, I also tried them with Wavelab, Sound Forge, Acid, and Cubase VST on Windows. Operation was rock-solid in all cases, but recording automation data in Cubase was particularly satisfying, in large part because the fine level of adjustability prevent zippering effects.

However, Spark is probably the ultimate host program for these plug-ins thanks to its FX Machine feature, which allows inserting plug-ins into a matrix (fig. 2) with both series and parallel connections. It follows the same conceptual model as patching stomp boxes together, but, of course, without the patch cords. There is one weirdness: When you load a plug-in into the matrix, edit it, close down the Edit window, then open it up again, the controls are in their default positions. This doesn't affect the sound, which remains as edited. [DSound tells us that this problem has been fixed.]

The Bottom Line

The DSound plug-ins are very cool sonically, and you certainly can't fault the graphics. Thankfully, despite their "purist" approach, DSound didn't emulate the more negative stomp box habits — nasty hiss, bypass switches that load down the inputs, AC hum, and batteries that give out in the middle of your big solo. However, there are some missed opportunities. For example, the Auto Wah doesn't have a mode where the filter frequency tracks the incoming envelope, as the envelope can affect only the LFO depth. It's hard to imagine how this popular effect got overlooked. Also, including an octave switch could have justified promoting this effect for bass.

The flanger doesn't allow for negative flanging (*e.g.*, being able to set regenerative feedback out-of-phase instead of in-phase), nor can it do through-zero flanging, where the difference between dry and delayed sounds reaches 0 milliseconds. Granted, no stomp box could do through-zero flanging either, but one of the supposed advantages of software is that it doesn't have to deal with the same limitations as hardware equivalents. (You can, however, obtain through-zero flanging in the Spark FX Machine by loading two flangers in parallel, then setting them for delayed sound only. Adjust one for a small, fixed delay with no LFO depth, and the other so that the highest point of the width's excursion is the same as the fixed delay.) And I would have loved to see the chorus delay time or LFO rate respond to envelope control. To be fair, this is again something not found in most stomp boxes — but then again, I'm the kind of guy who wants Hammond B-3 emulators to respond to pitch bend and velocity.

But it's fairer to evaluate what's included rather than what's not, and in that respect the DSound plug-ins are not only a lot of fun, they sound very clean and truly manage to give that "guitar vibe." I particularly like the Auto Wah filter effect — it's great on drums as well as guitar, and automating the filter frequency in VST gives real flexibility.

Admittedly, the DSound plug-ins are for a relatively specialized audience, but they fill a void that no one has filled up to now...besides, they definitely have a high fun factor.

Craig Anderton is the creative director of MusicPlayer.com, as well as the author of the classic book *Home Recording for Musicians*. He has produced, played on, mixed, and or/mastered 17 major-label releases.

By DJ Russ Reign • rmikowski@uemedia.com

Electrix EQ Killer and Filter Queen Frequency Isolator and Analog Filter Box

For live or studio filter applications, these boxes have what it takes

Electrix's recent release of the EQ Killer and Filter Queen modular performance effects has DJs and dance music enthusiasts around the world begging for more. The Filter Queen, basically a smaller, less expensive version of the company's original Filter Factory, is jam-packed with features and can totally change the face of a live performance or studio session single-handedly. The EQ Killer, a product that DJs have demanded for years, enables us to obtain interactive on-the-fly control of frequency envelopes with a wide array of tweaking options. I had the opportunity to field-test both units recently during a gig at the Tunnel nightclub in New York City.

Setting The Stage

I arrived at the club approximately an hour before doors opened to make the necessary connections and get the Electrix gear up and running. Since I would be using both pieces simultaneously during my performance, I needed to be a little bit creative: I connected the EQ Killer as a post-mixer effect by running the board's master output into the unit and then routing its signal to the club's amplifiers. Now I would be able to use the piece to effect the entire master mix, regardless of how many sources I'd be using at the same time. The rear panel of the EQ Killer contains six sets of RCA line jacks — inputs and outputs for traditional stereo setup, as well as an effects loop with send and return jacks.

The Filter Queen was next. Since this piece is more involved than the EQ Killer, I decided to connect it so I'd be able to cue-monitor in the headphones prior to making the effect live. I connected one of the turntables (once again via standard RCA cable) to the unit's input, and then ran the output back to the mixer. Now the Filter Queen's effects would be limited to that single source, but I'd be able to preview the reactions of different songs to the unit's diverse effects before letting them escape to the club's speakers.

The EQ Killer and Filter Queen are definitely "no frills" devices, which has enabled the manufacturer to keep them small and lightweight — a major positive when taking the units on the road. Each piece measures only ten inches in length (including mounting brackets), and provides two functional mounting options: The units can be placed securely on a flat surface thanks to gripping rubber tracks on their bottoms. Best of all, the angled face of the unit makes manipulation of surface controls a snap when using the pieces as tabletop effects. Electrix also includes removable brackets (complete with Allen wrench) that can be used to connect these two pieces to form a 19-inch, rackmountable dual unit.

ELECTRIX EQ KILLER & FILTER QUEEN

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

SUMMARY: Customizable, frequency-isolating kill box (EQ Killer) and digitally controlled analog filter box (Filter Queen).

STRENGTHS: Durable controls. Multiple mounting options. Excellent manuals. Applications as both studio and live performance components.

WEAKNESSES: EQ Killer only has RCA inputs and outputs (no 1/4-inch jacks). No power switches. No MIDI sync or automation support.

PRICE: $299.99 each.

EQ FREE LIT. #: 105

MASSIVE PASSIVE STEREO TUBE EQ

A PICTURE IS WORTH A THOUSAND WORDS...

Perhaps, but would photographs of our Variable Mu or VOXBOX have created their successes alone?

You have to hear this gear. You have to use this gear. Put your hands on the knobs and crank 'em.

Engineers who have already gotten hold of the MASSIVE PASSIVE have told us: "Why does it make everything sound so much better?", "It's organic and orgasmic.", "It's a f%#king powerhouse.", "It's unlike <u>any</u> other EQ.", "This is IT. The sound I've always dreamt of but couldn't ever get until now."

GOT THE PICTURE?

Craig 'HUTCH' Hutchison designed these monsters... The MASSIVE PASSIVE is a two channel, four band equalizer, with additional high pass and low pass filters. "Passive" refers to the tone shaping part of this clever new EQ design not using any active circuitry. Only metal film resistors, film capacitors and hand-wound inductors sculpt the sound, kinda like a Pultec EQ on hyper-steroids. Super-beefy, hugely-high-headroom Manley all-tube make-up gain amplifiers deliver your tunes into the next realm. You'll need to experience this.

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CIRCLE 31 ON FREE INFO CARD

EQ Killer

The EQ Killer offers three filter bands (low, mid, and high), each with its own control group. The band level controls can operate from full cut to 6 dB of boost for each frequency range. Each band also has the familiar Electrix Momentary button, which can be used to apply or "kill" a setting manually to the beat of the music, and a Band Kill button that will totally remove a frequency range from the mix upon activation. Perhaps the most impressive and fun-filled features of the EQ Killer are its low and high crossover controls. These allow the user to set the width of each band that the unit controls. Manipulating these knobs in real-time can create some impressive results, and the sound leaving the unit after all this remains clear.

I worked the low and high crossover controls relentlessly during the course of the night, often while one band (usually either high or low) was completely killed from the mix. By sweeping the crossover control knobs uniformly from one extreme to the other, you can create a custom-sounding effect that will make even the most played-out song come to life for your audience. The frequency isolation of the unit is precise as well. Hands down, this is the best band-isolating device I've used. Not only is it high quality enough for serious studio use, but the EQ Killer's durability makes it a live performance hit as well.

Filter Queen

The Filter Queen is an entirely different animal. Although comparable in design and layout to the EQ Killer, the similarities stop there. The unit's digitally controlled analog filters can create a vast array of effects. Despite its low price and harmless-sounding name, the Filter Queen is one powerful piece of equipment, and must be treated as such. I would highly recommend becoming extremely familiar with the unit's controls and their interaction not only with each other, but also with your music before using this unit in a live situation. The Filter Queen's powerful filter and phaser sweeps can be extremely impressive as live performance elements, but these effects can also destroy sound systems if used without caution.

That being said, the Filter Queen is flat out fun! It definitely takes a while to figure out how manipulating the controls will effect your mix, but once you get the hang of it, the possibilities are almost endless. The unit's angled front control panel houses knobs that control frequency, resonance, LFO speed and depth, envelope follower release and depth, and percentage of effect mix. Soft-touch buttons control effect band (high, mid, low), filter type (highpass, lowpass, bandpass, notch), waveform (sawtooth, inverse sawtooth, triangle, square wave, and random), momentary effect mix, and unit engagement.

The frequency knob and waveform setting can be used in conjunction to designate specific frequencies that will be filtered, which is excellent for emphasizing specific synth riffs or chord patterns in dance music. Since I had the unit connected in a manner that would allow for pre-mix monitoring, I could also tweak the envelope follower to create a "wakka-wakka"-type sound as described in Electrix's manual. The effect mix control was then used on the fly to alter the depth of the effect that the crowd was hearing. The trained ear can even get the effect in sync with the song's BPM by using the LFO speed and depth controls — pretty cool! Curious club goers were standing around the booth all night long, watching me work the Electrix pieces with wide eyes. I felt like a rock star!

Nuts And Bolts

For those familiar with the Electrix Filter Factory, MoFx, or Warp Factory, you'll find the same ceramic control knobs on the faces of these new siblings. These controls slide smoothly through their rotations and can definitely take a beating, a trait that is ultimately necessary in any live-performance piece. The Filter Queen has another function that lends particularly well to real-time use — 1/4-inch jacks allow for the connection of a standard momentary footswitch or an expression pedal that can be used to control the filter's frequency remotely.

Speaking of remote control, note that neither the EQ Killer nor the Filter Queen offers support for MIDI — for live DJs this probably won't be much of a concern, but for some studio production applications it would be nice to be able to MIDI these units up to a sequencer. Another minor complaint is that neither box includes a power switch; you'll have to turn them on and off using a power strip or by pulling the power adapter out of the outlet.

Both units also include great instruction manuals that can get you up and running quickly. Neither is particularly in-depth because you really need to hear for yourself how certain things operate on these pieces. But each manual covers all the necessary bases and includes detailed diagrams describing various connection scenarios. The Filter Queen's manual also has a "quick start" section that explains common applications and how to set the various controls to obtain them.

Sweeping Statements

At $300 each, the Filter Queen and EQ Killer provide a good value — they do what they say they'll do, and they do it well. Whether you're a DJ spinning in a club or crafting dance music in the studio, these two boxes should be welcome additions to your rig. ■

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By Rob McGaughey • soundsauna@earthlink.net

Akai DPS16 Digital Personal Studio

Akai's 16-track high-resolution studio-in-a-box offers plenty of power for music production

Over the past few years, a wide array of stand-alone hard-disk recording systems have hit the market. These devices are basically recording-studios-in-a-box that allow the user to plug in a guitar, keyboard, and/or microphone and record and mix a song with astounding sound quality. The Akai DPS16 fits squarely into this new market, offering up to 10 tracks of simultaneous recording, up to 16 tracks of simultaneous playback, full mixing capabilities, and built-in effects processors. In my evaluations, I had the DPS16 synced to Mark Of The Unicorn's Performer running on my Mac. The audio tracks were handled by the DPS16 and Performer handled all the MIDI sequences and automation. This combo worked well once I got a feel for how to get around on the DPS16. This feat did require frequent references to the manual for the first few hours (okay, several hours), as I didn't find the DPS16's user interface intuitive at first. But after a few projects I was getting around fairly quickly.

Tracking

Once you figure out the process for routing the physical inputs to tracks and the master bus for monitoring, the tracking process is pretty simple. The DPS16 allows the user to create, name, and save 100 locate points per song. You can also have up to 250 virtual tracks that can be recorded, edited, processed, and mixed with other tracks. This makes it easy to track parts, do overdubs, record multiple takes, and create composite tracks comprising the best elements of various takes. The punch-in/out functions also worked well and can be automated, performed manually, or controlled with a footswitch. The DPS16 provides phantom power to the XLR connectors on channels 1 and 2, which is a feature omitted by some of the competition. Input channel 8 allows the user to change the impedance to accommodate a standard line level device or a guitar/bass. Recording bass guitar direct sounded good, but electric guitar was less desirable; the DPS16 doesn't have a guitar amp simulator and I felt the distortion effect was weak. I did like the fact that you can use the three-band equalizers and/or effects in real time while recording.

I was impressed with the sound quality of the tracks that I recorded using the DPS16. The quality was good even at the old standard 16-bit, 44.1 kHz resolution, and was even better at higher resolutions.

AKAI DPS16

MANUFACTURER: Akai Professional, 4710 Mercantile Drive, Fort Worth, TX 76137. Tel: 817-831-9203. Web: www.akaipro.com.

SUMMARY: All-in-one digital recording studio including digital mixer, 16-track recorder, and up to four simultaneous effects. Will connect to a SCSI CD-R and burn an audio or data CD-R/CD-RW.

STRENGTHS: Excellent sound quality. Plethora of backup options. Comprehensive bit depth and sample rate choices. Phantom power. Dedicated monitor level knob. Faders feel good. Waveform editing screen works very well. Pitch correction. Tiltable LCD screen with easily accessible contrast knob. Lots of dedicated faders, buttons, and rotary knobs.

WEAKNESSES: Not always intuitive. Guitar distortion effect is weak and there aren't any guitar amp simulators on board. No sample rate conversion, dithering, noise shaping, or other word-length reduction mechanism. Manual could be better.

VERSION REVIEWED: v.1.11 OS, Controller Revision 3

PRICE: $2,795

EQ FREE LIT. #: 103

Mixing

I liked the feel of the faders on the DPS16. Many products today offer flimsy faders with virtually no resistance. The Akai faders, however, had a feel that was conducive to performing nice, smooth fades. There are 16 dedicated faders and pan knobs and a total 26 inputs available at mixdown (16 tracks + eight analog inputs + two digital inputs). These inputs can be routed through the three-band EQ on the channels and to four aux sends (routed to internal effects or externally for processing). The aux sends can be configured to operate pre or post fader, plus they can function as inserts for internal effects. The DPS16 offers a stereo master output with a single master fader, a stereo monitor output, a digital output, and

SPECIFICATIONS

Dimensions (mm)	515 x 127.5 x 358.5 (with LCD tilted down)
Display	320 x 240 dot FSTN graphical LCD with back light
Sampling Rates	32, 44.1, 48, or 96 kHz
Resolution	16- or 24-bit
Recording Media	10 GB internal IDE hard drive, external SCSI port for adding additional hard disks or CD-R/W, MO, Jaz drives, etc.
A/D Converters	24-bit 128-times oversampling, enhanced dual-bit delta/sigma modulation
D/A Converters	24-bit 128-times oversampling, advanced multi-bi delta/sigma modulation
Dynamic Range	>100 dB (100 Ohms terminated)
Distortion Ratio	–0.003% (1 kHz @ 44.1 kHz sampling rate)
Input level	–46 dBu to +4 dBu (max. +19 dBu, 1kHz)
Analog Audio Inputs	Ch 1–2: 1/4-inch TRS/ XLR combo jack (balanced) with switchable phantom power on XLR connection Ch 3–8: 1/4-inch TRS balanced/unbalanced connectors
Analog Audio Outputs	Master Out: (2) RCA — unbalanced Monitor Out: (2) RCA — unbalanced Aux. Send: (4) 1/4-inch — unbalanced
Output Level (Master Out, Monitor Out, Aux. Sends)	–10 dBu (max. +5 dBu, load impedance: 47 kOhms)
Headphone Output	Stereo 1/4-inch connector
Digital Audio I/O	S/PDIF on coaxial (RCA) connectors
MIDI	In, out/thru
SCSI	High-density 50-pin SCSI connector

a stereo "ping pong bus." The "ping pong" bus can be used to submix multiple tracks down to either one or two tracks. For example, you might have drums recorded on six different tracks that could be mixed down via the "ping pong bus" to a stereo mix to free up tracks, channels, effects, EQ, etc. I appreciated that both the monitor and headphone outputs had easily accessible level controls.

The mixer offers solo and mute functionality for each track and input. If you enter either solo or mute mode, you can select the desired channels. You can then globally turn off all solo/mute functions with a single button press.

I did most of my evaluation with the DPS16 synced to Digital Performer. I used Performer for sequencing MIDI tracks played on my Kurzweil K2500S and monitored the outputs of the keyboard through the DPS16. These tracks were then mixed with electric guitar, bass, vocals, acoustic guitar, and miscellaneous percussion instruments that I recorded directly onto the DPS16. This process worked well and I was able to use the built-in effects of the DPS16 to get a good final mix. I also used Digital Performer to automate levels and pan for the 16 audio tracks, the eight inputs, and the master outputs.

In addition to accepting MIDI automation control, the DPS16 can store up to 16 scenes in a song. Each scene stores channel on/off status, channel level, channel pan, channel aux send settings, channel EQ settings, channel signal routing to the "ping pong" bus, master bus output level, "ping pong" bus output level, aux send master level, and effects settings. These scenes can then be transmitted and stored as part of the sequence in an external sequencer such as Digital Performer. Thus an external sequencer is able to completely automate a mix on the DPS16. It took me some time to figure out how to make this work, but once I got it going, it worked beautifully. The manual didn't offer the detail I would have liked and had incorrect page references, misspellings, and omitted necessary information.

Editing

The DPS16 has a powerful, well-implemented cut/copy/paste editing engine. It has a large waveform editing screen and a jog/shuttle wheel to select edit points both visually and aurally. I found basic editing could be accomplished very quickly and accurately, even editing multiple tracks at once. You can cut or copy tracks, then overwrite other tracks, insert into tracks, or even insert silence. You can also stretch or compress the time (length) of the audio on a track. For instance, you can copy an audio segment that's 32 seconds long and insert it into a space that's only 30 seconds long. The DPS16 will speed up the track without changing its pitch to make it fit. I could time compress/expand up to 10% with minimal degradation; about what you'd expect.

The LCD screen on the DPS16 is fairly large, easy to read, and has a contrast knob right next to the screen (where it belongs). This screen can be tilted so you can view it no matter where you are in relation to it. This simple feature is really useful as the screen can be tilted to optimum position when you're over working at a computer or standing in front of a microphone

Effects

The DPS16 offers 44 different effects of which up to four can be accessed simultaneously. I thought the reverbs, chorus, flanging, and delay effects were good and I was able to get decent results with their basic editing controls. I didn't particularly care for the sound quality of the distortion effect, the compressor/limiter, or the plates. There's also a mono pitch corrector that I found worked surprisingly well on my voice. I used it to correct pitch inconsistencies and to correct notes that I slid into pitch on. Use of the pitch corrector uses all of the effects processing power (Akai tells us this has changed in version 1.5; other effects are now available simultaneously with the pitch corrector), so no other effect can be used at the same time as the pitch corrector. However, it's easy to use the pitch corrector and bounce the corrected track to a virtual track so you can use other effects.

The DPS16 allows you to create a beat map to store time signature changes and a tempo map, but the current operating system version doesn't offer a metronome. (Version 1.5 adds a metronome to the DPS16; see sidebar.) Creating and storing the beat and tempo maps is very easy. You can view timing information in absolute time, relative time (related to a user-definable offset) and/or bars/beats/ticks.

Creating Audio CDs/Backing Up

The DPS16 allows the user to take any track or pair of tracks and burn them onto a supported SCSI CD-R or CD-R/W drive. You can continue to add tracks until you fill the CD and then you can finalize the CD so that it can be played on a standard CD player. Unfortunately, my Toshiba CD-R drive isn't

SIMULTANEOUS RECORD TRACKS

SAMPLE RATE	32 KHZ	44.1 KHZ	48 KHZ	96 KHZ
16-bit	10	10	10	8
24-bit	10	10	10	6

SIMULTANEOUS PLAYBACK TRACKS

SAMPLE RATE	32 KHZ	44.1 KHZ	48 KHZ	96 KHZ
16-bit	16	16	16	8
24-bit	12	12	12	6

MISCELLANEOUS ITEMS/CHANNEL

SAMPLE RATE	32 KHZ	44.1 KHZ	48 KHZ	96 KHZ
Aux Sends	4	4	4	2
Effects	4	4	4	2
3-band EQ	16	16	16	6

DPS16 VERSION 1.5

As this issue went to press, Akai announced new version 1.5 software for the DPS16. The new OS provides five major enhancements: A metronome has been added, additional MIDI System Exclusive commands are supported for remote control/automation, the Pitch Corrector has been improved, the Remaining Time Display has been improved, and the operation of MIDI timecode slave sync has been changed.

DPS16 owners should contact Akai Professional or visit www.akaipro.com for further information and details on how to obtain this update.

supported by the DPS16, so I was unable to test this function. The manual states that you should contact your dealer or the Akai Professional Service Center for a list of compatible drives.

I am somewhat befuddled by why Akai offers 96 kHz sampling rates and 24-bit resolution in an all-in-one recording package when it has no function for converting these high-resolution audio files to 16-bit, 44.1 kHz for burning to a standard audio CD. Perhaps an update will offer a means to downsample and reduce the resolution of audio files.

Besides audio tracks, you can also use your CD-R, CD-R/W, Jaz, Zip, or MO drive for backing up projects. With removable media such as Jaz, Zip, or MO you can even back up a project across multiple disks with the DPS16. Another backup option is to use the S/PDIF digital I/O to back up projects to a DAT recorder. It's refreshing to find a stand-alone hard-disk recording system that actively supports multiple means of data backup.

Conclusions

Initially I found the DPS16 a bit cumbersome to work with. But after a few days, I started to get comfortable. I did frequently need to refer to the manual, which I found to be somewhat cryptic and frustrating.

On the positive side, the DPS16 really is a stand-alone, high-resolution studio-in-a-box — its 16 tracks, 250 virtual tracks, audio editing, mixing capabilities, and effects processor make it powerful enough to allow users to realize complex musical arrangements quickly and easily. Plus, the tracks I recorded on the DPS16 sounded fabulous — and isn't that what it's all about? ■

IN REVIEW SHORTS

MINDPRINT T-COMP

TWO-CHANNEL COMPRESSOR WITH TUBE SATURATION

By Mitch Gallagher

BASICS: The MindPrint T-Comp is a two-channel compressor that offers several "extras." Among these are variable tube saturation, insert points on each channel, a switchable 300 Hz sidechain filter, and defeatable Adaptive (signal-dependent) attack and release times. Input connections are "combi" balanced XLR/1/4-inch jacks. Two outputs per channel are offered, XLR and 1/4-inch; both can be used simultaneously. Each channel also has a 1/4-inch insert point for routing signal to and from external processors before it hits the compressor circuit. There is no external sidechain connection.

Housed in a single-rackspace chassis with a handy window to show off the glow of the tube, the T-Comp has a rear-panel slot that can accept optional 24-bit coax digital I/O (DiMod, $249).

PROS & CONS: After wiring the T-Comp up, the first thing I discovered is that it passes signal with power off! On investigating the manual, I found that the power switch indeed doubles as a hardwire bypass switch. In addition, each channel has its own bypass switch; this is great for dual-mono operation, but a pain for stereo signals. The bypass switches often click or pop when toggled; this is distracting when A-B'ing dry and processed signals.

The T-Comp can be used in either dual-mono or stereo (Link). In Link mode, Channel A's settings control both channels (although this isn't made clear in the manual). The unit uses a soft-knee compression approach. It would be nice to also have the option to switch to hard-knee processing at certain times. You can set the unit up manually (you're given control over threshold, ratio, attack, and release) or hit the Adaptive switch, which turns on signal-dependent attack and release time (the attack/release controls remain active). Adaptive worked well with lower ratios and light compression; with heavier compression, I had better luck setting things manually.

Sonically, I'd say the T-Comp falls into the "moderately colored" category — its compression adds some round lower-midrange fatness to signals. But there are two ways to push the unit into the "extremely colored" category: The Tube Sat control adds tube-generated upper harmonics to the signal when the threshold is crossed; an LED changes color to indicate how heavily the signal is being affected. If overdone on signals such as vocals, Tube Sat can get harsh and buzzy sounding — a little bit goes a long way. But on drums and percussion, the effect is excellent, making each hit jump out of the track with strong bite. Snare drum, in particular, was nicely enhanced by Tube Sat.

The second sonic tailoring option offered by the T-Comp is the Filter, which places a 300 Hz low-pass filter into the unit's internal sidechain, making the compressor less sensitive to low frequencies. This helps to balance bass-heavy signal's mid- and high-frequency response. Filter works well for tightening up the low-end of boomy tracks.

BOTTOM LINE: The T-Comp offers quality compression, good sound, and a number of useful extras. It ain't cheap, but if you desire controllable tube saturation with your compression, the cost is justified.

MINDPRINT T-COMP

PRICE: $1,099

CONTACT: MindPrint, dist. in the U.S. by Steinberg, 21354 Nordhoff St. #110, Chatsworth, CA 91311. Tel: 818-678-5100. Web: **www.mindprint.com**. *EQ* free lit. #112.

NATIVE INSTRUMENTS B4 VIRTUAL ORGAN

A SOFTWARE SYNTH FINALLY NAILS *THAT* SOUND

By Craig Anderton

BASICS: This Mac/PC plug-in, which also functions as a stand-alone application, models the sound of a Hammond B3 (including rotating speaker and tube preamp). It's compatible with Soundmanager, VST 2.0, MOTU MAS, and Digidesign DirectConnect (supports two output channels in dual mono; an update will support two mono and one stereo output). System requirements are reasonable — for the Mac, a 250 MHz PowerPC, 64 MB RAM, Mac OS 8.5 or higher (the manual says 8.0), and Opcode OMS or MOTU FreeMIDI and compatible MIDI interface; for Windows, 266 MHz Pentium, 32 MB RAM, Windows 9X/2000/NT 4.0, OS-compatible sound card, and MIDI interface. Copy protection

involves inserting the CD periodically for verification. Installing MIDI and audio connections was a cinch, as was running B4 as a plug-in under Cubase VST.

B4 works with ASIO on either platform, and supports DirectSound or (at the cost of greater latency) MME drivers on the PC. With ASIO, playing B4 was a very satisfying experience, with negligible latency and gobs of polyphony when using my 850 MHz overclocked Celeron.

The program includes 120 useful presets (10 banks of 12), and, like the original, you can save and recall programs with the inverse-color keys on the keyboard manuals. Speaking of manuals, the two manuals, as well as the bass pedals, can either respond to three different MIDI channels, or split from one channel. Regarding the *other* kind of manual, there is a printed version that's well written and concise.

PROS & CONS: This is one of those rare programs that has it all. The interface is highly literal, the software is easy to tweak, the sound is fabulous, and the emulation is spot on. You can use a MIDI fader box such as the Peavey PC-1600 to play with the "drawbars" — a real plus to hardcore Hammond fans. The keyboard even responds to velocity, in case you're one of those people who sampled your organ so you could have velocity control. And the voice-stealing algorithm is very clever: rather than steal notes, B4 steals the top partials of older sounds (each tone wheel is emulated individually). If you push polyphony to the limit (admittedly hard to do), you don't really notice anything missing, except perhaps a somewhat duller sound on some notes. A more off-the-wall feature is the ability to process signals through the distortion and rotating speaker. My wish list is short indeed: response to pitch bend and pressure (pressure is a wonderful controller for rotating speaker speed).

BOTTOM LINE: I've been to a lot of studios with a venerable B3, often lovingly maintained, off in a corner for availability on sessions. If you can't afford the real thing, or don't have the space, B4 is an uncannily satisfying substitute. I've heard a lot of organ emulations in my time, but this is the real deal.

NATIVE INSTRUMENTS B4

PRICE: $199
CONTACT: Native Instruments USA, 6477 Almaden Expy., Suite D2-F8, San Jose, CA 95120. Web: **www.native-instruments.com**. EQ free lit. #113.

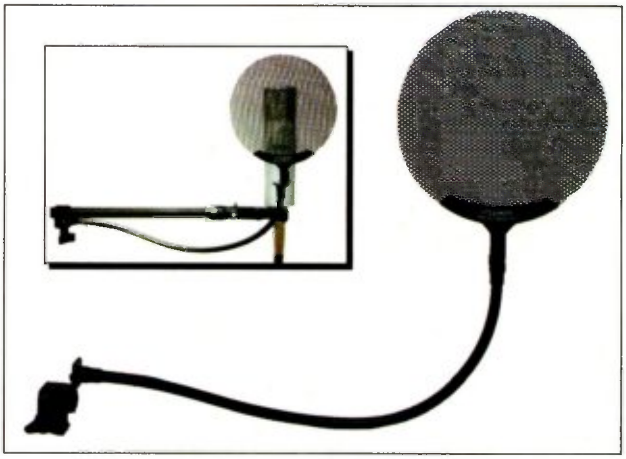
STEDMAN PROSCREEN PS101

HIGH-TECH POP SCREEN FILTER
By Mitch Gallagher

BASICS: Pop filters aren't an area where we've seen much high-technology development focused; for many recording engineers pantyhose stretched over a wire loop is considered cutting-edge. Stedman saw the need for a better, more professional pop filter. The result is the ProScreen PS101; a 4.6-inch diameter pop filter utilizing a patented "polarized" metal screen. The PS101 screen is attached to a 13-inch flexible rubber-coated gooseneck, which has a clamp for mounting to a mic or boom stand. (Also available for $49 is the ProScreen 100, the same pop screen filter as the PS101, but mounted to a threaded mic stand adapter instead of a gooseneck/clamp.) There's no "surround" or rim/frame on the PS101; just a circular sheet of wire mesh mounted to the gooseneck.

PROS & CONS: Stedman says the wire mesh in the PS101 is "polarized." By this I'm assuming they're referring to the fact that the mesh doesn't just disperse airflow, it redirects it in a downward direction, away from sensitive mic diaphragms. You can clearly verify that this is working; just blow through the PS101 while holding your hand on the other side. This "polarization" also means that there's one side of the unit that works well, and one that doesn't (the side with the Stedman logo is the "good" one).

I set up a comparison using a stock six-inch fabric filter and the ProScreen in front of an Audio-Technica AT4050cm5 mic. Some of Stedman's claims I don't buy (such as "increased comfort for the vocalist"), but others are right on the mark, such as



more effective pop elimination and sonic transparency. However, on severe pops (saying "puh" with an exaggerated "p"), the ProScreen added the barest hint of high-frequency "sss" to the tone that wasn't there with the fabric filter.

My other concern is durability; just because the ProScreen is metal doesn't mean that it's indestructible. I twisted a bit on the one I had, and the screen began to crack free from its mount. The metal mesh seems fairly resilient, but I'm betting you could bend it to where it would "crease" or wouldn't snap back (I didn't push my luck with the review unit). Under normal circumstances, it should be fine — just be careful no one sits or steps on it. (Stedman tells us the ProScreen is covered by a lifetime warranty, and that they've never had one fail.)

On the plus side, you're not going to accidentally push something through the metal mesh the way that you can with a fabric filter, and the ProScreen is *much* easier to clean and dry.

BOTTOM LINE: Fifty-nine dollars for a pop filter may seem steep, but keep in mind that, under normal use, the PS101 should last a lifetime. It's also easier to keep clean than a fabric screen. Most important, it really does seem to be more effective than fabric in diverting air blasts — and it does this in a sonically transparent way.

STEDMAN PROSCREEN PS101

PRICE: $59
CONTACT: Stedman, 4167 Stedman Dr., Richland, MI 49083. Tel: 616-629-5930. Web: **www.stedmancorp.com**. *EQ* free lit. #114.

SONY

SoundByte

FALL 2000
ISSUE #9

THE SONY PRO AUDIO NEWSMAGAZINE

www.sony.com/proaudio

Herbie Hancock Honored by Sony as a Media Master at AES 2000 in Los Angeles

Media Master

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Sony Music Studios Installs Second Oxford Console

Soundmirror Mixes NY Philharmonic on DMX-R100 • YES Uses DRE-S777

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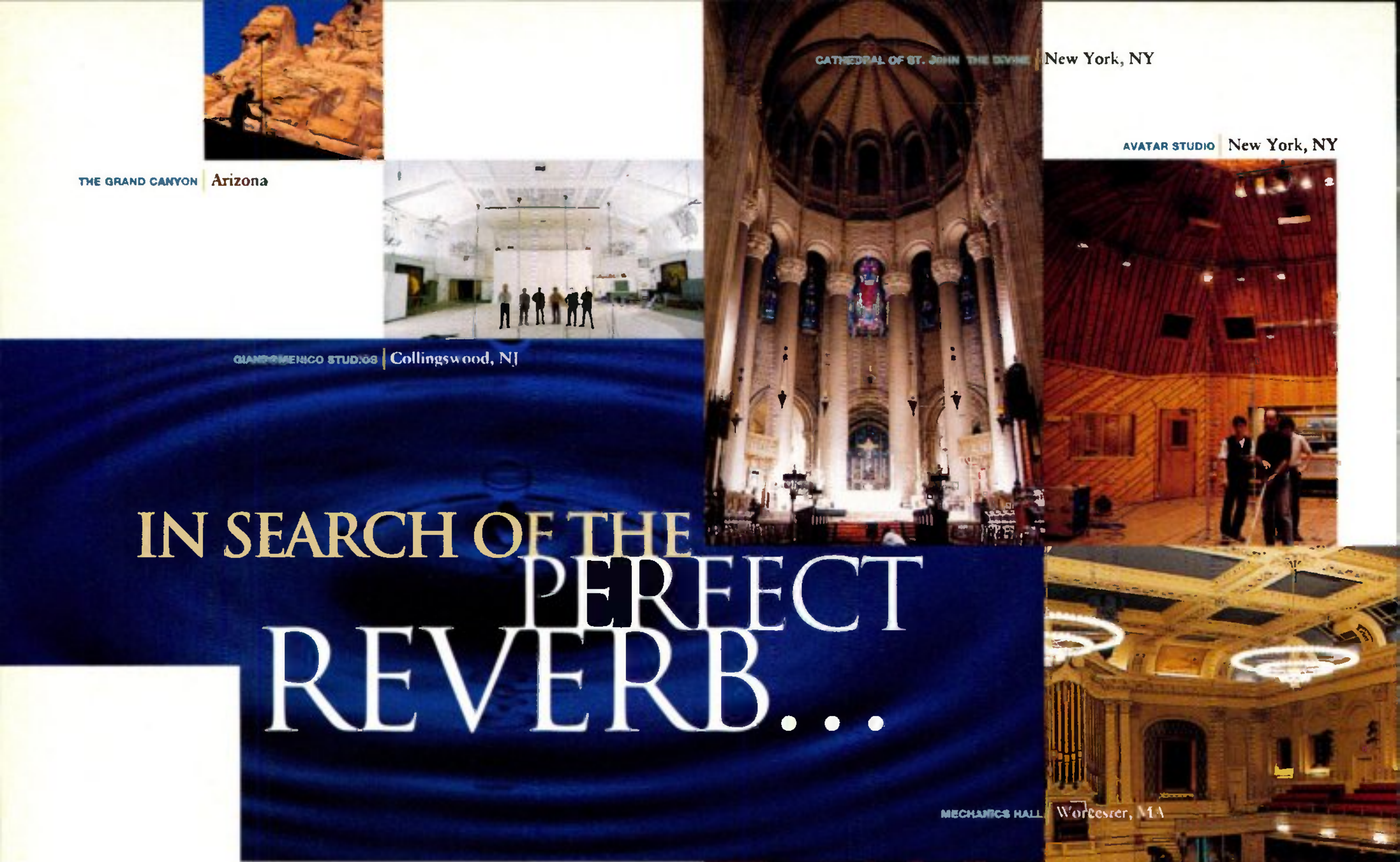

PRO AUDIO REVIEW 1999 REVIEWER'S PICK

DRE-S777

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

The DRE-S777 uses highly advanced signal processing to capture real sound spaces with incredible, detailed precision. The result? The most realistic reverb ever... because it recreates the "real" ambience of actual concert halls, cathedrals and studios. To quote Tom Jung from the 12/99 issue of Pro Audio Review, the DRE-S777 is "Second only to being there." An entire CD-ROM filled with some incredible sounding plates and spaces comes with the unit*.

A growing library of optional CD-ROMs is available including "European Halls & Churches" and the latest US release that includes some great American studios, churches, concert halls and the ultimate reverberant space, the Grand Canyon.

The proof is in the listening, so call 1-800-472-7669 ext. S777 today to order your **Free Demonstration CD and VHS video tape** and experience the difference between reverb effects and "real" space.

SONY

1-800-472-SONY ext. S777
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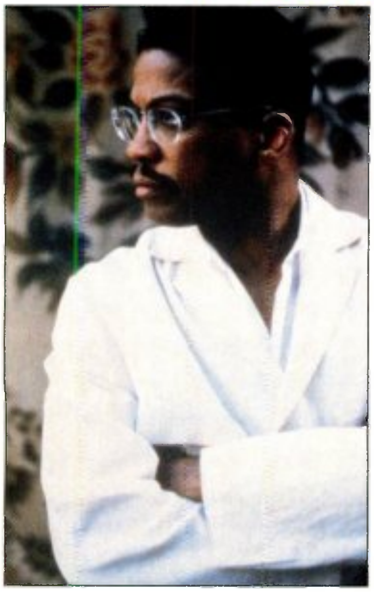
Herbie Hancock
Receives Media Masters Award



Commemorating 50 years of recording media innovation, Sony selected Herbie Hancock, platinum recording artist and keynote speaker of the Audio Engineering Society 109th Convention, as the recipient of the third Media Masters Award. The award presentation took place during the AES convention.

Created by the Media Solutions Company to celebrate Sony's 50th Anniversary in recording media, the Media Masters Award recognizes a select group of leaders in a number of industries for their pioneering creative efforts in the art of recording. Previous recipients include: CBS's Brian Fuss and *60 Minutes* creator and executive producer Don Hewitt.

A seven-time Grammy-winning jazz icon, Herbie Hancock recorded *Takin' Off*, his debut solo album which featured the top 10 hit "Watermelon Man," for Blue Note Records in 1963. Over the course of his career, Hancock has worked with a number of visionary artists, including the legendary Miles Davis. In 1973 he pioneered 'fusion,' an amalgamation of sounds which incorporate funk, rock, and jazz.

That year's platinum *Headhunters* album marked his first recording with a synthesizer and became the largest-selling jazz album in history. His score for the film *Round Midnight* won an Academy Award in 1986.

Audio Legacy

Al Jolson, Jr., owner of Masterlink Studios in Nashville, TN, (pictured below left with Sony's Art Gonzalez, Mike Poston of Equipment Pool, and engineer Chad Hailey), is carrying the high-tech audio legacy of his father into the 21st century. His studio recently acquired the Sony PCM3348HR digital multitrack recorder. The 48-channel, 24-bit system, which uses standard DASH tape, is proving to be a formidable bridge to the high-tech digital world, even for die-hard users of analog equipment.

Soundmirror
Mixes NY Philharmonic on DMX-R100

Blanton Alspaugh, engineer/producer at Soundmirror, a full-service recording/post-production facility in Sharon, MA, recently completed a mix project for the New York Philharmonic and Dawn Treader Productions (NY) with the studio's brand new Sony DMX-R100 digital console.

"The multitrack editing was completed here on a 24-track Sonic Solutions editing system," explains Alspaugh. "We then ran the material straight into the R100 and mixed it to a Sony PCM9000. The goal with an orchestral project like this is to achieve absolute sonic transparency. What you get out of the system has to equal what you put in with no changes in detail or ambience. On that score, the R100 really delivered."

The music from the New York Philharmonic will be featured on a CD for the American Cancer Society titled *Music of Hope*. Scheduled for release in early 2001, the CD contains additional musical selections from the London Symphony Orchestra, conductor Andre Previn, Billy Joel and Paul McCartney. Shown with the DMX-R100 in Soundmirror's Studio C are (l-r): Larry Rock, audio director, New York Philharmonic, Alspaugh, and Matt Singer, project producer, Dawn Treader Productions.

YES Masterworks Tour: All Good

Sony's Tim Derwallis (left) joins FOH mixer Dave Wilkerson of Clair Brothers in the control booth at the Jones Beach stop on the YES tour.

With a history that dates back over 30 years, and over 30 albums to their credit, YES has clearly carved out their well-deserved place in the musical history books. Always striving to achieve the finest sound both in the studio and live, the band used a new reverb on their sold-out summer Masterworks tour: the Sony DRE-S777.

Performing with a classic lineup that includes Steve Howe on guitar, Chris Squire on bass, Jon Anderson on vocals, Alan White on drums, and recent newcomer Igor Khoroshev on keyboards, the band called upon long-time Clair Brothers talent Dave Wilkerson to handle the FOH mixing chores. During the course of a career that includes working with such acts as U2, Fleetwood Mac, .38-Special, and The Tubes, Wilkerson had used almost every piece of gear out there. Then he got his hands on the Sony S777.

"My first impression was that it was the best sounding reverb, 'out of the box,' I have ever heard," he states. "It's really incredible sounding. Without having to tweak it at all, it gave me a three-dimensional sound through two speakers."

SoundByte caught up with Wilkerson at a recent show at New Jersey's PNC Bank Art Center, which was recorded by the Sheffield mobile truck to a Sony 3348 for a recent radio broadcast.

"I'm using the 'Concertgebouw' sample on Alan White's drums, and 'Musikvereinsaal' from Vienna on John Anderson's voice," explains Wilkerson. "The S777 is configured for mono in/stereo out, patched directly into the console. It's great that I have two stereo reverbs in one unit."

The S777 features four channels of analog output and, with the optional DSP card, it can handle a full four channels of AES/EBU digital output as well.

Will Alexander, keyboard technician and engineer for YES, along with Emerson, Lake, and Palmer, Herbie Hancock, and other notable keyboardists, also has high praises for the S777.

"I was speaking with Jon Anderson, and told him that I had heard this Sony Digital Sampling Reverb that sounded 'god-like'," explains Alexander. "He responded that he would like to hear a 'god-like' reverb. I then got in touch with the guys at Sony, and they kindly provided us one for the YES Masterworks Tour."

Alexander comments on what happened the day they received their new S777: "When Dave Wilkerson, FOH Engineer for YES, heard the S777, he was also blown away. He just unplugged the high-end digital reverb that was being used for the tour, and put it back on the truck — where it resided for the rest of the shows."

Interestingly, Alexander also notes that during his years working for Fairlight Instruments, he felt that the future of digital technology would bring about software-based devices that would revolutionize digital signal processing beyond the then current hardware-based technologies.

"One day a friend of mine showed me some software that mapped the floor of the ocean, giving a 3D terrascape of the con-

'Verbs

The band explores new sonic dimensions on the road with the help of Sony's DRE-S777 Digital Sampling Reverb.

tours," Alexander remembers. "I immediately responded that it would be great if it were possible to map existing audio environments, such as concert halls, cathedrals, or even the Grand Canyon. I had known that others were trying to achieve the mapping of an audio environment, but when I heard the S777, I said 'It's about time.'"

Now that the band had a new reverb in the racks, it was time to put it to the test.

"After the show on the first night I used the S777, Alan White had commented on the change in the sound of the drums, and asked what I had done," states Wilkerson. "I told him I have a new reverb. He noticed it from onstage, and heard that it had made a big difference."

The acoustic environments used by Wilkerson on the YES tour come from the European Halls & Churches CD-ROM.

Wilkerson, who also owns Right Coast Recording in Lancaster, PA, comments that he also liked the S777 because he feels its approach is different than other reverbs. "It's so much more than a generic room sound that you have to constantly adjust to your needs," he states. "This reverb actually has real rooms that you can just dial up. They are great sounds to start with, so you don't really need to go in and adjust anything. YES sounds great with the S777, and I can't wait to get my hands on the American Acoustic Spaces disc!"

New Software For The DRE-S777

The DRE-S777 Digital Sampling Reverb introduced by Sony Electronics last year is being enhanced with new optional software, including the DASK-S704 Sampling Function Software and two CD-ROM-based Sampling Reverb Software packages: DASK S702, "American Acoustic Spaces," and DASK-S703, "Japanese Acoustic Spaces."

The DASK-S704 Sampling Software harnesses the DSP power of the S777 to allow users to sample their own spaces and store them on Memory Stick media.

"The potential applications are limited only by the imagination," offers Paul Foschino, marketing manager for Sony Pro Audio. "Studios can capture their 'A' room and add it to overdubs done in a vocal booth or one of their other rooms. Film and video production people can capture the ambience of unique locations such as train stations or warehouses and use them later to match up ADR or Foley tracks with production sound. Live engineers can add the ambience of a concert hall to enhance the acoustics of a small theater or a club. The list of possible uses is almost endless."

DASK-S704 key benefits include:

• Reverb data may be uploaded and downloaded into a Memory Stick storage devices from the memory in the DRE-S777

• Large capacity Memory Stick media (64 MB for about 15 stereo samples)

Following the release of the DASK-S701 software, "European Halls & Spaces," late last year, the newest additions to the DRE-S777's growing support library, DASK-S702 and DASK-S703, offer a diverse mixture of pre-recorded spaces.

The "American Acoustic Spaces" disc includes reverb sample algorithms of well-known recording studios and halls. Among these are Oceanway Studios and The Enterprise in Los Angeles, Avatar Studios in New York, and Gian Domenica Studio in New Jersey. It also includes the Cathedral of St. John the Divine, Mechanics Hall in Worcester, MA, and the Grand Canyon.

The "Japanese Acoustic Spaces" disc offers unusual and fascinating spaces, including the Goto Planetarium Dome in Tokyo, the Hotaka mountain range, and a bathhouse named "Tamanoyu," also in Tokyo.

"Until now, the available sampled spaces were primarily concert halls and churches, most with pretty long reverb times," states Foschino. "With the introduction of the American disc, we now have a number of great sounding studios, with medium-decay-time and bright reverb characteristics that are perfectly suited for pop music production. And the Japanese disc adds some truly exotic ambient spaces that take the system to a new level."

Both the "American" and "Japanese" discs come with detailed documentation on the sampled spaces and an individually coded Memory Stick key that allows secure operation of the sample disc.

"These new discs support our plan to provide users with a diverse and growing array of sounds and locations," Foschino reports. "Building a library of sampled spaces around the world is time consuming and challenging, but the results are well worth the effort. These sampled spaces have a complexity and depth that set them apart from standard synthesized reverb. The S777 is the most natural sounding reverberation system we've ever made."

The DASK-S702, S704, and S704 are now available at a suggested list price of $930 per disc.

The Seco

When Sony Music Studios needed a versatile workhorse of a console for a recent studio upgrade, they (again) chose the OXF-R3 Oxford digital board.

Room 311, Sony Music Studios

At the corner of 54th Street and 10th Avenue on Manhattan's West Side, Sony Music Studios is perhaps New York City's single most impressive production facility. Considering the sheer volume and diversity of activity here, to call its range of services comprehensive seems like an understatement. From music recording, mixing, mastering and remastering, to live performances, video shoots, editing, and post-production for film and television, Sony Music Studios offers every conceivable service for the creation of entertainment content.

Given the staggering amount of work taking place every day at Sony Music Studios, the facility's versatility and technical excellence is essential. It's no surprise, then, that when the decision was made to upgrade Studio G, the console of choice was an OXF-R3 Oxford digital board. Installed in early Summer 2000, the OXF-R3 in Studio G is the second at Sony Music Studios, joining the Oxford in Room 311 in the classical music division.

In the last four years, DVD and surround sound have become the buzzwords in both the professional audio and consumer electronics industries. The theme of September's 109th AES Convention, in fact, is "Surrounded by Sound." DVD, furthermore, is the most successful product introduction in the consumer electronics industry's history. Offering convergence of high-resolution audio and video — and, of course, capable of delivering multichannel audio — DVD is a hit with consumers, more and more of whom are both equipped with sophisticated home theater systems and hungry for entertainment content.

For Sony Music Studios, where so much content is already generated, this equates to an even busier production schedule. And for Studio G, primarily — but not exclusively — an audio post-production suite, it means being equipped with a console that can handle whatever is brought to it.

"We do pretty much every facet of audio for television, film, and video," explains Michael Fisher, senior engineer, Sony Music Studios. "In the last four years, we've been extremely involved in DVD, and that is a good portion of what we do in this room now. We use a lot of surround capabilities of the Oxford, as well as stereo."

Adds Susan Pelino, director of audio post production, Sony Music Studios, and one of this year's TEC Award nominees in the category of Audio Post-Production Engineer for Television: "The room needed to be versatile, to handle a very, very wide spectrum of both music, documentary, and other television shows and specials. We needed a desk that could handle virtually anything we put in the room, including music mixing as well."

Case in point: *Sessions at West 54th Street*, a wildly popular

nd Coming

by Christopher Walsh

program featuring intimate performances by top artists (from the main stage at Sony Music Studios) is produced entirely in-house. Three seasons of *Sessions* have already aired, providing an abundant catalog of memorable performances. Recently, it was announced that selected episodes of *Sessions* would be released on DVD. Already, two "Best Of" compilations of *Sessions* performances have been mixed in 5.1 surround for DVD release. Sony's main stage has also hosted a variety of programs, including VH1's *Storytellers* and *Hard Rock Live*, *MTV Unplugged*, and *Live By Request*, which airs on the A&E Network.

"We've been doing so much music here at the studio," says Pelino. "A lot of the shows Mike and I have already mixed are now coming back to us for DVD."

Considered by many elite engineers and producers to be very well suited for surround mixing, the Oxford lends itself especially well for DVD projects, a growing source of revenue for high-end recording facilities. But audio professionals have more to say about the Oxford than its formidable flexibility and power. Regardless of geographic location or musical genre, time and again one will hear producers and engineers marvel at the Oxford's sonic superiority. "At Sony Music Studios, an exhaustive study was made of every large-format digital console on the market," explains Brian McKenna, director of audio operations and marketing, Sony Music Studios. "We took 'the temperature', of a lot of outside engineers, as well as our in-house engineers," McKenna recalls. "They pretty much all voted for the Oxford, as did the clients — the producers. You've got to go with that."

Studio G,
Sony Music Studios

As Pelino noted, Studio G has to be able to service traditional music mix projects as well as audio post-production, and, to that end, a pair of custom Augspurger monitors have been installed in addition to the Miller & Kreisel surround array. Studio G is also configured with a Pro Tools rig, the digital audio workstation now employed on a healthy majority of album projects.

"A lot of hip-hop and pop producers are primarily recording direct to Pro Tools now," McKenna points out. "And Pro Tools is 24-bit, so all they have to do is bring in their files or their own Pro Tools system, connect it directly to the Oxford, and begin mixing. We can make sure the systems match, they can bring their file, put it in, and they're good to go. It really works well."

If you are tempted to think that Sony Music Studios is under pressure to purchase Sony professional audio equipment, think again. Twenty-four rooms — yes, you read that correctly — are dedicated to audio at Sony Music Studios, and several other manufacturers are represented in the console installations here.

"Sony Music Studios is a very independent facility," emphasizes Courtney Spencer, VP, Professional Audio Group, Sony Electronics. "They buy what is right for the facility. Some people say, 'Sony studio, Sony console.' But, in fact, they buy what is right for them, as evidenced by the fact that they have other digital and lots of other analog consoles here."

David Smith, VP, Engineering, Sony Music, confirms this. He, along with Senior VP, Sony Music Entertainment Al Smith and engineers Fisher and Pelino, spent the better part of six months researching the many digital console choices available. "All along, Al's thinking was to be able to do music in the room, in addition to post," Smith notes. "The trouble with post consoles is that they're stripped down, they don't have all the bells and whistles that music mixing consoles have, primarily to do with signal processing. The Oxford has several different equalizers and dynamics sections that you can pick from. And I'll say this on the record: this desk is the best-sounding. Every desk has its attributes, but sonically, this is head-and-shoulders above."

"What David said is very true," Fisher adds. "I couldn't see any other digital console bridging the gap between the various format responsibilities that we have. I know it's satisfying the engineers in the classical music division. The flexibility of the EQ and compression is unmatched in the other consoles we looked at. The routing is extremely flexible. It all goes back to the fact that everything we looked for in a large console is here, even beyond what we would expect."

With several surround mixes for DVD already completed for artists, including Gloria Estefan and Bruce Springsteen, Studio G is heavily booked as a busy fall season heats up. Reflecting on the technical capabilities and the resulting diversity of work in the room, one realizes that Sony Music Studios' Oxford-equipped Studio G is, in fact, a microcosm of the larger facility. Sony Music Studios is a powerhouse in the recording business, and, as such, the best equipment is necessary.

"Having worked in studios all over the city," Fisher concludes, "I can say this studio is a unique environment. I don't know of any place that covers as broad a spectrum as Sony Music Studios."

Christopher Walsh is a New York-based writer and musician who has written for *Billboard*, *Pro Sound News*, *Gig Magazine*, *Videography*, and *Medialine*.

Sony Delivers at AES 2000

Sony launches a full array of versatile new products on the show floor at the Los Angeles Convention Center.

WRR-862B

Sony Unveils CD Recorder Offerings For The Pro Audio Market

At AES 2000, Sony Electronics is unveiling its first two CD recorders for pro audio applications. The CDR-W66 is designed for mid- to high-end recording studios and broadcast production (television and radio); the CDR-W33 targets more cost-conscious users, but offers most of the capabilities of its higher-priced sibling.

"Both units incorporate several unique features," offers Courtney Spencer, vice president, Professional Audio Group, Broadcast and

CDR-W66

Professional Company, Sony Electronics Inc. "These include selectable DSP functions like Parametric EQ, Limiter & SBM (Super Bit Mapping), and high-quality, 24-bit AD/DA conversion."

Physically similar, and sharing many of the same features, the innovative CDR-W66 and CDR-W33 offer CD-TEXT™ support, which allows disc/track names to be displayed and entered from the front panel AMS controller, the supplied remote control, or an optional PC keyboard. Remote transport control can also be accessed via Control-S or a PC-compatible keyboard. In addition, the CD recorders include a wireless/wired remote unit. The CDR-W66 will be available in January 2001; the CDR-W33 will be available in early October for a suggested list price of $799.

CDR-W33 Additional Features

- 32 kHz – 48 kHz built-in sampling rate converter
- Recordable and re-recordable recording media support
- FL display
- I/Os equipped with Coaxial Digital, Optical Digital, Analog Unbalanced phone jack
- 2U rack-mountable size in EIA.

In addition to functions found on the CDR-W33, the CDR-W66 offers such key features as:

- Word Clock interface
- 32 kHz – 96 kHz sample rate converter range
- Selectable SCMS modes
- AES EBU digital I/O, balanced XLR analog I/O
- RS-232C and parallel (GPI) control ports
- 2X speed duplication link for dubbing audio titles (using two CDR-W66 units)
- DSP functions available on digital inputs as well as analog

New Wireless Components

Building on the success of the 800 Series UHF Wireless Microphone System, Sony Electronics is introducing the latest additions to the wireless microphone family: the newly developed WRT-847B UHF Synthesized transmitter unit, its interchangeable microphone heads, and the WRR-862B UHF synthesized Dual Diversity Tuner.

WRT-847B

“Compatible with Sony’s existing receiving systems, the WRT-847B allows flexible simultaneous multichannel operation and operates over a 24 MHz frequency band,” explains Spencer. “Suitable for a wide range of broadcast, concert sound, and other applications, the WRT-847B is a really versatile product.”

Five types of microphone heads are available for use with the WRT-847B. The CU-F780, CU-G780, and CU-E700 optional microphone capsules are designed for vocal applications such as broadcasting and live concerts. The CU-E672 and CU-F117 microphone capsules are intended for interviews in news gathering and field productions.

The WRT-847B transmitter unit offers several important key features, including: selectable RF output level (10 mW for multichannel operation and 50 mW for long working distance); audio gain and attenuation setting from +9 dB to -12 dB in 3 dB steps; and an easy-to-read LCD that indicates extensive information on operating conditions such as channel number, wireless channel frequency in MHz, audio input level, compander time constant, battery status, and accumulated operating time.

The compact new WRR-862B unit also operates over a 24 MHz frequency band and it has two built-in tuner modules to meet the demand for two-channel reception in ENG and EFP applications. Designed so that it can be easily mounted on Sony cameras, the tuner’s magnesium diecast body is extremely lightweight and rugged.

“The WRR-862B can simultaneously receive two independent signals on two separate channels, “reports Spencer. “The space diversity system is employed to eliminate signal dropout and provide stable reception.”

Two SMC9-4S (Sony 4pin) audio output connectors are provided on the top panel.

“Since its introduction, Sony’s 800 Series system has been well accepted in the market for its wide audio dynamic range, low noise characteristics, stable signal transmission and reception,” concludes Spencer. “We are pleased to build upon and strengthen the wireless series.”

Additional Features

- LED and LCD indication to provide extensive information
- Long operating time-approximately five hours of continuous operation provided by four AA-size alkaline batteries
- Switchable RF squelch which can be easily turned ON and OFF
- Monitor jack for monitoring the output sound

Pro MiniDisc Recorders

Sony Electronics is also debuting two 1U-high rack-mountable MiniDisc recorders. The MDS-E10 and the MDS-E12 incorporate the latest ATRAC type “R” algorithm for superior sound and provide a host of new options. The new units replace the earlier 2U-high MDS-E58 and the MDS-E11.

Sharing many of the same characteristics, the MDS-E10 and MDS-E12 feature: 10 “Instant Start” memories that allow immediate playback of any 10 tracks; SPDIF coaxial and optical digital I/O, as well as analog RCA I/O; Long REC/PLAY (Max. 320 min.) using ATRAC3 REC mode; and versatile menu control of various functions including: HOT START, AUTO CUE, AUTO PAUSE, SOUND START PAUSE, VARISPEED, NEXT TR RESERVE, LONG REC MODE (320 min), AC TIMER REC, and DIGITAL REC LEVEL ADJUST.

“The MDS-E10 is ideal for radio broadcast and DJ applications,” states Paul Foschino, marketing, Professional Audio Group, Broadcast and Professional Company, Sony Electronics Inc. “The ‘Instant Start’ option stores the very beginning of the audio in RAM on up to 10 tracks which is great for triggering samples. Both models have pitch control as well. The MDS-E12 incorporates several additional options such as analog XLR I/O which makes it a higher-end recorder for broadcast pros, system contractors, and studio users.”

The MDS-E10 will be available in early October at a suggested list price of $599. The MDS-E12 will be available in November at a suggested list price of $899.

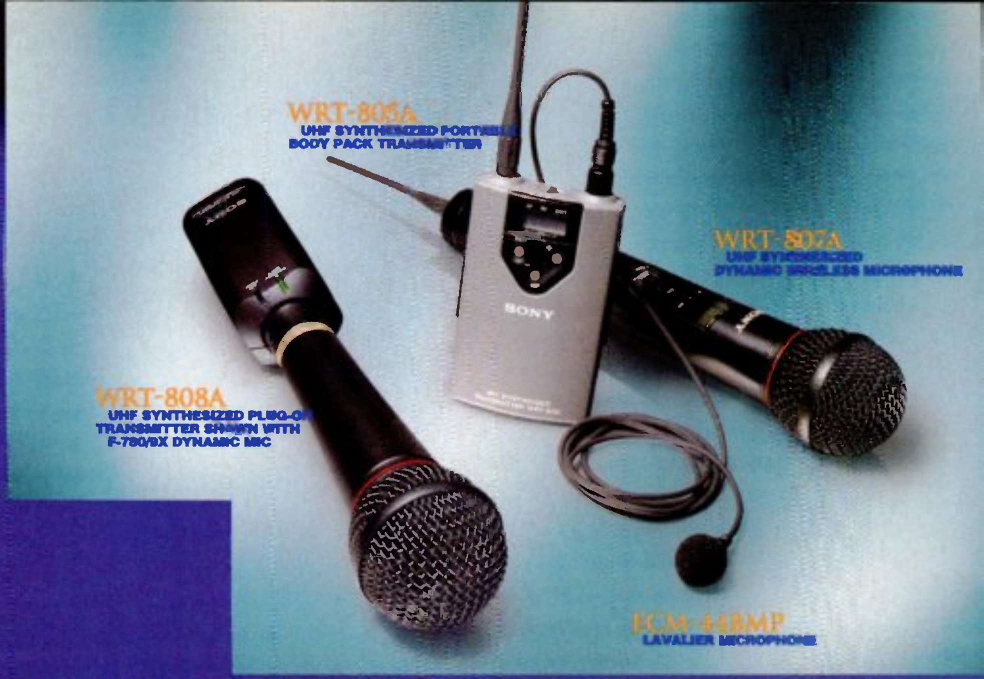
MDS-E10 and the MDS-E12 Key Features

- Transport controls can be accessed via Control-S (with supplied wired/wireless remote)
- RAM EDIT, which allows temporary, non-destructive editing of TOC files
- A-B ERASE
- Timer PLAY function
- PS/2 port on front panel, which allows a PC Keyboard operation
- Easy-to-read florescent display

Supplementary MDS-E12 features

- Cascade REC/PLAY functions accessible by using the control relay jack
- Built-in Parallel (GPI) port allows control from an external controller such as fader start. (An RS-232C port allows control via PC)
- Extremely long recording is available when using ATRAC3 REC mode and Relay REC mode combined on the MDS-E12

MDS-E12



Bay Roads Marketing Wins Sony Triple Crown Dealer Award

Luke Furr, president of Bay Roads Marketing in Sharon, MA, has been awarded all three of Sony's major pro audio rep awards for sales during the most recent fiscal year. Representing Sony professional audio products in the Northeast, Bay Roads took the Top Gun award in New York for selling the largest dollar volume of products; the Star Achiever award in New England for greatest percentage over quota; as well as Rep Of The Year based on their strong marketing efforts and dealer support. Bay Roads Marketing previously won the Top Gun award for fiscal years 1997 and 1998. However, this marks the first time that a single rep company has swept all three award categories. The honors were presented at a ceremony during Sony's annual Remarkable Rep event.

"In winning this 'Triple Crown,' Bay Roads has set a new standard for Sony reps," comments Courtney Spencer, vice president of Sony Professional Audio. "We are delighted with their record-breaking accomplishment."

Bay Road's Luke Furr (center) accepts award from Sony Pro Audio VP Courtney Spencer (left) and marketing manager Paul Foschino.

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SoundByte
Winner
1998 MERCURY AWARD
1999 BIG APPLE AWARD

SoundByte

Back Issues

Fall 1998 (Premiere Issue A-1011): AES 1998. Cover Story: Tony Bennett Uses Oxford For *The Playground* CD. Other Stories: Jorgenson and Johnstone Use Sony MDM-X4 on the Road. 24-Bit Digital Recording by Dan Daley. The Sound of Sundance by Philip Himberg. Q&A with Tom Jung. Plus: New Products at AES '98.

Winter 1999 (Issue #2 A-1027): Winter NAMM 1999. Cover Story: Jimmie Vaughan Goes Wireless. Other Stories: LSD on MDM-X4. Second Oxford at Ocean Way. Rolling Stones Engineer Uses Sony Wireless on Bridges to Babylon Tour. Orlando Opera Uses MDM-X4 for Local Radio Broadcasts. Oxford Scores Touchdown at HD Football Broadcasts. Plus: New Products at Winter NAMM '99.

Spring 1999 (Issue #3 A-1033): NAB 1999. Cover Story: The Vegas Sound. Other Stories: Muriel Anderson New Album. Barking Doctor Studio Oxford. The Massenburg Sessions. Wireless in the DTV Age by Howard Massey. Year of the Oxford: NMT Broadcasts Chinese New Year in HD. Plus: New Products at NAB '99.

Summer 1999 (Issue #4 A-1037): Summer NAMM 1999. Cover Story: James Brown Records Concert in Sony 24-Bit. Other Stories: Hot Summer Tours by Gary Eskow. Jimi Hendrix Red House Tour. Wynton Marsalis and Steve Epstein use Oxford for New Albums. New Staples Center to Feature Oxford. Plus: New Products at Summer NAMM '99.

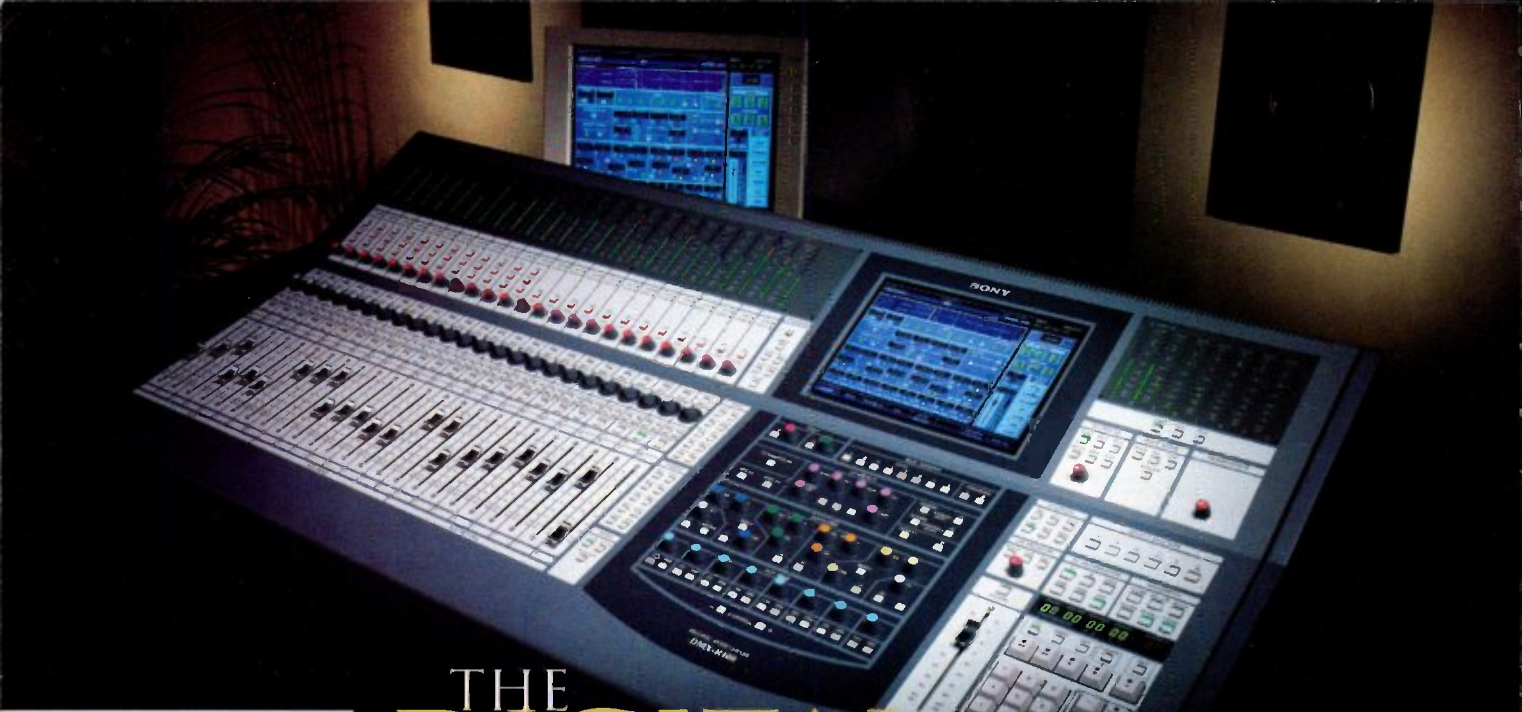
Fall 1999 (Issue #5 A-1040): AES 1999. Cover Story: Oxford Arrives at Ocean Way LA. Other Stories: WGN Radio Goes MiniDisc. Blair Witch Project Records Creepy Sounds on Sony DAT. Cher Tour Believes in Sony DPS-V77. Sony Wireless Products Rock the Hall of Fame. Tom Jung Reviews DRE-S777. Super Audio CD. Plus: New Products at AES '99.

Winter 2000 (Issue #6 A-1051): Winter NAMM 2000. Cover Story: Announcing DMX-R100, A Special Preview by Roger Nichols. Other Stories: Drummer Nigel Olsson Uses MDR-7505 & 7509 Headphones. Smooth Sailing With Sony Wireless at America's Cup. DRE-S777: The U.S. Sampling Tour Pictorial. NMT Uses Oxford at Village Jazz Festival. Plus: New Products at Winter NAMM 2000.

Spring 2000 (Issue #7 A-1054): NAB 2000. "The Ultimate Oxford Issue." Cover Story: Oxford Broadcasts at Staples Center. Other Stories: Oxford Studio Spotlight: The Hit Factory Criteria Miami and Loud Recording. Dave Reitzas Meets the Sony Oxford. Plus: Nashville Welcomes Sony DMX-R100. WDIV-TV NBC Goes Wireless. S777 Rises at Crescent Moon.

Summer 2000 (Issue #8 A-1063): Summer NAMM 2000. Cover Story: Ricky Martin – "Livin' La Vida Wireless." Other Stories: DMX-100 in the field. Al Jolson Jr.'s Masterlink Takes Delivery of PCM-3348HR. Mick Guzauski on DRE-S777. Sony co-sponsors All Star Guitar Night. Plus: New Products at Summer NAMM 2000.

To order back issues, please call: 1(800)472-SONY, ext. SoundByte



THE DIGITAL MIXER WITH AN OXFORD EDUCATION

DMX-R100

Full input module with a knob for every function

5.1 surround mixing and monitoring

96/24

96 kHz 24-bit capable

Introducing the Sony DMX-R100: a small-format digital mixer inspired by our Oxford console, considered by many industry leaders as the most advanced digital mixing system ever developed.

How does the DMX-R100 work? The way you want it to. You have a full input module with a knob for every function. Equalization and dynamics can be adjusted simultaneously. Your hand goes intuitively to the right knob. Your mixing session goes faster. You can concentrate on the mix, not on the technology.

The R100 can memorize your automation moves the moment you touch the high-resolution touch-screen fader. Don't tell the mixer to change modes. Don't think about it at all. Just touch it.

A color touch-screen is built into the control surface. Use the built-in router to assign inputs to faders. Select buses, sends, and directs to analog and digital outputs. View a complete input module or zoom in on the EQ and Dynamics sections.

Machine control with 9-pin and MMC interface is standard.

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27	28	29	30	31	32	33	34	35	36	37	38	39
40	41	42	43	44	45	46	47	48	49	50	51	52
53	54	55	56	57	58	59	60	61	62	63	64	65
66	67	68	69	70	71	72	73	74	75	76	77	78
79	80	81	82	83	84	85	86	87	88	89	90	91
92	93	94	95	96	97	98	99	100				

EDITORIAL

101	102	103	104	105	106	107	108	109	110	111	112
113	114	115	116	117	118	119	120	121	122	123	124
125	126	127	128	129	130	131	132	133	134	135	136
137	138	139	140	141	142	143	144	145	146	147	148
149	150	151	152	153	154	155	156	157	158	159	160
161	162	163	164	165	166	167	168	169	170	171	172
173	174	175	176	177	178	179	180	181	182	183	184
185	186	187	188	189	190	191	192	193	194	195	196
197	198	199	200	201	202	203	204	205	206	207	208
209	210										

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53	54	55	56	57	58	59	60	61	62	63	64	65
66	67	68	69	70	71	72	73	74	75	76	77	78
79	80	81	82	83	84	85	86	87	88	89	90	91
92	93	94	95	96	97	98	99	100				

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101	102	103	104	105	106	107	108	109	110	111	112
113	114	115	116	117	118	119	120	121	122	123	124
125	126	127	128	129	130	131	132	133	134	135	136
137	138	139	140	141	142	143	144	145	146	147	148
149	150	151	152	153	154	155	156	157	158	159	160
161	162	163	164	165	166	167	168	169	170	171	172
173	174	175	176	177	178	179	180	181	182	183	184
185	186	187	188	189	190	191	192	193	194	195	196
197	198	199	200	201	202	203	204	205	206	207	208
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ADINDEX

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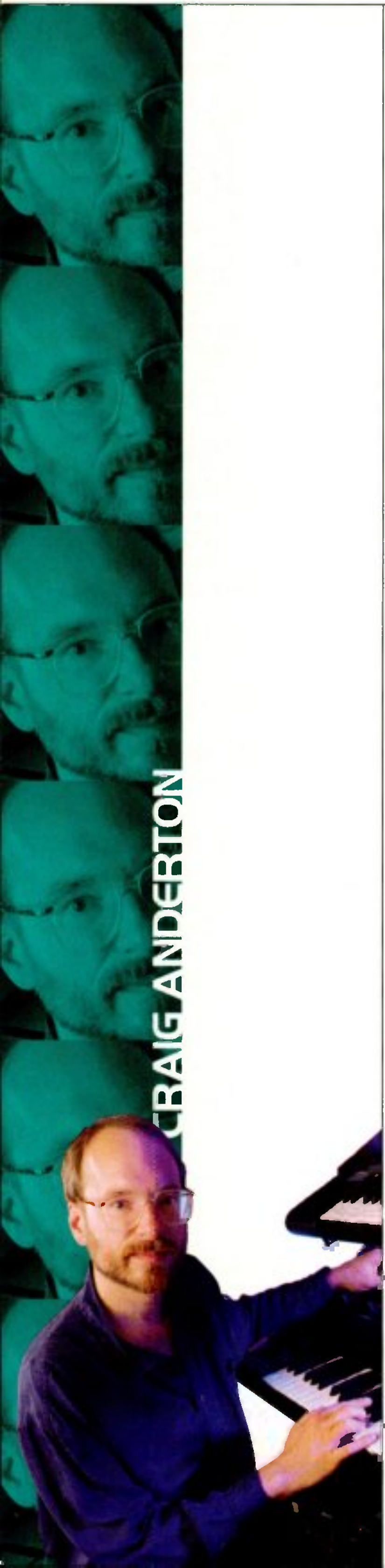
In the audio world,
Fall means one thing: another AES show

Trends at AES



For some, fall represents the start of the football season. For others, it's back-to-school time. But for those who actually care about variable bit rate encoding, tube transconductance, and whether higher sample rates really do make an audible difference, the coming of fall means only one thing: another Audio Engineering Society convention. This year, it was held in Los Angeles in late September.

The official show theme was "Surrounded by Sound," but I think the theme should have been the "Going to Be Big" show — as in, "Surround is going to be big, once people figure out where to put the speakers," or "24/96 is going to be big, once DVD-Audio takes off." There's also "Online distribution of music is going to be big, once broadband is in place," "The G4 is going to be really big, once companies optimize their software for dual processors and Apple figures out how to stream audio reliably over USB and FireWire," and, for the hardcore two-inch analog freaks, "Hard-disk recording is going to be really big, once there's a common file format."

Fact is, we're in a holding pattern. The planes are circling the airport, waiting for instructions on when conditions are favorable to land. But the hard truth is that, to stretch the analogy further, some of the planes are going to run out of fuel before they get clearance.

Take 192 kHz sample rates. One of the most interesting demos at AES was a comparison of analog, DSD (Direct Stream Digital), and PCM digital audio at 44.1, 48, 88.2, 96, and 192 kHz sampling rates. Unfortunately, the demo process was seriously flawed — instead of looping a section of music for comparison, a single piece of music played all the way through, with the different formats switched in and out. Even so, my take on the differences were pretty close to that of others whom I asked for opinions: 44.1 can sound brittle with high-amplitude, high-frequency passages; 48 didn't sound all that different from 44.1; 96 was more transparent; 88.2 wasn't that different from 96; and DSD was indeed close to analog in high-frequency "sweetness." As to 192, no one I talked to could hear, or feel, a difference compared to 96. That's not to say that with different material, on a different day, there might not be a difference — but we're talking quantitative, not qualitative, differences at best. It seems that 192 kHz may be technology whose time doesn't need to come.

I came prepared to be skeptical about DSD as well as 192, but left convinced that DSD is indeed a "better kind of digital." Will consumers flock to it? Doubtful, at least for the foreseeable future, because the player price will be high. But I think Mytek had the right idea when they billed their DSD four-channel hard-disk recorder as the way to finally break free from your analog half-inch. Of course, you might be using half-inch tape for its distortion characteristics, not its analog "sweetness," but that's another issue. I can easily see DSD-based machines supplanting DAT, because that format ("it's soooo 20th century") is clearly on its way to retirement in funny-format heaven.

The Ghost of AES Past

On the other hand, some of the promising technologies of previous shows are now firmly established. Twenty-four bits is a given, and its benefits are obvious — even 16-bit CDs sound better when prepared from 24-bit masters. Software synthesizers, which only a few years ago produced a few low-fidelity voices with unbearable latency, are now viable thanks to faster computers. Hard-disk recording has indeed taken over all but the niche market still owned by analog two-inch 24-track machines and high-end digital multitracks (*e.g.*, the Alesis M20). And Apple, despite its seeming cluelessness about the needs of pro audio, has made a bona fide comeback; their computers are once more a fixture at the show.

Physical modeling is another technology that's come into its own, as anyone who has heard the Native Instruments B4 (Hammond B-3 emulation software) or Line 6 Pod can attest. Even mics and speakers have gussied up their act in recent years, thanks to the advent of new technologies and design processes.

Surrounded By Sound — Or Apathy?

But these past successes do little to ease anxiety over the future. Surround is probably the biggest question mark. On the plus side, DVD has really taken off as a consumer format, and most of those machines are at least surround-ready, if not already containing decoders. And people who have surround love it; the difference compared to stereo is compelling and dramatic. Nonetheless, this

popularity has yet to translate into a push for DVD-Audio, possibly because it's hard to set up the typical living room for surround anyway (I'm sure most people are listening to their DVDs over stereo speakers). Besides, consumers may not be itching to sink bucks into more speakers and amps — let alone a subwoofer that may take up as much living room space as a coffee table.

Sure, surround-oriented audio sounds cool, but record companies are dragging their feet, too. There was supposed to be major DVD-Audio launch last Christmas, but now it looks like there won't be a big push for this Christmas season, either.

Bringing It All Back Home

So what does all this mean to the project studio owner? Well, we're pretty much in a holding pattern here also. You don't need to upgrade to surround immediately, although if your old mixer is about to retire, it would be prudent to replace it with a surround-capable model. Nor do you have to drop everything and go 24-bit. First of all, most new gear and software is 24-bit compatible anyway; sometimes all you need to do is tick the desired resolution check box under preferences somewhere in your software, then dedicate more storage space for the commensurately bigger files. Get yourself something like an Alesis MasterLink CD burner so you can archive your 24-bit masters until the world is ready for them (as well as to let you ditch that DAT player that's getting long in the tooth), but continue dumbing down your audio to 16 bits for general release on CD.

Arguably, the best upgrades you can do right now don't even relate to bleeding-edge technology: never have so many great-sounding mics been available at such low prices; mic pres — both tube and solid-state — have never sounded better; and a guitar modeling box can give you a much wider assortment of timbres than you can coax from even the best vintage amps. (However, there's still a "final frontier" for guitar amp modelers: they may sound great in a track, but *playing* through them is still a different experience compared to having some white-hot output tubes spit electrons through a massive output transformer into a tightly coupled speaker.)

Overall, the best thing about being on the cusp of change is that it gives us a chance to get better at our craft. You are hereby officially excused from having to buy the latest and greatest Mac (most software isn't optimized for it anyway), going to a new operating system (OS X for the Mac isn't ready for prime time, and Whistler won't be hitting your PC's hard drive for a while), or remodeling

> "TAKE THE TIME TO EXPERIMENT WITH NEW TECHNIQUES, READ SOME OLD MANUALS ON EQUIPMENT YOU ALREADY OWN, AND PUT THE MONEY YOU DON'T NEED TO SPEND ON NEW TECHNOLOGY INTO UPGRADING THE WEAK LINKS IN YOUR AUDIO SETUP."

your control room for surround. Instead, take the time to experiment with new techniques, read some manuals to pick up on features you might have missed in equipment you already own, take your spouse out to dinner and a movie, and put the money you don't need to spend on new technology into upgrading the weak links in your existing audio setup. Forget about tussling with technology for a little while, and make some music that reaches people's souls...the march of technology will pick up its pace soon enough.

Craig Anderton, creative director for MusicPlayer.com, is the author of *Home Recording for Musicians* and *Multieffects for Musicians*, both published by AMSCO. His tune "What Are You Waiting For?" was just remixed in Germany for inclusion in the Battery Park 2000 compilation.

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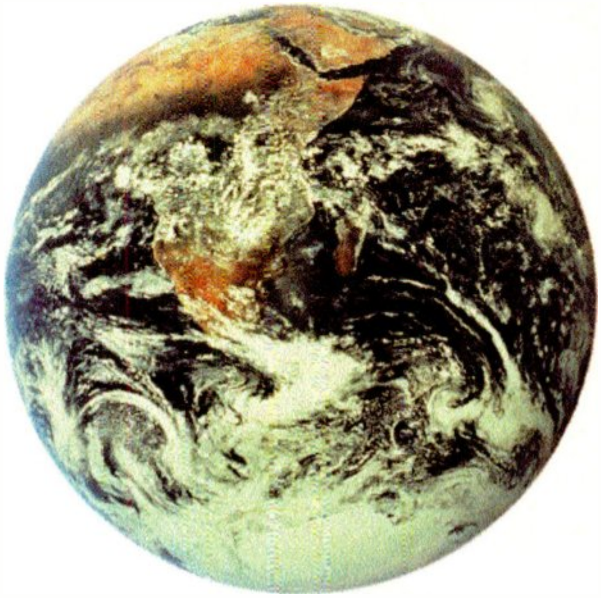
Have a question or comment for Craig Anderton? Visit his online forum at www.musicplayer.com.



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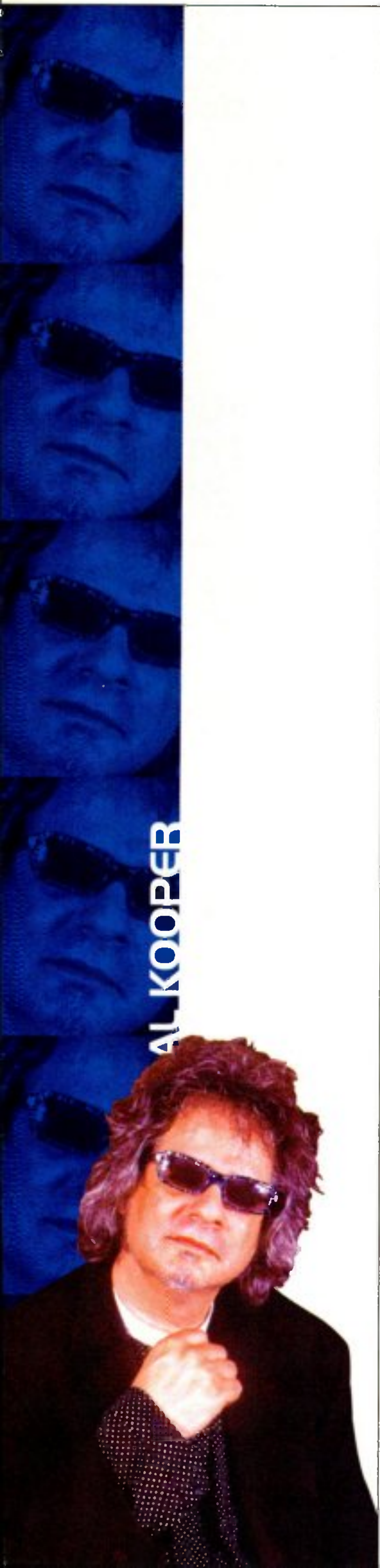
Dr. Al visits the AES show

Walk Around Heaven All Day



It was with great trepidation that I boarded that long tube with wings on September 21, 2000. Six years earlier, I had ventured from Nashville to Los Angeles for the annual Winter NAMM show and was almost wiped off the face of the earth by that nasty Northridge Quake of '94.

"I don't need to go back to California anymore, really...." I thought to myself, as I lay in my comparatively safe Nashville bed that night. When I say comparatively safe, I'm referring to tornadoes; the earthquakes of the south. "I've spent a lot of my life in California and I think I can exist without returning...." And so, in 1994, I vowed never to return to California in this life.

In the Summer of 2000, my only son and daughter-in-law birthed a grandchild in Northern California, where they dwell fearlessly. Suddenly my options multiplied. I had to go see my grandson — it's a priority thang. It worked out that the time-frame I chose to wing westward included the AES show and I made a pit-stop in L.A. to attend and amaze those who never expected to see me tread in la-la-land again. I walked in the doors of the Convention Center and took a deep breath: "I'm totally dead in this place if the Big One happens...." So I pretended I was in the Javits Center in New York City and went about my business. It helped a lot.

Twenty-four-track hard drives and their digital console progeny seemed to be the big trend at this show. TASCAM is showing a new console that integrates with its 24-track tapeless gizmo. Total price for 24 tracks of hard drive *and* a cleverly designed digital console? $7K. Times have changed, haven't they? My still-cool TASCAM analog console from 1992 cost $7,500 back then. Now I could have both 24-track *and* console (albeit smaller and digital) and still have $500 left to bet on the Knicks game.

iZ Technology, who built the RADAR system for Otari in the early '90s, is now distributing their own version of RADAR, aimed at the same market as the TASCAM. However, iZ's 24-track will set you back $7K on its own. But their technology is older and therefore, hopefully, less buggy on the first run. Mackie's also got their entry — it looks like an interesting battle for supremacy. Only time will tell.

EQ magazine had a brunch for its writers and forum hosts and it was an amazing cast they assembled in one room. Eddie Kramer, Geoff Emerick, Roger Nichols, Chris Stone, Alan Parsons, George Massenburg, Craig Anderton, Ed-

PHOTO BY MR. BONZAI

How many engineers does it take to take a picture? At the *EQ* forum brunch it took about thirty.

Cherney, David Reitzas, Jack Douglas, and a whole lot more all in one room actually getting along. How many producer-engineers does it take to change a light bulb? One — while the other 30 sit quietly and roll over in their minds how much better *they* could do it! There was none of that in the room — just some welcome camaraderie.

I've been holding out going digital ('cause I love analog), buying Pro Tools (always seemed too expensive), and recording to hard drive (this I can probably be talked into). Hey — when I started out in 1958, there was monaural, no monitors on stage, no remote controls, no Beatles, and no *EQ* magazine — we've come a long way, babies — and nothing illustrates that more lavishly than an AES show. Now, if I could just be a contestant on that show with Regis and have Chris Stone and George Massenburg as my lifelines....

WEBLINK

Have a question or comment for Al Kooper? Visit him online at the EQ Boards, www.eqmag.com.

nice income stream running down the middle), you'll have a *very different attitude on this subject.*

And, finally, what if your illustrious clients get caught? Let's say a guy from BMI walks through the exhibit hall just to see what's going on, or the wrong person catches that jingle you thought nobody would ever hear on a AM station in Montpelier (and believe me, it does happen). Well, when they get caught, you want to know what your clients are going to say? They're not going to say, "That's right, Mister Copyright Man, slap the cuffs on me, I'm a real bad dude." No, they're going to say, "Gee whiz, we don't know anything about the music bizness. The guy who produced it for us said it was all right, and he cleared it, and his name is (insert your name here) and his address is..."

But getting caught isn't the point. It just stinks. Can I get a witness? An amen?

WEBLINK

Have a question or comment for Jim Bordner? You can reach him at jim@gravitymusic.com.

MARILYN MANSON

▶ continued from page 64

but the recording of the drums was methodically thought out by himself and engineer Greg Fiddleman. Each song on *Holy Wood* used a different drum setup, and more than 20 kits were utilized. Great care was taken in miking the drums. One setup might include lots of vintage tube mics, a [Shure] SM57 or an SM7 on the snare, an AKG D30 or a FET 47 on the kick drum, with a tube U 47 hung in front of the kit for ambience, and U 67's as room mics. "Recording drums is all about tuning drums; I don't care what anyone says," Sardy states emphatically. "The main thing you spend your time trying to do is to get the drums to sound phenomenal in the room. So I don't know that getting solid drum sounds is ever by accident." The drum tracks were routed through vintage Altech compressors that Sardy bought from the Hollywood Bowl. "We also had some Dynamax compressors, some broadcast compressors, 1176's, LA2A's, some Gate Stay Levels, stuff like that."

Sardy, who comes from a strong engineering background, points out that, although Manson was extremely involved in the recording process, he was just technical enough to get them in trouble. When that happened, they turned to their chief tech on the project, Jonathan Little, owner of Little Labs, which makes Sardy's most loved piece of gear, the PCP box. "I used the PCP box a lot on this record. We ran everything through it; it was our DI, it was our way to do three amps at the same time, it's the only way to do it. It was phenomenal, and a lot of the sounds literally couldn't have happened without it," he enthuses.

Many of those sounds came from Twiggy, Marilyn Manson's multi-talented bass player. Sardy recalls recording Twiggy for the song, "Valentine's Day." "I would say we blew up at least 15 pairs of speakers working with Twiggy. There's just no way to get enough bottom, you know. At one point, I kid you not, we had gotten these big monitors in there — these huge B&Ws — and they have these holes in the bottom, like ports. The damn thing was blowing smoke rings out of this hole every time the kick drum hit! There were little puffs of round smoke coming out. All of a sudden we were asking each other, 'Do you smell something?' We looked down and the f*cking speakers were blowing smoke rings. You can't make this kind of stuff up...." ■

JON@FEZGUYS.COM
ALLEN@FEZGUYS.COM

New options for the online musician

Streaming and Scheming

So you've been plugging away at your Web site, working with downloads, graphics, and uploads to other sites. Now you've heard about Internet radio. It sounds intriguing: A free, all-your-music station, all the time? Okay! But guard against euphoria. Just because you put your music on the Internet doesn't mean anybody's noticed. If you've been developing your Web site in ways we've discussed in this space over these past four years, however, your fans may be ready to tune in online. Thanks to the venture-capital-funded company **Live365.com**, they can do exactly that. Live365.com is the biggest kid on the personal Internet radio block. They handle the grunt work, streaming almost any music you desire, whether it's yours or that of your favorite musician.

Let's "create your own Internet radio station." Go to: **www.Live365.com** and click on "Broadcast." Click to the sign-up area to create a member account. Pick a member name, enter a valid e-mail (this makes sense — they want to be reasonably sure we are who we say we are). Thankfully, the rest of the required fields are kept to a minimum (entering zip code and country doesn't feel too intrusive). Unless you're interested in Live365.com's news updates, unclick the newsletter and program guide subscription options. You can always change your mind and edit your profile later. Next, go to: "User Agreement." Under "Rules" (always interesting), Live365.com uses the phrase "At present," which we take to mean: "Who knows how long this will last?" The rules are generally meant to discourage audio file copying (see sidebar). If you want to create a program of music you don't own the rights to, be sure to study them carefully. Despite the current chaos and confusion consuming Internet audio, it's possible to be ethical without the unnecessary morality crap. Pirating songs isn't that big a challenge anyway.

We next accept the terms (do we have a choice?). An e-mail is then automatically sent to us by the site, arriving promptly and pointing to a log-in page to "create our account." After logging in, we're dumped back to the main page. We scratch our heads for a moment before clicking back to "Broadcast," where it says we are not signed up to broadcast. Huh?

Okay, here's what we do: We click on the link to turn our *member* account into a *broadcaster* account and now find ourselves on a page with three choices: "EasyCast" (Live365.com handles everything), "Live Broadcast" (we broadcast from our computer), and "Relay Broadcast" (we use a remote computer). EasyCast is what we want, and we select it. Instantly, 365 MB of storage space are available, plenty for a little program of MP3 tunes.

Before we can upload our MP3 music files to their server, we need to select a streaming bitrate. Most people don't have ISDN, DSL, or cable modems, so we opt for 24 kbps, which works fine over a 56k dialup modem. Live365.com offers a piece of proprietary software (PC only) called "EasyLoader" for uploading audio files to their site. It simplifies the upload process and also automatically converts existing higher bitrate MP3 files to the lower bitrates. This means that if we've already got our music encoded as 128 kbps MP3 files, we can use EasyLoader to convert them to our chosen 24 kbps bitrate as part of the upload process. Mac users don't have that luxury, and need to make sure the songs they upload are encoded at 24 kbps. Since we're using a Mac today we'll go the regular route, uploading our MP3 files (encoded at 24 kbps) one at a time, just like on regular upload sites. The link to upload through our Web browser, though available after we first registered as a broadcaster, can be a bit hard to find. From the home page, click on "Broadcast," then click on "Go To EasyCast." Option #6 provides a link to upload or delete songs through our browser. Resource links and tutorials are thoughtfully provided in case there are questions during the process.

We've uploaded some tracks and now click on "Go To EasyCast" again. A numbered list of options appears. Number 1, "Create A Playlist," is a scroll-down menu that contains more options for creating or deleting playlists, as well as containing the playlist of our uploaded tracks. We select our clearly marked playlist. The rest of the dialog boxes immediately change to reflect our new playlist. Broadcasters can choose a genre for their music, which is helpful if you like genres. Since the tracks we've uploaded are encoded at 24 kbps, we choose the 33 kpbs "Minimum Connection Speed." Below this are two boxes, one containing our playlist "Library" and one called "Current Playlist." We add the tracks from the library to the playlist and click "Save." Next, we click on "Broadcast Now." A dialog box tells us our playlist has been submitted and we should be up and running in about ten minutes. After ten minutes or so, we use the site's search option to look for our playlist by name, choose it from the resulting list, and *voilá,* we're an online streaming program. On their homepage, Live365.com has a useful feature called "10 ways to promote your station." It's worth checking out and includes tips

LIVE365.COM'S BROADCASTING RULES

Under the Digital Millennium Copyright Act (DMCA) (see www.fezguys.com/columns/026.shtml for more info), Live365.com's Internet broadcasters must comply with the following points. (Live365.com says they will be "obtaining licenses from the copyright owners" and have "abbreviated these rules to include only those likely to be relevant given the manner in which you are able to use the Live365.com system.") Keep in mind that if you are only working with music you own the rights to, you're free to do as you choose. Some of the restrictions limit the creative side here. Hopefully once the DMCA settles down (or is modified in some way), some of the rules will be lifted. The rules are designed to protect copyright owners, but the legal landscape must fearlessly enter the doors opened by recent technologies. At the same time, listeners should be able to request a band's song, and that band should, obviously, be compensated for it.

1. Your program must not be part of an "interactive service." This means that you cannot perform sound recordings within one hour of a request by a listener or at a time designated by the listener.

2. In any three-hour period, you shouldn't intentionally program more than three songs (and not more than two songs in a row) from the same recording. You shouldn't intentionally program more than four songs (and not more than three songs in a row) from the same recording musician or anthology/box set.

3. Continuous looped programs may not be less than three hours long.

4. Rebroadcasts of programs may be performed at scheduled times as follows: Programs of less than one hour: no more than three times in a two-week period. Programs longer than one hour: no more than four times in any two-week period.

5. You should not publish advance program guides or use other means to pre-announce when particular sound recordings will be played.

6. You should only broadcast sound recordings that are authorized for performance in the United States.

7. You should pass through (and not disable or remove) identification or technological protection information included in the sound recording (if any).

and also the URL you can use to link to your show from your Web site.

Some Thoughts On Internet Broadcasting

The long-term financial health of companies streaming unlicensed music is uncertain. At some future point, Web sites like Live365.com will have to pay back-licensing fees as a result of the DMCA. These back fees scare the hell out of potential investors. Given the rapacious nature of copyright Collection Societies, it's plausible the past-due balance may exceed the current (and future) worth of the entire company.

Until these fees are decided (this is a good time to be a digital rights lawyer), amateur Internet radio stations are swimming in murky waters. If Internet radio broadcasting services like Live365.com get run out of business by huge licensing fees, your personal band streaming service will disappear. No offense to independent musicians, but companies providing streaming service for indie music will be hard-pressed to show profit. The service could possibly fold into a larger business model (*á la* MP3.com). The FezGuys

▶ continued on page 142

THE FEZ GUYS

▶ continued from page 141

actually spent a pleasant lunch looking for other funded (read: venture capital money) companies providing free streaming services for indie music. We couldn't find any. (If you know of any, please tell us!) This appears to reinforce the idea that potential investors are standing by to see whether or not this particular business model (such as it is) will work. The Internet broadcasting sites mentioned above are visible enough that it's unlikely new sites will chance starting from scratch.

The FezGuys had a consensual FezDream last night. We saw a future of thousands of grass-roots arts organizations working together, providing streaming services for local bands, all paid for by grants from the National Endowment of the Arts. Then we woke up.

Bottom Line on Live365.com: They are a group of plucky folks bold enough to provide this service in the current climate. The FezGuys say: "Go for it."

A Subscription Overview

Got an audience who might pay you for access to your music? As Napster continues to shake things up, musicians are toying with subscription services as a way to make a little cash using the Internet. Sometimes called fan clubs, this idea has been around for ages. Many bands (and even, gasp, major labels) have used them successfully to both build community and drive revenues. Consider the profit margin involved in the Dave Matthews Band taking in $30/person annually from 30,000 online fans before shipping anything out (we made this up, but it's possible). Musicians such as Todd Rundgren have been testing subscription waters for years, and younger bands are breaking new ground, quickly building up databases of online users to market their music directly to fans.

One popular model asks fans to buy a low (we mean cheap, like, one dollar) monthly subscription (paid yearly) to your online music and art. Musicians must make some important decisions first. A little work must go into your subscription model. The most irritating decision is how you will accept the money (For more info on payment options see: "Collecting The Funds" at **www.fezguys.com/columns/008.shtml**. Also check out newer payment options: **www.paypal.com**, **www.tipjar.com**, **www.e-gold.com**, **www.cybercash.com**, and, of course, standard credit card transaction systems. These last ones can be purchased as off-the-shelf packages — search around online). Keep a safe (from prying eyes and from accidental loss) database (backed up frequently) of your users. Then you must decide what you will deliver and how you will deliver it. How secure does it need to be? If you're advanced (or have techie friends), you can create password-protected accounts. If you have a small user base (up to 300 people) and trust them not to give their passwords away, this should be good enough. If you have 100 subscribers or less, you might even be able to get away with e-mailing them all a "hidden" (no password and no link in to any other page or site) page containing the goods. The more subscribers you have, the more likely it is they may share their password and so the more complex it will be to provide a solid technical solution. If you've got a lot of fans, you'll want to team up with a tech programmer or perhaps even a small company who can provide these services at a realistic price.

To get the best response, deliver something on a regular (monthly is good) basis. A combination of news, photos, and music is a good start. Even though it's delivered online, consider creating a collection of the same materials in a familiar physical package on a yearly basis, included in the price of the subscription.

Take the long view. Be creative. Do some oddball packaging rather than a boring old CD jewel box. Remember those fans willing to pay a little money in advance for your art are the truly dedicated ones, but also the most demanding. Fans of this nature will alternately be your best grass-roots promotion team and your harshest critic (after you, of course). Treat them as you'd like your favorite musician to treat you. Too many times we've seen a musician turn on a dedicated online fan. It only has to happen once, but the ripples travel infinitely. Also, if you care whether or not your subscribers toss your music onto Napster, consider using a watermarking technology to track down those who leaked it. For one watermarking option take look at Cognicity's AudioKey software at **www.cognicity.com**. They offer a free 20-day evaluation period. Have a friend "steal" a song and post it. Follow directions to discover whether you can actually see the trail.

The subscription model offers myriad choices. There's no "right" way, and that means creative freedom. Have fun with it. Your fans will thank you.

DIY everybody! www.fezguys.com

▶ WEBLINK

www.wiredplanet.com (recently acquired by listen.com) allows you to create your own rudimentary streaming program containing only Wiredplanet-sanctioned music.

www.mp3.com will stream your uploaded band tracks, though it simply plays each song and then ends, which isn't radio. MP3.com's early entry into customized major (and indie) label music streaming, My.MP3.com, ran aground on the rocky shore of litigation and needs one more ridiculous multi-million dollar deal with a major label before it returns, probably as a fee-based subscription service.

SURROUND SOUND

▶ continued from page 92

which allows the producer to set and embed various "downmix coefficients" (such as level, panning, etc.) within each track, it's best not to rely on it. Instead, you should plan on doing a separate stereo mix in addition to your surround mix. Fortunately, since most of the space on a DVD-Audio disc is reserved for audio data, there's plenty of room to store both mixes, and data compression can always be used if the program material is too long to fit. Most of the surround sound monitor controllers mentioned above allow you to hear the (usually disappointing) results of downmixing to stereo or mono at the touch of a button; Kind Of Loud's Tweetie plug-in also provides one-click downmix previewing from within Pro Tools.

If your music will be released on DVD-Video, the surround mix must first be encoded into Dolby Digital, MPEG, and/or DTS formats (an encoded data stream is optional for DVD-Audio releases). All of these algorithms are "lossy" perceptual encoding schemes (as opposed to MLP, which is a "loss-less" data compression scheme), meaning that some of the audio data is actually discarded. In theory, the data that's thrown away is audio you wouldn't hear anyway (for example, low-level signals buried beneath louder ones, or portions of sounds masked by others in the same frequency range), but no matter how great a job the algorithm does, the audio is going to be compromised to some degree.

Fortunately, there are steps you can take in the mixing process to minimize the degree of degradation — for example, frequency-specific limiting in some bands can sometimes help reduce "spittiness" — so it's helpful to be able to preview your mix through an encoder in order to make these compensations. Standalone hardware encoders are available from Dolby and DTS, and there are a number of computer-based MPEG encoder cards (most of which run only under Windows NT) that allow real-time monitoring, but these all tend to be relatively pricey. A less expensive alternative is provided by the non-real-time software encoders available from a number of manufacturers, including Sonic Foundry and Minnetonka, as well as Kind Of Loud's soon-to-be-released SmartCode Pro plug-ins.

Howard Massey heads up Workaday World Productions, a full-service surround sound project studio. He is also the software reviews editor for *Surround Professional* magazine. Special thanks to Tomlinson Holman, Bob Margouleff, Chuck Ainlay, Elliot Scheiner, George Massenburg, Will Eggleston, Richard Elen, Peter Chaikin, Buzz Goddard, and Suz Howell for their assistance in preparing this article.

TONY VISCONTI

▶ continued from page 82

also go up higher, around the 2k range, and find where the beater is and just tweak a little bit there. I'll assign most of the low end to the bass guitar. Again, it depends. I haven't done a lot of hip-hop, and I think the reverse would apply there.

So you're saying, if the frequency you want to boost doesn't work because of other things in the mix, go an octave or two octaves higher, and that'll give you the same psychoacoustic effect. Does that work for all instruments?

Yeah, play with the octaves. In opera, a voice has its tessitura — its range where it sounds most pleasant. A singer might have a four-octave range, but there's maybe one octave that's golden, and you want to sing things in that register. The same thing with instruments. Technically, with its undertones, a guitar will go down to 40 Hz because that's an octave lower than the E string. But does it sound good down there? Do you really want to boost those frequencies, or do you want to go where the guitar sounds most pleasant, to the area it was built for? The same thing with a kick drum, the same thing with the bass; every instrument has got its range that's beautiful, and that's what you want to find. That's my philosophy, and that's what you want to draw out of a mix. And quite often you'll find a conflict, say with a guitar and a piano, because, in rock, they pretty much are playing around the same area, though a piano might be a little higher and a little lower.

What mics do you like to use for vocals?

The Neumann U 87 and the [Audio-Technica] 4050, which I like mainly for female vocalists; it seems to just open up their voice more. The '87 is a very masculine microphone; it's thick and it sounds really good. For rock vocals, I'll use a '57 and a '58. And recently, [I've been using] the Telefunken ELAM 251, a very expensive tube microphone. I used it on Bowie's voice, and he just sounded beautiful on it because he's got low end in his voice, but the high end on that mic is brilliant, too, so the vocalist leaps out at you. That's a dream mic.

Do you ever record vocals with compression?

Yes, all the time, though I try not to let it go beyond 10 dB of compression at a 3:1 ratio. Beyond that, you're going to really hear the compression. But most of the stuff I do is rock, so that's acceptable.

Will you compress it again when you're mixing, or will you do manual gain riding of the vocal tracks?

On a rock record, I'll let the compressor do the riding. It works — it's the sound you want — but then, even on a lot of rock records, you still have to do manual leveling for one word or for one syllable. With Pro Tools, I'll just go into the sound file and I'll boost the words that are mumbled; I'll even put EQ on it just for those brief words so I can hear the t's and d's of a word. That's a whole new level of mixing — that's micro mixing.

A common problem, especially for novice recordists, is getting a vocal to sit in a mix.

The vocal is probably the most important part of the mix. If the vocal is poorly placed in the mix, it's going to defeat the purpose; it's not going to sell the music. People will remember vocals more than they'll remember the guitar licks — most people buy a record because it's a great song. My way of doing that is, again, to see where the vocal lives sonically. See where it lives and play with the EQ until you find where the vocal is going to be really hot. An expert at this is George Massenburg — I just cream when I hear his vocal sounds; he really does that absolutely right. So, if I can, I'll use a Massenburg EQ — he's found that thing that makes vocals sound great. It's kind of in the upper middle and the extreme highs, the sound from the back of the throat. Instead of just the tone, you hear all the physical things that make up the tone, too.

Compression will help, and don't ever be afraid of putting the vocal too high in the mix. Sure, there's a point where it's ridiculous — I've heard some people put it too high — but you've got to hear every word, and what you should do is ask someone not involved in the production if they can hear every word. That's really the acid test. That's a real pitfall — if you've been working on an album for three months, you know every word of every song, and, psychoacoustically, you think it's all there. Then a person comes in the room, and you say, "What do you think?" and they say, "Oh, it's very nice, but I couldn't understand one word the vocalist was saying." It's a shock — a real jolt — when you hear that.

I'm a firm believer in de-essers, too, because you need to have a lot of brightness on a vocal track. But if you just start tweaking, the s's are going to rip your ears off, so what I'll do is, I'll take a feed directly off tape and hit the de-esser first, before adding EQ. Then I'll patch it into my channel, and I'll start tweaking the high end.

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

STEINBERG NUENDO

▶ continued from page 103

transfer load meter rarely even lit up. The computer remained responsive and snappy, there was no feeling of sluggishness.

This is impressive performance — and it was without attempting to optimize any audio system settings or the computer's performance. I simply loaded Nuendo into the computer with Dell's factory Windows 2000 install, created the project, and hit play.

Naturally, your mileage will vary. But you can safely say that Nuendo is powerful enough to handle large projects and professional situations.

Conclusions

So can Nuendo play in the big leagues? In my opinion, the answer is a resounding "Yes!" The program offers an almost endless list of features and abilities (I've truly only scratched the surface here), and, more importantly, lets you get done what you need to get done in the course of doing audio productions — whatever final medium or format the audio will end up in.

It's not perfect yet. I suffered a few crashes and lock-ups in the course of this review — but its stability was good for a v1.02 program. And because Nuendo can be set to Autosave backups, none of the crashes were catastrophic. Besides this minor instability, I had a few complaints, but overall the program was solid, and lived up to its claims.

Nuendo isn't the cheapest native audio software on the market, but it offers an excellent value. Even when combined with Steinberg's companion audio hardware (or whatever compatible hardware you want to use), the price still falls far below the competing proprietary hardware-based competition.

When it was all said and done, I got a lot of work accomplished using Nuendo — and I had a good time doing it. This one is a solid winner; it should be on the short list of must-see products for anyone searching for a professional audio production package. ■

NUENDO VERSION 1.5

At the Fall AES show, Steinberg unveiled version 1.5 of Nuendo. In addition to numerous audio editing enhancements, version 1.5 adds support for the Macintosh platform, as well as full support for ReWire and VST 2.0, allowing compatibility with virtual synthesizers.

Mac requirements: G3 processor or better, System 9.0.4 or higher (OS X will be supported), 128 MB or more RAM, USB port.

ACROSS THE BOARD

▶ continued from page 162

reverb processor. It has as much power in the basic model as four 480 reverbs. You can actually split it up that way if you want so you don't have to rent more 480's.

TC Electronic has been shipping the System 6000 for about six months now. They claim that their new algorithms are so powerful that they need to be processed on two DSP chips in parallel. The spaces sound pretty good, so maybe they're on to something.

Sony was displaying their top-end DRE-S777 reverb as well; it can sample a space and build up a reverb model to match. Also, Yamaha has been in the reverb business for more than a decade, and their new SREV1 sampling reverberator shows that they are not about to be left out of the race.

Best of Show

Off to the side of the show floor was a booth with one guy and a computer. The company was Webber Tapes Ltd. from the UK. The name of the product was Audio Compare. I grabbed a poop sheet and moved on. At lunch I read the overview of Audio Compare. My mouth fell open and part of my AES Burger escaped. This guy has built an analyzer that can tell the difference between two different audio sources. The system uses sophisticated ear-modeling techniques to show on paper what some of us have been hearing when we listen to audio gear. "Why do these sound different when they measure exactly the same?" Now we may be able to figure it out.

The box consists of two separate 700 MHz Pentium-based processors — each looking at one of the audio sources. The audio can be a digital file or it can be digitized at whatever sample rate you desire. The software then tells you what the difference is between the two audio streams. They had a copy of the good and bad Steely Dan pressing that they were using as a test.

This is not for the squeamish. The box starts at $50,000 plus options. Warner Brothers and MCA were looking at it for CD plant checking. I hope every CD plant gets one so I never have to go through the same thing I did a few months ago with the Steely Dan pressing problems. Webber is sending me a complete analysis of the Steely CDs, and when I know exactly what caused the problem, you will be the first to know. Wait, I guess I will be the first to know, but you will be second. No, I will probably tell Donald and Walter, Warner Bros., a friend who works at Fox in L.A.—okay, you will be nearly the last to know, but at least I will tell you!

So What

You know what I noticed most about the entire AES show? There was plenty of opportunity to spend money. However much money you have in the bank, there was the perfect item that cost exactly that much. A couple of years ago at the AES in San Francisco I was in the Sony Oxford demonstration area. When the salesman told me that the console was around $900,000 I said, "I will wait for the $20,000 version." This year that same salesman came up to me and said, "Well, here is your $20,000 version of the Oxford," pointing at the DMX-R100. I had no choice.

WEBLINK

To find out more about the companies and products mentioned in Roger's column, visit www.littlelabs.com, www.panasonic.com/pbds/proaudio.html, http://bpgprod.sel.sony.com/matrix.bpg, www.lexicon.com, www.tcelectronic.com, www.yamaha.com, and www.webbertapes.com

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DIGI001 Digital Audio Workstation For Mac And PC

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

FEATURES–
- 18 simultaneous, 24-bit ins and outs with support for 44.1 and 48 kHz sample rates
- 20Hz - 22kHz freq. response ± 0.5 dB
- 2 channel, XLR mic/1/4" line inputs with -26 dB pad, 48v phantom power, gain knob, and HP Filter at 60Hz
- 6 ch. line inputs (1/4") TRS balanced/ unbalanced w/ software controlled gain
- +4dB balanced 1/4-inch Main outputs
- Balanced 1/4" monitor outs with front panel gain knob
- 1/4-inch unbalanced line outputs channels 3-8
- Headphone output with independent gain control knob
- 2 channel S/PDIF coaxial digital I/O
- 8 channel ADAT optical I/O can also be used as 2 channel optical S/PDIF

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

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

MOTU AUDIO Hard Disk Recording Systems

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

MOTU PCI-324 AUDIO

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

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

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

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

CD RECORDING/MASTERING

ALESIS

Masterlink ML-9600
High-Resolution Master Disk Recorder

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

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

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

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

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

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

Includes
- Infra red remote control and rackmount brackets

marantz

CDR-631 Professional CD Recorder

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

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

MICROBOARDS

StartREC Digital Audio Editing/ CD Duplication System

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

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

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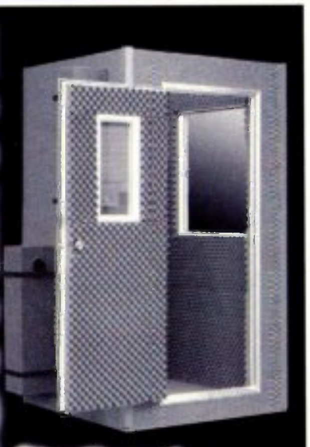

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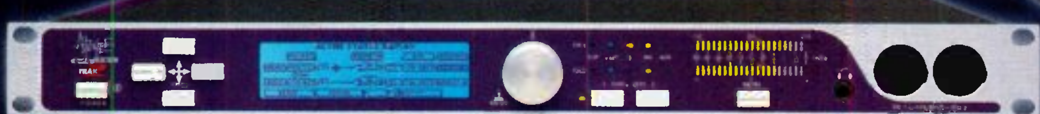

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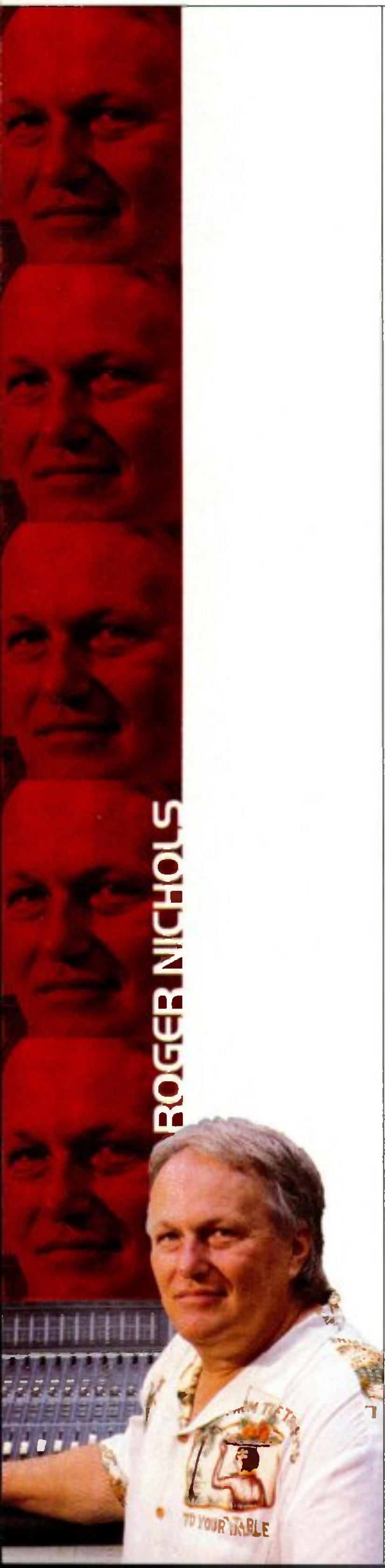

Everyone is back from the AES show, except for maybe the guy on the street I hired to hijack the Meyer truck full of X-10 speakers and the load of Sony DMX-R100 consoles on their way back to Sony. I haven't heard from him, so he is either enjoying the spoils in his new project studio located under the 9th Street crossover, or he has been put safely away somewhere where they are bribing him with regular daily meals so he will snitch on whoever put him up to it.

AES Stuff

I know you've already heard a lot about this year's AES show, but here's a partial list of what I found most interesting.

Little Labs

A friend grabbed me and showed me a booth I had passed by 40 times without noticing. It was a little booth with a little sign, saying "Little Labs." They were showing two products that I really liked. One was a direct box/splitter for guitars. You plug in the guitar and the signal can be sent to the tape machine and up to three guitar amps at the same time. The signal is split with precision transformers so that there is no change in impedance when more than one guitar amp is involved. An additional feature of the box is the ability to send a line level signal from tape through the box to the aforementioned trio of guitar amps. I have been recording guitars direct and sending them back out to the amp for 35 years. The trick is to match the impedance and the level so the guitar amp thinks it is seeing a guitar...what a concept. I built a box that I still have kicking around somewhere, but now I found the perfect way to replace it.

I am not done with Little Labs yet. They also had a digital audio router that worked like a video selector switch. You can have four output devices and select among any one of the five inputs. The digital signals are buffered and regenerated for each output, so there is no loading down of the signal when one source is sent to multiple outputs. Both boxes cost about the same as the new shoes my wife just bought.

Panasonic

Panasonic is re-structuring the way their pro audio division works. They have dumped all of the consumer-grade products and are focusing on high-end audio. The first product to hit the streets will be an eight-channel mic pre with 24-bit/96 kHz A/D conversion. They have spent lots of money and lots of man-hours to come up with a top-notch product. It looks good on paper, and I will let you know as soon as I hear it in a studio environment.

The new software for the Panasonic DA-7 digital console was drawing a crowd. The latest software revision includes a HUI emulation mode. This will allow control of Pro Tools faders as a page on the DA-7. Lots of guys are buying consoles for fader automation and hard-wired EQ and compression to free up Pro Tools for the esoteric plug-ins. I work this way a lot.

Sony

Okay, see what happens when you spend a little more for a console? The Sony DMX-R100 was pushing the envelope for small format digital consoles. The two biggest features are 1,024-step faders and the color touch screen. For surround panning, you just touch the picture of a room wherever you want the sound placed. I have only one problem with the color touch screen. If anyone put their grimy hands on mine, I would break their hand!

The faders are the best part. The Sony DMX-R100 is the only small-format console with actual touch-sensitive faders. All of the other consoles use a method whereby the fader needs to actually be moved for it to be considered a "touch." But here, if there is a vocal ride coming up that you want to erase, all you do is touch the fader. The fader stays right where it is, and the computer re-writes the fader data until you release the fader. This is the way that large-format consoles work. Of course, the automation for large-format consoles costs about $1,000 per fader plus $70,000 for the computer.

The 1024-step part is good, too. The large-format consoles use this step size. I think the Oxford uses 2048 steps. This allows you to make very small 0.1 dB changes in the sweet spot, and gives you a finer resolution when the fader is down at –20 or –40 dB. These ultra-fine kind of fader trims are common in large-format consoles, and you can hear a 0.1 dB difference easily when comparing the level of two sounds.

High-End Reverbs

The race is on for the next level of reverb processing. Lexicon was showing the new 960

▸continued on page 146