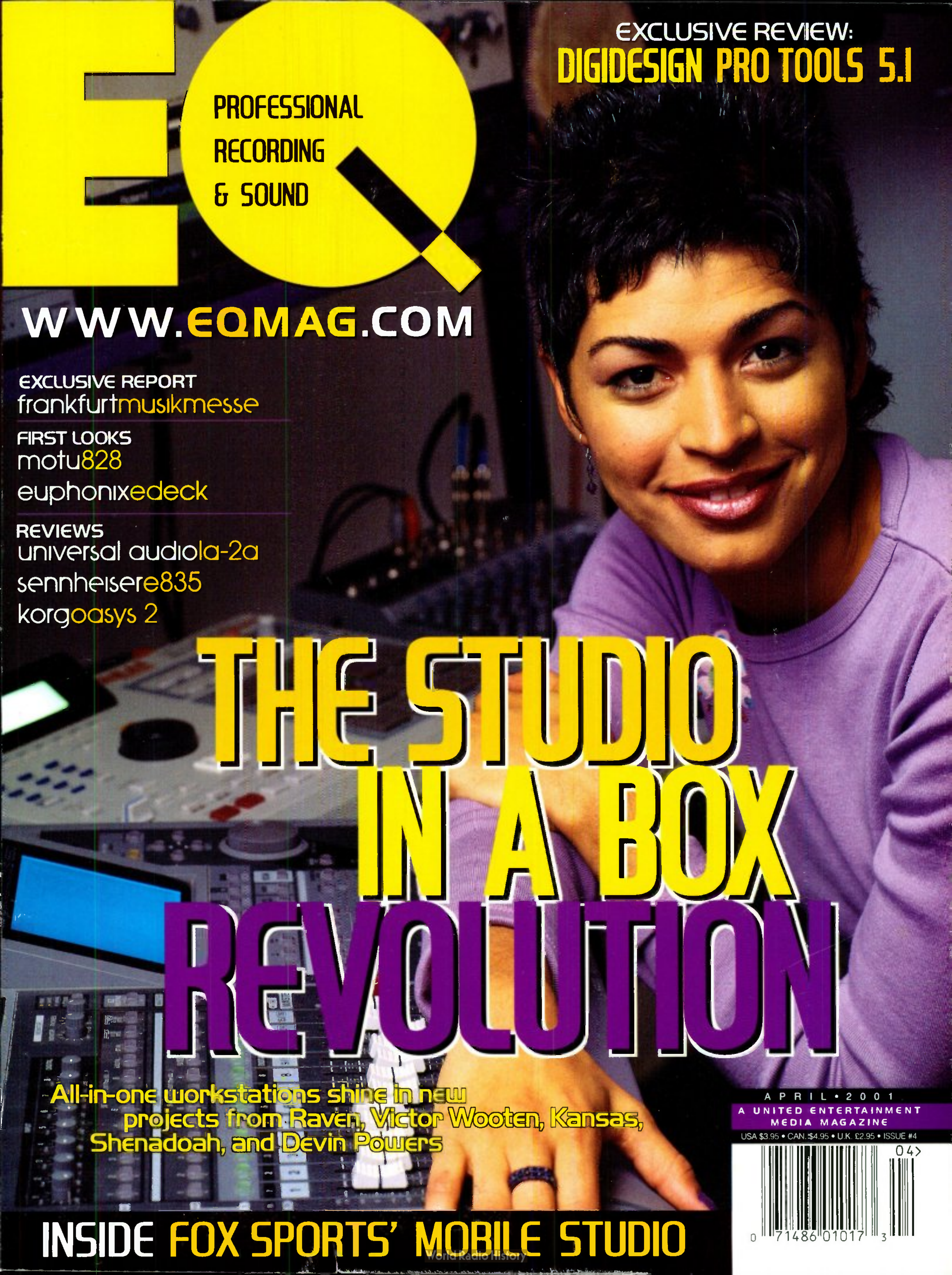

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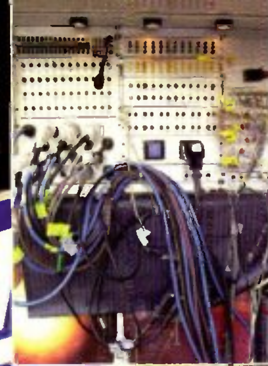

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RAVEN SITS IN FRONT OF HER ROLAND VS-1680.
PHOTO BY EDWARD COLVER

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EDITORIAL

Quest for the Best

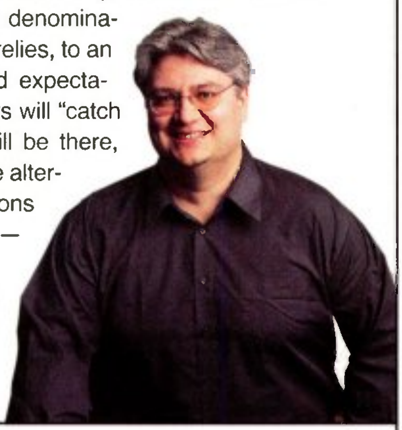
The other night I had the pleasure of attending a panel discussion on surround sound presented by Dolby Labs at the Hit Factory studios. I've made a few forays into the world of surround mixing myself, and was interested to hear where the discussion would lead. The panel was stellar: Frank Filipetti, Phil Ramone, and John Kellog from the production world, and Wendy Carlos and three members of Blue Man Group representing artists and composers — all highly articulate and rational speakers. The combination was intriguing; normally these kinds of discussions tend to lean toward the tech-y side. But this time out, the topics stayed firmly on the creative side. The artistic passion expressed by the panelists was truly inspiring — these are people who have seen potential for surround production as an artistic palette.

But not everyone was so firmly committed. In fact, there were several nay-sayers in the audience who clearly felt that surround sound is more of a marketing ploy than an expanded expressive medium. And that's okay — dissenting views provide a reality check as well as an opportunity for those on the other side to really think through why they're such strong proponents — a chance to look past the "gee-whiz"" and find the substance.

There was also, as is to be expected, discussion of the "right" way to do surround, especially how to deal with the LFE and center channels — not that anyone claimed to have the answers. The center channel discussion quickly dissolved into the expected camps: those who use it in their mixes, and those who don't. The reason for this is the question of whether consumers will have five equal speakers, and thus a center channel transducer capable of reproducing the desired quality, or if (as is more common) they'll have a center channel speaker that's compromised compared to their left and right speakers.

It really comes down to whether you're creating with the current lowest common denominator in mind, or if you're willing to allow some listeners to hear the end results in a manner less than faithful to the original concept.

My opinion is, if you're going to go for it, go for the finest production you can. If that means assuming the listener has five equal speakers, then so be it. If the consumer chooses not to use five equal speakers, that's their prerogative. As Wendy Carlos pointed out, when an artist paints a picture, he has no guarantee that it will be displayed with proper lighting or in an ideal environment. Fortunately, our medium allows for a continually rising lowest common denominator — one of the benefits of an art that relies, to an extent, on technology. The hope (and expectation) is that, as time passes, consumers will "catch up" with the ideal, and your music will be there, waiting to be enjoyed in all its glory. The alternative is to forever have your productions optimized for a compromised system — not a viable alternative in my mind.



—Mitch Gallagher
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"ON A GOOD NOTE, MATT SERLETIC SEEMS TO BE THE CLOSEST TO AN ACTUAL POP MUSIC PRODUCER, USING A GOOD COMBINATION OF REAL AND SYNTH INSTRUMENTS."

—IAN GRAHAM, SMALL DOG STUDIO

MEDEA FRENZY

I read Mitch Gallagher's Medéa AudioRaid review in the February issue, and thought it was very helpful. In the review, however, he says that it's "pricey." What does he have in mind for a good price? I would love to know because I'm about to buy Pro Tools Mix Plus.

Jayson Valencia
via Internet

[*Mitch Gallagher replies:* I meant expensive compared to just going out and buying off-the-shelf drives — with drive prices these days, you could get 120 GB for quite a bit less than the Medea's list price. But even though you'd have the same amount of storage space, you wouldn't be getting the performance benefits of the AudioRaid, which is the trade-off.]

PLEASE, JUST SAY NO

Jim Bordner's column on sound-a-likes [November 2000] was excellent. I have faced the same issue, and agree that the only sensible thing to do is to say, "I'm sorry, but it's way too risky." Sometimes, "I'm too busy right now," is the solution as well.

One point that the article didn't mention is that most ad agency music rights agreements contain a little thing called a "hold harmless" agreement, in which the composer/music guy agrees to "indemnify and hold harmless" the client from all claims arising out of the produced music. This would, of course, include copyright claims. Such clauses often require the music company to not only reimburse the client if he is forced to pay damages, but to assume all costs of legal defense.

Indeed, you may wind up paying for legal defense costs by the ad agency's silk-stocking law firm, in addition to whatever exemplary damages you are forced to cough up.

I've often wanted to say to clients putting the pressure on, "Hey, tell you what, I'll think about it, but first I want a written contract, guaranteed by you, individually, and your spouse — and secured by a mortgage on your house — that you'll personally indemnify and hold me harmless from any judgment against me resulting from this lift. And since the corporate legal department says the lift is okay, I want an opinion letter from corporate counsel, signed by him/her personally, which spells out exactly what song is being lifted, and for what purpose, and that the attorney is certain that the lift is not illegal, and that he will also indemnify and hold me harmless."

Les Schefman
via Internet

LESS POP, MORE MUSIC

Your December 2000 article on pop music opened my mind to how little regard the artists, producers, and record companies have for the actual value of music and its artistry, and more toward the almighty dollar. "We sold only four million of this song and gotta go for five on the next." It's always been that way and will always be that way.

David Foster's comment on "the days of going out and playing in clubs and learning your instrument are gone" couldn't be more wrong. Every day I work with new bands whose talent and dedication amazes me. But the industry shoves this pop "music" down our throats day in and day out, leaving no airwave time for something of value. These are the bands that deserve a break. David Foster and the other producers are not helping the industry realize this goldmine of talent.

Half of these pop artists can barely hold a note and lip sync their songs at concerts. Where's the talent in this, and why is the industry supporting this? If the artist sells X-million records, the producer is also acknowledged as being great. But it all sounds the same — factory music.

The real question, to me, is, "Why would these producers ever need a world-class studio?" It seems to me that a good mic, synths, and a computer would do the job. That way the money wasted in the big studios could be applied to the bands with talent.

On a good note, Matt Serletic seems to be the closest to an actual pop music producer, using a good combination of real and synth instruments. For the others, let's see them actually mic up a drum kit and use a *real* drummer. Get a *real* guitar player, perhaps with a MIDI synth converter. Play the keys, not sequence them. Lay some funky *real* bass down. I'm sure they can. And write a song that matters and will be around longer than the artists' careers. Then I will acknowledge them as great producers.

Ian Graham
Small Dog Studio
via Internet

CORRECTIONS

In the February issue's Bonzai Beat column, Eddie DeLena's nickname was reported as "Booth." His actual nickname is "Buttah," which we all agree makes much more sense.

Also, on page 78 of the same issue, the Pro Tools studio pictured in the "Riding The Tidal Wave" article is no longer part of the Tidal Wave facility. The studio pictured belongs to L.A. producer/engineer Dave Carlock, who was partners with Greg Ladanyi in Tidal Wave from fall 1999 to fall 2000. Currently, Dave and his studio are located in Hollywood. You can email him at davecarlock@davecarlock.com. ■

GETTING A SPACIOUS LEAD VOCAL

I'm trying to achieve the "commercial" lead vocal sound that not only hits you dead center, but also gives you a sense of "spread" on the lead vocal.

—Kyle

Try this: Mult the vocals to two aux tracks. Put 30 ms of delay on one and pan it hard left. Put 15 ms on the other and pan it hard right. This gives you a nice early reflection and spaciousness without sounding like an "effect." I usually also mult the vocals to a few more aux channels and put some more delays (in tempo to the track — one to eighth-notes, one to quarter-notes, etc.). I'll add them in just to the point where you can "sense" that they're there without really "hearing" them.

—Jim

SUBTRACTIVE EQING TECHNIQUES

I'm wondering why subtractive EQing techniques are better than boosting techniques. Do they both not introduce phasing problems?

—dchinsang

Subtractive EQ isn't necessarily "better" than additive; they both have their strengths. Once you've mastered finding and correcting the odd "spike" or resonance (or mode) in a signal, you'll often find that you need to do a lot less when you use one or two surgical "dips" rather than peaks. Also, for a given power spectrum, a dip has a much lower "Q" than a peak, and this means you're hearing less of the EQ (which, for Q's greater than about 1.0, start contributing a ringing) and more of the signal itself.

Additive EQ is certainly as useful as anything else; you just have to know when to deploy it. A sharp Q will definitely have ringing, and a musical "sound" associated with it. Also, I use shelves wherever and whenever possible; in my book (which is not to say that others don't interpret this differently), they're defined as having a Q of no more than 0.5, which means they're very neutral.

Finally (and I know that this is not going to go over very well), I haven't noticed that "minimum phase" EQ has any great advantage over EQs with a predictable phase shift. I think to whatever extent we talk ourselves into thinking that it's adding some kind of magic, we might listen harder and make a better recording, but nothing more. It's been demonstrated time and time again that phase shift in and of itself (not in stereo, or summed with another signal) is pretty darned hard to hear. Rather than worrying about internal phase shift in EQs, I'd spend the time on trying for better mic positions and on listening objectively (by far the hardest work we do) to different EQ approaches.

—George Massenburg

MASTERING: BEFORE AND AFTER

Being new to this game, I like to always have a reference CD that's close to what I'm working on. However, being that it's a finished product, my reference has been mastered. How close should I try to get to the finished CD sound? Sometimes when I listen to a CD I find myself wondering if certain things happened at the mix or the mastering process.

—Hack

It depends on the material you're working on, and the references you choose. I've never found a reference that exactly equates to the material I'm working on. I've found that they'll net me an overall reference point if and when I feel I'm losing focus, but are never a "road map" for the music at hand.

Get the mix sounding as right as you think you can make it sound, then let the mastering engineer add their special touch.

—Fletcher

Another tool you may find helpful is a spectrum analyzer. See what the response is of your favorite ref, and try to mirror that (with everything except maybe overall level). At least response-wise, you can get the gist of the direction they were shooting for.

—Nancy Matter

I would go for as close a match as you can, especially the overall bass-to-treble spread. Then print the mix *without* the big treble boost; your mastering engineer has a nicer, sweeter EQ than you do.

If the ref is a close match stylistically, check your product against it for things like vocal to band level, bass to kick level, etc.

Don't try to get it as loud as the competition. A/B your mix with the competition's after getting a similar apparent level in the monitors. Watch out for too much compression or EQ at the two-bus!

—Mark Lemaire

CAN EXPANSION UNDO COMPRESSION?

If I come across a CD that I think has had way too much compression applied to it, can I undo the compression by using an expander? Can an expander restore some of the squashed dynamic range?

—onhwy61

The short, safe answer is that one-sided expansion (unlike various Dolby's, which are bi-directional) falls woefully short of being able to restore original dynamics.

—George Massenburg

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.

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Mark Of The Unicorn 828

By Steve La Cerra

We've been hearing about FireWire (a.k.a. IEEE 1394) and its ability to move data at rates up to 400 Mbps, but the pro audio world is just starting to see FireWire applied to digital recording. One of the new FireWire-based products is the 828 from Mark Of The Unicorn. A rackmount audio I/O that interfaces to your computer for disk-based recording, the 828 includes features that make an external mixer unnecessary.

The 828 provides eight channels of balanced or unbalanced analog I/O. Channels 1 and 2 employ Neutrik Combo connectors for mic or line level input; 48-volt phantom power is available on these jacks, switched from the front panel. The remaining six channels utilize 1/4-inch TRS connectors for line- or instrument-level input. Each input delivers 40 dB of gain adjustable in pairs via front-panel trim controls. Converters are 24-bit for a dynamic range of 105 dB. Optical input and output ports may be used as eight-channel, 24-bit ADAT lightpipe, or as a stereo S/PDIF optical I/O. These ports may simultaneously be used for lightpipe input and optical S/PDIF output (or vice-versa). This is useful when a mix requires optical input from an ADAT, but also stereo S/PDIF optical output to a mixdown deck. A rear-panel 9-pin sync input jack ensures sample-accurate audio transfers with any device supporting ADAT sync.

One of the problems facing users of hard disk recording is monitoring latency — the delay between the time when a musician plays a note and when they actually hear the monitor output from the interface. The A/D conversion, routing from the interface through the computer's software, and from the software back through the D/A converters, requires a certain amount of time — enough to put a musician "out-of-sync" with the tracks to which he's overdubbing. To solve this problem, the 828 incorporates MOTU's CueMix Plus for no-latency monitoring. It provides the ability to select an input via software and mix that input with the main outputs of the 828 to create an in-sync monitor mix. The result is that there's no delay when you "patch through" a live instrument. This mix is routed out of the 828's front-panel headphone jack, as well as the main audio outputs on the rear panel. Independent level pots allow control over the main out and the monitor out.

Along with ASIO drivers, MOTU ships the 828 with one of the industry's first WDM drivers, enabling the 828 to act as front-end-compatible software (such as Cakewalk's SONAR and Sonic Foundry's upcoming WDM release of Vegas) under Windows 2000. In addition, the WDM driver can emulate Wave driver operation under Windows Me and Windows 2000. Also included is MOTU's AudioDesk software, an audio workstation program for Mac OS that facilitates recording, editing, mixing, and mastering of projects. AudioDesk also includes MOTU's PureDSP time-stretching and pitch-shifting technology, as well as dozens of real-time, 32-bit effects plug-ins. Users requiring integrated MIDI sequencing are given an upgrade path to MOTU's Digital Performer.

Additional features of the MOTU 828 include an internal power supply and a rear-panel footswitch jack that can be assigned to perform any two computer keystrokes via the 828's Control Panel software. Front-panel metering includes LEDs for signal present on each analog and digital I/O, plus clock status LEDs. ■

MARK OF THE UNICORN 828

WHAT IS IT? A FireWire-based audio interface for computer recording.

WHO NEEDS IT? Anyone who wants FireWire audio I/O for their computer.

WHY IS IT A BIG DEAL? The 828 is one of the first available FireWire-based audio I/Os.

SUGGESTED RETAIL PRICE: $795

CONTACT: For more information, contact Mark Of The Unicorn at 617-576-2760 or visit www.motu.com. *EQ* free lit. #101.

By Bobby Owsinski

Euphonix E-deck and Listen-In

Did you ever have to spend unproductive time waiting around for the artist, producer, or A&R person to show up to approve a mix? Or, worse, have to do a mix over again because someone in the band was out of town and couldn't be at the session? If so, the new E-deck from Euphonix may soon put an end to those problems by bringing the studio as close as an Internet-configured computer.

Although primarily known for hardware such as the R-1 digital recorder and CS-3000 and System 5 consoles, Euphonix has teamed with Rocket Network to produce a software product that could have significant impact outside traditional recording studio boundaries. In a nutshell, E-deck enables audio files of mixes (from stereo up to 5.1 surround mixes) to be accessed and auditioned anywhere in the world via a secure server and a personal computer. Better-than-CD-quality playback is provided, with PCM stereo capable of up to 24-bit/96 kHz, and 24-bit/96 kHz DVD-Audio-quality 5.1 surround mixes available. WAV, AIFF, MP3, and Windows Media files are supported; future audio formats can be supported through decoder plug-ins.

Here's how it works: In the studio, mix files are recorded, encoded, and then uploaded to a secure server on the Internet using the E-deck Studio module. In order to retrieve and audition these mix files, the artist, producer, or A&R person simply boots up a standard PC with any soundcard and an Internet connection, logs on using an access code, downloads the files, and chooses the mix to listen to using E-deck Pro. (E-deck uses ASIO drivers for soundcard compatibility.) Mixes can then be moved directly to a record label or manufacturing facility for archival and production.

Although E-deck is intended to be an audio-only application, it's also possible to send large graphics and text files, allowing these types of files to take advantage of the security features as well. A byproduct of E-deck Studio is that the engineer can also create better-sounding MP3 files (if needed) thanks to the high-quality encoder engine.

Yet another feature of E-deck is a function known as "Listen-In." Listen-In allows real-time remote monitoring of a live studio session in progress through a password-protected section of the software using Windows Media Server technology. This means that a producer, collaborator, or (heaven forbid!) a record company person can hear exactly what's going on during a session without being there. The recording studio sets up a "Listen-In" broadcast session and you simply log on to hear what's happening. Listen-In requires a high-speed connection to operate, and while it's not quite CD-quality, it's more than sufficient for near-critical listening (about the same as FM radio).

Pricing for E-deck Pro is structured in a manner similar to that of a cell phone; you pay a standard monthly fee of $235 for online storage capable of holding about 25 CD-quality mixes. If you exceed that amount, you pay more. That works out to less than $10 per mix — obvious savings compared to the cost of sending a DAT or CD via FedEx! ■

EUPHONIX E-DECK & LISTEN-IN

WHAT IS IT? E-deck enables audio files to be accessed and auditioned anywhere in the world via a secure server and a personal computer.

WHO NEEDS IT? Artists, producers, record label personnel, and songwriting collaborators. Anyone who has to hear what's happening in the studio but can't physically be there.

WHY IS IT A BIG DEAL? It beats waiting for FedEx to arrive or listening to a mix on the phone.

SUGGESTED RETAIL PRICE: $235 per month for E-deck Pro, which includes a Rocket Network account. E-deck Studio price to be announced.

CONTACT: For more information, contact Euphonix at 650-855-0400 or visit www.euphonix.com. *EQ* free lit. #102.

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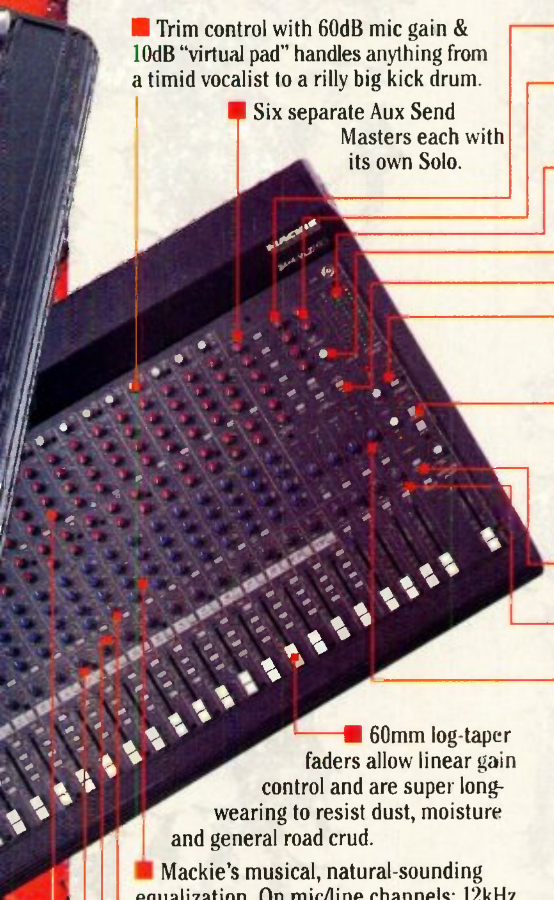

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- Mackie's musical, natural-sounding equalization. On mic/line channels: 12kHz Hi shelving, peak midrange sweepable from 100Hz to 8kHz (so it can also be used as a 2nd HF or LF control) and 80Hz Lo shelving. On stereo line channels: 12kHz Hi shelving, 3kHz Hi Mid peaking, 800Hz Lo Mid peaking and 80Hz Lo shelving.
- Sharp 75Hz 18dB/octave infrasonic filter on all mic channels cuts wind noise, stage rumble, mic clunks and P-pops.
- Super-twitchy –20dB signal present and overload LEDs on each channel.
- Constant loudness pan control.
- Six aux sends per channel. 2 pre-fader, 2 post-fader and 2 pre/post switchable.
- Stereo Aux Return 4 Master can be assigned to Buses 1-2 or 3-4.
- EFX to Monitor lets you send different effects or effects levels to stage monitors without screwing up your main PA mix.
- Easy level setting with In-Place Stereo Solo. Just solo a channel & adjust the Trim 'til the meter flickers at the Level Set arrow.
- Separate Tape Return level control.
- Global Aux Return Solo switch.
- Separate Solo section with level control, global AFL (post fader) or PFL (pre-fader) mode switch & Aux/Sub Solo LEDs.
- Separate Talkback section with level control, LED and switches for assigning talkback to Main Mix or Auxes 1 and 2. There's also a separate mic preamp input on the back of the mixer so you don't have to tie up a channel.
- Tape to Main Mix routes tape inputs to main outputs for music during breaks.
- Each Submaster bus has Solo switch, Pan control and Left/Right assign switch.
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- Inserts on all mono mic/line channels
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- 60mm long-life logarithmic-taper faders
- 6 aux send masters with individual solos
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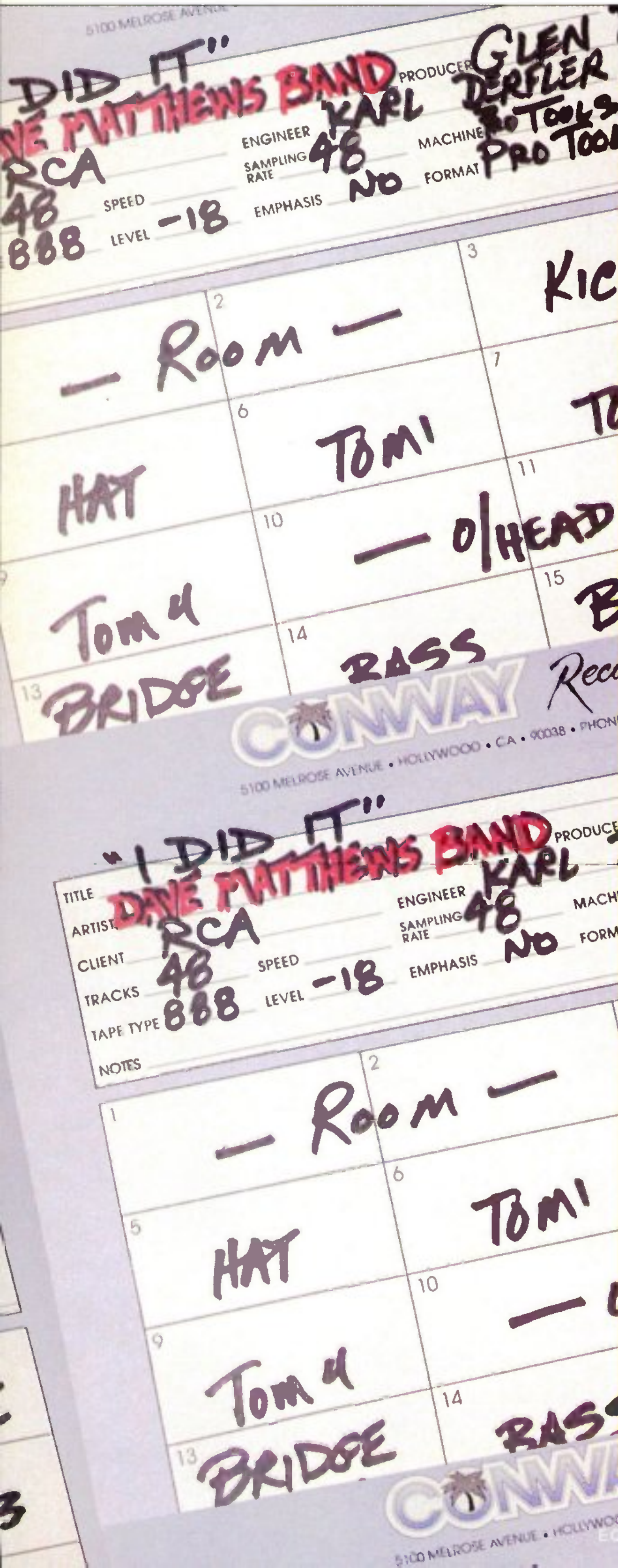
CIRCLE 58 ON FREE INFO CARD

Dave Matthews Band "I DID IT"



by Lisa Roy

DATE: October 5, 2000
STUDIO: Conway Recording Studios, Studio C
LOCATION: Hollywood, CA
ARTIST: Dave Matthews Band
PROJECT: *Everyday*
TRACK: Dave Matthews playing electric guitars for the song, "I Did It"
PRODUCER: Glen Ballard
ENGINEER: Karl Derfler
ADDITIONAL ENGINEER: Scott Campbell
ASSISTANT ENGINEER: John Nelson

SIGNAL PATH

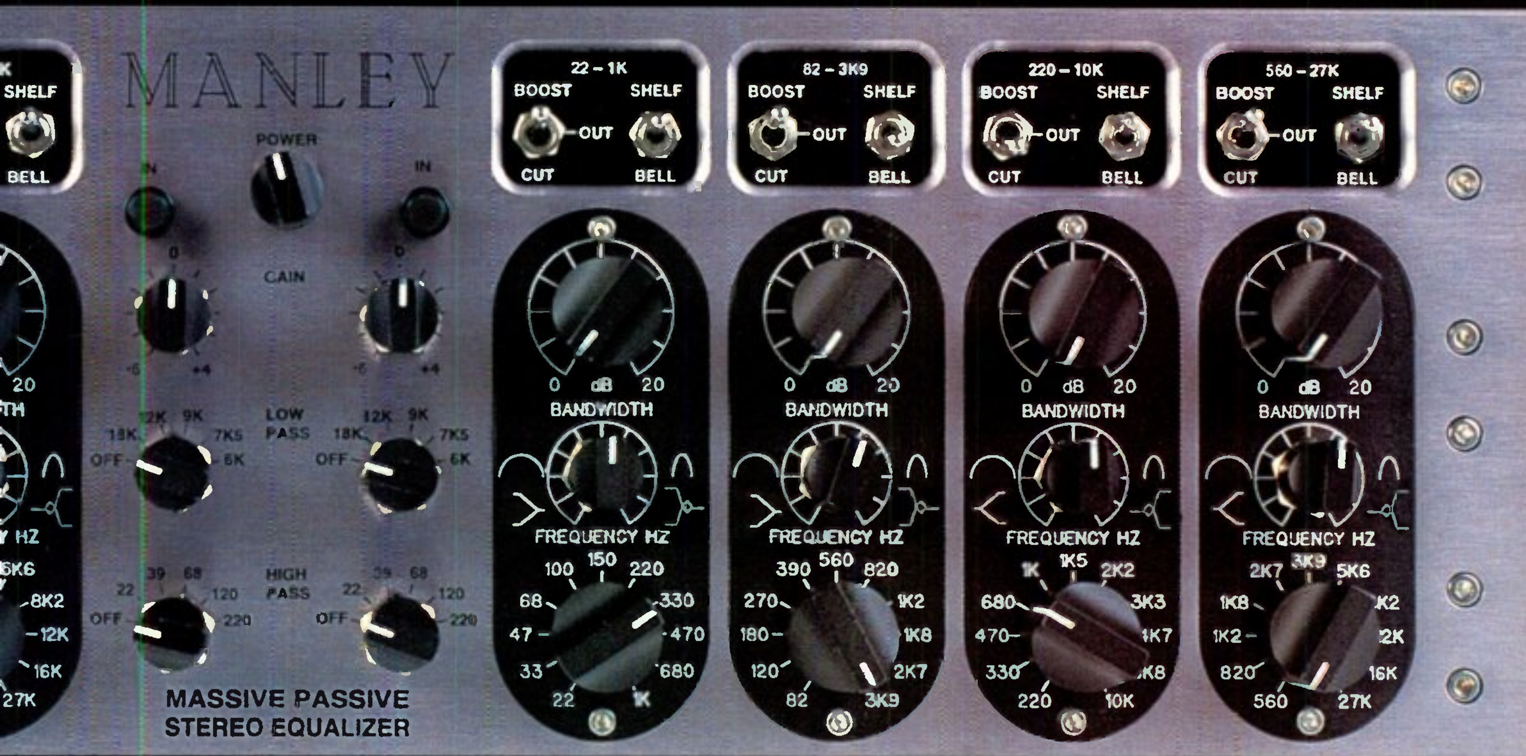
Karl Derfler says, "For the main rhythm part of the song, Dave was using a black six-string 'Jerry Jones' baritone guitar. For the chorus, he used Glen's '79 Fender Strat. Both were played through Glen's Matchless head and speaker cabinet. For mics, I used a combination of a Sennheiser MD409 and a Shure SM57 going through Neve 1084 preamps. Everything went to Pro Tools."

Before they stepped into Conway, Scott Campbell was at Glen's home studio, Aerowave, on October 2nd. recording a writing session for "I Did It." They ended up keeping the main guitar on the master track as part of the guitar sound. Scott explains, "The Jerry Jones baritone guitar was plugged into the HC30 Matchless amp, going through a single 12-inch Matchless cabinet with a Shure SM57. That went to a Euphonix CS3000 mic pre. Insert out, right into Pro Tools."

MIC POSITION

According to Karl Derfler, "We were using a close-miking technique; they were positioned about a half-an-inch off the speaker cone. The '57 and '409 were butted up to each other, side-by-side. I was going for a very present tone, so I wanted little ambient interference and just a very up-front, in-your-face sound. I used the combination of these two dynamic microphones because I find that, combined, I get a true representation of the amplifier's sound. The trick is making sure that the phase relationship between the two mics is perfect."

MASSIVE PASSIVE STEREO TUBE EQ



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CIRCLE 31 ON FREE INFO CARD

Scott Campbell adds, "For the demo session, I had the '57 not quite touching the grille cloth — you could slip a piece of cardboard in between the grille cloth and the mic. The mic was aimed straight on at the center of the cone. I find that I usually capture a guitar sound with character with that position."

PROCESSING

Karl Derfler explains, "The bridge guitars are interesting because basically what Dave did was hit a chord, which I put through a 201 Drawmer gate. I fed a metronome — like a sixteenth-note click-track — to the key input of the Drawmer. When he hits the chord, the gate is keyed by the click-track, and you get this little *da-da-da-da* sound without having to actually play sixteenth-notes or whatever. So you get this tremolo effect in perfect time with the song. For the bridge, we had three guitar amplifiers running simultaneously. I would let Glen pick which combination of amps he liked. We had the Matchless, a Vox AC30, and a heavily modified Marshall head with a Celestion-loaded speaker cabinet.

"After we did the performance, I used the auto-pan plug-in in Pro Tools to do tight, in-time auto-panning. It just moved across the stereo spectrum constantly."

TRACK NOTES

Karl Derfler says, "Glen and Dave did the demo of this song with Scott Campbell at Aerowave in Encino on October 2nd. That was our template that we used to build the song around. So we kept that original demo main guitar track on 'I Did It.'

"Dave is known for his acoustic guitars, but he played electric on this. My knowledge of it is that when Glen and Dave were writing, Dave picked up the baritone guitar and just started playing it — he really enjoyed playing that guitar. This song was written around the baritone guitar —there's a lot of bottom end, which is why there's such incredible depth to the sound of the song.

"Originally we set up the guitar amps in the room just lined up, with no baffles between them, and somehow we were getting this strange, boxy, nasty tone — it was very unpleasant. That night I asked [assistant engineer] John Nelson to find big foam baffles and put them between each amplifier for isolation. I didn't want to put the amps in boxes; I usually don't like that sound. We used these giant one-foot-thick foam baffles, and the sound of the guitars really tightened up. I also had more control over each amplifier.

"As far as recording to Pro Tools is concerned, for producers and artists, it's the most wonderful production tool. It gives you such a wide palette, sonically and arrangement-wise. We'd fly things around or make arrangement changes so easily, which you can't do in the analog world without major effort."

Elaborating on his October 2nd session, Scott remembers, "Dave loves the baritone guitar. We weren't necessarily looking for a particular sonic quality; it was more of what inspired him: What instrument is he going to come up with great riffs on?

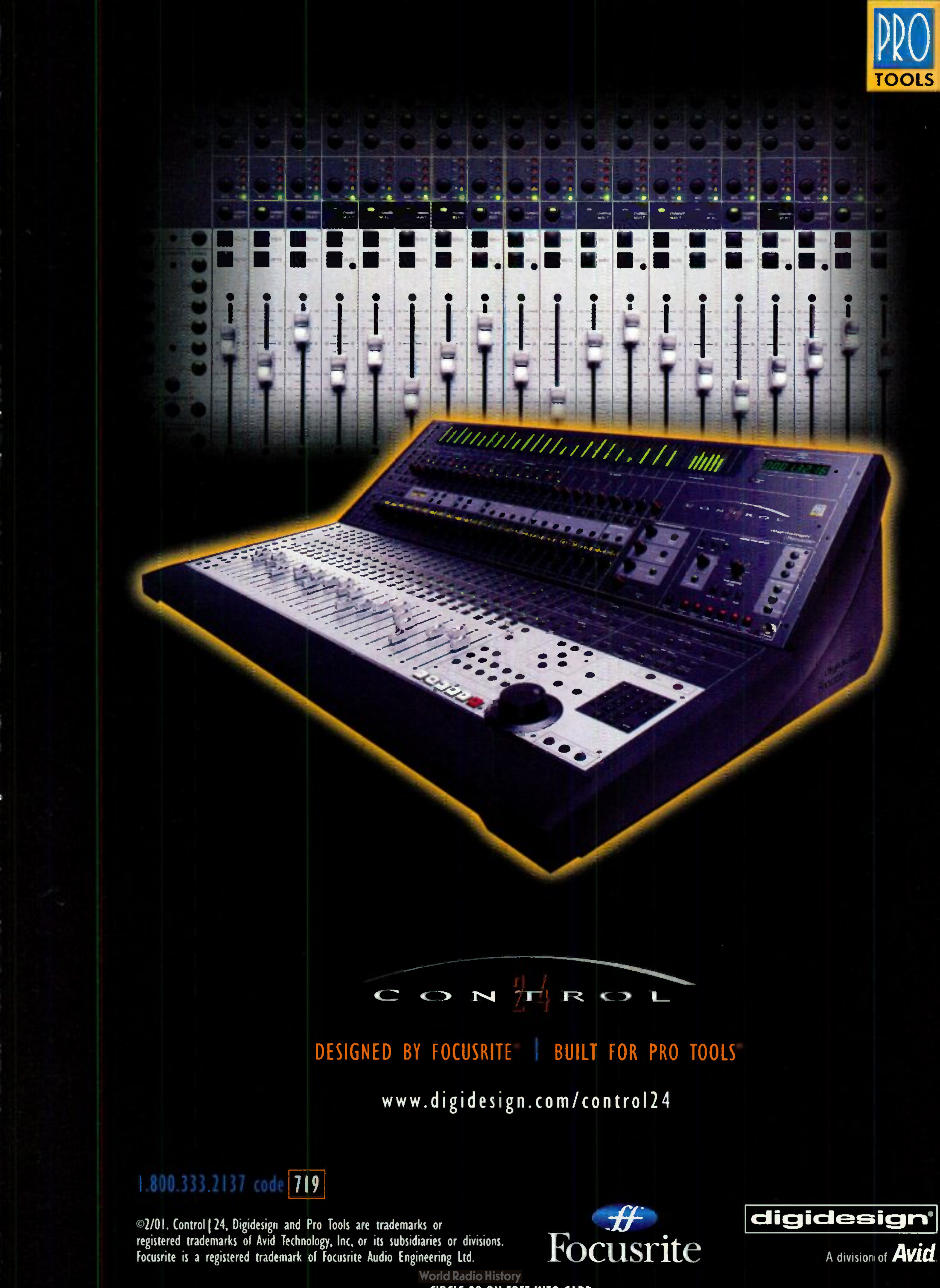
"At Aerowave, the Matchless is always wired up with the '57 on it; it comes up normalled to the console. We work really quickly, and a lot of times that's our guitar sound.

"I think the Shure '57 is a great mic for a lot of things, especially electric guitars like on 'I Did It.' I've miked them with other things, but the '57 is a good standby, especially for that dirty guitar with a little swagger in it — it really works well.

"The reason I used the Euphonix mic pre is because it has a lot of headroom; it can handle a lot of signal without clipping. In using Pro Tools, I've noticed you're dealing with a different gain structure than analog tape. You tend to drive equipment harder, so I need the headroom. We did this song very quickly without a lot of thought — just went with our gut and treated the recording process as a performance. I think it shows — the final record has a vibe that we originally started with in the demo stage. When working with Glen and Dave, speed is always of the essence. It's got to be done immediately. I developed my own templates for everything, and I just had to get it done like that because they were in the flow.

"Regarding Pro Tools versus analog, the thing that makes the song sound good is not the format, it's the key it's in, the tempo, the arrangement, and the energy that the performer is putting into it. The reason that we use Pro Tools is because it's very quick in the writing stage. You put an idea into it, make an arrangement, do some quick editing on it, and get your final arrangement together. I think it sounds great."

So how do they keep up with the lightning pace? "During the sessions, Rachel [Ballard's assistant] made these killer margaritas — the recipe was handed down from an Eagles' drum tech. They're called '6-6-6's.' I can't tell you the recipe — it's a secret." ■



CIRCLE 09 ON FREE INFO CARD

RISING STAR

Kevin Page

EQ questions an up-and-coming engineer

by Howard Massey

Memphis's House Of Blues studios has long been the stepping-stone for great talents, so it's no surprise that staff engineer Kevin Page is rapidly building a reputation as one of the top up-and-coming talents in the recording world.

Page's duties at HOB include everything from setting up sessions for outside engineers and producers to "driving the bus" himself. He comments, "Working different styles of music has really helped train my ears and has taught me about the common ground between the various musical genres. A typical week for me could mean 80 to 100 hours of work, but it flies by so fast that I really need more time in the day."

EQ: How did you land your current gig?

KEVIN PAGE: While interning at House Of Blues, I needed to make some additional money. A local contractor provided me with some early morning work so I could be at the studio in the evenings — even all night long! One 110° day, they called me off the studio's black tar roof and threw me into my first second-engineering gig. It was sink or swim for me, but I swam my nervous butt off!

How did you get started in engineering?

I was raised in a suburb of Detroit, listening to my dad's rock 'n' roll band rehearsing in the basement. That, and listening to the old eight-tracks playing on his stereo, is where I developed my love for music. I never wanted to be onstage, but I always loved creating music, and that's what led me to engineering. I started out at Full Sail in Orlando. Then I went on to intern at House Of Blues Studios in Memphis; I've been there for seven years now.

Who are your heroes in engineering and record production?

George Martin, Bruce Swedien, Berry Gordy, Quincy Jones, Mutt Lange, Hugh Padgham, and Don Was.

What are your favorite current recordings, and why?

I tend to listen to the genre of music that I'm working with at the moment. When I was working with Downpour, I was hooked on the sonic qualities of the *A Perfect Circle* record. That record sounds really amazing — it hits hard but doesn't hurt. Right now I'm really getting into Reuben Blades' *Tiempos*. I love the way all of the different ear candy lays together in the mix.

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

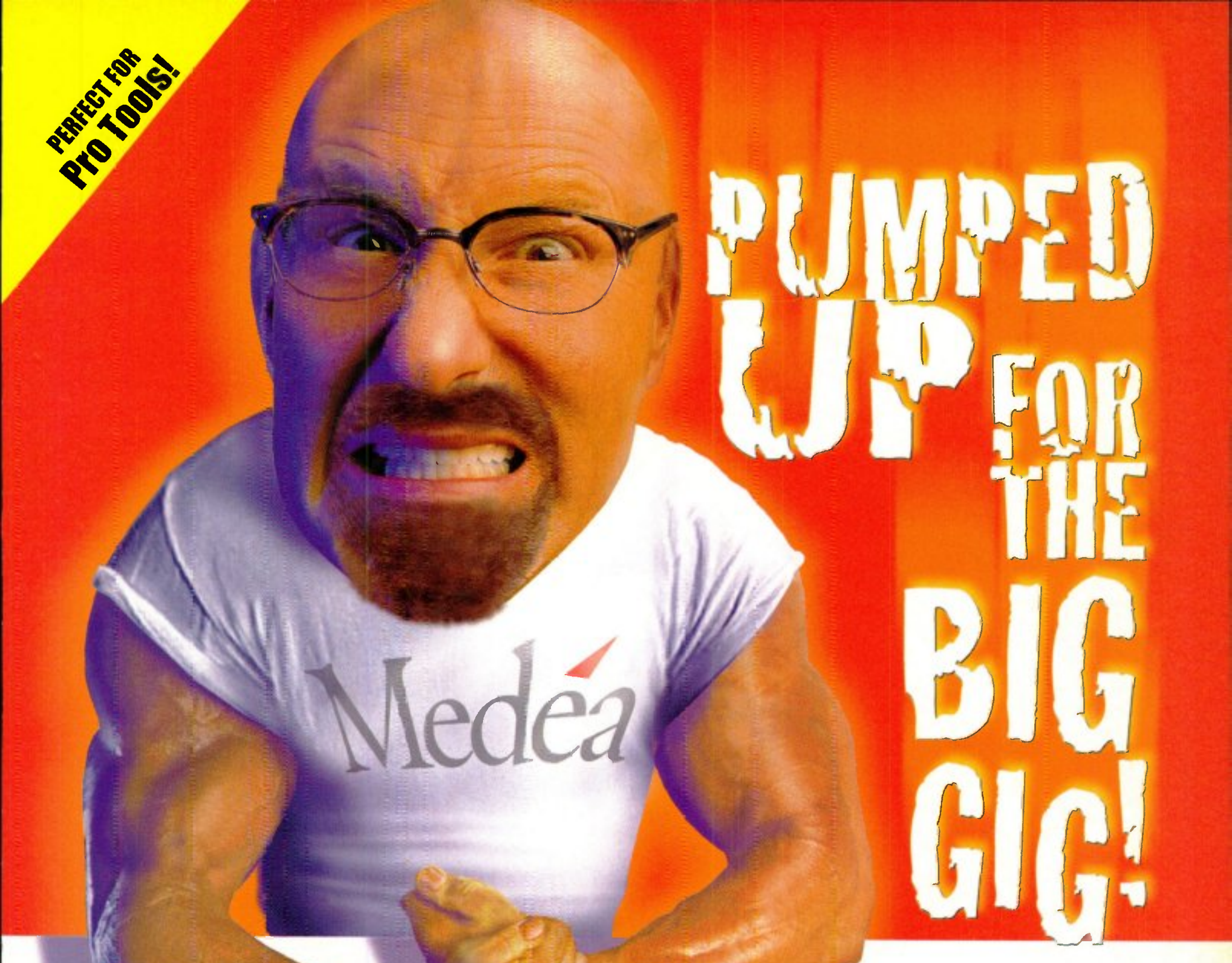
Keep every tracking vocal. I engineered a project for producer Malcolm Springer, and he made me keep them all. It was amazing how those tracks smoked the overdubbed vocals we attempted.

What's the best advice anyone ever gave you?

Producer Greg Archilla once told me, "Hang your feelings up at the door when you walk into the studio. Don't take it too personally." That advice has kept me from losing my mind many times! ■

WEBLINK

Email Kevin at kpage@midsouthrr.com.



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WEBLINK

Nancy Matter may be reached at NMatter@earthlink.net. You can check out the Moonlight Mastering Web site at www.MoonlightMastering.com.

PHOTO BY EDWARD COLVER

By Howard Massey and Steve La Cerra

Moonlight Mastering

Nancy Matter masters the art of mastering

STUDIO NAME: Moonlight Mastering
LOCATION: Burbank, California
KEY CREW: Nancy Matter
CREDITS: Olivia Newton-John, Tim Easton, Hugh Reagan featuring Clark Terry, Bonnie Pointer, Triloka Records, D.H.Peligro (formerly of the Dead Kennedys), Miche'le (Death Row Records)
MONITORING EQUIPMENT: Genelec 1031A, 1092A
RECORDERS: Panasonic SV3900 [2], Sony PCM2700-A [1], Marantz CDR630, Plextor and Yamaha CD burners, Exabyte, Nakamichi MR1
OUTBOARD: Manley Variable Mu compressor, TC Electronic M5000 (loaded with three DSP engines), Weiss EQ1MK2 24/96 equalizer, Audio Control RTA SA3051, Diffet Buss Changer
COMPUTERS: Apple G3 with OS 8.6, 128 MB RAM, Maxtor 45 GB IDE and 12 GB internal IDE drives, Seagate Barracuda 9 GB and Cheetah 18 GB [2] drives, Adaptec ADA 3940UW SCSI accelerator card, 21-inch ViewSonic monitor; Apple G4 with OS 9.0.4, 384 MB RAM, 18 GB internal IDE drive and Adaptec ADA 3940UW, 21-inch ViewSonic monitor
DAW EQUIPMENT: Lucid 8824 digital-to-analog converter
SOFTWARE: Sonic Solutions Version 5.4, Sonic Solutions HD Version 1.5

Moonlight Mastering owner and chief engineer Nancy Matter is rapidly gaining a reputation as one of L.A.'s top mastering gurus. A graduate of the Trebas Institute of the Recording Arts, Matter started her career working in the mastering studio of a CD and cassette manufacturer. Working her way up the ranks there, she became chief engineer before transitioning to Walt Disney Records, where she was put in charge of various in-house mastering projects. Following a long stint as in-house producer for Disney, Nancy decided to roll the dice and set up her own independent mastering house, offering professional-quality results from a project studio environment designed by Chris Pelonis.

EQ: Do you think that never having been a tracking or mixing engineer has hampered you at all?

NANCY MATTER: No, mainly because I've done a lot of live sound. I know enough about tracking and mixing that I can direct an engineer and we can collaborate.

Was this what you always wanted to do?

It was a fluke, but it ended up becoming a craft that I love. It's a true art form.

I've often thought that mastering engineers have to be almost audio microsurgeons.

Sometimes. But because I deal so much with the independent market — and a lot of the independents are unsure of the mastering process — I find myself being an educator as well, and I really like that angle.

Have you ever gone back to the engineer and said, "Look, I really think this needs a remix"?

Yeah, I have. People with less experience tend to over-compress everything. What happens is that I can't do anything with it because it's already at its maximum. So I've occasionally requested that clients take material back home and reduce the compression so that I have more ability to do what I need to do.

I gather that normalization is your enemy.

Let's put it this way: I would rather be doing it here. Once in a while people do normalize, but (especially among the indie market) a lot of people don't go that route. Instead, they throw in a lot of compression — you know, it's got to be louder, louder, louder. At that point, they're basically into distortion. I would say that's the main reason that I turn projects away. I like it better when it comes in fresh.

A lot of times I have communication with my clients up-front, and I highly recommend that they not use pre-mastering tools. Pre-mastering and mastering will simply be fighting each other at that point.

If maximum loudness is the Holy Grail, how do you achieve it?

The one thing I do is to use frequency-dependent compression, then go into an EQ, and then I come back out and run the signal through another compressor. Even though it's an odd combination, it works. When somebody comes in with a mix and they say, "We need this to sound like Papa Roach," I can hook up this kind of chain, and [snaps fingers] it's done. I have to fine-tune it to make sure I don't over-compress, and I have to play some other little tricks that are a little out of the norm, but they work.

Is it a matter of experimentation, or do you think to yourself, "this is going to work better with this piece of gear"?

I kind of know which one's going to work. But I'll still try popping things in and out to see what will happen. You can tell instantly, yes or no.

Usually, when a client comes in, I try to get as much preproduction information as possible: How did you track this? What kind of microphones were you using? Did you use any compression? Not only does it help educate me as to what the process was, but it also helps me create a relationship. If I'm going to be sitting here with someone for ten hours, it's good to have that open dialog right from the beginning; it builds a comfort level.

What formats do you usually receive mixes on?

DAT and half-inch. Sometimes I get CD-Rs, but there can be problems with them — especially if there's a label on the disc. It may not perform the same way. Ninety percent of the time it does, but for that 10%...it's, "Hey, do you have a DAT?"

What tips can you give our readers in terms of preparing their material for mastering?

One thing is, don't over-compress! That makes things very, very difficult. The other thing is to really listen to your mixes. Take them into other environments. Listen to them on lots of different speaker systems.

What do you do when a project comes in that may be technically adequate, but you don't personally like it?

That's a tough question. Fortunately, I like all genres of music. Because I started out in the manufacturing process, then doing hundreds of albums with Disney, I was literally working on raindrops, rivers flowing, things like that. You kind of develop not so much a like or dislike, but a sonic treatment of what you can do to make something sound like a commercial release.

I've had very, very few experiences where I literally cringe. I'm just listening to the material; sometimes I don't even catch what the lyrics are saying because I'm only listening to the tone of the voice. I'm focusing more on the technical aspects.

A skilled mastering engineer can make a huge difference to the sound of a recording, but isn't that also a function of how much time the client is willing to have you expend on a project?

With a major name artist, they're going to let you go for as long as you need. But with clients that are on a shoestring budget, they pretty much tell me up front how much they can pay me, and I try to work out a package deal with them. I'll say, "Okay, you've got six songs; in general, this is how long it takes me, and this is how much I'm going to charge you." Usually they're fine with that. I know pretty much how long it's going to take.

Have you ever gotten a project where you literally didn't have to do anything?

Not a whole album, but there were a couple of tracks where I didn't have to do anything. It's very rare. [Laughs.]

What advice can you give our readers who want to become mastering engineers?

Learn as much about technical aspects as you can. Sometimes mastering houses hire interns, so there's a way to break in. The bottom line is to just learn, and it takes years. Persevere — keep working at it. Ask lots of questions because no question is a dumb question. ■

PHOTO BY WES BENDER

RCA BK-4A

MICROPHONE NAME: RCA BK-4A
FROM THE COLLECTION OF: Bob Paquette, The Microphone Museum, Milwaukee, WI
YEAR OF MANUFACTURE: Circa 1950
PRICE WHEN NEW: $185
TYPE OF MIC: Ribbon
POLAR PATTERN: Non-directional
FREQUENCY RESPONSE: 70 Hz to 15,000 Hz
OUTPUT LEVEL: –60 dBm at 1,000 Hz, ref. to a sound pressure of 10 dynes/square centimeter
HUM PICKUP: –125 dBm, maximum at 60 Hz, 0.001 Gauss
ELECTRICAL IMPEDANCE: 250 ohms
MOUNTING: Half-inch pipe thread
FINISH: TV gray
DIMENSIONS: 12 inches (length) x 1.25 inches (maximum diameter)
WEIGHT: 15 ounces, less cable
MIC NOTES: The BK-4A holds an extremely important place in the development of RCA's single-ribbon, "polydirectional" or multi-pattern microphone technology. It was during development of the BK-4A that Dr. Harry F. Olson (an RCA engineer) discovered a construction "defect" that ultimately led to the multi-pattern technology behind RCA's enormously popular 77DX.

Connected to the rear of the ribbon assembly in the BK-4A is a folded, acoustically damped pipe intended to form an "acoustic resistance termination" for the diaphragm. In other words, the pipe acoustically "loads" the diaphragm much like an enclosure would load a loudspeaker. During the course of his work, Olson found a defect in the pipe. An acoustic "leak" or opening in the pipe was modifying the polar response of the BK-4A from its intended omnidirectional response. This discovery formed the basis for the concept of using a variable aperture in the chamber behind the diaphragm to modify a microphone's directional character.

RCA's original catalog number for the BK-4A was MI-11005.

Technical and historical data courtesy of Arthur Garcia.

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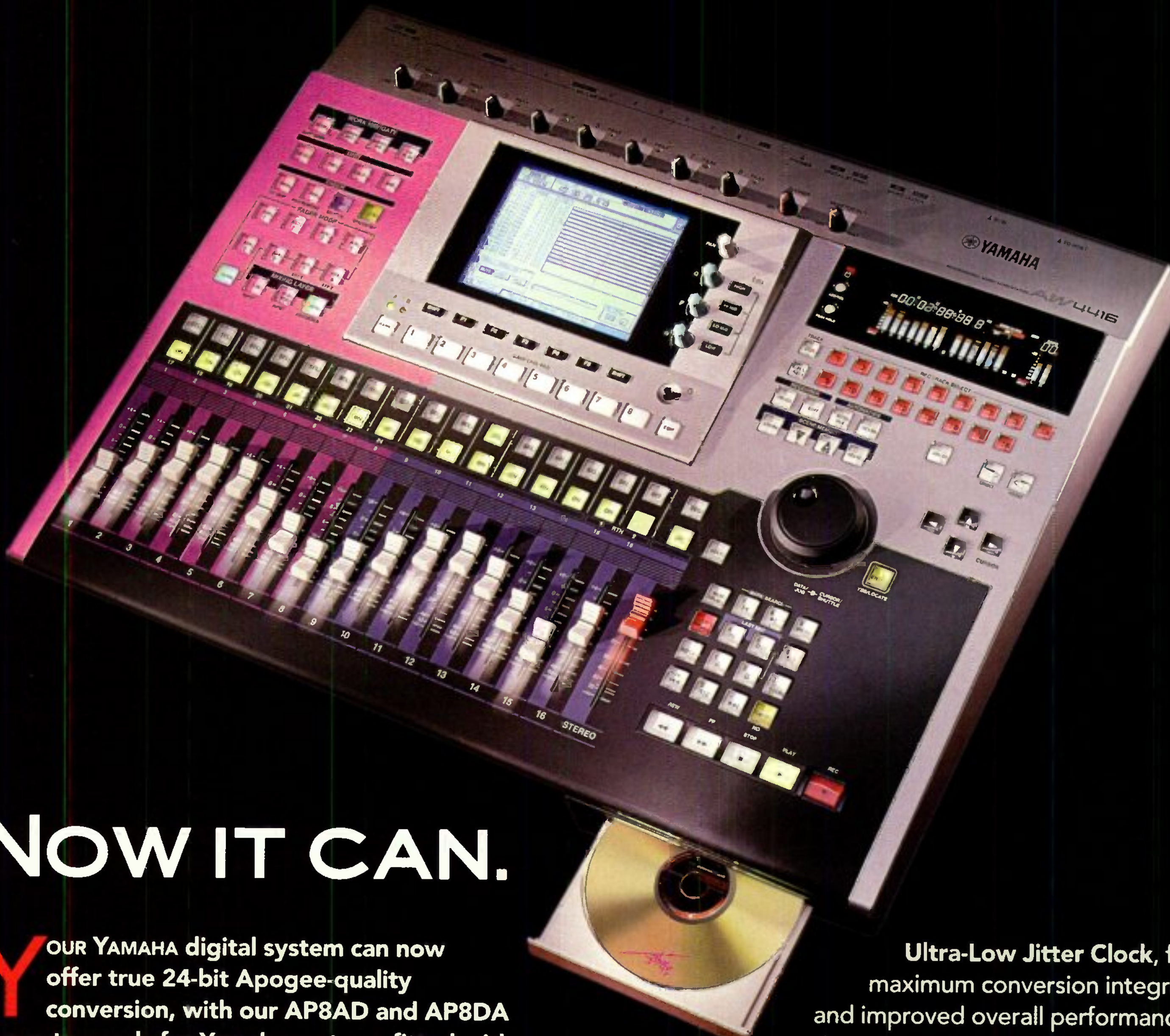

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Apogee's Yamaha-compatible converter cards also fit full-size YGDAI slots.

Apogee's AP8AD and AP8DA cards are available from Apogee authorized dealers, and from selected Yamaha dealers.

*Compatibility and maximum sample rates are subject to host system's support – please check compatibility before purchase.

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* based on current U.S. list and pro audio dealer "street" prices at the time of ad production and on the assumption you will buy a CRT-type SVGA monitor and not an ultra-pricey flat panel model like our art director insisted on using for this ad.

** based on average of length of current pop songs using 24 tracks @48Hz/24-bits and a liberal number of extra regions and virtual takes. Does not apply to extended trance remixes.

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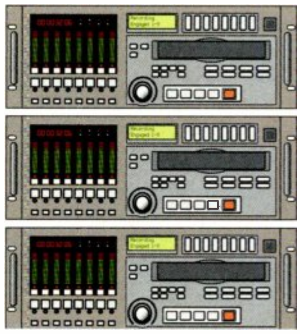

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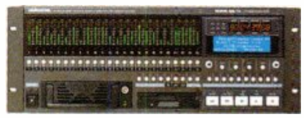

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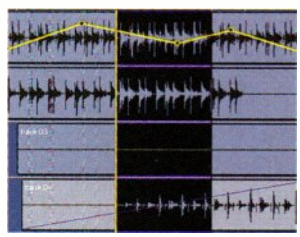

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Capturing the performance

John Paterno

SUSPECT:	John Paterno
ANCESTRY:	Italian and Irish
OCCUPATION:	Engineer/mixer
BIRTHPLACE:	Glen Cove, NY
RESIDENCE:	Los Angeles, CA
VEHICLE:	2000 Dodge Grand Caravan
DIET:	Italian food and red wine
PET PEEVES:	Telephones in the control room; covers of Beatles songs being used to sell products; hearing a mix on the radio and it not being on the record.
CREDITS:	Suspect has often worked with producer Mitchell Froom and has engineered for Joan Osborne, Los Lobos, Vonda Shepard, Tim McGraw, The Blue Horn, Bonnie Raitt, Ted Hawkins, Box Set, Julia Darling, Suzanne Vega, Sheryl Crow, Eros Ramazzotti, etc.

EQ: What was your first recording gig?

JOHN PATERNO: I went to the University of Miami, so I got a little practical experience there, but my first real gig was in L.A. as an assistant at Cornerstone Recorders in Chatsworth.

What was your first job as a solo engineer?

I think it was the Ted Hawkins record *The Next Hundred Years* with Tony Berg producing.

What kind of gear do you own and why?

Basically I look for things that make it easier and faster to do the gig. I try to only buy things that I know I'll use on a regular basis or that most studios don't have. And, believe it or not, I really obsess over whether or not to buy something. The [Empirical Labs] Distressor, the ADL 1000, the Alan Smart CL2, and the EAR 660 are all things you don't necessarily find in a lot of studios. I got the [Teletronix] LA3's recently because they're one of my favorites, and I'm always bummed when there isn't one around. I guess I like compression, huh?

Mics and guitar pedals are also fun and practical purchases. The Altec 633 (salt-shaker) is just a fun, ready-made crunch mic. The [Neumann] KM 86 is my "desert island" mic — as long as the island has phantom power. I just picked up the Soundelux U95S — it takes a lot of SPL and definitely has that [Neumann] U 47 midrange quality that I'm into. For pedals, I'll try just about anything in any situation. I've got quite a few to choose from. All of the Voodoo Lab pedals I've bought sound great. I use them all the time for tracking and mixing. The Proctavia on a background vocal is a very cool sound. And the ZVex stuff is insane! We used a Fuzz Factory on "Love is Alive" on the Joan Osborne record, direct into the console. The Little Labs PCP is the box of choice to interface the pedal world with the line-level world. There's nothing worse than blowing up a guitar pedal you spent a ton of cash on. The built-in DI sounds great, as does their stand-alone DI. It's also a great guitar splitter, among other things.

What studios do you like working in?

My favorites are Sound Factory, Henson/A&M, Sony Music Studios in Santa Monica, and Sound City. They all have three things in common, in varying degrees: great sounding rooms, great maintenance, and a great staff.

What type of console do you prefer?

A working one! It depends on the project, but generally an old Neve or API for tracking. For mixing, I've really gotten into the SSL sound, E series, and also the 9000.

How do you compare analog and digital technology?

They're two different sounds. They both put their sonic signatures onto things. Tape

does its compression thing, and digital, to me, changes the size — makes it a little less deep to my ears. If I have to use digital, I like to try to get it on analog first, especially drums, and then transfer it to [Sony] 3348 or [Digidesign] Pro Tools. My tape of choice is BASF — it sounds like I expect it to sound on playback.

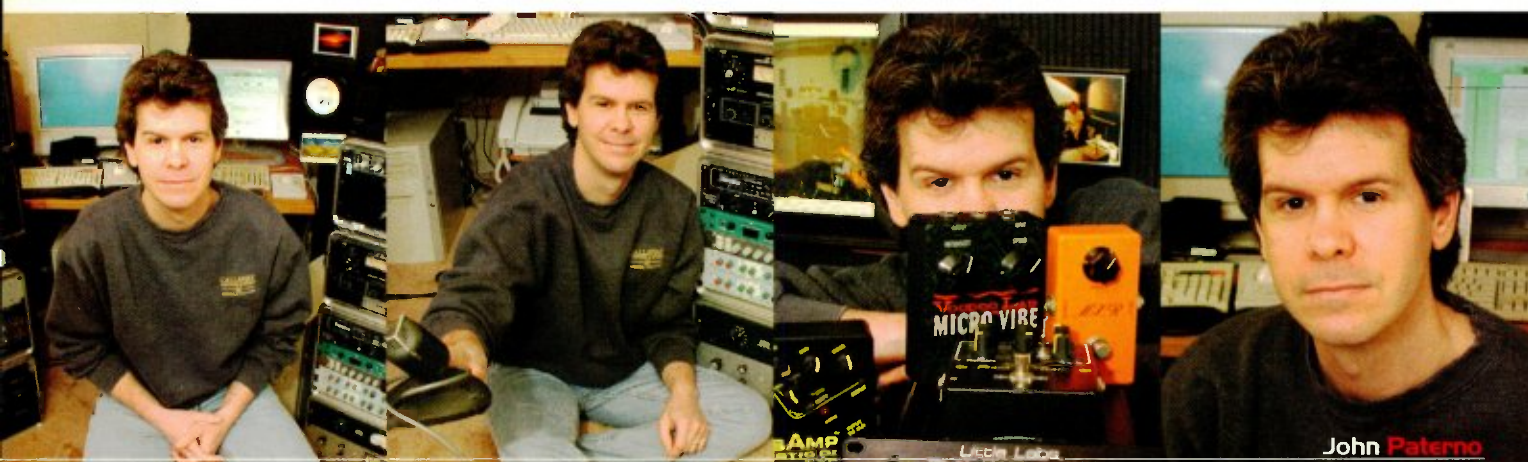
Is there something novel or peculiar about your use of Pro Tools?

As much as I like the possibilities Pro Tools gives you, I try not to use it to manufacture performances — although sometimes you have to. Once I worked with someone who couldn't sing to an existing track for some reason. The producer and I had the artist sing it wild, just singing and playing an electric guitar to minimize bleed. We did six passes. We picked the lines, and I took it line by line and put it together. It came out okay — it gave the illusion of performance — but it took a long time because I'm always anal about making it sound like one. But it's *never* as cool as if it were a real one. I also pretty much refuse to edit drums. I'll cut tape to put together a complete track, but I won't sit down and go through and cut drums against a grid. It goes against everything I got into making records for, which, in essence, is capturing great performances.

What did you learn from your years as an assistant to Tchad Blake?

I'll try to be brief. From a technical standpoint, let me just say that I learned that making sure it sounds good in the room is as important as what you do to it once it comes up on the console. From a broader, more philosophical view, what you do outside the studio has a direct effect on what you bring into the studio. Whatever he gets himself into — engineering, binaural recording, photography, sculpture, single malt scotch — he goes in 100% and is completely into it. He doesn't settle. He always seems to translate those experiences into new ways of looking at things in the studio. That has had much more of an influence on me than his [Tech 21] SansAmp settings, although I do have some of those written down somewhere....

Let's touch on the new Joan



John Paterno


SHURE
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Osborne album, produced by Mitchell Froom. In a previous interview, he referred to the band's attractive sound — that it doesn't sound like studio musicians.

First, they do play together as a band — both with Vonda Shepard and as their own trio. Second, they're very aware of their sounds. The sound of Pete Thomas's drums is very important to him. He'll go out of his way to get what he feels is right for the song, both by his drum choices and tuning choices. Davey Faragher on bass, Val McCallum on guitar, and Mitchell are all the same way. They're not afraid. It almost always sounds good in the room before I even move a fader. It's also a big help that I've worked with these guys before; it feels like a team effort when we work together.

How did the basic tracking proceed?

We recorded the record pretty much live in Studio A at Sound Factory, which has a main room, a piano iso, a "live" iso area, and a machine room that can also be used for iso. The console was a 36-input API, and I used Neve 1073 modules for the drums. Drums and bass were set up in the main room. Drum mics included a [AKG] D112 and [Neumann] U 47 fet for the kick, a [Shure] SM57 and [AKG] 452 for snare, [AKG] 414's for toms, and 452's for overheads and hihat. Toms were recorded into the overhead tracks to conserve space on the tape. There were several room/close kit mics, including an Altec 633, [Neumann] U 87's, and a [Sennheiser] 441. The bass was a Little Labs DI and a SansAmp acoustic DI. Guitars were done in the "live" iso with a '57 and KM86. Mitchell's keys were done through various pedals and DI'd into the console. Joan's vocal was done using Mitchell's Telefunken 251 through a Hardy M1 mic pre and an ADL 1000 compressor right to tape. The tape was BASF 911, 15 ips, with Dolby SR. We stayed on one machine, except for the strings.

The main thing about working with this group of players is that they are extremely fast. The plan is to be prepared for anything, because you never know what will make its way to the final version. My assistant, Joe Zook, did a great job keeping up with me. What basically happens is they sit in a group, learn the tune, and work out an initial arrangement. If there's a loop involved, we'll work that up in Pro Tools so that it can be played live and printed to a few tracks while the musicians are doing the take. Then they all go out to their instruments at the same time and start playing. It's a bit of chaos at first, because we're working out sounds and parts at the same time. But it all seems to fall together quickly most of the time. I usually at least get a basic drum thing

▶ continued on page 136


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CIRCLE 05 ON FREE INFO CARD

Live recording guru Steve Remote reveals his tricks of the trade

Drum Miking for Live Recordings

by Steve Remote

As one of the most famous jazz venues in the world, the Blue Note in New York City has played host over the years to a who's who in jazz music: George Benson, Ron Carter, Larry Carlton, Chick Corea, McCoy Tyner, David Sanborn, and many others. Blue Note co-owner Steve Bensusan brought in Jack Kreisberg to form Half Note Records, a label that's producing a series of CDs recorded live at the club. Recorded, engineered, and mixed by Steve Remote of AuraSonic Ltd. and produced by Kreisberg, these releases include live performances by the Paquito D'Rivera Quintet, Elvin Jones, Will Calhoun, the Irwin Mayfield Sextet, and the David Morgan Trio, to name a few. Remote is using some interesting mic placement techniques to ensure that the sound quality of the recordings is at a maximum, while onstage leakage is kept to a minimum — a critical concern when recording in a live situation. Here he discusses his approach to capturing drums for live recordings.

Recording a live performance can be a challenging gig because you want to achieve the best possible recording without changing the artists' world or making them uncomfortable. It's a balance between the ultimate placement of a microphone versus what I call the "virtual gobo" — especially for drums. Obviously we can't always put gobos on stage to isolate musicians during a live performance, but there are some simple ways to take advantage of microphone pickup (polar) patterns to increase isolation and reduce leakage problems. By getting it right in the recording process, there's no need to "fix it in the mix." By the way, you can use these ideas for sound reinforcement situations as well.

The Half Note recordings are multitracked to TASCAM DA-88's through API 3124 mic preamps. I try to keep the number of mics on the drum kit at a minimum since these are the most likely to pick up other instruments on the stage. Lately I've been using a Beyerdynamic M88 on the kick, Neumann TLM 103's for the snare and toms, and Beyerdynamic M160 ribbon mics for overheads. In some situations I might add an Audio-Technica AT4050 for the hi-hat, but, for a jazz group I typically don't need it. I also may put up what I call a "sweet spot" mic for the entire kit — a Neumann TLM 103 placed right behind the drummer's head, about six feet high and tilted down toward the toms by a few degrees.

A BAFFLING SITUATION

Although there's no carpeting or baffling around the drums, we're able to achieve pretty good isolation of the drum mics from the bass, guitar, and piano — even where the M160 overheads are concerned. In the case of a trio, we might have the drummer on stage left, bass player in the center, and the piano player on stage right.

My drum miking technique has been fairly consistent regardless of

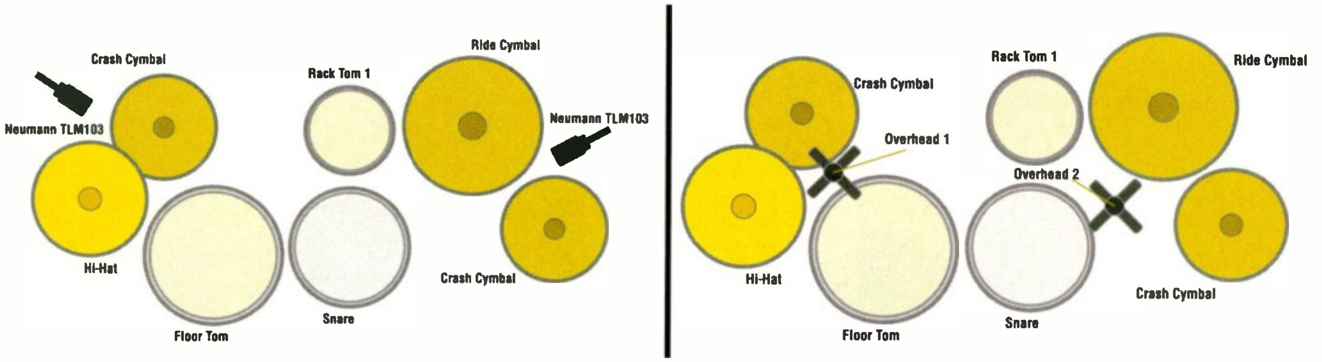

FIGURE 1 (LEFT) and FIGURE 2

the drummer and kit. The basic idea is that I position the mics to take advantage of their pickup patterns, and reduce leakage as much as possible. One of the TLM 103's is placed on the hi-hat side of the kit, at roughly the height of the top of the snare drum. I aim that mic toward where the drummer is sitting, but rather than close-mic the snare, I back the mic off anywhere from six to twelve inches away. Backing it off avoids that close-miked snare sound (which really isn't appropriate for a jazz drummer), while giving me some of the hi-hat and rack tom. Since the TLM 103 is a cardioid mic, maximum rejection is at the rear of the mic. I try to rotate the mic so that the rear is pointing toward a stage monitor or whatever instrument might be to the drummer's left.

On the other side of the kit I place another TLM 103, about the same distance away from the floor tom, ride cymbal, and rack tom so that it picks up that side of the kit (see fig. 1). This mic is also rotated slightly, so that the rejection point is facing toward the bass amp, minimizing leakage from the bass amp. To check phase between the mics, I solo various combinations of mics in mono and listen carefully to whether the low end becomes weaker when they're added together. If the low end does weaken, then I may have to move a mic to another position.

UP AND OVER

Overhead mics can be a source of unwanted leakage due to the fact that they're generally placed around the kit pointing in "unfriendly" directions. For me, the key to isolating the M160's is placing them high up, pointing straight down. Looking at the floor tom side of the kit, I try to center one mic over the floor tom and cymbals (see fig. 2). On the other side of the kit there might be one or two cymbals plus the hi-hat, so I'll make sure that this overhead mic is centered somewhere between the snare and the cymbals.

What's interesting to me about this arrangement is that the higher I place the overhead mics above the kit, the more cymbals and drum kit I get, while the less bleed I get from the stage. It's the reverse of what you'd expect, but the higher they go, the better the isolation of the kit from the rest of the stage. Bringing the overheads down close will give you more of the bell of the ride cymbal, but remember that you're also closer to the other instruments, monitors, and amplifiers.

Sixty percent of my drum sound is a balance between the two overheads and the kick drum mic (see sidebar for details on miking the kick drum).

OPTIONS OUT OF THE GATE

Even though I have about 32 channels of gate in my truck, I rarely use them during recording because they can chop off some of the subtleties that a drummer might play. In a rock or pop (sometimes jazz) recording situation, I usually use Sennheiser MD409's for the toms. By placing those mics an inch away from the skin, no surrounding sound is going to be as loud as that drum; it acts as its own gate. By eliminating the gates, you hear a natural sound instead of a chopped-off sound.

If you have a drum wedge sitting on stage blasting away toward a mic, there are ways to reduce leakage from the wedge without using a gate. Change the position of the mic to take advantage of its directionality. Fig. 3 shows a typical close-up placement for a snare drum mic. This is fine for recording, but the mic is likely to pick up sound from the drum fill monitor

CAPTURING THE KICK

Over the past 16 years, Steve Remote and I have worked on countless AuraSonic projects. We're constantly sharing ideas and concepts that span our common interests — drum sounds included!

Let's focus on kick drum; occasionally we use multiple mics, but usually one mic on the Half Note live recordings. Since this article deals with jazz, let's presume that the drummer is using a two-headed drum (*i.e.*, front and back heads without a mic hole). For instance, Elvin Jones uses both heads on his bass drum, which are always tuned very tight. He also uses a wood beater for plenty of attack and click, without using another mic at the batter head. On his 73rd Birthday Tribute at the Blue Note, we used a Beyerdynamic M88 approximately one inch away from the front head on the audience side, placed halfway between the center of the drum and the rim.

If there's a hole in the front skin, we tend to place the M88 just inside it. When isolation is needed and gobos aren't practical, Steve uses Popper Stoppers modified with cardboard — and sometimes foam and cardboard — to create a mini-shield.

—Andy Bigan, Keeptimeny@aol.com

FIGURE 3 (left) and FIGURE 4

shown near the hi-hat. By moving the back of the snare mic to the position shown in fig. 4, you can still get the great sound of the snare but — with the rejection point of the mic facing both the hi-hat and the drum fill — you'll increase the isolation of the snare drum from the rest of the stage. In a loud situation, where you have electric guitar and bass amps on either side of the drums, you can point the speaker cabinet mics slightly away from the drum kit to help improve isolation of the amps from the drums.

SOUNDCHECK

When we're about to do a soundcheck, the drum tech or drummer will ask me if I want to work on the drums. I'd rather do a line check of all the inputs and then hear the entire band play a tune so I can get the sound while everybody is rockin'. That's when I'll know for sure if the sounds are working together. Plus, I can check the leakage and phase between the mics in about 20 minutes while the band is playing. At that point I want to identify anything that might need to be changed to make the sound of the entire band work. I hate the "fix it in the mix" or the "throw a gate on it" mentality. That's not what it's about. It's about getting the sounds right on tape (or disc), maintaining the phase coherency, and making sure that you minimize the "bad" leakage.

Steve Remote may be reached at Aurasonic@aol.com or 718-886-6500. Web: www.AuraSonicLtd.com.

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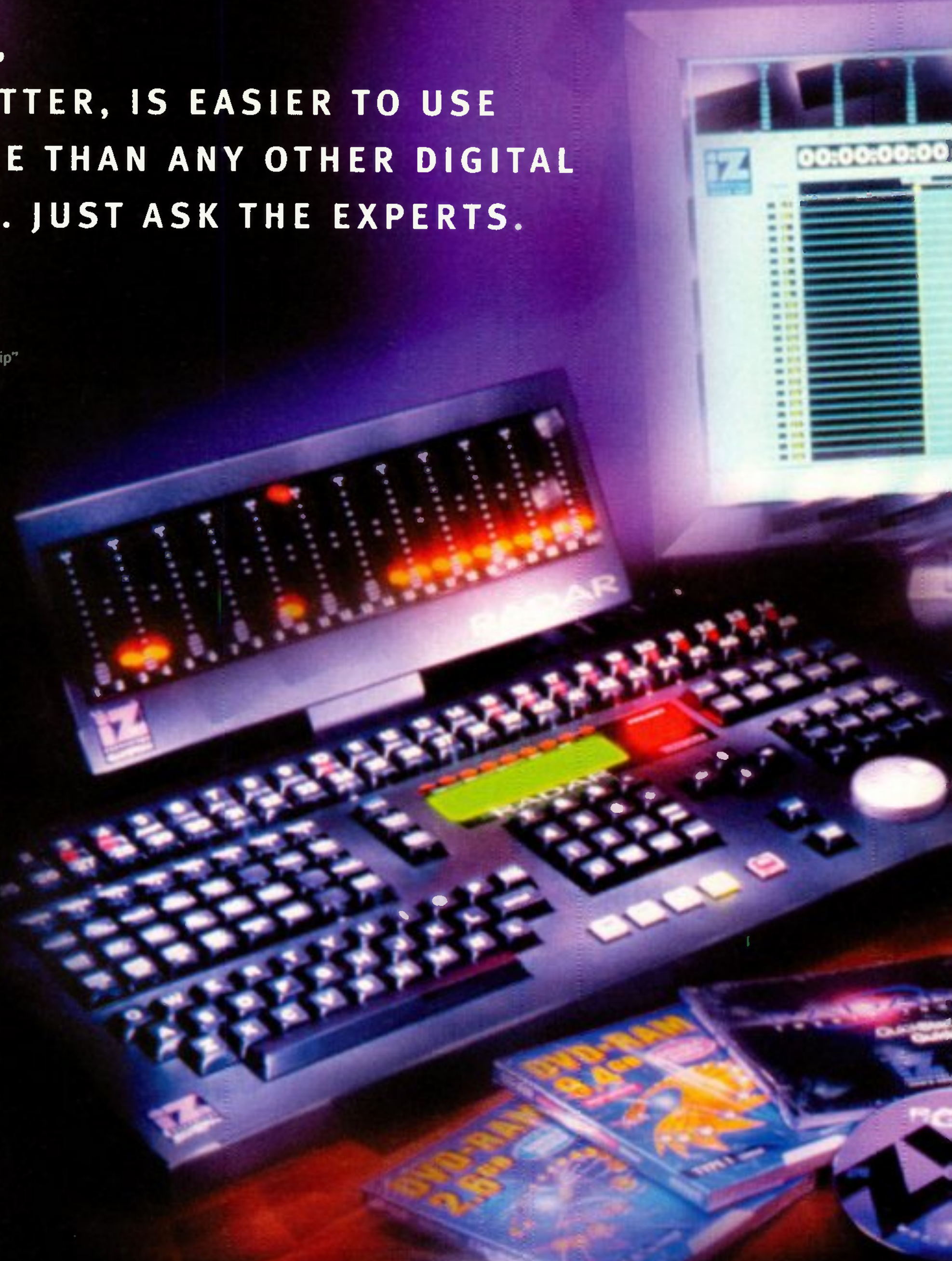

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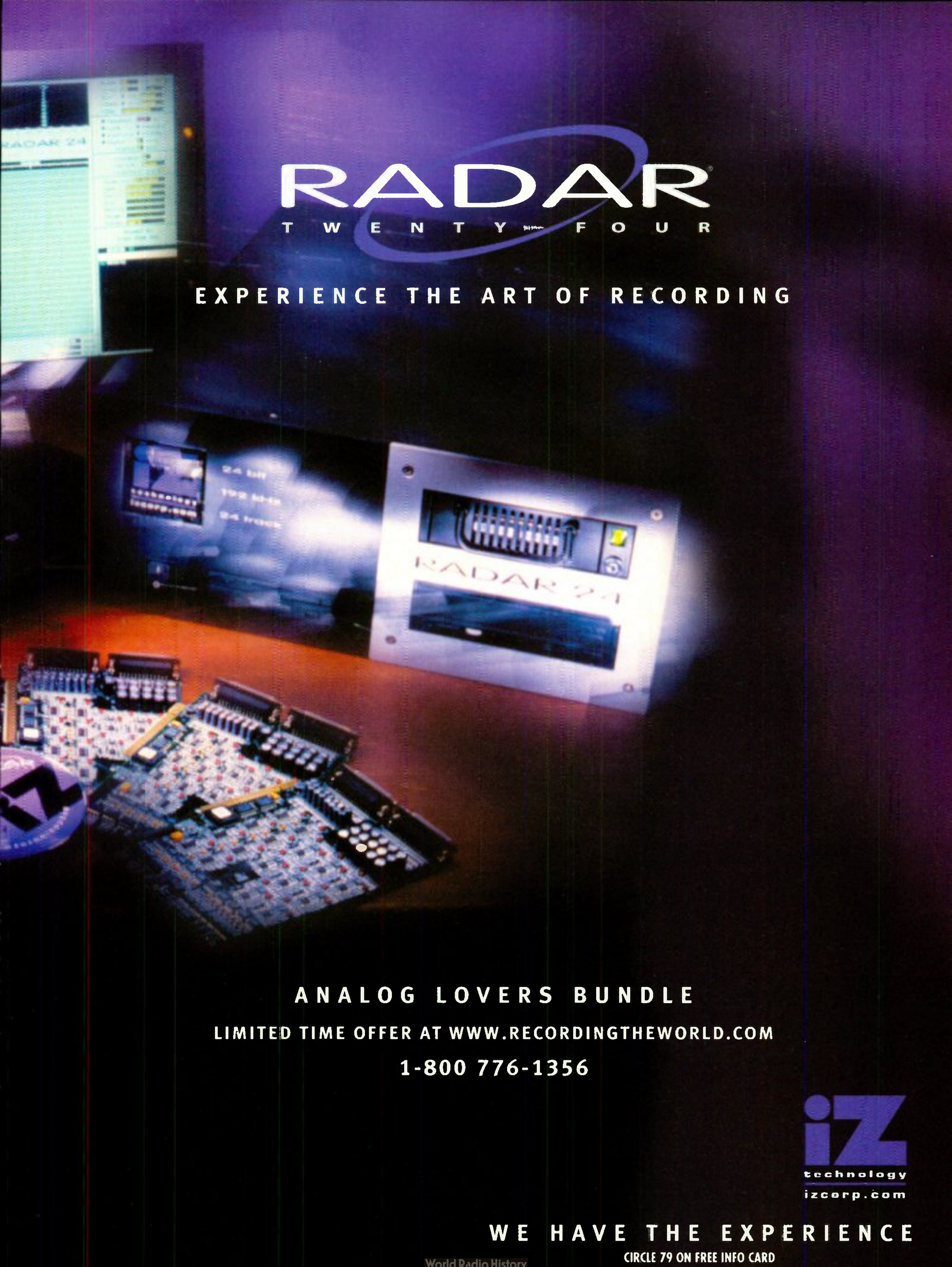

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Avoid being surrounded by mistakes

Surround Master Preparation

by Bobby Owsinski

Surround sound brings a new level of complexity to the entire process of recording, mixing, and mastering. With at least four extra channels, there's now that much more room for error. After all, if some people can still screw up stereo (switching left with right, one channel out of phase, etc.), imagine what can happen in surround. One thing's for sure, though: master media prep has become more critical and necessary than ever before. Here are some suggestions and guidelines to follow while mixing in order to avoid any unforeseen difficulties later.

CHANNEL ASSIGNMENTS

During a surround mix, sooner or later the question of channel assignment on the master recorder (be it tape or hard disk) always arises. Just what is the correct track assignment? Actually there's always been several ways, with some good reasons behind each one. Just to get things off on the right foot and cut through any confusion, let's start with the one that's quickly becoming the *de facto* standard. A dedicated stereo mix, or Lt/Rt (which means Left TOTAL and Right TOTAL and is a mix of all the surround tracks to stereo), can be recorded onto Tracks 7 and 8.

This format is the SMPTE/ITU standard (see Chart 1), as well as the assignment matrix suggested by Dolby. The configuration transfers easily to the corresponding four audio tracks (L, R, C, LFE) of the most widely used video formats today such as DigiBeta or D5, with the idea that if something happened to channel pair 1 and 2, the all-important center channel (the one with the dialog in film or video) still remains intact. This format is also recommended by Dolby, as it is the common pairing of channels in Dolby Digital encoding (although the AC-3 encoder can actually be configured to any track configuration). The surround products of Panasonic, Mackie, and Martinsound, to name just a few, now support this configuration as well.

The following two assignment methods are also used, but you see them less and less as the SMPTE/ITU standard becomes more widespread: Chart 2 shows a configuration that many film studios use, although it's seen in some music production as well. It seems to make sense in that it's a somewhat visual representation of the way the speakers are laid out, but falls short when it comes to logical track pairings.

Chart 3 is the channel assignment method preferred by DTS. Again, the pairings are logical, but the placement is different from the SMPTE/ITU standard. Tracks 7 and 8 usually contain the stereo version of the mix, if one is needed.

There are other assignment permuta-

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Client ____________ Producer ____________

Studio ____________ Engineer ____________

Sample Rate	☐ 48kHz	☐ 96kHz	☐ Other	
Bit Resolution	☐ 24bit	☐ 20bit	☐ 16 bit	
Reference Level SPL	☐ 85dB	☐ 79dB	☐ Other	
Tape Format	☐ DTRS	☐ Optical	☐ Hard Disc	☐ Other
Time Code Format	☐ 29.97DF	☐ 30 NDF	☐ Other	
Surround Level SPL	☐ Equal to Front		☐ -3dB to Front	
Leader Contents	☐ Audio Slate ☐ 2 Pop ☐ Other		☐ 1kHz tone @ -20dBFS ☐ Pink Noise @ -20dBFS	
LFE Lowpass Filter	☐ On		☐ Off	
☐ Original Master	☐ Simultaneous Production Master			☐ Protection Dub

Track	1	2	3	4	5	6	7	8
☐	L	R	C	LFE	LS	RS	LT	RT
☐ Other								

TIME CODE	PROGRAM TITLE	Mix	Extraction
: : :		☐	☐
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: : :		☐	☐
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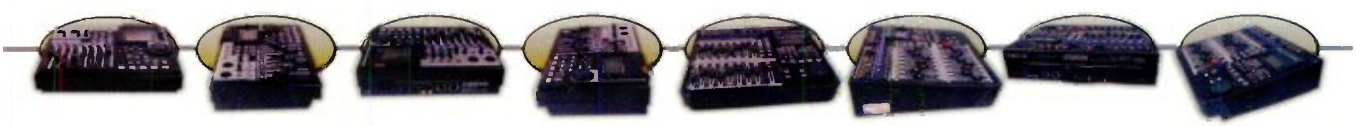

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CIRCLE 44 ON FREE INFO CARD

CHART 1: SMPTE AND ITU TRACK ASSIGNMENT STANDARD

1	2	3	4	5	6
Left Front	Right Front	Center	LFE	Left Surround	Right Surround

CHART 2: PREFERRED FILM TRACK ASSIGNMENT

1	2	3	4	5	6
Left Front	Center	Right Front	Left Surround	Right Surround	LFE

CHART 3: DTS TRACK ASSIGNMENT STANDARD

1	2	3	4	5	6
Left Front	Right Front	Left Surround	Right Surround	Center	LFE

tions that are occasionally used, but all seem to be falling quickly by the wayside as the SMPTE/ITU track assignment method gains favor.

MEDIA PREP

It's imperative that as much information about your project as possible is documented since a surround project goes through many more steps than a CD and therefore has a lot more potential for error.

• **Slate the master.** More than ever before, it's important to not only properly document the master tape or disc, but to prep the master in order to make sure that there're no questions as to the actual track assignments. We've all had instances where the documentation doesn't align with what's on a tape, and this can lead to disaster in surround. Plus, with surround, you can never really be sure that what you're hearing is actually what's been documented.

The best way to avoid confusion is to go back to the admittedly low tech — but foolproof — method of using an audio slate on each channel, indicating the channel assignment (*i.e.*, "Channel one — left front," "Channel six — right surround"). This only takes a minute and provides an ironclad certainty that the proper tracks find their way to the proper channels.

• **Print a test tone.** Print at least 30 seconds of 1 kHz tone at –20 dBFS, which is the SMPTE standard reference level, across all tracks. A 1 kHz tone is a pretty good way to discover if there's any clock discrepancies that will affect the purity of the signal and result in clicks and warbles that might not be heard during the actual program material.

Also keep in mind that any program on tape media should start no earlier than two minutes into the tape, since most errors and dropouts usually occur in the area up to that point.

OTHER THINGS THAT SHOULD BE DOCUMENTED

Proper documentation of a master tape or disc means that there's never a question that can't be answered or a mistake made because of confusion or omission of data.

• **Is the LFE channel filtered, and at what frequency?** This makes it easier to figure out which is the subwoofer channel if the assignment documentation is lost.

• **What's the reference level in SPL?** This refers to the reference level that you used while mixing, which helps the mastering engineer to better approximate what you were hearing during the mix if there's a problem down the line.

• **What's the sampling rate?** Since there's a number of sampling rates now available to mix to (192, 176.2, 96, 88.2, 48, and 44.1 kHz), documenting this helps to avoid any clock or sync issues that may arise during mastering or authoring.

• **What's the bit resolution?** Generally this will be 24, 20, or 16, but an indication is necessary in order to set dither correctly if it's required. Dither is a noise signal that's intentionally introduced into the digital word in order to remove unused or unwanted bits at the end of the digital word (*i.e.*, when a 24-bit program must be converted to a 20-bit program). To simply lop off the bits (truncate) at the end of the word sounds bad, so dither is used instead.

• **If timecode is included, what's the format?** Everything in surround is time-stamped via SMPTE timecode, since it's eventually linked to picture in the delivery format. The mastering or authoring house will absolutely love you if you deliver the audio pre-striped. If you're already synced to picture, then you must be using the same flavor of SMPTE as the picture. Otherwise, I recommend that you get in the habit of always using 29.97 drop frame, which is the NTSC broadcast color standard.

• **Are the surround channels calibrated equally to the front channels or –3 dB?** In film-style mixing, the surround channels are calibrated at –3 dB. Music mixing has the surrounds equal in level to the front speakers.

• **What's the media format and how many pieces are there?** The master may actually be on several pieces of media on several different formats. A media count here can eliminate the confusion of an incomplete mastering or authoring job later.

• **How long is the program?** This is necessary because it determines if data compression must be used.

• **What's the intended final audio resolution (*i.e.*, 24-bit/96 kHz or 20-bit/48 kHz)?** A combination of sample rate and word length determines if data compression is used and how it's set.

• **Any glitches, distortion, dropouts, or anything unusual?** It's always a good idea to indicate this information on any master tape. This stops the mastering engineer from wasting time (and your money) checking his own equipment when a defect actually lies in the master.

Yes, multichannel provides us with multiple chances to screw things up. But, if we just take the time to do some media prep and simple documentation, we can eliminate any problems before they happen.

Bobby Owsinski is a surround mixer and DVD producer, and the author of *The Mixing Engineers Handbook* and *The Mastering Engineer's Handbook*.

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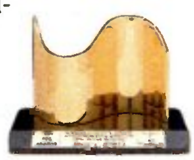

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CIRCLE 35 ON FREE INFO CARD

THE STUDIO-IN-A-BOX REV

When you're putting a recording studio together, you have two choices: One approach is to go for piles of application-specific pieces of gear (preamps, mixers, recorders, computers, software, effects, etc.), then find a room to put it all in, spend time wiring it all up, fixing incompatibilities, troubleshooting, and so on. This has traditionally been the way in which "pro" studios have been assembled — and there's certainly nothing wrong with it. In fact, it allows you to choose exactly the gear you want for each application, make your set up as flexible as you want, and optimize the rig for the sound quality and performance you require.

But, in the past few years, another approach has taken on increasing appeal: The all-in-one digital "studio-in-a-box." Descended from the cassette-based, self-contained recorder/mixers of old, early digital units such as those from Roland and Fostex opened a new chapter in portable

OLUTION

THINK SELF-CONTAINED DIGITAL WORKSTATIONS ARE JUST FOR DEMOS? FIVE PROS PROVE OTHERWISE....

recording. These days, self-contained digital workstations such as the Akai DPS series, Boss BR8, Fostex VR800 and VF16, Korg D series, Roland VS series, TASCAM 788, and Yamaha AW4416 offer tremendous flexibility, tons of processing and editing power, loads of real and virtual tracks, sophisticated mixers, and often the ability to burn a final audio CD right from the unit itself — to say nothing of easy portability. All you need to add is a microphone or sound source or two, a set of monitors or headphones, and (lest we forget) a tad of talent.

But can these "studio-in-a-box" units truly be used to produce professional quality audio? To explore this question, EQ tracked down five Roland VS-series users who are putting their digital recorder/mixers to work on top-level projects. —Mitch Gallagher

YIN-YANG

If you think it's not possible to make a professional quality, commercially releasable album on a portable digital audio workstation, think again. Bassist extraordinaire Victor Wooten's latest double CD release, Yin-Yang, is the proverbial exception to the rule. Recorded and mixed entirely with a pair of Roland VS-1680's operating




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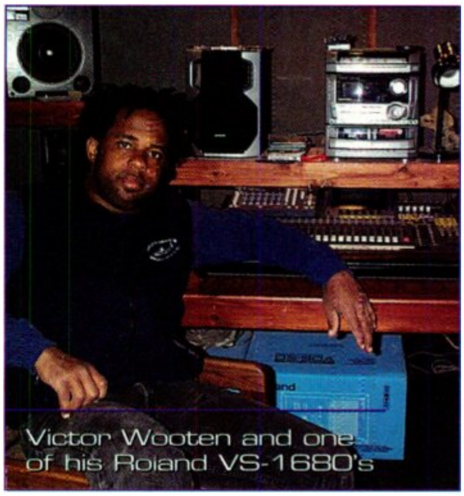
CIRCLE 41 ON FREE INFO CARD

at 20-bit/48 kHz, the only ancillary gear used was a few outboard mic preamps during basic tracking sessions at Nashville's Masterlink and Monkey Finger studios (plus a day's work at Bootsy Collins's home studio for the track "What Crime Is It?"). Wooten — who tours extensively both as a solo artist and with Bela Fleck and the Flecktones — recorded the entire album on VS-1680's "at home and in dressing rooms and hotels across the country."


Victor Wooten and one of his Roland VS-1680's

Engineer Kurt Storey recounts the basic procedure: "I used really good mics and mic pres during tracking, but we recorded directly to the 1680, using its onboard A/D converters. I had to use two machines because you can only record eight tracks at time, so I would have eight tracks of drums on one machine and a couple of more tracks of drums and basic keyboards and guitars on another machine.

"Once that was done, I mixed down the drum tracks off the first machine onto the second one to make a slave file. Then we would overdub on the two machines anywhere we happened to be." A pair of external Seagate Barracuda 9 GB hard drives was used for storage, and archiving of the data was done on a daily basis to a CD-RW drive.

The VS-1680's portable nature made it a natural choice to use for overdubbing while on the road, but the obvious question is why Storey and Wooten chose to use them for tracking and mixing, as opposed to the more traditional options such as multitrack analog or digital tape or the ubiquitous Pro Tools. With disarming candor, Storey confesses, "Victor just has his own way of doing things. We had used a VS-880 on a previous project, and he said, 'Man, I think we can do the whole record on this.' I said, 'Well, okay….' — but it was kind of funny because at the time I hadn't had a lot of experience with this. Initially I had gone into the studio thinking, 'I better run a couple of [TASCAM] DA-88's just to back up with in case we lose something,' but the 1680 sounded so good that I thought, 'Man, I'm just not going to worry about that and just get on with it.'"

> "FOR MOST OF THE TRACKS ON THIS ALBUM, VICTOR JUST PLUGGED DIRECTLY INTO THE GUITAR INPUT JACK OF THE 1680, WITHOUT USING ANY DI BOX OR PREAMP AT ALL."

Victor Wooten's bass sounds are at once distinctive and far-ranging. Many songs on *Yin-Yang* have multiple bass tracks — sometimes as many as eight of them. But Storey is insistent that most of the sound comes directly from Wooten's fingers, as opposed to any magic DI boxes or signal processing. "For most of the tracks on this album, he just plugged directly into the high-impedance guitar input jack of the 1680, without using any DI box or preamplifier at all. There was a Telefunken mic pre over at Monkey Finger that I used on one solo, but that was the exception. I just try to go for natural tones and stay very much in the background."

Storey doesn't even use much compression on Wooten's bass. He explains, "Vic is really more of a soloist, and you don't want to compress that; you want to capture the full dynamics of the instrument. At any given moment he's going fast and loud or slow and gentle and everything in between." For the most part, the only time the bass is compressed is where there are multiple bass tracks and one track is contributing the low bottom-end only. In those cases, an Anthony DeMaria tube compressor was sometimes used on the input side, or gentle compression from the VS-1680 was applied during mixdown.

Vocals (by Wooten and guest artists Bootsy Collins and Tabitha Fair) were recorded almost entirely with AKG SolidTube microphones, and MS miking was used extensively for ambience on the drum tracks.

The mix process was also done entirely within the VS-1680's themselves; no external console or outboard gear was used. "I did most of the mixing while Victor was on the road — just me working alone in the living room of my apartment, monitoring on a pair of Meyer HD-1's," Storey explains. "I spent a total of a couple of weeks on the mixes, then we met on the road and brought the VS-1680's into a studio in Seattle to check some of the mixes."

One of Storey's strengths as an engineer is his tasteful use of reverbs and ability to keep them out of the way of the instruments. "All the reverbs you hear on this record are internal effects from the VS-1680; no outboard signal processors were used. I used a soft ambient patch to enhance the drum sounds a little bit, or on some of the ballads I might use a hall reverb. In general, I might adjust the pre-delay or room size of the factory presets, depending on just how big I wanted the reverb to sound, but I never EQ'd the sends or returns. The main thing I did was to ride the returns a lot for dynamics.

"I had two machines, so when I mixed, I'd have the bass and drums on one machine and the vocals and keyboards on the other. That way, I'd have one dedicated reverb for the drums, another one for the solo bass, and a third one for the vocals. Whatever DSP was left over would be used as inserts for compression or whatever."

All the editing — and there was extensive slicing, dicing, and comping work — was done by Storey solely within the VS-1680. "It worked," he comments, "although I have to admit I sometimes felt it might be easier or faster in Pro Tools. Doing all that editing on the LCD did a number on me, though — my girlfriend's a massage therapist and she kept telling me, 'Man, your neck is messed up....'" [*Laughs.*]

Mastering, done by Randy Leroy at Final Stage in Nashville, was equally interesting. "I brought the VS-1680's into the mastering room," Storey recalls, "and just played back the automated mixes right then and there, so it was first-generation real-time, going from the S/PDIF output straight into Randy's SADIE system.

"Actually, we were tweaking the mixes even during mastering!" he laughingly recalls. "About a week before our mastering date we had pretty much everything done, but then Victor realized that we had ten instrumentals and nine vocal songs and said, 'Man, we have to add one more track just to make it even, like Yin and Yang, right?' We had done this track with Bootsy Collins a couple of years back, and somehow the master got lost. Victor only had part of a drum track and a basic vocal track. So I had to recreate the bass loop and then Victor took that up to Bootsy's home studio and retracked the drums and added some more vocals. It was so last minute; I not only had to mix it over headphones in the lounge of the mastering room, but Victor was literally doing bass overdubs while I was mixing!"

> "THEY HAD THREE 1680'S LOCKED TOGETHER, MIXING ON HEADPHONES, WITH VICTOR PLUGGED IN AND DOING BASS OVERDUBS!"

Leroy recalls the rather bizarre scene vividly: "They were actually mixing some songs in the room next to me while I was mastering other ones! And, at that point in time, I was thinking, 'Oh my God, what have I got myself into?' What I'd heard was sounding really good, but the thought of a mix that was going on in a room next to me — they had three 1680's locked together, mixing on headphones, with Victor plugged in and doing bass overdubs! It was something else; I was stunned and I was scared." [*Laughs.*]

But when it came to sound quality, Leroy says, "I was surprised. It was that particular project that really opened my eyes as to what the VS-1680 was like. I get a lot of projects in that are done on 02R's, ADAT's, Mackies, DA-88's, and things like that. Generally, those projects don't quite compare sonically to projects that come in that were tracked on [Sony] 3348's and mixed to 1/2-inch tape or stuff like that. But when you compare the 1680 to the mixes that I get in from 02R's and things like that — all those dedicated tape machines and dedicated midline consoles — they didn't have anything on that all-in-one box, not at all."

Another important advantage was that, because there was no two-track media, there was no intermediate step or conversion process. Leroy recounts his mastering signal chain: "We had one 1680 in the room that was playing automated mixes. I took its S/PDIF digital output into a ZSystem router, which converted the signal into AES/EBU. From there it was routed to Apogee PSX-100 D/A converters. Processing was done in the analog domain, through a Manley Stereo Mastering Variable Mu compressor and a GML 9500 Mastering EQ. The signal was then routed to a Wyse ADC1 Mk II A/D converter for conversion to digital again, and finally into my Sonic system for assembly. To control peaks, I would sometimes put a little bit of limiting on with a Wyse DS1 working in the digital domain." D/A conversion was done at 88.2, 24-bit, which was then sample-rate-converted to 44.1/24-bit. The 24-bit words were then dithered down to 16 bits using UV-22 processing in another Apogee PSX-100.

Leroy sums up his experience mastering *Yin-Yang* this way: "When they started playing back those mixes, I was, like, damn! Effect-wise, tonality-wise, stereo definition, depth of field, everything — the VS-1680 totally surprised me." Proof that quality musicianship plus quality gear in the hands of a skilled engineer is the winning formula for professional results. *—Howard Massey*

LEARN MORE ABOUT VICTOR WOOTEN'S BID FOR BASS SUPREMACY AT WWW.VICTORWOOTEN.COM.

THE VOICE OF MAGIC

Last year, producer David DelGrosso (Open Heart Records) called me to help him prepare the material of a great new R&B vocalist named Raven, who had recorded and produced her album herself using Roland VS Series workstations. "I started with the 880," states Raven, "and then I upgraded to the 1680, so some songs were done on one machine and some on the other. The 1680 was great. The format has prepared me to work with other hard-disk formats like Pro Tools because the features are similar. Plus I was very pleased with the internal effects of the Roland."

However, the material needed mastering, and I agreed to help. We booked a weekend of studio time beginning late Friday night and finishing early Monday morning. David and Raven arrived late Friday night and we decided to get right into it. I listened to each song first to get an idea of how it sounded and what it needed. Most of the songs were on DAT with a couple on CD-R. I was surprised at how different each song actually sounded from a sonic point of view. Even though the 1680 was used to record most of the material, it wasn't always used to mix it. In fact, a couple of the songs had been mixed on an SSL. Our mission was to give each song a consistent "sonic signature" — glue it together and make it not only better, but also more cohesive.

David and Raven were sure of what they wanted. It was my job to find the technology and make the moves that would realize their ideas. The first phase of mastering was the transfer of the material into Pro Tools. I decided to record each of the songs through an SSL Compressor (G384) and then through an Apogee AD8000SE

PHOTO BY EDWARD COLVER

Raven and daughter enjoy her Roland-based home studio.

converter. Only the songs that had been mixed through an SSL were *not* recorded through the G384; those songs were digitized into Pro Tools through the AD8000SE using Soft Limit (the other songs didn't have Soft Limit "on"). After each song had been loaded into Pro Tools, it was time to perform the sonic surgery.

The biggest challenge in smoothing out each track was the sound of the low end. The low end differed so much from track to track that I literally had a different "aux" for each song. We loaded up to five plug-ins per stereo aux input and routed the appropriate song through it. That way each song could have its own unique signal chain. When listening from song to song, we could quickly and easily make changes. On the majority of songs, we used a Waves C4 multi-band compressor/EQ to control the low end. We would vary the amount of compression per song as well as the cut-off point of the bass content (70 Hz, 80 Hz, etc.). On some of the tracks, we used the DUY Tape and Valve plug-ins to add a specific type of warmth.

Our goal was to make each song have as much "slam" as possible, while maintaining a smooth and cohesive balance from the beginning to the end of the record. We were constantly making little tweaks to the low end, burning a CD, and then listening to those changes on a boom box, in the car, and on a Walkman. Raven and David knew exactly what they wanted changed, and by how much. We were making 0.5 dB and 1 dB tweaks the entire weekend!

When we finally had a balance that David and Raven were happy with, they went back to L.A. and decided that they wanted changes. I felt that it didn't make sense for us to get into a FedEx fling, so I recommended that Brant Biles (Mi Casa Studios) do the tweaks. He did a great job on the final touches, and the project was finally completed. I hope that a lot of you have the opportunity to hear this CD. The songs are great, Raven sings great, and you'll be able to appreciate the sonic consistency that we were able to get from a wide variety of stereo masters!

I have to give credit to the Roland VS-1680. It really is one hell of a box. In mastering, we were able to get the "meat" out of Raven's songs that she and David were after. Had the 1680 not been able to capture those sonics, then "it" wouldn't have been there to bring out.

I know that Raven feels very confident that, in the future, her material will be recorded more consistently. She is now in control of working on the 1680 and knows how to get the most out of it. And that's not all she learned: "I learned about the wonderful gift of mastering and what it can do for a project. I also learned to never underestimate a song — even if you think you might have to record it over. Some of the songs that I thought I was recording only for reference wound up on the record. Now I make sure all the recording conditions are right — just in case."

I can't wait for Round Two.

—David Frangioni

RAVEN'S ALBUM WILL BE RELEASED IN EARLY SUMMER. VISIT RAVEN ONLINE AT WWW.MUSICYOUCHOOSE.COM.

SHENANDOAH 2000

With 25 charted singles spanning a 12-year career (not to mention a Grammy award), Shenandoah certainly has done their share of recording. Yet their latest release, *Shenandoah 2000,* marks a departure from prior recording projects. The release was almost entirely recorded at drummer (and *de facto*-engineer) Mike McGuire's house on a pair of Roland VS-1680's. No iso booths, no fancy mic techniques, and no proper control room. Even in this age of at-home recording, *Shenandoah 2000* sounds like a million.

"We've been making records for 12 years," McGuire begins, "but I've never sat behind the controls of a board. So while I was learning how to use the Roland VS-1680, I asked a lot of questions and read a lot of books on recording. The biggest challenge was making sure everything was recorded to hard disk properly. I was the engineer for the tracks, so I was careful not to over- or under-compress things too much to disk. I also

made sure I didn't do anything weird with the EQ that couldn't be undone. I pretty much cut everything flat, trying to make it sound good from the source. I figured that as long as I didn't screw it up, Brent Sparks could make it sound even better [Sparks cut vocals and mixed *2000* at The Playground in Nashville, TN].

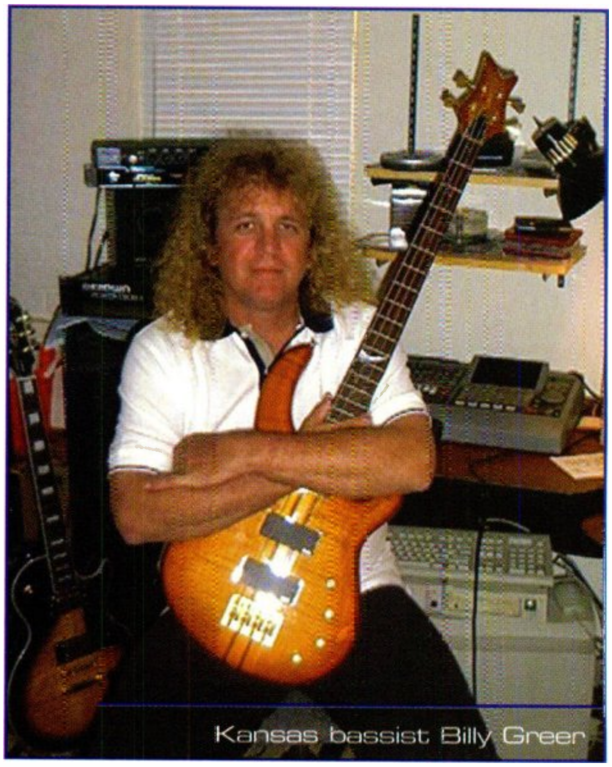
According to lead vocalist Brent Lamb, "Before recording any of the rhythm tracks, we agreed on a tempo for each song. Then we'd record a click in Digital Performer and onto the 1680 as a reference. Mike played Roland V-drums to the click, and recorded the MIDI data into [Mark Of The Unicorn] Digital Performer [running on an Apple PowerBook G3]." Using the V-drums allowed McGuire to edit the parts via MIDI data, and also allowed him to change his sounds instead of, for example, wishing he had "recorded a song with a deeper snare drum rather than the piccolo snare." For many of the songs on *2000*, McGuire laid down a straight groove for the band to record against and re-recorded his parts toward the end of the process — after hearing what the other band members played. He cites "Tied To A Tumbleweed" as an example where guitarist Jim Seales took the song in another direction during his solo, and McGuire re-cut drums to complement Jim's solo.

Shenandoah used one VS-1680 to record drums, percussion, and bass, while a second machine (locked to the first via MIDI timecode) was used to record electric and acoustic guitars, keyboard overdubs, and an occasional scratch vocal. When it came to recording guitars, Brent maintains that "95% of the electric guitars on *Shenandoah 2000* came from the Roland VG-8, so we didn't have to worry about the noise situation. But we recorded all of the acoustic guitars in one of Mike's bedrooms. It was a very quiet room with carpet on the floor and a lot of bedspreads and blankets — very dead sounding. We used a Shure SM81 into the Avalon VT-737 to the VS-1680. For monitoring, we ran a headphone cable into the bedroom."

When it came time to record lead vocals and mix, *2000* moved to The Playground. McGuire digitally transferred tracks from the 1680's to an Otari RADAR via S/PDIF (two tracks at a time). Brent's lead vocal was then recorded through a CAD E-300 and a custom Requisite mic pre. The songs were primarily mixed at The Playground through the RADAR, but McGuire reveals that if "Hillbilly Hotel" is the second single from *2000,* he might use the home mix. While the RADAR mixes were recorded to DAT, the at-home mixes were recorded in stereo directly onto two auxiliary tracks of one of the 1680's. The aux tracks served as a "mixdown" deck, allowing McGuire to simultaneously play 16 tracks from that same VS-1680 while recording the mix onto two virtual tracks. McGuire also used the 1680's internal "Mastering Tool Kit" algorithm to tweak overall EQ and apply split-band compression on the mixdown tracks.

"Recording *2000* was the most fun I've ever had making a record," McGuire admits. "Having this kind of creative freedom is so rewarding, and it really let our personality come through." *—Steve La Cerra*

VISIT THE SHENANDOAH WEB SITE AT WWW.GOSHENANDOAH.COM.


Kansas bassist Billy Greer

SEVENTH KEY

In addition to his duties as bass player for the hugely successful band Kansas, Billy Greer has been working on a number of other projects, including Kansas vocalist Steve Walsh's recent release *Glossolalia,* the self-titled album from The Sign, as well as his own project, *Seventh Key* (available May 2001 on Frontiers records). For *Seventh Key,* Greer is singing lead vocal and co-writing the songs with Mike Slamer, making extensive use of the Roland VS-1680.

> WITHIN A FEW HOURS OF SETTING UP THE 1680, I WAS TRACKING AND OVERDUBBING — PLUS I DIDN'T HAVE COMPATIBILITY ISSUES.

"I started out on the VS-880 and then decided I needed the extra tracks, so I upgraded to a VS-1680," Greer begins with a chuckle. "Steve (Walsh) has a VS-1680 as well as [Mark Of The Unicorn] Digital Performer, so when he did *Glossolalia,* he'd transfer files from Digital Performer into his 1680 via S/PDIF, burn a disc, and send it to me. I'd load the file into my 1680, work on a bass track, burn a disc, and send it back to him. It was almost the same scenario with *The Sign.* The guys were recording the CD in New York using ADAT's. They actually sent me an ADAT and a tape of the songs with a couple of open tracks to do the bass. I burned the tracks over to the 1680, recorded my bass tracks, bounced them back to the ADAT, and then sent it back to New York.

"It worked out great. I was able to record the bass tracks from the comfort of my own home in Florida and spend the afternoons on the beach. I tried hard disk recording directly into a PC, but having never fooled around with a computer in general — let alone for hard disk recording — I found it very frustrating. The creative curve took a severe dive and I found myself staring into a computer screen and scratching my head a lot! The VS-1680 was easier for me to relate to; within a few hours of setting up the 1680, I was tracking and overdubbing — plus I didn't have compatibility issues."

Greer used his VS-1680 as his major writing tool for *Seventh Key,* demoing and developing ideas with Mike Slamer. "Mike produced, engineered, and mixed the record. He recorded the drums at his studio in California, and then sent me the files in Digital Performer. I'd load his DP files into my computer, bump them over to the 1680, and work on my overdubs (right now I don't record into Digital Performer). Since I couldn't burn a DP file

Devin Powers and his *Blind Date* studio setup

back to him, I'd send Mike an audio CD of my tracks that he'd fly into Digital Performer. The CD burner connects to the 1680 via SCSI. I can access the CD-R functions without a computer simply by going to 'Write CD,' and assigning the tracks to burn to a disc."

> "I'VE ALSO DONE A LOT OF STACKED VOCALS USING THE 1680'S VIRTUAL TRACKS. I'LL RECORD SIX OR EIGHT TRACKS THEN BOUNCE THEM TO A STEREO PAIR."

Greer reveals that he gets a "great bass sound using the onboard effects from the 1680." He uses Dean basses with an effect called "DI Bass" and another called "Miked Bass." "I used those sounds with Kansas, *The Sign,* then *Glossolalia,* and now *Seventh Key.* I plug the Dean bass into an Alembic FX-1 tube preamp, and record one track of just the output of the tube preamp, plus a second track using the onboard effects in the 1680. Then I mix the two together for my bass sound. I also use the onboard compression from the 1680, but only on the Alembic preamp; the DI channel is uncompressed so that it's pretty open. I try to control the level just enough not to cause any ugly digital distortion or kill the dynamics." Greer monitors at home using a pair of Yamaha NS-10M's with a Crown PowerTech amp.

Some of Greer's vocals for *Seventh Key* were recorded with Slamer in California, mainly because "he has a beautiful old Neumann U 67 that really sounds sweet — there's not a bad frequency in this mic!" At home, Greer uses a Peavey PVM-T9000 tube mic. He generally works alone, which he finds "distracting." Reaching for the controls, singing, doing the punches, all the while trying to keep the sound coherent. "It's tough," he says. "I had a friend, Chris Mucci, help engineer while I sang. It's not that big of a pain when it's a demo, but for a final product

I don't want to worry about pushing buttons.

"Depending on how my voice is feeling," Greer continues, "I'll sing a song through a few times. Then I can dump a couple of vocal tracks into Digital Performer and comp the lead vocal. It's so easy to edit a vocal line together in DP — not like the old days on analog, where we'd make a map of which track would be used for which lines. Then we'd try to switch faders on the fly, or hit the mute buttons (which always made a noise anyway). Now I can cut and paste non-destructively and know it's always there on the hard drive. I've also done a lot of stacked vocals using the 1680's virtual tracks. I'll record six or eight tracks — doubling and stacking harmonies — then bounce them to a stereo pair. Again, the original tracks are still there if I need to make a change or adjust the balance between the parts.

"Transferring files back and forth runs up a pretty hefty bill with FedEx," Billy admits, "but when you compare that with flying to Los Angeles, it's not a bad trade off!" —Steve La Cerra

CHECK OUT BILLY GREER'S WEB SITE AT WWW.BILLYGREER.COM.

BLIND DATE

After a successful record release and tour with The Vents in 1997, composer/guitarist Devin Powers made the move into music for television. For the past two seasons, Powers has been producing music for Universal Television's Blind Date, largely relying on his Roland VS-1680 to record and mix songs for the show.

"Basically," Devin says, "I'm using a VS-1680 as my main recorder and an MPC2000 as my main sequencer. Everything I've recorded for television over the past two-and-a-half years has been done on the 1680. I have a pair of Avalon VT737's that I use as preamps and for mastering the mixes. I turn the Roland preamps all the way down, take my sources through the VT737's, and then into the 1680. Using the analog EQ and compression from the Avalon units warms up the sound of the 1680.

"There's a DI on the front of the 737 that I sometimes use for bass or guitar. While I'm working on a track, I'll switch back and forth between just the 737 and going through the 737 to a [Line 6] Bass Pod. If I need a clean

> IF I WANT A LITTLE MORE EDGE, I'LL DRIVE THE INPUT OF THE [AVALON VT]737 HARD, COME OUT OF THE 737, AND GO INTO THE BASS POD USING AN SVT-TYPE SETTING.

'Sting' sort of bass sound, I go right into the 737 using my [Fender] P-Bass. If I want a little more edge, I'll drive the input of the 737 hard, come out of the 737, and go into the Bass Pod using an SVT-type setting. If I use the Bass Pod (or any kind of amp simulator) and the 737 together, I get a very warm sound. It's the same for guitar as well. The guitar sound for the *Blind Date* theme is essentially the Pod on the 'High-gain Marshall' setting — into the 737 to roll off some of the harsh upper-mids.

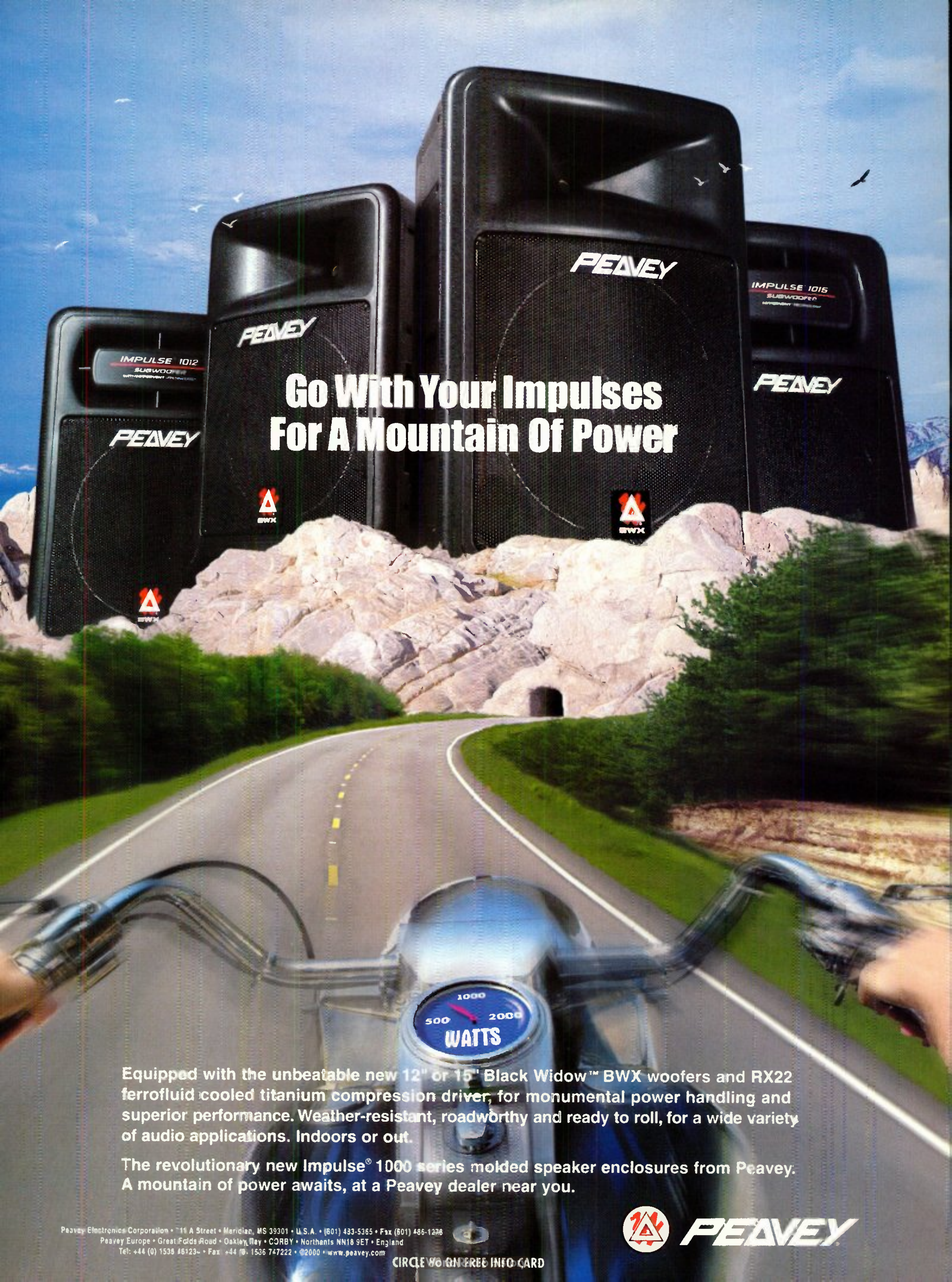
"Depending on how much time I have, I might run drum sounds from the MPC2000 or keyboard sounds from my Roland D50 through the Avalon as well, but those usually get recorded straight into the 1680 because their sounds have already been processed. When I do a piece with 'real' sounds, such as acoustic guitar, shaker, tambourine, claves, or (occasionally) a vocal piece, I use a Manley Cardioid Reference microphone into the 737.

"Once I've recorded and mixed the song, I patch out of the Roland 1680's analog outs (I don't use the digital ins and outs at all) and into the 737's for a mastering or sweetening effect. The outputs of the 737 go into an HHB CD recorder. During the second season of *Blind Date,* I started using the Vari-MU and Massive Passive from Manley Labs, which are also great sounding units. At this point, I have the 737's set up as mic pres on their way in to the 1680, and the Vari-MU and the Massive Passive are patched between the outputs of the 1680 and the inputs of the HHB recorder. I'm always ready to record, plus I don't have to re-patch for the mixdown.

"A convenient aspect of the 1680 is that I can do mixes very quickly for the show using autolocates and automation. I burn a mix to the HHB, and then do variation mixes with different solos: maybe a slide, acoustic, or a wacky lead guitar solo. Obviously, the heavier solo would give the track an edgier sound, the acoustic a lighter sound, and the slide a more bluesy sound. Then I might do a mix of just bass and drums, or acoustic guitar, tambourine, and slide guitar. I can record a two-minute cue, and — in about fifteen minutes — have eight different versions of that cue. This allows the editors to use it in the show several times as sort of a mini-score, without playing the same thing every time.

"Once my mixes are on CD-R, I put the CD-R into the iMac in my office, which is networked with the Avid systems used to edit the show. I have a 120 GB Medéa hard drive with 80 GB of cues. I'll upload the mixes, label them, give them a cue number, version, and name like 'Solo,' 'Minus Solo,' or 'Bass and Drums.' Then I use Avid to turn the cues into movie files so that an editor can preview the cues. The editor pulls up what we call a 'Music Store' and goes shopping for his or her cues on that hard drive. They can click on a category name, pull up a list of cue titles and numbers, and preview the cue. If they like the cue, they just drag it onto their hard drive for use in the show." *—Steve La Cerra*

YOU CAN HEAR DEVIN POWERS'S MUSIC AT WWW.DEVINPOWERS.COM. IF YOU'RE TIRED OF SITTING ALL ALONE IN YOUR STUDIO, VISIT *BLIND DATE* AT HTTP://BLINDDATE.EXCITE.COM.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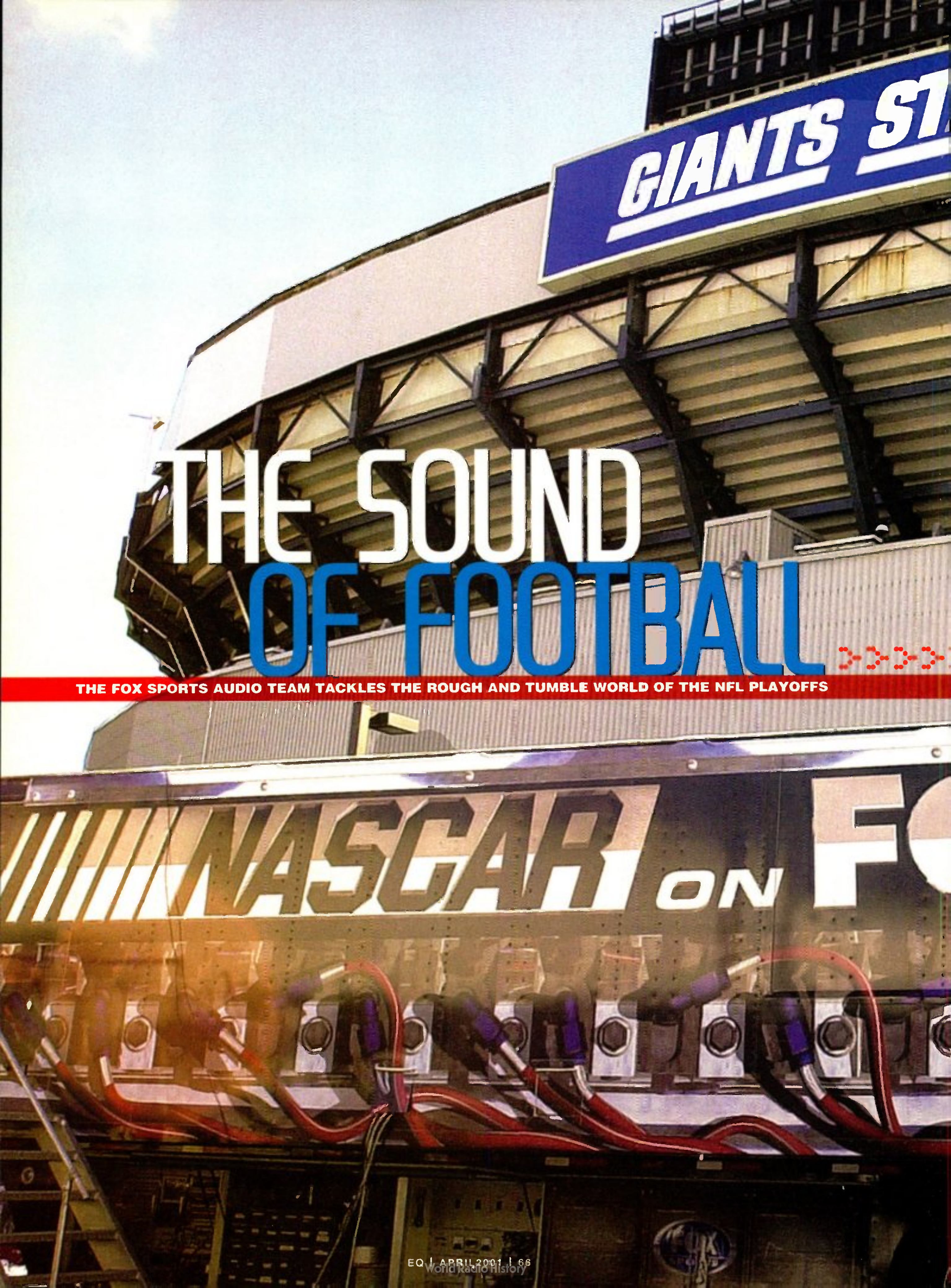
THE SOUND OF FOOTBALL

THE FOX SPORTS AUDIO TEAM TACKLES THE ROUGH AND TUMBLE WORLD OF THE NFL PLAYOFFS



BY STEVE LA CERRA • PHOTOS BY WES BENDER

Think you've got a high-pressure gig? Then consider this: Every week during the NFL football season, millions of people focus their attention on what's considered by fans to be the "prime" game of the week. Broadcast by FOX Sports and featuring the enormously popular sportscasters John Madden and Pat Summerall, FOX Sports' weekly NFL game has almost 100% penetration in the United States. And, since it goes out live on the air, there's very little room for error on the part of the audio crew. Masterfully organizing this audio mayhem on a weekly basis is FOX Sports' senior audio consultant Fred Aldous. We caught up with Fred at the Meadowlands in New Jersey the day before FOX Sports broadcast an NFC playoff game between the New York Giants and the Philadelphia Eagles. Fred was surprisingly calm....

THE EASY STUFF

For any NFL game aired on FOX Sports, there are two production trucks on location: the "A" unit, which houses audio, production, live video, and video tape; and the "B" unit, which accommodates graphics, maintenance, and the virtual first-and-ten line. The trucks arrive on the Friday prior to the game for park, power, and cabling. Aldous lives with his audio gear in the A unit (furnished by NEP, Pittsburgh, PA), literally acting as "audio control central" for the entire show.

"Any audio connection involved in the show comes through this room," Aldous begins. "I have feeds from live microphones for Pat and John upstairs in the booth, a microphone on each and every video camera, parabolic mics on the field, goalpost mics, two pairs of stereo crowd microphones, a mic for the field reporter, and a microphone on the umpire to fill in audio at the line of scrimmage. I also take audio feeds from 16 videotape devices, consisting of four channels each, for when we do instant replays." Live bump and highlight music arrives via two 360 Systems Digicarts.

"I use DPA 4023 mics in an X-Y configuration on the field for the crowd, and a pair of DPA 4052's in A-B configuration outside the announce booth (also for the crowd) so I can get a fuller sound. These stereo mics are routed to an Ashly MM508, forming a stereo submix that comes into my Calrec Q2 console. The mic on the NFL umpire is a DPA 4061 lavaliere mic with a Sennheiser SK250 wireless transmitter. The ump stands in the defensive backfield, and, since he's facing the offense, I get the quarterback cadence plus the initial surge of crushes, grunts, and groans — but only for a short amount of time. He has a switch for turning that mic on or off at his discretion. Then there's a NFL liaison known as a 'green hat,' who is on the sideline with a kill switch at the RF receiver (which essentially ground pins two and three at the output of the wireless receiver). The NFL dictates and controls when that mic is open or closed, due to the possibility of foul language being used by the players. We're allowed to open that mic from the clap of the huddle through the initial surge. Then I transition to the parabolic mics, which are full-time. So even if the ump mic isn't active, we still pick up the field sounds from the parabs."

Each camera interfaces with a Camera Control Unit (CCU) located in video production. In Fred's "control room," he has remote control over gain settings for all of the camera mics so that he doesn't risk overdriving the front end of his mic pres before the signal reaches the Calrec desk.

MAN IN A BOX

Assisting Fred in the mixing process is audio submixer Kevin McCloskey, who is responsible for four on-field parabolic microphones, two goalpost mics, and all of the wireless receivers and transmitters. Kevin also directs the parabolic mic operators on the field, making sure they're positioned where the action is happening.

McCloskey's location changes every week, depending on the facilities at a given stadium. At the Meadowlands, Kevin was set up in a press box that had a spectacular aerial view of the entire stadium, making it easy for him to see an overall picture of the game. "ABC puts their submixer in a truck," Kevin reveals, "but then the submixer can't see the parabs. I move the parab operators around the field, anticipating where the ball will be so I can pick up the sound of the play. That's half the battle. I have four parabs on the field (two on the near side of the field and two on the far side), each with a Neumann KM 183 microphone mounted in the dish. I'm not allowed by the NFL to move the parabs into the box that's formed between the two 30-yard lines because that's a 'safe zone' where the coaches and players live during the game. Video and still cameras are not allowed there either. So I generally have one parab on each side of the field between the 30 [-yard line] and the end zone, and mirror that on the other end.

Kevin McCloskey's mix position in the press box.

"When the ball moves past the 30, I move all four parabs down to one end, so the coverage is pretty good. Once they're down into 15- or 20-yard lines, the crowd gets sonically bigger and interferes more with what I'm trying to capture, so I'll put two parabs in the end zone, which helps out. If I'm facing the parabs at the field from the sidelines, fans are right behind the play. But with the parabs in the end zone, they face the play and the fans are way down the length of the field. Believe me, with 70,000 fans yelling, the crowd noise is still there."

To compensate for potential crowd interference, Kevin carefully worked out the focal point (the distance of the mic to the center of the dish) of the parabolic mics: "I had a tough time finding the focal point because, usually, the closer you get to the dish, the tighter the coverage. It was the opposite with the KM 183, and took me a game or two to tweak. The Neumann's are about five inches outside the dish, and they connect to Sennheiser wireless transmitters. The crowd produces this low-mid roar that just swallows everything up. I try to cut through that with the KM 183. Although the parabs do most of the work, I also have a Crown PCC160 on the crossbar of each goalpost. I hardly open those at all during the game — maybe when a team is nearing the end zone, say at the 5-yard line or so. The goalpost mic is really cool if a kick hits the cross bar. That happened a few weeks ago and they replayed it three times because the sound was awesome!"

Kevin's mics are routed through his Mackie 1604, out of which he sends a mono submix down to Fred's truck. The reason for a mono mix is that there's no

LIP SERVICE

Due to the nature of some of the video processing and routing, there's an inherent delay in the video, relative to audio, during a live FOX Sports NFL broadcast. Here's Fred Aldous's take on it:

"Audio and video can be out of sync, especially when the microphone is being routed analog, direct to my console. Digital video sources introduce small delays into the video signal. For example, you have a camera operator with a RF camera. There's about two frames of (video) delay in the processing for the RF transmission. Then the video signal goes into a frame synchronizer and a digital switcher in the truck, each of which add another frame of delay. But the mic on that RF camera is coming to me in real time, so it's ahead of the video signal — they're out of sync. As a result, we have to hold all the audio back while the video processing is happening, or we won't have lip sync. After that, we add the virtual first-and-ten line — which in itself creates a delay of about 20 frames The magic number for me is 746. I have to delay my audio by 746 milliseconds to sync it with video.

"We do a lip-sync test where we put a tape into a video tape machine, and on that tape is a beep sound matched with a video flash. When we line up the beep with the flash, we know we have lip sync. Once I have lip sync here, our composite signal (audio plus video) goes out to the FOX Sports Studio in L.A. If L.A. receives the program out-of-sync, then we know the audio and video somehow became separated in transmission, and they can re-sync it from there. But that rarely happens anymore because we're using fiber-optic lines to send the signal to L.A."

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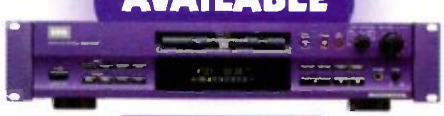

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easy way for McCloskey to get a left/right perspective relative to a camera. Because the director makes so many camera cuts, and Aldous likes to keep a full sound through the duration of the show, he runs this mono feed into an Orban 245F stereo synthesizer, using it to fill out his stereo and surround mixes.

McCloskey's world also includes a rack containing Sennheiser EM 1046 UHF wireless receivers for the parabs, the umpire mic, and a mic for FOX Sports field reporter, D.J. Johnson. Many college football broadcasts utilize wired parabolic mics, which Kevin calls a "nightmare. My guys strap on their parabs and can go anywhere. Freddy sends me a program mix from the truck, including Pat and John, so I can get a perspective on the whole game. I send that down to the parab guys on the field from a one-watt wireless transmitter with an interrupt feed. I can interrupt that program at any time — talk to them and direct them up- or downfield. D.J. Johnson uses a wireless mic (a Sennheiser E835S hand mic with a Sennheiser SK250 transmitter) that comes up to here. Plus, we send him an in-ear monitor mix via a Sennheiser SR300 transmitter and Sennheiser EK300 receiver.

"The producer can interrupt D.J.'s mix to give him direction, and because the producer has D.J. on PFL, D.J. can let him know where he's at or when he's ready for a report. At that time, the announcers will throw it down to the reporter. Fred opens the fader on the producer's cue, and D.J. does the interview. When he's done, the fader is closed to air, but he's still on a PFL with the producer in the truck for communication purposes.

"Altogether, we're using nine wireless mics and five (mono) wireless sends for communication, all on different frequencies. The wireless receivers are all located up with McCloskey, and the feeds go from him down to Fred in the truck via hardwire." To ensure that wireless transmission and reception between his position and the field will be at maximum

NEP TRUCK GEAR LIST

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Calrec Q2 console with 60 channels and 120 inputs
CCU mic preamp
ATI 8mx
Ashly MM508

RECORD/PLAY
Denon DN-C680 CD player with remote
360 Systems Digicart 2 digital audio recorder [2] with RC 220 remote.

PROCESSING
Aphex Dominator II
Aphex Tube Expressor [2]
dbx 120XP subharmonic synthesizer
dbx 160A [8]
dbx 900 rack with 929 noise reduction [2]; 904 gate [1]; and 903 comp/limiters [4]
Dolby Model 740 Spectral Processor
Orban 245F Stereo Synthesizer [2]
Rane AVA 22 Dolby Time Link digital delay [3]
Sony DPSV77 effects
TC Electronic DB Max

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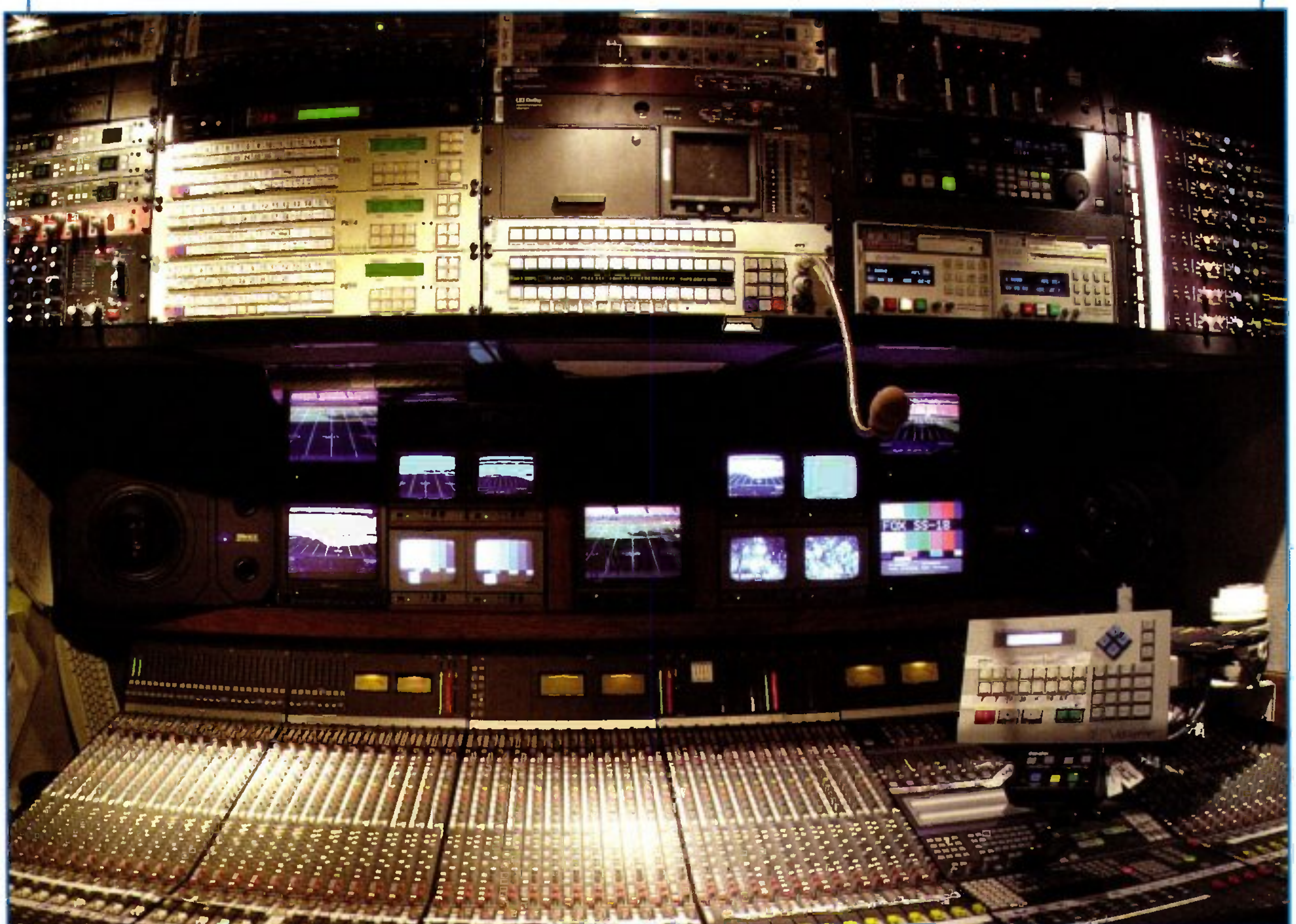
strength, Kevin runs eight remote antennas situated out on the ledge of his press box.

LET'S TALK ABOUT THIS...

For a FOX Sports NFL production, audio program is only half the story. Perhaps equally as important is communication between the show's producer, director, associate director, audio crew, video crew, camera operators, and field reporters — all of whom are in different locations at the venue. With the exception of the previously mentioned wireless program feeds to the parabolic mic operators, all of the communication lines are hardwired.

"One of my biggest roles," Fred explains, "is making sure that people who need to communicate with each other get the right connection. I have twelve isolated communications channels extended outside of the truck for particular groups of people on what we call 'PL' or party line channels. For example, one of the PL channels is a camera conference channel where the director talks to the technical operators. This is primarily the director's common channel to the entire crew. Then there's another channel for the producer to talk to the associate director, stage manager, and tape operators. We also have a separate channel for the statisticians. If everyone lived on one channel, it would be total chaos! All of these iso-channels are for people who need to talk to each other but don't necessarily need to speak to people outside their group. Then we have another 24 to 26 channels within or between the two trucks. Inside the truck, every operator has a master station, so it's all point-to-point communication."

One of the most important areas is the announce booth from where Madden and Summerall deliver the show. "The announce booth is always on one of the upper levels of the stadium at the 50-yard line," says Aldous. "I have a separate communication line up to Paul Neisen, my booth audio assistant or 'A2.' Paul sends me John and Pat's headsets, their handheld mics, Sennheiser E835's with Sennheiser wireless in-ear monitors (same as the field reporter), which we use for the open and close of the show, and all the communications to and from the booth. I send Pat and John a stereo program mix to their headsets (Sennheiser HMD25's), which can be interrupted by the producer or director for cueing purposes. We call that an IFB (Interruptible FoldBack). Paul has spare headsets up in the booth so, if we have a problem, he can swap one out. The connections from the announce booth are all hardwired to me via three 12-pair cables — which are pretty full at this point!



Fred Aldous's mobile control room.

"In terms of EQ, I don't put as much bottom end in a broadcast voice as I'd put on a vocal in a music mix; I want the voice to cut through during a broadcast. I tend to mix a bit bright for the announcers as opposed to a music vocal where I'd mix a bit darker. With 72,000 fans screaming out there (plus 22 players on the field doing their thing), I make sure that the bandwidth for my announce mics is such that their voices don't get muddy."

For Aldous, "patching communications is a minimum one-day affair. The house that we're working in on any given weekend has cabling pre-installed by the networks. I interconnect the truck with an I/O panel at the stadium. From the I/O panel, connections are pre-wired to various locations in the stadium. Then we run extensions from the breakout panels on the field, announce booth, or press boxes to the gear we need to connect.

"Generally, we do our interfacing on Friday. A local A2 (audio assistant) comes in and sets our hardware on the field, Paul sets hardware upstairs in the announce booth, and Kevin does the same. Then we check all of the connections between the truck and the hardware. We generally run fourteen 12-pair mults for the show. When the NFL playoffs come around, we typically add another 48 pairs to accommodate extra cameras. Adding just one camera to the game requires several lines: communications to and from the camera operator, data to the camera, and usually, when we add a camera, we also add a video tape machine for that camera. So there are another four channels of audio (in and out), plus a program feed to the camera operator."

There's not as much redundancy on the audio lines as one might expect because, as Fred describes it, "This show runs at almost 100% all the time. We have enough spare gear to rob parts and pieces to make something work. But for the most, there's very little room for error. There's rarely a piece of gear that's not being used. Last week we had some pair problems in our multi-pair cables, so I re-patched connections to different pairs. Part of what I need to do this week is reset my patches before starting a facilities check." As a result, Aldous checks every audio connection every week by checking continuity from the truck down the line to its destination.

THE ELEMENTS

Unique to NFL production is the wide range of weather extremes that must be dealt with on a weekly basis. Aldous explains, "We have sunny days when everyone's out in their T-shirts having a great time, or we can have –20° weather in snow — which brings its own problems. In severely cold weather, cables get brittle, freeze, and break. Severe cold has its effect on microphones as well: the diaphragm isn't going to function as well in –20° temperatures as it was does in 65° or 80° temperatures. Some of the frequency response is lost because I don't get full use of the diaphragm. It's a

Due to space constraints in the mix room, rack gear is mounted in a bridge floating above the console.

severe cutoff toward the low end, and dynamic range closes right up. Other things we run into with bad weather are more obvious, such as wet-gear failure. Wet-gear failure is my biggest concern in the elements. If there's water in a connector, I get crackle, snaps, pops, hisses, hums, and buzzes. I have to do my best to foresee these problems and weatherproof."

LET'S KICK THIS PRODUCTION OFF!

When asked whether the producer and director give him specific requirements regarding sound for the show, Fred replies, "To a degree, but with this particular show being the 'A' NFL show on FOX, I get a lot of latitude as to what I deliver to the viewer. For me, mixing the show is the payoff. That's my creative time, my playtime, so to speak. Setup can sometimes be a nightmare because we have to deal with the elements, but mixing is exciting for me, especially since I know that we're going out live. On Sunday, I get out of the tech world from Friday and Saturday and I get into the creative world." For an "average" show, Aldous uses most of the available inputs on his Calrec Q2 desk. The console has 60 channels with two input paths per channel: one line level and one mic level (120 faders total). He uses monitor returns for feeds from the VTR's and other line-level sources such as the synthesized stereo feed from Kevin's Mackie 1604.

"The most important aspect of the show," Fred continues, "is that the director will call camera cuts and I have to match audio with whatever you're looking at on the air. I either follow his cuts or try to anticipate where he's going. I scroll through cameras constantly to see what's going on at the field, trying to stay ahead of what the producer or director is doing. If I know their style, and I see something interesting, I can somewhat anticipate what they'll do with their video cut. That allows me to be ready and *on time* when they call for it. It's a pretty instantaneous process…'Standby camera one, take one. Standby camera two, take two,' etc. When he is saying 'take,' it means that particular camera will be on-air. At that point in time, the director is talking to the entire crew on a common channel. The technical director has all of the cameras, video machines, and graphic sources patched into his video switcher so when the director calls a cut, the technical director takes it visually with his video switcher. I take the audio source to match. If he takes a camera on the field, I'll mix that audio source to the program feed. Sometimes I'm involved with troubleshooting a problem and it becomes a juggling act between trying to track down a problem and doing the mix at the same time. It's a difficult thing that I've learned, just like listening to five sources at the same time while I'm mixing!

"It's the rush of live TV. It has to be right the first time. While I'm mixing the show, I'm listening to my mix of the game in surround, a director, a producer,

MAKING THE LEAP INTO TELEVISION

Fred Aldous has been working in television since 1981, a time when he considered himself "a starving musician." Aldous explains, "I got a job at a local TV station, started doing live shows, and then got into sports by accident. CBS had come to town to do a ski show. I made a contact with them and started freelancing around 1983. In 1986, I quit my job and I've been freelance ever since.

"I love the music industry, but it became a grind sitting in the studio day in and day out. The thing I like about this is that I get to travel and see different parts of the world. I have challenges everywhere we go due to the elements, so it's never the same old thing. I thoroughly enjoy the variety in what I do. Interestingly, I'm not a huge sports fan, and you know what? I can't be! If I get caught up in the game, then I'll miss my cues from the director and producer. I'll be distracted from being creative and mixing. What I've had to learn over the years is to disassociate myself from the game. This has to be right. It has to be right on, and it has to be in sync.

"The big difference between this and music production is that, in music production, you typically have time to be as creative as you want because you're not in a live situation. You don't have to worry about matching your audio to video because, for a music video (usually), the music track is laid first. Then the video director matches the picture work to the audio. In live sports, it's exactly the opposite — I match my audio to the director's cut. If you're doing music production in a recording studio, you can go back and re-cut a guitar lick or a drum track.

"I believe that, eventually, I will go back to music production. I've been doing this long enough now that I'm almost ready for another change. The grass isn't greener, it's just different!"

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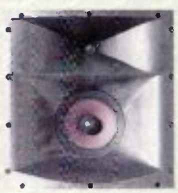

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and an associate director, plus I also listen to my audio guys so that, if there's a problem, I have communication with them. Sometimes, in my world, the elements — or even the fans — are against us. I did a college football game a few years ago, and my parab operator at the time set his dish down to go use the bathroom. When he returned, his parab was gone! We figured that it ended up in a frat house with goldfish in it! We have things like that to worry about."

WEB LINK

- Visit the FOX Sports Web site at www.FOXsports.com.
- Fred Aldous may be contacted at aldousaudio@juno.com.
- Kevin McCloskey may be reached at irishtoadl@aol.com.

OKAY FRED, YOU'RE SURROUNDED!

In addition to stereo and mono on-air feeds, FOX Sports has been airing NFL games in Dolby Surround every week since 1994. FOX Sports and Fred are "very proud of that fact, and the shows sound great in surround. We do it live, on the fly. The most important part of that is making sure all of the formats work: mono, stereo, and surround. If you go with a huge surround mix, then your stereo and mono mixes suffer. On the other hand, if you go too light on surround information, then the surround mix suffers. I bleed a little of the surround channels back into the mono mix, so if a sound hits a surround channel, some of it is panned to the center channel. That prevents the surround mix from being drastically different in content from the mono or stereo mixes. FOX is the only network I know of that specs all of their sports broadcasts in surround."

According to Aldous, "Mixing the NFL for FOX Sports has really honed my skills as a mixer, in terms of making sure I know what needs to happen, when it needs to happen. Fortunately, we have the luxury of setting up ahead of time and checking out all of our sources. After doing 13 years of NFL, I pretty much know the stadiums and facilities. I know what the crowd will be like, as well as my announcers — what my levels need to be for them, what their voices sound like, and what their intensity level is. It's a rush to know that millions of people across the country are listening to me at any point, and it has to be right the first time. Once it hits the stars, it's gone forever!"

WHAT'S THE FREQUENCY KEVIN

Since Kevin McCloskey is in charge of the wireless systems every week for the NFL games aired on FOX Sports, part of his responsibility is making sure that the frequencies planned for wireless use are free and clear. McCloskey explains, "Every city has its own coordinator from the FCC — not just for us, but for TV stations, radio stations, and anyone coming into town using wireless gear. Before we even get to the stadium for the game, we send an email to the frequency coordinator saying that we're going to need X amount of frequencies. We specify our frequency range (for example, my parabolic mics are in the range between 506 to 525 MHz), and they assign channels for me to use within that range. It has to be planned ahead of time because each team uses about 30 wireless frequencies for things like coach headset communications. In addition to the teams, NFL Films and other stations may be at the game using RF systems. And with digital TV broadcasts increasing, stations are popping up in our band and taking frequencies away from us. We can't possibly know all of them. So we always check ahead to see what's available. Last week in Philadelphia, I got stepped on prior to the game, but I just changed frequencies and we were okay. I always have a few spare frequencies that I know are open and ready to go."

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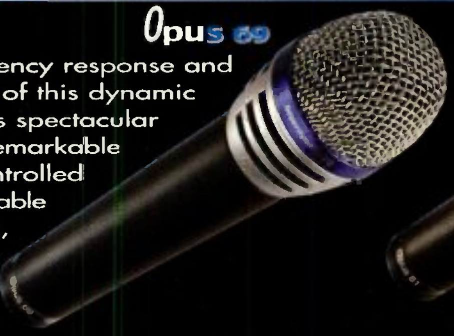
Opus 69



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Opus 81

Advanced technology and extraordinary engineering has produced this state of the art condenser microphone. The Opus 81 can be used on the stage or in the studio. It produces a smooth, warm sound, accurately reproducing the rich textures of a vocal performance.

Instrument Microphones

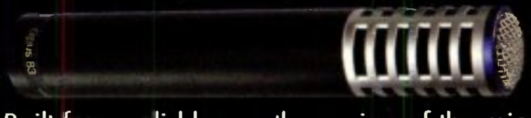
Opus 51

A boundary microphone, the Opus 51 captures direct and reflected sounds from an instrument in perfect balance and at an equal level. This microphone is ideal for capturing the full sound within an instrument such as the piano or bass drum.

Opus 65

The Opus 65 is built with a 1.5 inch diameter diaphragm within a powerful neodymium magnet. The Opus 65 requires little or no equalization to produce the natural sounds of low frequency producing instruments, virtually eliminating the problems incurred by other kick drum microphones.

Opus 83



The Opus 83 is a condenser microphone ideal for drum and percussion overheads, cymbals, and acoustic instruments. Built from solid brass, the casing of the microphone protects the sensitive element from the rigors of the road.

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FRANKFURT MUSIK

We're not in Anaheim any more, Toto…turn on German MTV, and you'll see big-budget ads for Cubasis. Visit the local equivalent of Circuit City, and you can buy sets of four or five sample loop CDs for around $25. And consumer-oriented music creation software such as eJay's techno and hip-hop studios (with recording, sequencing, editing, and a generous supply of samples to get you started) are as commonplace as word processing software at Office Depot.

Part of this is because the dance/electronica/experimental music scene is big enough to exert a gravitational field that has pulled a younger generation to hardware and software companies, adding a much-needed injection of fresh thinking. The music technology scene in Europe is vibrant, thanks to audiences that are more willing to accept non-commercial music, a societal emphasis on the arts, and the fact that a musical revolution is going hand-in-hand with a technological one. Is it any wonder that I love to gig over there?

Okay, enough background, let's get to the gear. Despite Frankfurt's close temporal proximity to NAMM, there was a raft of new products, some of which were held back from NAMM so they could get a more friendly reception on home turf.

GIMME KNOBS

Human interfaces are on the rise, and one of the hottest products at the show, **Logic Control,** was co-developed by **Emagic and Mackie Designs.** This sleek-looking, eight-channel interface uses Penny & Giles motorized, 1,024-step, 100-mm touch-sensitive digital/optical faders. There's hands-on control over hundreds of MIDI and audio functions; dedicated sections include transport controls, cursor keys, automation, and function modes. Bank Switching functions access additional channels, but for hands-on control of up to 128 channels, there's the eight-channel Logic Control XT expansion module.

On a less grandiose scale, the palm-sized **MIR (MIDI Interactive Remote)** from **C-Mexx** is ideal for programming MIDI parameters without having to squint into an LCD or bend over a rack. It incorporates a 2x16-character backlit display along with several knobs and switches, and can control up to 39 different functions per preset. Software for Mac/Windows allows for rapid editing and parameter setup.

EMAGIC & MACKIE LOGIC CONTROL

My review of Propellerhead Software's Reason (EQ, February 2001) said, "I wouldn't be surprised to see someone come up with a dedicated hardware controller." Well, it's here: **Novation's ReMOTE** not only has a semi-weighted three-octave keyboard action with aftertouch, but also includes transport buttons, a bank of configurable controls for real-time tweaking, and a built-in power amp with stereo speakers.

MULTITRACK RECORDING: CHANGES IN THE MAKING

Mackie's D8W Integrator, a software/card package, integrates Mackie's Digital 8•Bus and HDR24/96 hard disk recorder/editor. All components are synced with sample accuracy; controls on the D8B work the hard disk recorder, so it's now possible to use one mouse, one keyboard, and one monitor to control both systems. The Integrator card

CRAIG ANDERTON REPORTS ON THE SIGHTS AND SOUNDS AT THE ANNUAL MUSIKMESSE TRADESHOW

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installs in the HDR24/96 and features MIDI I/O ports for MTC and MMC, and a port for connecting to Mackie's Universal Time Synchronizer (which adds multiple Sony nine-pin control ports, VITC read/generate, bi-phase lock, word clock distribution, video generation, and SMPTE clock).

In other mixing console news, **Trident Audio** announced the release of **Series 80-5.1** console. The mic preamp and EQ in the new board are identical to those in the original Series 80, but the desk also incorporates modern features such as 5.1 panning and drop-in compatibility with Oram channel strips. The board's center section includes Oram's Hi-Def EQ and dual-channel compressor circuits. Also announced was a collaboration with iZ Technology, which will allow a RADAR 24 remote and monitor to be built into the console.

In software-land, **Emagic** announced **Logic Audio 5.0,** whose revamped automation system handles 32-bit fader values and is tied to individual tracks, rather than arrange window sequences and audio regions. Bottom line: When you move or copy these objects, the automation data goes with them. Other features include more than 50 plug-ins in Logic Audio Platinum, a mixer sidechain function so external signal inputs can serve as control signals, support of audio instruments' individual outputs, hardware-independent audio scrubbing, high-end dithering (courtesy of POW-r Consortium), and REX 2.0 support.

Samplitude has always been popular with Europeans, even though — as the company admits — it's a difficult program to learn because of its depth. But **Samplitude 6.0** introduces a clever feature to get around that problem: you can choose user interface "presets" limited to specific tasks (*e.g.,* CD burning, mastering, surround editing, multitrack recording, etc.), which restricts the menu options and toolbars to only those functions required to accomplish those tasks — brilliant! Other features include the ability to work with video and MP3 files directly, 32-bit floating point internal resolution, 24/96 support, surround, Pentium 4 optimization, a new mixing console, and removal of the 2 GB file-size limit. Also, Samplitude has now become part of the Magix family of "rich media" software, holding down the pro end of the line.

Roland's CDX-1 DiscLab is a combination phrase sampler, CD-R/RW burner, mixer, and eight-track recorder/sequencer. Up to 512 samples can be recorded from audio inputs or imported (as audio or WAV files) from CD, triggered from the onboard pads, processed with the onboard dual effects processors, time-compressed, looped, truncated, mixed, and sequenced, then burned on CD. Onboard storage is 16 MB (up to seven minutes of phrases), expandable to 128 MB using DIMMs. Up to eight tracks can be recorded (two simultaneously) and saved on CD-RW; after mixing down audio tracks and samples to a master track, eject the CD-RW, replace it with a CD-R, and create a finished master (mastering tools such as multiband compression and four-band EQ are included). CDs can also be burned directly from the audio inputs. Audio I/O includes XLR, TRS, 1/4-inch high-impedance guitar input, and stereo outs; digital I/O includes optical and coaxial S/PDIF.

HARDWARE SYNTHESIZERS

Yamaha's Motif family of music production synthesizers integrates a sampling sequencer, real-time control surface (with a remote mode that allows controlling Logic Audio, Cubase VST, Cakewalk, and Pro Tools — mute tracks, run

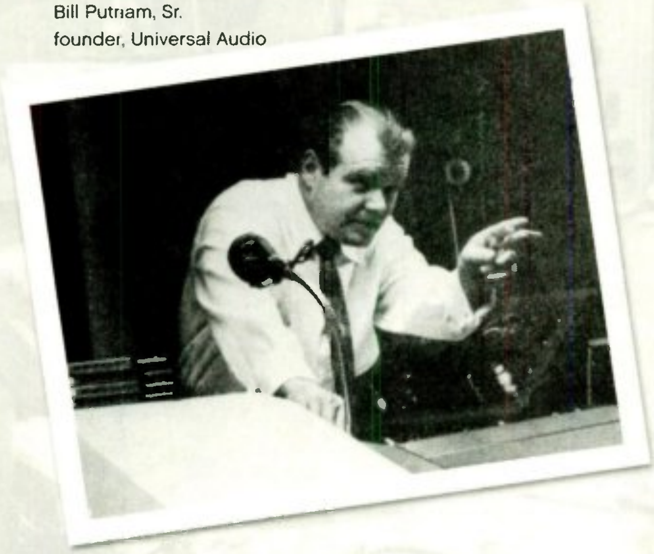

KORG ELECTRIBE-M

the transport, mix MIDI and audio tracks, etc.), and accepts hardware plug-ins to provide different types of synthesis (available plug-ins include FM synthesis, physical modeling, and vocal harmony processing). The sequencer offers up to 200,000 notes of MIDI recording over six minutes of stereo audio recording; supported audio file formats include AIFF, WAV, Akai S1000/3000, and Yamaha A3000/4000/5000. Furthermore, an interactive arpeggiator provides "human" patterns such as guitar strumming and flute trilling. Onboard expansion includes a USB MIDI interface, SmartMedia card slot, SCSI port, and optional mLAN board (an alternative to mLAN; the AIEBI expansion board, provides six assignable analog audio outs as well as optical and coax S/PDIF outs).

Clavia's 5U rack version of the Nord Lead 3 is called (not surprisingly) the **Nord Rack 3,** while **Waldorf's** Micro Q rack has spawned the compact **Micro Q Synthesizer,** which sports a three-octave keyboard. **Korg's Electribe-M** offers drum and bass line generation. The sound engine provides 144 dance drum waveforms and 50 synthesizer PCM waveforms, including synth, bass, pianos, simple periodic waveforms, pads, etc. For processing, there are 11 insert effects and overdrive; other effects include BPM delay and the "motion sequencer" incorporated in other Electribe models. Another fun box for groove fans, **Roland's D2 Groove Box,** uses the touch-sensitive D Field pad interface, which has three modes: sound (for playing sounds effects and patterns directly from the pad), XY (provides control over two parameters), and Spin for scratching-type effects. Total storage is up to 40,000 notes. Other features include six-voice polyphony, along with reverb/delay/multieffects.

The dark horse of the crowd is **SoundArt's Chameleon Synthesizer,** which uses a general-purpose

Bill Putnam, Sr.
founder, Universal Audio



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Motorola DSP engine to allow the creation of various types of synthesizers and effects processors (anyone remember the Peavey DPM-3 from 1989, which was based on the same concept?). Tools for software developers to devise new algorithms are expected to be available from the company's Web site.

SOFTWARE SYNTHESIZERS

The Messe crowd went nuts over software synthesizers. We'll start with **VirSyn** (Windows), available as a stand-alone synth or VST plug-in. You can program up to 12 modular synths for simultaneous use, but they all share a common interface structure. Effects such as distortion, echo/delay, and chorus/phaser/flanger are available, as is a 64-step sequencer that can arrange up to 256 sequences.

CreamWare introduced the **Volkszämpler,** a VST native plug-in for Windows/Mac that supports mono and stereo samples with resolutions up to 32 bits and sample rates up to 96 kHz. It's compatible with Akai S1000/S3000, SoundFont2, WAV, and AIFF file formats.

NATIVE INSTRUMENTS FM7

Native Instruments showed **Absynth** (Macintosh), a very cool "analog" synth capable of making deep and interesting sounds, and the Windows/Mac-compatible **FM7.** Based on FM synthesis, the FM7 can import programs in the standard FM sound formats (DX7, DX7II). However, you can further shape the sound with filter and

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distortion operators. Both new synths are compatible with VST2.0, DirectConnect, MAS, and ASIO; FM7 is also DXi-compatible.

Emagic announced the **EXSP24,** a VST 2.0-compatible sample player that can play samples ranging from 8- to 24-bit depths, and can import WAV, AIFF, SDII, and SoundFont2 files. Also, the **ES2 (Emagic Synthesizer 2)** for Logic Audio features up to 16-note polyphony per unit, with three oscillators per voice. Two self-resonating filters, which run in serial or parallel, shape the timbre. Modulation sources and targets are defined via a "Router" display, which can be switched to graphically indicate LFO and envelope shapes, or, when switched to "Vector," can provide for dynamic oscillator mix control and control of two freely selectable parameters.

Modularing 3.0, from **Mil Productions,** is an interactive music system with arpeggiator, synthesizer, network synthesizer, sample player (with real time stereo sample loop editing and drag and drop from audio libraries), router, delays, General MIDI programmer, several sequencers, trigger pads, synchronizer, etc. Meanwhile, **Steinberg** collaborated with **Waldorf** to create the **Attack VST** drum synthesizer (Mac/Windows). It offers 24 sounds spread over two octaves. In addition to analog sounds (two oscillators with nine waveforms), there are also three samples of hihats and crash cymbals. A filter section allows six different varieties, including resonance with self-oscillation and overdrive; other effects include ring modulation and FM. Being a VST 2.0 instrument, all Attack parameters can be MIDI-controlled.

SIGNAL PROCESSING

Of course, plug-ins rule — but hardware help is coming on strong. **CreamWare's Pulsar XTC** is a hardware/software package for Mac/Windows with six 32-bit SHARC DSP's and a collection of XTC DSP plug-ins (dynamics, modulation, reverb, etc.), along with the Volkszämpler software synth. The combination is designed to complement programs such as Cubase and Logic by allowing the use of additional plug-ins that don't load the host processor. All plug-ins are VST 2.0-compatible, so they can be automated within the recording software; four optional I/O packages are

STEINBERG ATTACK VST

also available. In a similar vein, **Emagic's ESB TDM (Emagic System Bridge TDM)** links host-based sound creation such as VST 2.0 sources and TDM-based systems (when used within Logic Audio Platinum Mac). Pro Tools users can now combine both native and TDM processing; Logic Audio's native outputs appear in the TDM mixer by use of a send/return-system, and, furthermore, ESB TDM allows integrating up to 32 EXS24 software sampler units in the channels of Logic Audio Platinum's TDM mixer.

Speaking of Emagic and plug-ins, in addition to standard vocoding, **Evo 20 (Emagic Vocoder 20 band)** can track the pitch of incoming audio and deliver results that "sing." It can also serve as a formant filter bank with volume faders for each band. Meanwhile, **Prosoniq** showed the **Magenta Advanced Pitch Resynthesizer,** a VST/RTAS-compatible plug-in that provides polyphonic real-time pitch resynthesis, simple reverb and delay, a circular virtual controller that can tweak two parameters at a time (and can also be controlled via MIDI), and filters/envelopes/LFO for additional processing.

Steinberg's Voice Machine (VST Mac/Windows) is designed specifically for vocal processing. It generates up to four additional voices (triggerable from MIDI note on/off events) and can change melody or correct intonation without altering the voice's character. As pitch-shift and character functions are separate, it's possible to do tricks such as simulate other singing voices or impart a male voice's character onto a female vocalist.

In this high-tech world, rack units still exist — **Mindprint's DTC (Dual Tube Channel)** is a 3U, two-channel front end offering transformer-coupled mic and line preamplification, high-impedance instrument input, equalization, dynamics processing, and an optional digital interface (with 24-bit/96-kHz, optical/coaxial S/PDIF/AES-EBU digital input and output). Also, **XTA Electronics' SIDD (Seriously Intelligent Digital Dynamics)** is a dual-channel processor that packs a compressor, gate/expander, dynamic equalization, brickwall limiter, delay line, harmonics generator, and six fully parametric bands of EQ (switchable to high/low shelves) into a 1U rack. I/O includes two balanced ins, four balanced outs (AES/EBU option available), and four interfaces (RS232, RS485, MIDI, and PCM-CIA card).

Well, that's enough gear for now. At the show, two trends stood out above the rest: musicians want more hands-on control over software-based devices, and, increasingly, computers are taking over functions that used to be handled by outboard gear. I'm sure these trends will only accelerate in the years ahead.

Craig Anderton is creative director for MusicPlayer.com (visit his forum and say "hi"), guitarist for the Cologne, Germany-based group Rei$$dorf Force, and author of the classic text *Home Recording for Musicians* (AMSCO).

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CIRCLE 65 ON FREE INFO CARD

PICKING A

Recording School

EQ'S GUIDE TO FINDING THE RIGHT AUDIO-ENGINEERING SCHOOL FOR YOU

While none are easy, there are number of roads to becoming a recording engineer. *EQ* magazine is full of success stories that began in myriad ways: on-the-job training, university programs, or recording schools. One of the best ways to begin, as with many careers, is by achieving an education appropriate for your chosen field. The good news is that potential students have a number of options. The revolution in accessible recording gear, and a demand for quality, affordable audio education, in a field where quickly changing technology is the norm, has led to a recording school selection that is widening every year. The challenge is to make the right choice. Finding a school that fits your needs and abilities will take some research.

To take the sting out of your quest, we've included a list of some of the recording school options available in North America. Before calling or visiting the various schools' Web sites (not a bad way to begin), establish a list of criteria that will help you judge which program suits you best. Here are six guide-points to consider.

FINANCE: Determine if the program can fit your budget. Also be sure to inquire about any financial assistance that's available. Scholarships or grants are sometimes offered to students with the right merits and profile, and such help might bring some of the more pricey programs within reach.

FACILITIES: This will be one of the most important and deceptive aspects of your search. First, make sure the schools you're considering have the right equipment to train you. Archaic gear and poorly supplied control rooms can lead to potential recording engineers that are unprepared for real-world engineering. This isn't a situation you want to find yourself in!

Also, before deciding on any school, make sure that you know their student-to-facility and student-to-faculty ratio. It's possible for a program to have a first-rate setup, with every useful piece of gear since the compressor, but if you can't get sufficient hands-on time due to overcrowding, the value of the program takes a swan dive. This should be one of your main areas of inquiry.

FOCUS: Make sure that the school you're considering stresses your specialty of choice (e.g., digital audio, mastering, or studio business management). More importantly, make sure that the school makes the same priority out of audio engineering that you do. A "multimedia" school that offers only cursory coverage of recording technology could leave you short-changed.

LOCATION: Having the school right in your backyard is nice, but relocation for the right education can be well worth the trouble. One thing to consider is that an "off-market" school might not have the same potential for apprenticeships or internships in area studios. Also, location should be a factor in setting up your education budget. Make sure that any cost-of-living changes will be reflected in your financial planning.

TRACK RECORD: One of the most dependable yardsticks of school quality is their track record for professional placement of alumni. Also, consider the ratio of students who complete the course to those who don't.

FACULTY: Who teaches you matters. Make sure that your instructors are qualified. Remember that their credentials need not include formal training. There's nothing like learning the tricks of the trade from an A-list producer or engineer — whether he has a fancy degree or not. ■

READ ON FOR A LIST OF NORTH AMERICAN SCHOOLS

BY BRITT STRICKLAND


EX'PRESSION CENTER FOR NEW MEDIA

Web: www.fullsail.com
Notes: Offers a Recording Arts Associate of Science degree and a Show Production degree program.

FUTURE MEDIA CONCEPTS
NEW YORK: 305 E. 47th St., New York, NY 10017
Phone: 212-888-6314
BOSTON: 43 Thorndike St., Cambridge, MA 02141
Phone: 877-362-8724
PHILADELPHIA: 325 Chestnut St., Philadelphia, PA 19106
Phone: 215-922-2500
Web: www.fmctraining.com
Notes: Manufacturer's Certificate of Merit. Courses range from five days (two-part introductory course), to ten days (master class).

INTERNATIONAL COLLEGE OF BROADCASTING
6 So. Smithville Rd., Dayton, OH 45431
Phone: 800-517-7284
Web: www.icbroadcasting.com
Notes: Associate Degree program in Applied Science of Communication Arts in Television and Radio, Associate Degree program of Applied Science in Video Production/Recording, Audio Engineer Diploma Program in Recording/Audio Engineering, Diploma Program Broadcasting.

INSTITUTE OF AUDIO RESEARCH
64 University Place, New York, NY 10003
Phone: 212-777-8550
Web: www.audioschool.com
Notes: 600-hour Recording Engineering and Production Program. Offers Diploma in Recording Engineering and Production.

LOS ANGELES RECORDING WORKSHOP
5278 Lankershim Blvd., North Hollywood, CA 91601
Phone: 818-763-7400
Web: idt.net/~larw
Notes: Recording Engineer Certificate (900 hours), Audio-Video Production Certificate (900 hours).

MADISON MEDIA INSTITUTE
One Point Pl., Suite 1, Madison, WI 53719-2809
Phone: 800-236-4997 or 608-829-2728
Web: www.madisonmedia.com
Notes: Associate of Arts Degree — Recording and Music Technology (60 credits/60 weeks), and Multimedia Technology (60 credits/60 weeks).

MUSIC TECH COLLEGE OF MUSIC AND RECORDING ARTS
304 Washington Ave. N., Minneapolis, MN 55401
Phone: 800-594-9500 or 612-338-0175
Web: www.musictech.com
Notes: Both diploma and degree programs available. Associates of Applied Science Degrees are available in recording technology and music production. Industry Pro Outreach program for possible internships.

OMEGA RECORDING STUDIOS SCHOOL OF APPLIED RECORDING ARTS AND SCIENCES
5609 Fishers Lane, Rockville, MD 20852
Phone: 301-230-9100
Web: www.omegastudios.com
Notes: Omega Recording Studios School Certificate (approved by the Maryland Higher Education Commission), 226 clock hours.


LOS ANGELES RECORDING WORKSHOP

ONTARIO INSTITUTE OF AUDIO RECORDING TECHNOLOGY
502 Newbold St., London, Ontario, Canada N6E 1K6
Phone: 519-686-5010
Web: www.oiart.org
Notes: Three semester Audio Recording Technology college-level degree (164 class hours, 144 lab hours).

RECORDING ARTS CANADA
ONTARIO: PO Box 11025, 948 Hwy. #8, Stoney Creek, Ontario, Canada L8E 5P9
Phone: 888-662-2666
QUEBEC: 34, Chemin des Ormes, Ste-Anne-des-Lacs, Quebec, Canada JOR 1BO
Phone: 450-224-8363
Web: www.recordingarts.com
Notes: One year Audio Engineering and Multimedia Production diploma.

RECORDING INSTITUTE OF DETROIT
14611 9-Mile Rd., Eastpointe, MI 48021
Phone: 888-683-1743
Web: www.recordinginstitute.com
Notes: Recording Engineer Program (three classes), Associate of Recording Engineer Program (seven classes), Recording Engineer/Producer Program (eleven classes).

RECORDING INSTITUTE OF TECHNOLOGY (DEPARTMENT OF MUSICIAN'S INSTITUTE)
1655 McCadden Place, Hollywood CA 90028
Phone: 323-462-1384
Web: www.mi.edu
Notes: Offers Recording Institute of Technology certificate (six months) and a Recording Arts for Producers certificate (six months).

THE RECORDING WORKSHOP
455 Massieville Rd., Chillcothe, OH 45601
Phone: 800-848-9900
Web: www.recordingworkshop.com
Notes: Recording Engineering and Music Production Program (five weeks), Advanced Recording Engineering and Music Production Program (one week), Studio Maintenance and Troubleshooting Program (one week), NewTech Computer-based Audio Production Program (one week).

SAE
NEW YORK: 269 W. 40th St, 2nd Flr., New York, NY 10018
Phone: 212-944-9121
NASHVILLE: 7 Music Circle North, Nashville, TN 37203
Phone: 615-244-5848
Web: www.sae.edu
Notes: SAE Diploma (nine-/18-month).

SF AUDIO NET
39 Gilbert St., San Francisco, CA 94013
Phone: 415-863-6883
Web: www.SFaudio.net
Notes: Two-month Music Production program.


SAE NASHVILLE

SHEFFIELD INSTITUTE FOR THE RECORDING ARTS
13816 Sunnybrook Rd., Phoenix, MD 21131
Phone: 410-628-7260
Web: www.sheffieldav.com
Notes: 290 hours/six-month full-time Audio Engineering Program.

SOUND MASTER RECORDING ENGINEER SCHOOL AUDIO/VIDEO INSTITUTE
10747 Magnolia Blvd., North Hollywood, CA 91601
Phone: 213-650-8000
Web: www.engrsnd.com
Notes: Ten-month program. Recording Engineering Certificate (720 clock hours).

SYNERGETIC AUDIO CONCEPTS
8780 Rufing Rd., Greenville, IN 47124
Phone: 800-796-2831
Web: www.synaudcon.com
Notes: Two-day Sound System Setup and Optimization Seminar, Three-day Sound System Design Seminar.

TREBAS INSTITUTE
VANCOUVER: 112 East 3rd Avenue, Third Floor, Vancouver, British Columbia, Canada V5T 1 C8
Phone: 604-872-2666
TORONTO: 410 Dundas St. E, Toronto, Ontario, Canada M5A 2A8
Phone: 416-966-3066
MONTREAL: 451 Saint-Jean St., Montreal, Quebec, Canada H2Y 2R5
Phone: 514-845-4141
Web: www.trebas.com
Notes: Nine-month diploma programs are offered, including Music Business Administration, Audio Engineering Technology, and Recorded Music Production. All can be used as one year of credit toward a BA in Sound Technology at the Liverpool Institute for Performing Arts in England. Assistance with job placement.

UNITY GAIN RECORDING INSTITUTE
1953 Riccardo Ave., Fort Myers, FL 33901
Phone: 941-332-4246
Web: www.unitygain.com
Notes: Audio Recording Program with certificate of graduation (taught in four 12-week sessions).

CONFERENCE PROGRAM*

Saturday, May 5, 2001

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

Keynote

GAVIN presents the birdseye view of the recording industry from the prospective of the record label executive/producer. As both industries change, can we predict where we will be in the future, and how these two symbiotic industries will continue to interact?

Speaker: To be announced

THE DIGITAL STUDIO, PART 1

The proliferation of hard disk and optical recording systems has changed the operations of today's studios. An in-depth discussion of the technical and business implications.

Moderator: Roger Charlesworth, Technology Consultant

9:40 a.m. - 10:30 a.m.

Whose Drive Is It, Anyway?

A big question nowadays. When a client is provided with a workstation and work drive(s), what happens when the session is over? Who's responsible for the information on the drive, and for making the drive ready for the next session? What constitutes the submission of a project? With all the rules in production changing, record labels can be in a quandary as to what exactly constitutes a complete set of masters. In what form and formats do track and mixdown masters get submitted? These questions and more will be addressed.

Panelists: Bill Dooley, Extasy Recording; Dino Elefante, Sound Kitchen; Allison Booth, BMG Nashville; Michael Davis, Digital Audio Post

10:30 a.m. - 11:00 a.m.

Morning Break/Demos

11:00 a.m. - 11:45 a.m.

SANS, RAID Arrays, and Network Technology

Exploring the business aspects of making it easy and transparent for your client to move between rooms; addressing the need for backups, offering archiving services, and capitalizing on new opportunities — are these new revenue streams, or mandatory expenditures you hadn't counted on?

Panelists: Wade Norton, NRG; Jason Hollowell, Four Seasons; Jeff Greenberg, Village Recorders; Erinn Thorp, Crawford Communication

11:50 a.m. - 12:30 p.m.

File Interchange

Are you moving data between platforms? An analysis of OMF, Open TL, AES 31, and other protocols. Who can interconnect with whom and how? Will universal interconnectivity happen — and when? Our manufacturer's panel discusses the state of interconnectivity, and offers attendees the chance to voice their needs and concerns.

Panelists: Jeff Geidt, SADiE; Ted Wolfe, Euphonix; Rob Hill, Steinberg; Digidesign; Timeline/TASCAM

12:30 p.m. - 1:30 p.m.

Networking Luncheon/Demos

ATTRACTING AND KEEPING CLIENTS

The competitive landscape of the modern recording industry means traditional business strategies must be rethought and refreshed. We present two views on the subject — that of the seeker, and that of the sought after.

1:30 p.m. - 2:30 p.m.

The Producer's View

What drives a producer to — or away from — a particular facility? What do producers expect? A look beyond typical customer service issues — our powerhouse panel speaks out.

Panelists: Ben Fowler; Joe Chicarelli; Norro Wilson; Buddy Cannon; Michael O'Reilly

2:40 p.m. - 3:30 p.m.

The Studio View

Keeping the client and encouraging lost clients to return. Creating new revenue streams and service centers in your facility. Turning any potential space into a profit center. Anticipating your client's needs at all levels. Alternative clients, who are they and how do you attract them to your studio? Buying into new technology — how early do you adopt it, and how do you choose what's right for you? Our studio owner/manager panel helps lift the veil.

Moderator: Andrew Kautz, Emerald Entertainment

Panelists: Warren Rhoades, Sound Stage; Jennifer Rose, Sound Kitchen; John Fry, Ardent; Tim Butler, CRC (Chicago Recording Company)

3:30 p.m. - 4:00 p.m.

Afternoon Break/Demos

4:00 p.m. - 4:50 p.m.

So Your Clients Want to Book a Surround Session...

Building a Surround room, or outfitting a new or existing room for Surround — the basics and the extras. Our panel offers their hands-on experiences as case studies.

Moderator: David Amlen, Sound-on-Sound

Panelists: Jim Kaiser, Mastermix; KK Proffit, JamSync; Chris Fogel, Engineer/Studio Owner

5:00 p.m. - 6:00 p.m.

NARAS P&E Panel

Changes in the professional audio industry are the catalysts for change in the methods and environments used to produce audio content. An all star panel from the Producers and Engineers wing of NARAS addresses what these changes mean for the creative community.

Panelists: To be announced

6:00 p.m. - 7:00 p.m.

Reception

Sponsored by EMTEC

Sunday, May 6, 2001

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

The Consumer View

A review of purchasing trends and attitudes among American audio and electronics consumers.

THE DIGITAL STUDIO, PART 2

The technology of digital recording. How hard disk, optical and tape-based systems, and facility interconnectivity is affecting studios of every size.

Moderator: Frank Wells, *Pro Sound News*

9:40 a.m. - 10:30 a.m.

Multi-Facility Sessions

Transporting digital recordings between facilities and backup solutions. The days where you could walk into any studio in the world with a roll of two-inch tape are over. What your client shows up with may be incompatible with your facility's systems, or, you could be blamed for the failure of compatibility when a project moves between facilities. Learn how to anticipate and avoid these problems.

Panelists: Lynn Futson, 3D Audio; Mike Poston; Jim Jordan, StarStruck Studios

10:30 a.m. - 11:00 a.m.

Morning Break/Demos

11:00 a.m. - 11:50 a.m.

Networking Technologies and Alternatives

A practical seminar with manufacturers and service providers of the tools to interconnect and stay ahead of the pack.

Panelists: Gary Holladay, Studio Network Solutions; Paul Levy, Advanced Audio; Glyph; Medea; Rorke

12:00 p.m. - 1:00 p.m.

Networking Luncheon/Demos

1:00 p.m. - 1:50 p.m.

DVD-A: A Smooth Session Primer

When your client wants to work on a DVD-A project, can you advise them what audio formats, documentation, and other materials they need to turn in to the label? What does it take to give an authoring house what they need? What questions do you need to ask your clients?

Speaker: Paul West, Universal Music Group

2:00 p.m. - 3:00 p.m.

Business Essentials

Run with the big boys. Experts in the areas of marketing, finance/leasing, insurance, and operations illustrate with stories and solutions traditional and innovative ways of handling business fundamentals.

Panelists: Keith Hatschek, Hatschek & Associates; Joe Montarello, The Recording Studio Insurance Program

INREVIEWTOC

TABLE OF CONTENTS

Digital Audio Essentials

from the digital audio interface experts

NEW! ADAT ←→ S/PDIF conversion

z-8.8a Lightpipe Detangler is a Lightpipe patch bay and also converts bidirectionally between ADAT Lightpipe format and S/PDIF (AES/EBU optional). S/PDIF inputs also feature defeatable sample rate conversion, allowing four asynchronous stereo digital sources to feed an eight-channel Lightpipe destination.

Digital Detangler automated digital audio patchbay

Available with 8x8 or 16x16 inputs and outputs. Supports AES/EBU, S/PDIF optical & coaxial, ADAT Lightpipe optical (no format conversion to and from ADAT format). Mix and match port configurations to build the perfect solution for your studio (standard configurations also available).

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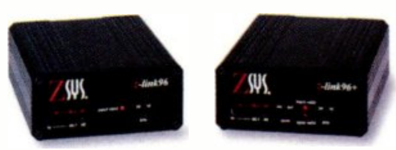
With a full range of input and output formats, input and output wordwidths up to 24 bits at sample rates up to 96 kHz. Available in two-channel (z-3src) and eight-channel (z-8src) packages.

Digital Detangler Pro automated digital audio patchbay

Available in 8x8, 16x16, 32x32, and 64x64 configurations. Can be controlled from dedicated remote controller, Mac, or PC software (network versions available).



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CIRCLE 52 ON FREE INFO CARD

Studio OBSERVER

PUBLISHED BY EMTEC PRO MEDIA, INC.
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STEVE ALBINI

ANALOG GURU

Steve Albini feels strongly – very strongly – about his preference for analog as the ultimate medium for recording music. However, his passion stems not from the stridently ideological perspectives often espoused by purists; rather, they are couched in terms that are at once practical and intellectual, but no less heartfelt.

"It's not a matter of arguing over the relative sonic merits of analog versus digital; it's simply a matter of recognizing the fact that analog is a mature format, one that's undergone constant refinement over the last fifty years," Albini explains. "That's why analog is truly the only choice as both the master recording and archiving format. The earliest analog recordings are still playable, whereas there are recordings on digital formats that were made just a few years ago that can never be heard again. Not only because loss of data makes them unrecoverable but also because some of the formats they were made on have become obsolete."

> *"That's why analog is truly the only choice as both the master recording and archiving format. The earliest analog recordings are still playable..."*

Steve Albini's extensive discography gives any observations he makes on the recording of music significant weight. It ranges from classic recordings for artists such as Nirvana, Jimmy Page & Robert Plant, and Robbie Fulks, to the seminal voices of the alternative era of pop music, including P.J. Harvey, Killdozer, Silver Apples and the Pixies. But his influences add another dimension to his perspective: Albini cites the groundbreaking field recordings of Alan Lomax, and the American musical archives of the Smithsonian Institution and labels such as Folkways Records as having had as much effect on his approach to music as any individual or technology.

"What you realize is that, when you make a record, what you're actually doing is making a historical document of a creative event," he says. "What people like Lomax did was as much inspirational to me as it was influential. They were recording music that was being made anyway, not that was specifically created for a studio. So, as a producer and an engineer, I think my job is to allow the artist to make a record that represents them. I see my task as a purely technical one: getting someone's creativity on tape without creating any obstacles in the process. My role is to be a conduit for the creative impulses of other people, and to make a permanent record of what might be a fleeting moment of creativity."

And that's where the analog format and Steve Albini's vision come together. The Montana native, who moved to Chicago in 1980 and became deeply enmeshed in the city's thriving alt and underground music landscape, grew up on analog, learning the craft of engineering by recording his and friends' bands, often on borrowed equipment.

FROM THE TOP

21st CENTURY SOFTWARE

On Oct. 1 of last year, EMTEC Pro Media completed the implementation and integration of the comprehensive SAP integrated business software system. And believe me, this is music to everyone's ears.

SAP's brilliant software allows companies to integrate and access every aspect of their operations — sales, customer support, distribution, inventory, shipping, manufacturing and cost information — from a central server. Furthermore, it allows every department in a company to tap into all of that information on a real-time, up-to-the-minute basis from any PC-based computer on the system.

The benefits to EMTEC's customers are real and immediate. Every type of media facility using any type of media will share the advantages of this kind of power with EMTEC. We can respond to price fluctuations, changes in demand, and other critical business variables faster and more precisely than ever before. And very soon, our implementation of SAP will be taken a step further, with an Internet connection.

Integrating SAP into EMTEC's operations was a complex and challenging process. But it reflects the level of commitment that we've always had for our customers. Media users are out on the cutting edge of technology. We want you to know that we're right there beside you in that effort.

Joe Ryan, President • EMTEC Pro Media, Inc.

He opened his 8-track studio in 1986, and eventually purchased a 24-track machine — a vintage Studer A80 once owned by Phil Ramone. Since then, his personal recording arsenal has grown considerably, housed in a two-studio, 11,000 — square-foot facility dubbed Electrical Audio.

Describing analog as an "impeccable" format, Albini points out that it has survived and thrived over the course of a half century, and that in the process has formed an unbroken road from past to present to future. "I'm certainly glad that everyone from Nat Cole to the Beatles to Buddy Holly made their records on analog tape," he says. "It's a bit facetious, but the point is, had digital technology been available to artists like that, we likely would not have access to them anymore. There are dozens of digital formats that have come and gone during analog's lifetime, and what that's created is an entire orphan generation of masters that can never be played again, unless they were also archived to analog."

"In completely quiet passages, digital may be more apparently quiet, but the reality is, it's quiet in a different, less pleasant way."

The real problem with digital, Albini believes, is that, unlike analog, which has formed an almost spiritual partnership with music over the decades, digital technology is not fundamentally an audio domain, and thus is directed by any number of other influences, most of which have nothing to do with sound. "There's a complete overhaul of media choices every time there's a change in the computer paradigm, which makes it impossible to have a permanent, consistent record of events," he explains. "Computer data storage changes when industries like the insurance business demand changes in the archival technology."

Albini considers every recording a "master" recording, whether it's the multitrack or the stereo mix master. He likes the fact that analog provides a consistent format for every step of the process, as well as the documented fact that analog lasts and provides for future access to all levels of the recording process. "As long as you take reasonable measures to store it properly — the key is to remember to look at it from the beginning as conservation rather than restoration," he adds. "But that's far different from a digital sound file. Files will eventually disappear."

Albini also likes the fact that the analog domain offers more critical options than often meet the eye. He likes EMTEC/BASF's SM 900 maxima 2-inch tape formulation for recording, citing its exceptional capacity to handle high output signal. On the other hand, Steve uses EMTEC/BASF SM 911 for mixing and archiving. "The SM 900 certainly handles high output well, but what sold me on it was its reliability and playability," he explains. "For mixing and long-term storage, the issue of higher output handling is less of an issue, but what you get by using two different formulations is the equivalent of a backup. So if over time there's a problem in storage, there's a completely different formulation to fall back on."

Digital's proponents cite that format's advantages in several areas. Albini, characteristically, has a ready response for every point. On the topic of noise floor, he notes that, as long as music is playing, "A properly biased and aligned machine — we use 500 nanoWebers per meter here — has no noise issue, even with 24 tracks wide open. In completely quiet passages, digital may be more apparently quiet, but the reality is, it's quiet in a different, less pleasant way. It begins to get grainy because the word length gets shorter. In terms of dynamics, analog is a forgiving medium with a lot of flexibility and softer boundaries, perfect for music.

Digital just clips — it's on, it's off. And I think that procedures like multitrack edits and spot erasing are actually more cumbersome to do digitally."

Albini's no digital Luddite; he's happy to use formats such as DAT and CD-R for applications such as reference copies and transfers. "I'd just never use them to make records," he laughs. "Because you can never forget that in the end what you're doing is creating a historical record of a creative event. And analog shows the most respect for both the history and the creativity."

Andre Wahl and Garth Richardson at the Armoury

IN SESSION WITH GARTH RICHARDSON

Garth Richardson came up through the audio engineering ranks the old-fashioned way: via apprenticeship. It didn't hurt that one of those that Garth, a Toronto native who now lives in Vancouver, got to learn from was his father, Jack, a producer whose discography includes critically acclaimed records for artists such as Poco, the Allman Brothers, Michael Bolton, Alice Cooper, the Guess Who and Bob Seger. In fact, Garth's first session as a second engineer was on Seger's classic "Night Moves." Nonetheless, Garth Richardson remembers that he spent plenty of time mopping floors and cleaning ashtrays, the time-honored route through the business. But learning the basics from the bottom up has been a foundation stone for his own discography, which includes engineering and remix work for, Motley Crue and White Lion and others, as well as producer credits for Rage Against The Machine, the Melvins, the Supertones, Sword and more.

SO: How did you become an engineer?

Garth: I learned from others. My father, Jack Richardson, was a record producer. Early in my career, I started working a studio in Toronto called Phase One Studio, which launched a lot of great engineers, including Randy Staub (Metallica), Scott Humphrys (Rob Zombie) and Bill Kennedy (Nine Inch Nails). After that I moved to L.A. for twelve long years - and I mean long years. But my first real breakthrough was doing demos for the Bullit Boys - I helped them get their deal. My first big record as an engineer was Mother's Milk by the Red Hot Chili Peppers.

SO: How do you approach your productions?

Garth: The way things sound is critical to getting the sense of the artist across on record. I always get the artists to spend a day or two with all of my amps and let them play for a while, just playing with and looking for sounds.

SO: As much as it seems characterized by crunchy guitars, metal has also been one of the great genres for vocal sounds. Can you tell us a little about how you record vocals? Which microphones do you prefer?

Garth: When I did the first Rage Against The Machine record, which I engineered and co-produced, we used a Shure SM 58 on the lead vocals. I wanted (lead singer) Zack de la Rocha to feel as through he were performing a live show. Sometimes it's a matter of establishing the vibe in a studio, getting the artist in the setting they're most comfortable with, even if it's just in their mind. I was working with a band called Joydrop, and their singer, Tara Sloan is a great vocalist. What I'm doing with her is using two mics - an AKG -12 and a Shure SM 7 - at the same time. I put the C-12 on top of the SM 7, and I have her position to sing into the C-12. That combination gives you all the nice top end of the C-12 and all the mid-range body from the SM 7.

SO: Group vocals are always a challenge. How do you approach that craft? Is layering a big part of it?

Garth: For that, like for a lot of things in life, the greatest thing is to work for someone that has been doing it for a long time. I got my training from producer Michael Wagener (White Lion), who is the king of layering - as many as 20 tracks for a song. The key is to balance them by taking a slightly different approach to each track - sing some kind of growly, others more smoothly. That gives you a lot of tonal combinations to choose from.

SO: When it comes to microphone placement on amplifiers and instruments, do you still like to experiment, or do you have tried-and-true techniques that you default to? What are your favorite instruments to mic?

Garth: "It depends on the band and the style of music, and I admit that there are a few basic approaches that always work. If I want an in-your-face sort of drum sound, I tight-mic every drum and guitar amp. If you want that big, open sound, you move the mics about three feet back. The one thing that I try not to do is to keep doing the same thing over and over again. Andre Wahl, the engineer I've worked with on the last five records I've produced, and I discuss mic placement in detail before each production, because as I mentioned earlier, the sound of a record has as much to do with its success as the song, the singer anything else."

SO: Do you have any favorite studios? Which ones and what about them makes them special to you?

Garth: My three favorite studios are Sound City (Van Nuys, CA), the Warehouse Studio, and the Armoury (both in Vancouver). The Warehouse, which is owned by Bryan Adams, has one of my favorite consoles: it was custom-built by Rupert Neve originally for George Martin. But all three studios have great vibe and all three have something else that I consider critical - excellent maintenance. I don't mind paying for a studio where everything works. In fact, when the rate's too low, I warn everyone: check phase, check this, check that, all the time. It's like anything else - you get what you pay for.

SO: You've expressed a high regard for the EMTEC/BASF SM 900 maxima-- what its it about that formulation that works so well for you?

Garth: SM 900 is simply a great-sounding tape. The bottom end is true and the high end is crystal clear. I love when you hit it with level and you get that great-sounding compression, yet with minimal print-through. SM 900 is one of the reasons that analog will be around for years to come.

SO: How have you adapted to the integration of analog and digital technologies in the studio?

Garth: Nothing beats the sound of tape compression. But that said, I do like to combine the sound of analog with the flexibility of digital, of being able to shift things around in an instant. Also, while I like to record most everything to analog initially, I use digital for a lot of overdubs to avoid "polishing" off the top end of the tape. But you do have to stay up to date. I remember my dad coming home one day in 1968 and saying, "Now they have three tracks! What's next?"

TAPE TALK

by Jean Tardibuono

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

Often lost in the discourse about audio formats is the fact that, no matter what format audio professionals choose for music, the need to make back-up copies of everything is fundamentally critical.

On the surface, it might seem like the issues about backing up are purely economical and technical. But as Steve Albini points out so eloquently in this issue's feature article, accessible musical archives have enriched the lives of millions, and the loss of even a few notes is a loss on the same magnitude. Analog media has been a positive and enduring force in this application.

Analog bridges both technical and chronological canyons because of its universality and proven reliability. But the bottom line is that we at EMTEC want to see awareness of archiving and back up raised across the board, regardless of format. That's one of the reasons EMTEC's array of media products is as broad and diverse as it is, from our analog tapes and digital linear media through our recently released DVD-recordable optical disc lines.

EMTEC's commitment has led us to take positive action, such as our $10,000 in-kind grant to Queens College in New York to support the preservation of Louis Armstrong's recordings, and our most recent effort, as a participating sponsor of the Save Our Sounds project, led by the Smithsonian Institution and the Library of Congress.

In contributing towards the preservation of a relative handful of notes in the ocean of a century's worth of music, the real implication of these efforts is the heightened awareness of what all of us can do to preserve our musical legacy.

By Mitch Gallagher • mgallagher@uemedia.com

Digidesign Pro Tools 5.1 Digital Audio Workstation Software Upgrade

The latest and greatest version of Digidesign's omnipresent DAW

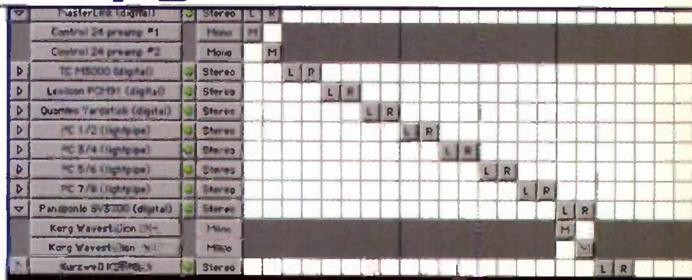
It will come as no surprise for me to tell you that Digidesign's Pro Tools DAW system has become standard issue for many audio professionals. Pro Tools systems are used in all aspects of audio production, and at all levels, from home studios to high-end commercial facilities to esoteric postproduction houses. So when there's a significant update to the software, it's big news. That was certainly the case at the AES show held in Los Angeles last fall, when Digidesign unveiled Pro Tools 5.1 — a major upgrade to the product that introduces numerous new capabilities as well as a number of functional enhancements.

Since Pro Tools has been covered extensively in these pages in the past, this article will focus on what's especially noteworthy among the new features in version 5.1.

AUDIO EDITING

1. Beat Detective (TDM systems only): One of the most exciting new features in Pro Tools 5.1 is Beat Detective, which can analyze an audio track, extract tempo and bar/beat information, and conform audio to the extracted data. You can start out calculating the average tempo of a selection, and generating tempo and bar/beat maps. At this point, you can use Beat Detective to separate one audio region into multiple regions based on detected beats, then conform the multiple regions to the previously generated tempo map.

Beat Detective can automatically fill and crossfade any gaps created by conforming regions, and it works across single or multiple tracks — perfect for multitracked drums. It can also work in "Collection mode," which lets you analyze multiple tracks separately, then consolidate all the triggers detected.



▶ DIGIDESIGN PRO TOOLS 5.1

MANUFACTURER: Digidesign, 3401-A Hillview Ave., Palo Alto, 94304. Tel: 650-842-7900. Web: www.digidesign.com.

SUMMARY: Pro Tools version 5.1 provides an array of new, desirable features, as well as several "it's about time" features. More than just a "surround" update, the new version also includes enhanced MIDI, editing, mixing, and functionality/convenience features.

STRENGTHS: Host and DSP processing supported. Stereo and multichannel tracks. Multichannel and multi-mono plug-in support. Insert re-ordering. Output paths. Beat Detective. Tab to Transient. Improved Strip Silence function. MIDI Event List editing. Multiple plug-in windows open at once. Duplicate inserts and tracks. Sixteen levels of undo/redo.

WEAKNESSES: Not yet supported for PC-based systems. Some features not offered in LE version. No MIDI scrubbing. Can't turn off individual speakers in surround panner. RTAS plug-ins can only be used on disk tracks.

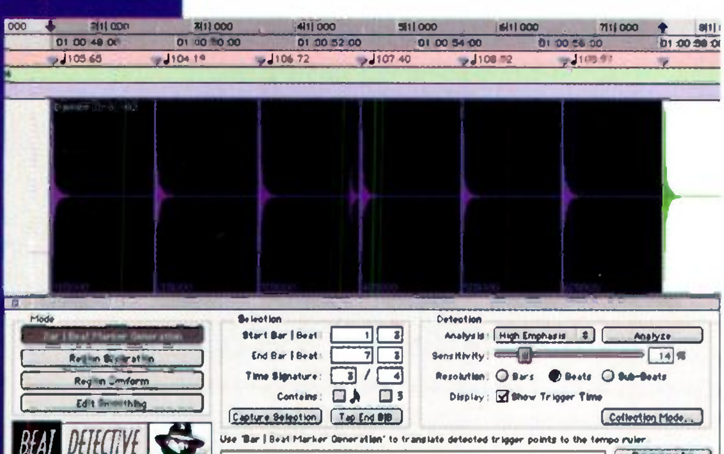
SYSTEM REQUIREMENTS: TDM: Mac-based Pro Tools|24, Pro Tools|24 Mix, or Pro Tools|24 Mix Plus system. LE: Mac-based Digi 001, ToolBox or ToolBox XP system, or Audiomedia III card.

PRICE: Varies; contact dealer or manufacturer

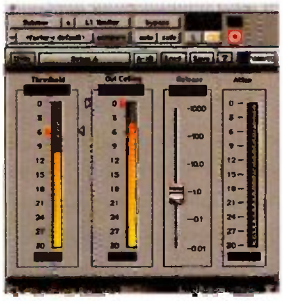
EQ FREE LIT. #: 110

In practice, Beat Detective works amazingly well, given source material with sufficiently discrete transients. It can analyze down to the "sub-beat" level, and can detect sixteenth-notes and triplets. At the sub-beat level, it can also be used to superimpose one track's "groove" or rhythmic feel onto another audio track.

In the example shown here, I've analyzed a short segment of a very loosely played djembe track and generated a tempo map based on the analysis (the pink bar above the waveform). My next step was to break the region up based on the detected beats (shown by the blue vertical lines). Next, I deleted the



Beat Detective window



The Strip Silence Preview (above) and Multi-mono L1 windows

tempo map and set the session's tempo to 105 BPM. Beat Detective then conformed the beats to the new tempo. The last step was to use Beat Detective to fill in and crossfade the gaps and overlaps in the results. Worked like a charm!

2. Strip Silence: The Pro Tools Strip Silence command has been enhanced and made easier to use. The biggest change is that the new version provides a preview display that shows how the audio regions will be created when silence is stripped out. Very nice.

3. Tab to Transient: In the upper-left corner of the new version's Edit window is an insignificant little icon that, when clicked, engages a very powerful new feature: Tab to Transient. This feature automatically jumps you to the next detected transient peak in a track when the Tab key is pressed. Tab to Transient is clearly a great tool when editing drum tracks, but it's also useful for making selections, setting start/end and in/out points, and so on.

In practice, it works very well. My only suggestion would be the addition of a user parameter for adjusting the sensitivity of transient detection.

4. Universe Window (TDM systems only): New to Pro Tools version 5.1 is the Universe window, which provides an overview of the entire session. You can click in the window and instantly locate without scrolling. That's a great feature, but it would be even nicer if you could drag in the Universe window and have the selection correspondingly zoom to fill the Edit window.

You're given limited display options; you can stretch the window out horizontally and the track display stretches to follow, but there's no way to make the very thin track bars any larger (taller) vertically. As it is, if you only have a few tracks recorded, it can be tough to see them. A vertical zoom control would be a useful addition.

Despite the preceding two wishes, the Universe window provides an easy way to navigate around a session quickly. Once you start using it, you'll wonder how you got along without it.

PLUG-INS

1. Duplicate and Re-order Inserts: Plug-in handling in version 5.1 has been greatly enhanced by two long-awaited features: Option-dragging allows you to create a copy of a plug-in in a different location on the same track or on a completely different track. You can also drag a plug-in instantiated on a track to a different insert location; if you place it on top of an existing plug-in, the old one will be replaced.

GET RIGHT TO THE POINT...

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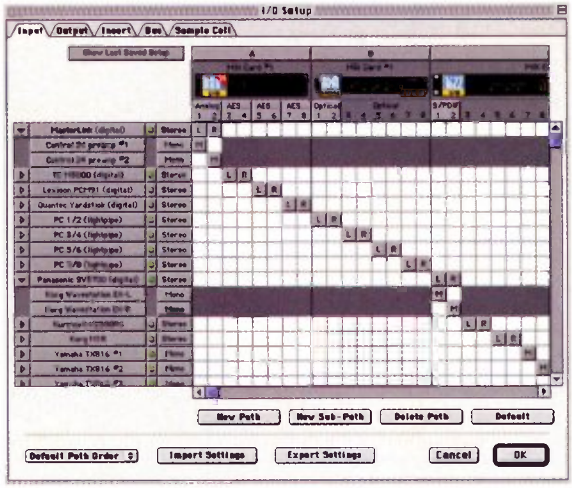
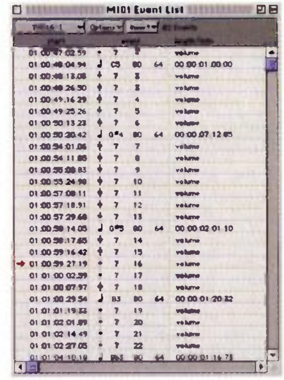



Input Routing (left) and MIDI Event List windows

Both operations retain all settings and automation data.

2. Multichannel and multi-mono plug-ins: Part of Pro Tools 5.1's surround mixing support includes multichannel processing capabilities, which come in two forms: true multichannel plug-ins, such as the new DigiRack multichannel plug-ins included with the update (Compressor, Limiter, Expander/gate); and behind the scenes support for turning any "regular" plug-in into a multi-mono plug-in.

With multi-mono plug-ins, when you select a plug-in, enough instances are automatically loaded to process all the channels in a track (e.g., six limiters to process a 5.1-channel track, as with the multi-mono Waves L1 limiter shown here). Pro Tools automatically links the multiple instances under the control of one assignable channel (e.g., the left channel provides the settings for all six channels in a 5.1 multi-mono plug-in) or you can choose to have each channel in a multi-mono plug-in operate independently. Note that the channels of the plug-in aren't linked by sidechains, etc. (For that you need a true multichannel plug-in.) You're simply given the option to handle them as one easy-to-load and control multichannel plug-in rather than six (or however many) separate ones.

3. RTAS Plug-ins in TDM Sessions (Mac only): One of the most significant new features in Pro Tools 5.1 is the ability for TDM systems to also take advantage of host-based RTAS plug-ins. There are a few limitations: RTAS plug-ins must run on disk audio tracks and don't work on aux inputs or master faders; RTAS plug-ins must be inserted before any TDM plug-ins on a given track; and tracks using RTAS plug-ins must be auto-voiced rather than hard-assigned to a specific voice.

But these are all minor "just be aware of it" things compared to the benefits of getting a bunch of DSP power basically for free — your computer is just sitting there waiting to give it to you. Many RTAS plug-ins don't introduce processing latency into the signal as TDM plug-ins do.

The DSP-hungry among us can only wait in anxious anticipation for Pro Tools to support multi-processor computers....

4. Multiple Plug-in Windows: A much-requested feature has arrived! Multiple plug-in windows can now be open simultaneously.

5. Trim Plug-in: Version 5.1's new Trim plug-in allows you to adjust the gain of an audio track, aux input, or master fader over a range of –INF to +6 dB, as well as flip the polarity of and mute the signal. As a bonus it does this without using any DSP power.

MIDI

1. Record to multiple MIDI racks: Pro Tools 5.1 supports the ability to record multiple MIDI devices or MIDI channels to separate MIDI channels — nice if you're transferring data from an external sequencer or recording from, say, a keyboard and percussion MIDI controller at the same time.

2. MIDI Event List: Ever since Pro Tools gained enhanced MIDI sequencing capabilities, users have been clamoring for event editing capability, and it's arrived with version 5.1. You can edit event start time, note number, velocity, release velocity, and length, as well as controller number and value. You can also use the pulldown at the top-right of the window to insert notes or controller events.

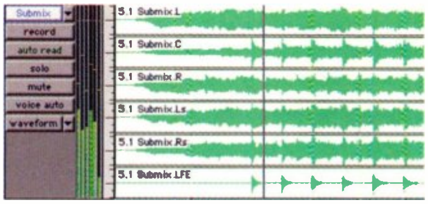
TRACKS/MIXER

1. Inactive Tracks: With any Pro Tools rig, managing system resources can be an issue. In version 5.1, you can make tracks, I/O assignments, and plug-ins inactive. Inactive items are simply "turned off" — they keep all their settings.

This seemingly minor feature is actually one of my favorites in the new version. In the past, if you bounced a track with plug-ins, and wanted to keep the original around in the session "just in case," you had to turn off the track's voice assignment and remove any of its plug-ins to keep from using unnecessary DSP power. The new "inactive" feature lets you keep the track around exactly as it was without tying up precious DSP power. It's also great if you're tight on DSP and need to A-B two versions of a track or plug-in.

2. Duplicate and Import tracks: Two handy track functions have been added in this version: You can now duplicate — create an exact copy of — a track within a session, and you can import a complete track from one session into another. In both cases, all assignments, plug-ins, settings, automation, bussing, audio files and regions, and fades are included.

3. I/O Routings: Pro Tools 5.1 offers tremendously improved input, output, insert, bus, and SampleCell routing capabilities. You can save, name, and import and export your I/O configurations, which is nice if you work on multiple systems or if you're converting older sessions for use with a new or different system. The routings to various hardware ins and outs are called "paths"; each path can also contain sub-paths. In the Input Routing example shown here, the connections for my three Pro Tools hardware interfaces are shown; where the interface has both analog and digital connections, you can switch which one you're using. On the left of the window are the paths I've set up so that the software can route to the hardware. The first one (at the top of the list) shows that the MasterLink is the path; it's connected to inputs 1 and 2 on the 888 I/O interface. Sub-paths are used to route two mic



A multichannel audio track (top) and a Surround Panner window

preamps from the Control 24 to the 888's analog inputs 1 and 2.

4. Surround Mixing Support (Pro Tools 24 Mix and Mix Plus systems only): There are numerous new features in Pro Tools 5.1 for surround

▶ continued on page 136

By Mikail Graham • mgraham.EQ@onemain.com

Zoom PS-02 Palmtop Studio

A glimpse at the future — palmtop studios

Ten years ago, Zoom turned the world upside down with the 9002 multieffects processor, an engineering feat that packed an unheard of amount of signal processing power into a box almost small enough to carry in your pocket. How appropriate then, at the start of the 21st century, for Zoom to introduce the PS-02, the world's first true palmtop studio — one that actually does fit in your pocket. Incorporating a pattern-based drum/bass machine, multitrack audio recorder, mixer, and digital effects processor into one compact box, the PS-02 would seem ideal for those who need a portable writing/recording rig.

I first heard about the PS-02 through a Zoom ad (in EQ, of course) featuring the Alessi Brothers, who had recorded a demo using nothing but the PS-02. I downloaded the MP3 demo and was very surprised by what I heard. I knew I had to check it out.

NOTHING BUT THE FACTS

Accessing the various PS-02 functions is done via eleven buttons, three sliders, a volume wheel, a four-way touch pad, and an easy-to-read backlit LCD. Up to 100 songs can be created using any of the 200 preset rhythm and bass patterns, all of which use sampled PCM drums, percussion, and bass. There are six stereo drum kits including Standard, Rock, Jazz, Analog, Power, and Funk, and no less than five basses, including some really fine-sounding finger, pick, slap, acoustic, and synth samples. For the bass, you can set the key and select any of fifteen chord progressions for each pattern, including major/minor/diminished/augmented, etc. — essentially all the chords for almost any idea you might have.

You can also use the PS-02 as a stand-alone effects processor. Zoom's Variable Architecture Modeling System (VAMS), originally developed for their GFX-8 Guitar Effects Processor, lets you get a multieffects processor with 50 different algorithms, of which you can use up to six simultaneously. There are 60 factory presets, 60 user presets, and a built-in chromatic tuner. There's a wide variety of amp/cabinet simulations, extensive modulation algorithms such as auto wah, pitch shifter, exciter, phaser, chorus, flanger, and tremolo/vibrato, a compressor,

▶ZOOM PS-02

MANUFACTURER: Zoom is distributed in the U.S. by Samson Technologies Corp., P.O. Box 9031, Syosset, NY 11791-9031. Web: www.zoom.co.jp.

SUMMARY: A highly portable multitrack recorder, drum and bass machine, and stand-alone multi-effects processor.

STRENGTHS: Small size. Amazing amount of power and potential to create great sounding demos, or to work out ideas on the road. Excellent reverbs and delays.

WEAKNESSES: Too much data packed into small display. Can't currently create your own drum/bass patterns. No loop mode for audio tracks. One tempo per song.

PRICE: $625 (includes one 8 MB SmartMedia card)

EQ FREE LIT. #: 105

SPECS AT A GLANCE

- 20-bit internal DSP
- 20-bit, 64-times oversampling A-D
- 20-bit, 8-times oversampling D-A
- Zoom DSP ZFX-2 (internal 24-bit processing)
- 31.25 kHz sample rate
- 100 songs
- 200 drum/bass patterns
- 6 simultaneous effects
- 50 effect algorithms
- 120 effect patches
- Integrated omni condenser microphone
- 1/4-inch mono input –10 to –50 dB w/adjustable pad
- 1/8-inch stereo aux Input
- 1/4-inch stereo output –10 dB
- Headphone output (stereo mini-jack)
- 3 AAA Batteries or supplied AC adapter
- SmartMedia storage
- Weight is 140g (minus batteries)

three-band EQ, noise reduction, useful stereo mixdown FX, and a reverb/delay module that rivals boxes costing twice as much.

There's more: The PS-02 also includes a three-track digital recorder with a sample rate of 31.25 kHz. Up to ten takes per track can be recorded in mono using either the built-in microphone or line-level jack, a stereo source using the aux in, or even a combination using both the aux and mono inputs mixed together. Pre-roll plus punch in/out points can be set, plus all three audio tracks can be bounced down to a stereo or mono track if you need room for more ideas. SmartMedia cards are used to store songs, audio recordings, drum/bass patterns, custom multieffects settings, and all system info. Using a 64 MB card (approx. $80 street price) will get you 33.5 minutes of record time in hi-fi mode and an amazing 67 minutes in long mode. Zoom says you can work up to four hours (it was more like three in my tests) on four AAA batteries, or you can use the included AC adapter.

For this review, Zoom also supplied me with a pre-release of their brand-new Hip-Hop set, which completely changes the internal drum/bass set by adding six new bass patches, 147 patterns, and some very funky urban dance grooves. Plus, there are 30 preset songs and fourteen new effects patches that are specifically prepared for use with the built-in condenser mic. The new Hip-Hop set is already up on the Zoom worldwide Web site (**www.zoom.co.jp**), and a new rock set will soon be available as well. Both are free for download from the Zoom site or they can be purchased on SmartMedia cards from Zoom dealers.

USING THE GEAR

During a recent four-day power outage (due to a big winter snowstorm), I had a chance to work extensively with the PS-02. I found it quite easy to create grooves that moved from one chord change to the next, transposing and repeating my favorite bits. I did, however, find that you can't use the pattern repeat command to repeat audio tracks, and, unfortunately, there's currently only one tempo available per song. Laying down multiple guitar tracks, vocals, and harmonies as well as using the unit's internal effects is a whole lot of fun. When it came time to do a stereo mixdown, the PS-02's built-in pan controls for each audio track and the Mix FX presets, along with its three sliders for audio, bass, and drums, made it a snap.

I did find a bug in the manual, which states that the built-in tuner can be called up by pressing and holding the Effects Bypass button while in any mode, but I found that it's only available while in the effects mode.

Just days before I had to finish this review, a representative from Zoom sent me a prerelease of their new Macintosh based PS-02 Card Manager software. With this nifty program, you can export your song data to a Standard MIDI File, import or export audio (mono AIFF or WAV), and, most importantly, restore or backup all of the data in the PS-02. A Windows version is already available, and the Mac version should be posted by the time you read this — both are available as free downloads.

WISHES

Automation would be nice for volume and tempo, but, at this price point and size, you really can't complain. The only other things I'd like to see are an option for cut/copy/paste of a song section, including audio, to another location, and a way to create your own custom drum and bass grooves. In regard to the latter, Zoom is providing new PS-02 pattern data plus drum and bass sounds at their Web site. All you need to use them is a Mac or Windows computer and an inexpensive SmartCard reader. In addition, a new Zoom product, the upcoming RT-323, will be able to create drum/bass data that can be read by the PS-02.

THE FINAL WORD

The PS-02 is a traveling musician's dream come true. True, it's a bit awkward getting used to the small backlit display, and knowing which buttons to push for certain edits will take some time. But Zoom has laid things out logically so that, after a few sessions, you'll be able to find your way around without even using the dual-duty mini tutorial/manual. The quality of the audio, starting with drums and bass, is quite impressive, and the various effects sound great and offer a level of programmability not found in similarly priced dedicated multieffects boxes. But the capper for me is the three audio tracks and built-in condenser mic — the quality of which is nothing short of amazing for a device this size. I found the PS-02 to be one of the most amazing wonders of modern technology I've had the pleasure of working with thus far. It delivers what it advertises and more. The unit sounds great, and will change forever the way we think about recording musical ideas on the run. ■

SMARTMEDIA STORAGE CAPACITY

CARD SIZE	HI-FI MODE	LONG MODE
8 MB	3 min, 12 sec	6 min, 24 sec
16 MB	7 min, 30 sec	15 min
32 MB	16 min, 11 sec	32 min, 23 sec
64 MB	33 min, 33 sec	67 min, 7 sec

By Mikail Graham • mgraham.EQ@onemain.com

Spectrasonics Vocal Planet Five-Disc Vocal Sample Library

A treasury of vocal samples

Ever wished you had one of those Bobby McFerrin Voicestras full of soulful male and female R&B singers to lay down some hot riffs whenever you needed them? Or maybe a mix of Tuvan throat singers, Bavarian yodelers, and Himalayan and Serbian melodic vocalists just for the fun of it? If so, then this is the collection for you.

Spectrasonics, creators of Symphony Of Voices (a four-disc set of symphonic and classical vocal samples), has done it again with a massive five-disc companion set entitled Vocal Planet. A project more than four years in the making, the discs include over 12,000 samples with support for a variety of samplers, including Akai, E-mu EOS, Kurzweil, and Roland. Other sample players such as the EXS-24, GigaSampler, GigaStudio, Sampletank, and Unity can also read the Akai/E-mu format discs. The attention to detail is impressive, with some selections making use of "Chromazones" and "Groove Control" — two Spectrasonics innovations designed to help samples sound and feel more musical. Chromazones is Spectrasonics's name for recording a particular sampled phrase in all twelve keys — for example, Vocal Planet's R&B disc makes great use of this technique to avoid the usual "munchkin" effect. Groove Control is their proprietary method of slicing up mono or stereo samples so they can be played back at various tempos using a MIDI sequencer without losing timing or pitch. Like most recent Spectrasonics releases, Vocal Planet makes full use of this feature. There are Groove Control templates included for just about every major Mac/PC sequencer available, as well as Standard MIDI File format so just about anyone can get into the groove.

▶ SPECTRASONICS VOCAL PLANET

MANUFACTURER: Spectrasonics, PO Box 7336, Burbank, CA, 91510. Tel: 818-955-8481. Web: www.spectrasonics.net.

SUMMARY: An amazing collection of more than 12,000 vocal samples compiled on five CD-ROMs. Exquisitely recorded in a variety of styles including jazz, gospel, blues, dance, vocal percussion, and effects.

STRENGTHS: Stunning audio quality. Tremendous variety and selection.

WEAKNESSES: Not enough stereo samples.

PRICE: $399

EQ FREE LIT. #: 103

Most samples are recorded in mono, and sound great. Those that are in stereo, such as the jazz and African sections, are even more powerful and sonically inspiring. Almost all samples are presented without effects or reverb, making it easy to add whatever special sauce you might need. Though there are quite a few Distorted Reality-type selections in the FX category, I'd sure love to hear more — perhaps even an entire Bizarre Vocals volume.

The European section includes some lovely Celtic, Gaelic, Welsh,

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and Scandinavian phrases with both male and female vocalists. The vocal percussion loops and effects are ripe for dance tracks, and would make great additions to any basic rhythm track. Though well conceived, my least favorites were the spoken phrases in the blues, R&B, and soul categories, which — for me at least — will be hard to find a place for. Then again, a few of them may actually get me to try composing in some new directions. In contrast, the jazz samples are out of this world, moving from haunting, free-form phrasing and scats to group multi-samples that swing and sway with such class that they just beg you to use them. The

> "THE GOSPEL SAMPLES ARE SO FULL OF PASSION THAT THEY MADE ME WISH I'D GONE TO A CHURCH LIKE THIS WHEN I WAS GROWING UP."

gospel samples are so full of passion that they made me wish I'd gone to a church like this when I was growing up. The Combo Grooves section bops and bounces so hard that it has "play me now" burned into each and every one of them.

Spectrasonics mentions that it would take "more than eight hours to play this entire library from start to finish," making the collection a great value for $400. Add in the included lyric and musical reference guides, Groove Control quick tips, and full manuals for each sampler (all in PDF format), and I'd say you've got one of the most all-inclusive vocal libraries on the market. The five discs are just bursting at the seams with creative ideas, and when coupled with Spectrasonics's Symphony Of Voices, you'll have access to just about any earth-based vocal style imaginable — all from the ease and comfort of your favorite sampler. ■

By Martin Davidson

Universal Audio LA-2A Re-Issue Electro-Optical Limiter

The venerable LA-2A compressor/limiter returns

The LA-2A has a long and honorable history in the recording industry. It was the first commercially viable electro-optical limiter. Before the LA-2A, compressor/limiters typically used tubes as the gain-reduction element, and though other methods of controlling dynamic range had been developed, the circuits were rather slow and weren't particularly transparent. In contrast, the reduction element in the LA-2A consisted of an electro-luminescent light source and cadmium sulfide photocell, which not only had a much faster attack time than its predecessors, but also had a release time that allowed for a more natural sound. It was an immediate hit with engineers, and remains so nearly 40 years later.

The LA-2A was originally produced by Teletronix and passed through several companies before the rights were acquired by Bill Putnam's Studio Electronics Corporation shortly before he changed the company's name to UREI. Under the auspices of UREI, production of the LA-2A continued until 1969. More than twenty years later, Bill Putnam's sons, Bill Jr. and Jim, founded Universal Audio to carry on their father's legacy. The LA-2A was the second product introduced by the company.

LAYOUT

The appearance of the re-issued LA-2A is similar to the original. The only front-panel changes, other than a small "UA Classics" logo over the VU meter, are the thumbscrews holding the front panel have been replaced by Phillips screws, and the Compress/Limit switch has been moved from the rear of the unit to the front. (In truth, the latter is a fairly common modification on vintage units; I've worked with several vintage LA-2A's that had the switch moved to the front.) The rear of the unit is also very similar to that of the original — the fuse holder and attached AC line have been replaced by an IEC connector with an integrated fuse holder and XLR plugs have been added, making interfacing in today's studio environment more convenient. The Universal Audio version does retain the barrier strip connections used on the original LA-2A, though whether this was done for verisimilitude or because anyone would actually use them is anyone's guess. It's a nice touch, though.

▶UNIVERSAL AUDIO LA-2A

MANUFACTURER: Universal Audio, PO Box 3818, Santa Cruz, CA 95063-3818. Tel: 831-466-3737. Web: www.uaudio.com.

SUMMARY: Delivers the same features and character of the venerable LA-2A original.

STRENGTHS: It's a great reproduction of the original with great sound quality.

WEAKNESSES: No hardware bypass. Peak reduction and gain control knobs do not display calibrated gain and gain reduction.

PRICE: $3,495

EQ FREE LIT. #: 106

For those who haven't worked with vintage audio gear, the shape of these boxes can be a little disconcerting. Where a normal rackmounted processor is pretty much an enclosed rectangle with the rear panel containing the I/O and power connections, the back of an LA-2A is open, with tubes and transformers accessible. The enclosed chassis is only a couple of inches deep, which means that a rackmounted LA-2A won't block the airflow of equipment mounted below it.

THE SOUND

There are actually two questions here (or two answers, depending on how you look at it). The first is, "How does this box compare to a vintage Teletronix LA-2A?" And the second is, "How does this box sound when used in the studio?" First, let's say that the Universal Audio LA-2A compares very well with the original unit. Sonically, there are some noticeable differences between original and new LA-2A's, but, as those who work in rooms with more than one Teletronix LA-2A can attest, no two vintage units

sound quite the same either. As to why a Teletronix and a Universal LA-2A sound different, there are several possible causes: component fatigue and quality control (of the vintage units, not the Universal), modifications or component replacements done to the originals over the years, and the condition of the tubes — all of these can (and do) make a difference in the sound of vintage audio equipment.

But, for most of us, the sound of a processor in our own working environment is more important than the sound of a unit compared with an old box with the same model number (largely because most of us don't have an original to compare to). And, judged on its own merits, the Universal Audio LA-2A is a superb piece of audio equipment. Whether it's used on vocals, bass, kick drum, acoustic instruments, horns, or amplified instruments such as guitars, the LA-2A makes the instrument somehow sound like it's "on a record." (In reflection, this isn't that odd — most of the classic records probably used an LA-2A in the same applications discussed here.)

SPECS

MAX GAIN REDUCTION	40 dB
OUTPUT LEVEL	+10 dBm nominal, +16 dBm peaks
FREQUENCY RESPONSE	essentially flat from 30 Hz to 15 kHz
NOISE FLOOR	75 dB below +10 dB output level
DISTORTION	less than 0.35% at +10 and less than 0.75% at +16 dBm output
INPUT AND OUTPUT IMPEDANCE	600-ohm balanced.

Another characteristic of the LA-2A is that it has its own kind of magic. When tracking, the LA-2A was particularly effective on vocals, bass, and horns. In mix situations, I found myself using it in a few other places as well — on kick drum, snare, and steel guitar — anywhere I wanted the result to be full and smooth. I wouldn't characterize it as "transparent" since it does indeed change the sound put through it, but the gain reduction is so smooth that when setting it up and dialing what sounded like "a little" compression, I often found that I was hitting 5-7 dB of reduction.

Of course, nothing is truly perfect, not even the Universal LA-2A. I have a few minor complaints with the re-issue, and although all of these can be tracked back to the original unit, they're worth mentioning. The lack of a hardwire bypass switch is one. Also, the markings on the peak reduction and gain control knobs are fairly useless, since they go from 0–100 rather than showing calibrated gain and gain reduction. A more serious issue is that the input and output transformers are designed for 600-ohm balanced loads. I couldn't plug one of my favorite pieces of gear into the LA-2A without experiencing severe distortion due to an impedance mismatch between the two units. (This was solved by putting another piece of gear between the two, but it was still a disappointment.)

But, really, it all comes back to the sound, and, on that level, the Universal Audio LA-2A doesn't disappoint at all. At a retail price of $3,495, the unit isn't inexpensive, but if you're seriously committed to the highest quality recorded sound, it's an expense worth considering. ■

By Steve La Cerra • slacerra@uemedia.com

Sennheiser e865 Evolution Series Microphone

Sennheiser's newest hand-held mic provides top-of-the-line sound quality

Sennheiser's e865 is the most recent addition to the company's Evolution Series of microphones intended for onstage applications. A pre-polarized condenser microphone, the e865 occupies the top slot in the line of hand-held vocal mics. The e865 is designed to provide premium sound quality with a minimum of handling noise, and can withstand SPLs up to 150 dB. Along with a microphone stand holder, the e865 is furnished with a padded zipper-type case. The e865's body is a dark, non-reflective steel-blue with a black basket, which proved unobtrusive on stage.

I used the e865 at a series of shows in venues ranging from small clubs with a capacity of 500 to theaters accommodating 2,000 people. I immediately noticed that the e865's gain requirements were higher than many other condenser microphones I've encountered in live sound situations; gain settings were closer to those for dynamic mics. This is actually a plus for the e865 because it allows the mic to be used with a variety of consoles regardless of whether there's a pad on the mic input or not. (Some consoles need the pad to prevent condenser mics from clipping the input.)

If there's a word to describe the sound of the e865, it would have to be *fat*. The mic has a smooth low-frequency response, plus a rich proximity effect, which adds weight to male vocalists. It also helps prevent female vocalists from sounding shrill. Within the context of a live mix, the mic has presence without sounding peaky in the upper-midrange.

When I used the Sennheiser to mic a singer/acoustic guitarist, I paid close attention to the amount of guitar leaking into the e865. (I had no need or desire for the e865 to pick up the guitar.) Judging from the input meter, the sound of the guitar was down about 35 dB from the vocal — which I'd consider very good. When the vocalist opened up his pipes, the Sennheiser mic sounded unstrained. Male vocalists need to be careful about singing too close to the e865; within about 1.5 inches, the mic's proximity effect can become quite prominent, tipping the response toward overly low-end-heavy. Given this, the e865 is probably better off in the hands of singers who have a clue about proper mic technique.

Handling noise was minimal, making it a non-issue for the e865. In light of the mic's extended low end (40 Hz), and the aforementioned proximity effect, it'd probably be smart to use a low-frequency cut filter to make sure no subterranean junk is picked up by the mic. Off-axis rejection is excellent, and the e865's hypercardioid pattern is well controlled — so much so that the timbre of a voice will begin to change at around 45° off-axis. It's precisely this control that helps the e865 reject stage and monitor noise, making it effective on a loud stage. ■

▶SENNHEISER E865

MANUFACTURER: Sennheiser Electronic Corporation, One Enterprise Drive, Old Lyme, CT 06371. Tel: 860-434-9190. Web: www.sennheiserusa.com.

SUMMARY: Hand-held vocal condenser microphone for onstage use.

STRENGTHS: Smooth frequency response. Great rejection of off-axis sound. Wide dynamic range.

WEAKNESSES: Proximity effect can be a bit strong at very close distances.

PRICE: $399.

EQ FREE LIT. #: 107

SPECIFICATIONS

Frequency response	40 Hz to 20 kHz
Polar pattern	Supercardioid
Nominal impedance	200 ohms (@ 1 kHz, balanced)
Sensitivity	3 millivolts/Pa, equivalent to –70 dB referenced to 0 dB = 1 volt/µbar
Front-to-back	greater than 25 dB
Rejection at 1 kHz	at 120 degrees off-axis

ADINDEX

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By Craig Anderton • anderton@musicplayer.com

Korg OASYS
PCI DSP Synth, Effects, and Audio Card

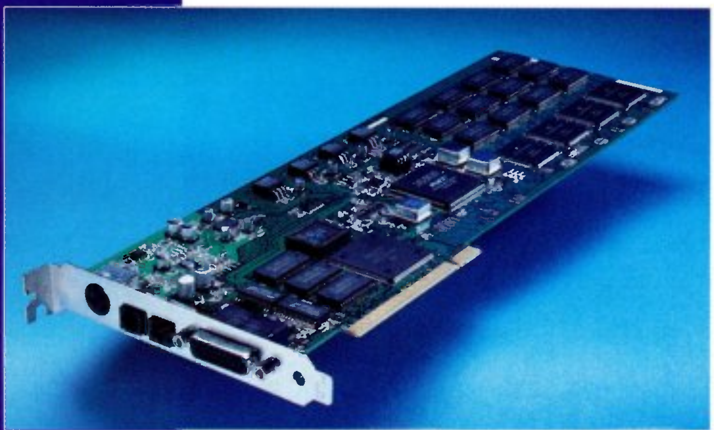
Is Korg s OASYS a mirage or the real thing?

The Korg OASYS PCI card combines a software synthesizer, effects processor, and audio interface on a standard PCI card. This is not a souped-up SoundBlaster, but uses a boatload of DSP to deliver some serious synthesis, while placing a minimal load on the host processor.

Originally Mac-only, OASYS PCI is now cross-platform, and has graduated to version 2.0. Despite the steep-sounding $2,200 list, a recent, significant price drop to dealers means a typical street price under $1,000 — a nice discount, considering what you get in return.

While host-based processing and synthesis has become viable due to faster processors, adding DSP on a separate card extends what a computer can do, while minimizing latency (with proper drivers). OASYS PCI uses five Motorola DSP chips to implement its selection of 44 synthesizer algorithms, 133 effects, and mixer. But don't expect to run too many of these simultaneously: Korg decided to "spend" the available DSP power on pure sound quality, rather than to play a zillion voices or set up huge Multi's. Those with basic studios might prefer quantity over quality, but for those seeking high-end timbres, going with OASYS PCI might be a much wiser move than, for example, buying a new hardware synthesizer.

Note that OASYS PCI doesn't do voice allocation other than setting a limit on the number of voices a patch can use. These reserved voices, even if they're not in use, aren't available to other patches. Fortunately, when used in a sequencing environment (as I did with Cubase 5.0), you can set OASYS PCI to be a MIDI track output, then route the OASYS audio stream back into Cubase (using ASIO) and record it as an audio track, thus freeing up the DSP for other tasks.



▶KORG OASYS

MANUFACTURER: Korg, 316 S. Service Rd., Melville, NY 11747. Tel: 516-333-9100. Web: www.korg.com/oasyspci.htm.

SUMMARY: Put a whole bunch of great sounding synthesis and effects algorithms inside your computer, along with an audio interface.

SYSTEM REQUIREMENTS: Mac: 200 MHz 604e (233 MHz G3 when used with other programs), System 8.5.1. Windows: 200 MHz PII (300 MHz PII when used with other programs), Win 95/98.

STRENGTHS: Great sound quality. Flexible audio interface. Extensive modulation options. All parameters can be automated. Imposes no significant load on host CPU. Cross-platform operation. Updatable. Includes multi-sample editor along with sample playback.

WEAKNESSES: Not enough DSP power for very complex Multi's or extensive polyphony. Unimaginative (though solid) editing software. Response lag when loading programs. No dynamic voice allocation.

VERSION REVIEWED: 2.0

PRICE: $2200

EQ FREE LIT. #: 108

Limited DSP power is also an issue when playing OASYS "live" (though few people drag their computer on stage along with a MIDI keyboard to play synthesizer), because of a dropout of a few seconds when you load new programs.

Note that the OASYS PCI includes a zero-latency monitor mixer so you can monitor sounds as they go into the card — not with whatever latency your computer added when the sound hit the output. This is very helpful if you have a slower computer.

The package consists of the card, two breakout cables, and a CD-ROM with driver and synth editing software. Installation, and updating from version 1.0.1, worked as expected. For some users, the card's 12.25-inch length might be a problem — make sure there's enough space in your computer. Although I couldn't test OASYS with a Mac, it's OMS- and FreeMIDI-compatible, and there's also

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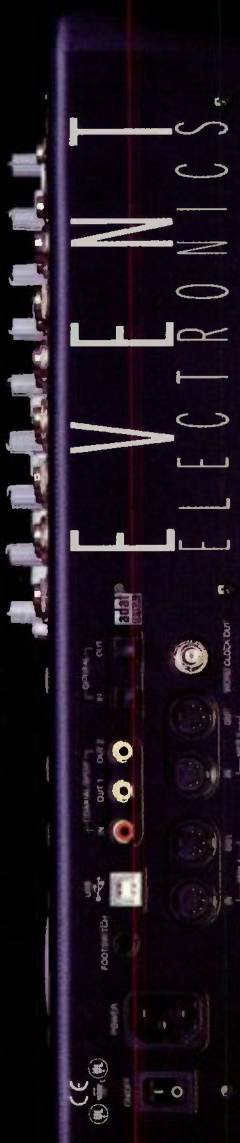

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EFFECTS

The effects categories (followed by the number of models) are:

- Amp & Speaker emulation [2]
- Chorus [14]
- Compressor & Gate [8]
- Delay [19]
- Distortion [4]
- Early Reflections [2]
- Ensemble [2]
- EQ [14]
- Filters [12]
- Flanger [8]
- Modulations [8]
- Organ FX [2]
- Other [7]
- Pan & Tremolo [9]
- Phaser [9]
- Pitch Shifter [5]
- Reverb [8]

drivers for "vintage" Macs that use serial ports for MIDI communication. (Given the OASYS card's modest system requirements, you may be able to use it with an older "boat anchor" computer that's been gathering dust.)

Documentation consists of a printed installation guide; the rest is on PDF files. Read them, or you'll miss out on some of the card's most important features.

The most basic element, the Patch, is a particular synthesis algorithm. Editable parameters are different for different algorithms. Patches don't do anything until you insert up to two of them into a Program. This sets parameters for each patch such as the number of voices, level, tuning, pitch bend, scale (yes, you can create your own microtunings and intonations), voice allocation, key priority, and split-type functions (key range and velocity triggering).

The next step is combining Programs into Multi's. While two splits/velocity-switched layers will be enough for some people, Multi's allow far more complex sounds, providing the DSP can handle it (there's a reason why most of the demo Multi's are limited to four Programs with a couple of effects). Multi's save essentially everything you see on the screen — mixer settings, controller assignments, patch settings, program assignments, etc.

The modulation options are outstanding, particularly as they allow for sophisticated automation. Each program and effects parameter is modulatable by up to two modulation sources (there are 36 available, including goodies such as MIDI tempo), with variable modulation amount. Upon assigning modulation, the parameter name turns red and underlines, making it easy to see what's being modulated in any Patch. Furthermore, all control panel knob twisting and slider moving can be recorded to, and played back from, sequencers as sys-ex information.

The OASYS 12-channel mixer is the system's traffic director. It speaks ASIO, so the inputs can come from OASYS synths, audio streams from ASIO-compatible digital audio programs, or the various card inputs (e.g., ADAT, analog, and/or S/PDIF). You can also do tricks such as send four ADAT tracks into the card, process them, then bounce the results into the remaining four ADAT tracks via the card's output. Channel inputs can be mono or stereo, so, technically, you can end up with more than 12 tracks coming into or exiting the mixer.

Each mixer channel holds one program. Program parameters include:

- Four "slots" for insert effects (Effects-only programs are available that can store all the insert effects for a channel. Thus, if you like to use certain effects on, say, vocals, you can call them up as a block.)
- Two post-insert mono sends with pre/post fader and mute buttons
- Output assignment
- Pan
- Solo
- Mute
- Fader
- MIDI channel assignment for the included program (32 channels)

Channel output choices are any stereo audio pair (analog, S/PDIF, or ADAT track pairs), but there's also a shortcut — you can specify any pair as a master output bus and just set the assignment to master.

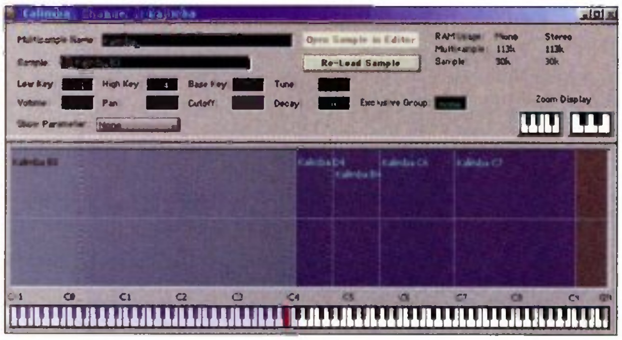


FIGURE 1: Editing the Folk Guitar Multi preset. The topmost window is the Program. Directly underneath it is the editor for the plucked string Patch, and, below it, the editor for the phaser effect. The rearmost window is the mixer; only two channels are actually being used.

Mixer automation is available for 11 of the mixer channel parameters (volume, pan, pan width, solo, mute, both send levels, both send mutes, and both pre/post buttons); these are fixed MIDI controllers, and each channel responds to data over its assigned MIDI channel. To squeeze out every ounce of processing power, you can turn off graphic updating of mix controls.

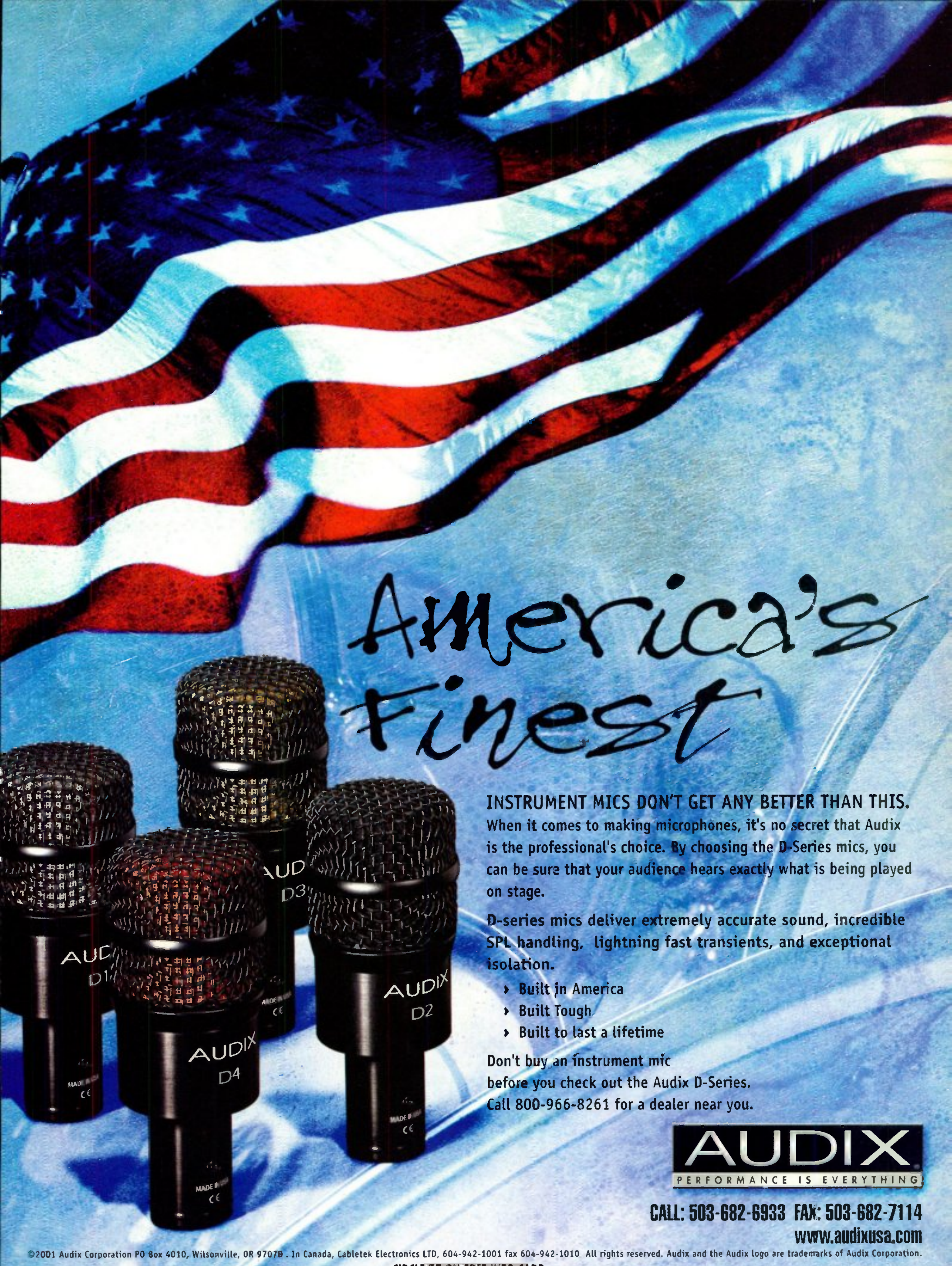
The OASYS PCI offers eight algorithm categories: analog synth emulation (nine models), electric piano, guitar/bass (four models), organ (two models), PCM (16 types), percussion (three models), variable phase modulation (four models, basically variations on two- and four-operator FM), and Waveguide (five models of wind instrument and voice). If you have a sense of déjà vu as you listen to the sounds, that's no surprise: variants of the OASYS engine have found their way into the Electribe Series, Z1, MOSS expansion boards for the Triton, and other Korg products.

IN AND OUT

The OASYS audio interface recalls the Korg 1212. Due to the small panel size, most connections are handled via supplied breakout cables. I/O includes:

- ADAT in
- ADAT out
- Stereo analog in (24-bit)
- Stereo analog out (24-bit)
- S/PDIF input (24-bit)
- S/PDIF output (24-bit)
- Word clock input (BNC)
- Word clock output (BNC)
- ADAT sync input
- ADAT sync thru

Because of the included ASIO drivers, ASIO-compatible programs can access the card's I/O, mixer, and effects processing.



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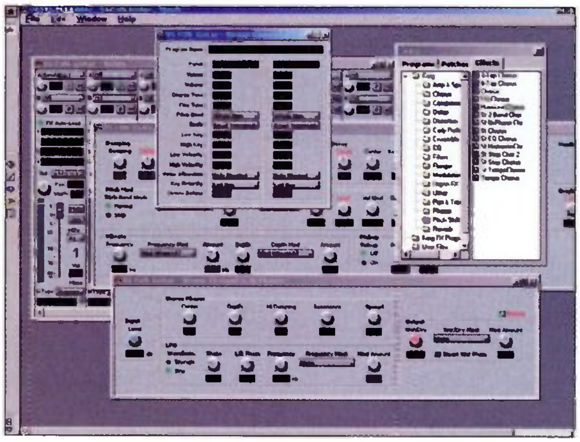

FIGURE 2: The multi-sample editing screen, where you can change tuning, decay, volume, sample key ranges, add/delete samples, etc. Being able to use samples and multi-samples is one of the main new features in version 2.0.

So how do they sound? If you're familiar with modern Korg products, you know that Korg has its act together with this type of technology. The bass is full without being muddy, the highs bright without being strident, and the midrange balances well with the frequency range extremes. The overall sound is clean, but not clinical; sonically, OASYS PCI is without reproach.

Korg had claimed an upgradable OASYS architecture, and in Version 2.0 they've added a sample playback engine as well as 16 new synthesis algorithms. For convenience, you can click a button to open up and edit samples in an external audio editor.

This function is pretty complete: manage samples in a multi-sample and use a two-pole filter and amp (both with ADSR) for shaping the overall multi-sample (this processing affects all samples in the multi-sample, not individual samples). The only significant limitation is that, due to the way that RAM is distributed among the DSP chips, samples for a single patch must fit in 6 MB of RAM. Given the 24 MB of onboard RAM, this yields enough room for four 6 MB patches.

The total of 133 effects (see sidebar) cover the usual bread-and-butter types, but there are also more "out there" effects such as ring modulation, random filtering, vocal formant effects, lo-fi modifiers, etc. This is serious stuff, and, given the low system latency, you can use your computer as a giant effects processor if desired. While the scope and quality of the effects is impressive, the ability to modulate every parameter is aces. This is a card with sounds and effects that can be shaped and twisted in real time, which (at least to me) is essential if you want to go for truly expressive soundmaking.

Incidentally, we've mentioned only channel insert effects so far, but effects can also go in the four send busses and output bus.

THE SOFTWARE

The synth patch editors are Spartan. Those accustomed to the graphic richness of products from Emagic, CreamWare, and Native Instruments may find the look of the OASYS dated. There's no graphic representation of envelopes, and parameter calibrations aren't in real-world values (just arbitrary numbers). However, there are some nice user interface touches, such as being able to drag the mouse left/right or up/down to change knob values — none of this "rotate the knob with the mouse" foolishness.

When you make significant edits, such as add or remove a Program or effect, the OASYS DSP recalculates the sound from scratch. This can take a few seconds, so, if you're doing a succession of changes, there's a "Disconnect" option that frees the DSP from enacting every change you make. When you select "Connect," all the changes you made are recalculated at once.

OASYS PCI is packed to the brim with synth and effects options. Including a quality audio interface that sweetens the deal, but sounds are what this baby is all about.

If you like the sound of contemporary Korg instruments, here's a way to get pretty much the same results — as well as a ton of new sounds — at a much lower cost. Or, if you're the kind of person who simply is up for something new, this is an excellent answer — especially once you've warped the audio with some of the effects. When version 1.0 came out at $2,200, the buzz was "nice card, but pricey." With the increased feature set and lower street price, OASYS PCI has remade itself into an extremely attractive synthesis tool. ■

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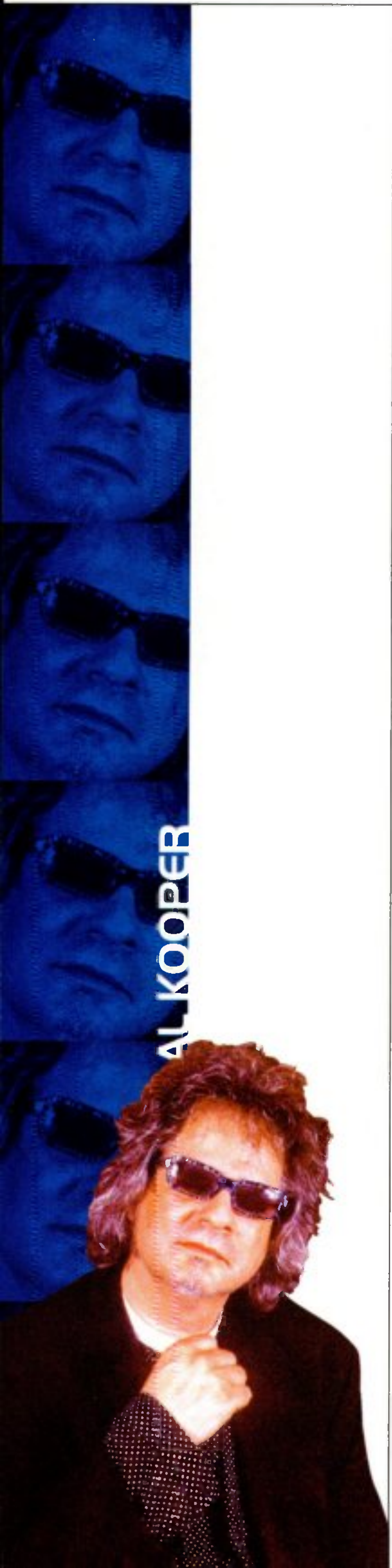
Dr. Al examines the format changes that surround us

Cautiously Waiting for 9.1



I remember when Quadrophonic Sound was foisted upon an unsuspecting public in the mid-'70s. It was available on vinyl discs and eight-track tapes. The new configuration was championed and patented by Columbia (Sony) Records.

I was an artist and producer at Columbia at the time. They gave me all the Quad software and an eight-track Quad system. I had a bachelor apartment stashed clandestinely in Hollywood with a four-poster bed; a speaker on each post faced inward. The girls' jaws used to drop in awe as the strains of "For The Love Of Money" by the O'Jays impressively swirled around the bed like a soul hurricane. I couldn't play, for instance, the *Super Session* album that they re-mixed for Quad 'cause it was pretty wimpy in that format. Four instruments and a vocal didn't provide any exotic echo or panning opportunities like the full band and background vocals on an O'Jays track did. It just sorta laid there by itself, like I did on many a night in that Hollywood apartment.

I have a theory why Quad failed so miserably. In the time-frame it was released:

Housewives killed it.

"You wanna put *how many* speakers in *my* living room?!"

Pure and simple: it's high concept versus actual reality. Today, it's another story entirely. Now, there are six to eight speakers to place in one's home theater (formerly living room), and harried homemakers, hardworking husbands, hardworking homemakers, and harried husbands can't wait to place myriad speakers in the walls and ceilings of their domestic *kinos*.

At present, it's more about having "chateau surround" than fretting about speaker placement. The difference between the '70s and the millennium is thus: people wanna have the most cutting edge technology NOW — screw everything else.

The problem for me is the third of the Three Unchanging Constant Truths voiced by the late Hoyt Axton:

1. A cowboy hat blows off in a stiff wind.
2. Sex rules the world.
3. *Things change.*

Today they change so suddenly that one must carefully research the purchase of any of the fore-mentioned luxuries. Spend $3,000 for the latest 5.1 amplifier and preamp, only to read about 7.1 technology before you can break the boxes open. While I sit here and rant, record companies are stockpiling catalog audio DVDs in 5.1 surround sound. Sony (formerly Columbia) Records has now proposed 7.1, a former cinema-sound-only system, for home theater and surround audio formats. While it looks like their last invention, Super Audio CDs, will go the way of their short-lived Betamax and Quadrophonic nightmares, they already have a new contender in the ring.

The problem with 5.1 and 7.1 surround audio DVDs is simple: Unlike most of the formats before it, the existing catalog is largely unsuitable for this format. Mono Elvis albums ain't gonna cut it in surround sound; Benny Goodman will stall; Little Richard will be six or eight times shriller than usual. CDs worked because they disposed of the clicks and pops of vinyl. Therefore, it didn't matter if they were mono. Surround is a spatial media, not one of specific clarity. Mono is doomed in surround. In a perfect world, if this bird is gonna fly, *current* projects need to be mixed in it. If Eminem mixes his next project in surround, his demographic will most certainly increase.

People snapped up any damn record when stereo first came out on vinyl. My first stereo album was Max Roach with the Boston Percussion Ensemble. It wasn't what I went in to buy, but it was the *only* stereo LP they had in the store. This situation is now about to be duplicated. Consumers are quickly purchasing DVD players, and movie disc software sections are rapidly dwarfing VHS aisles. Most lay consumers don't even know that surround audio DVDs exist, and that they're compatible with a system they may already own. As I type this, folks are in the studio mixing The Eagles' *Hotel California* in 5.1, anticipating the surround audio boom. They *should* be mixing the next Britney Spears CD for surround instead. ■

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The first in an ongoing series on dealing with the tough ones

Taming the Difficult Client

Back in the day, before I opened the doors at Gravity Music, I used to work in advertising agencies — writing copy, producing TV spots, occasionally getting the fun of producing a jingle project or video score. When the creatives would start grousing about this idiot client or that Philistine product manager, the account executives would cool our jets by saying, "If there weren't any jerks, we wouldn't have any clients."

Sure, it's a cynical joke, but it contains a grain of truth. When we put on our business hat, we're inviting everybody with a need for audio and a valid checking account to give us a try. As a result, we're going to end up working with a lot of different personality types. And a lot of them aren't going to match up well with our own. In fact, some of them are going to mix like water and pure sodium (which, if you remember your high school chemistry class, is a mighty volatile combination).

Since many of us were musicians before we were engineers, we may come into this with a rock 'n' roll expectation of the business. In a band or recording setting, we're used to working with people we enjoy, guys or gals we don't mind hanging out with, regular Joes with language, humor, and clothing similar to our own.

But, if we refuse to work with everyone we wouldn't invite to dinner, we're not going to make a living for long. So a Warrior comes to deal in a calm, friendly, and efficient manner with the most difficult of clients. Here's a primer on how to manage and cooperate with some of the worst offenders.

FRUSTRATING CLIENT #1: THE 500-POUND GORILLA

Input: A Gorilla wants things done exactly the way he tells you to do them, right or wrong (although they're *never* wrong). Impervious to suggestion — if you make one, he'll tell you immediately why it won't work. Make a decision, and he'll override it. Sneak a change by him, and he'll catch it. Most infuriating of all, he's often right.

Signal-to-noise: "Hey, who's signing the checks here?" "I liked it better before you did...whatever you just did."

Clipping factor: +18 dB. Hey, what are you, chopped liver? You didn't spend all this time and money becoming an audio expert to have someone tell you how to do your job, right?

Output: You've only got two choices. One is to prepare a perfect argument, with no holes and nothing you haven't thought of. (If a Gorilla can ask you even one question you can't answer immediately, they'll discount everything else you say.) It's possi-

> "IF WE REFUSE TO WORK WITH EVERYONE WE WOULDN'T INVITE TO DINNER, WE'RE NOT GOING TO MAKE A LIVING FOR LONG."

ble to sway a Gorilla, but, obviously, you should pick your battles carefully. There's no point in expending the mental energy and raising hackles if all you're arguing over is an EQ setting. The second choice, and usually the much easier way, is to simply hold your nose, get a mental image of the paycheck, and do it the way they want it.

FRUSTRATING CLIENT #2: TOM SWIFT

Input: He found out what your gee-whiz DAW can do, and he's just gassed by the "Amazing World of the Future" we live in. Now he wants to try everything, on every track, in every session.

WEBLINK

Have a question or comment for Jim Bordner? Email him at jim@gravitymusic.com.

Signal-to-noise: "Can we try that plug-in thing you used on the other track on this one? Let me hear it all five ways." "Boy, I love coming here…this is way more fun than the office."

Clipping factor: +6 dB. In a way, it's kind of charming. But, hey, the clock is ticking, and you've got better things to do than demo your software.

Output: You can usually curb this client's enthusiasm by simply reminding him of the expense he's incurring by trying everything. Just say, "You know, I think we're spending more time on this detail than it really deserves…the cost-effective thing to do is to make a decision and move on." You'll be amazed at how a Tom Swift personality will respond to this nudge.

FRUSTRATING CLIENT #3: THE MUTE

Input: Almost none. The problem with this guy is his reluctance to tell you what he's thinking. Non-committal, never critical, rarely approving, he listens and nods and lets you finish an entire album mix without so much as a peep. Then a week later he calls and says, "We used the wrong reverb on all the vocals — and the drums suck."

Signal-to-noise: "Sure." "Sounds fine." "Great." "You're the expert." One week later: "You know, I've been thinking…"

Clipping factor: +12 dB. Well, if you didn't like it, why didn't you just say so?

Output: He didn't say so because he's insecure. Look at all those blinking lights and switches and dials — there you are navigating it all like there's nothing to it. You must be smarter than him, so he'll just keep his mouth shut. That is, until he leaves and isn't confronted by The Great and Powerful Oz at his control panel. It's your job to put him at ease and draw him out during the session. When I start hearing those half-hearted affirmations — "Sounds fine," etc. — I like to josh them out of it. I'll say something like, "Hey, that doesn't sound very enthusiastic — nobody leaves this studio until they're deliriously happy!" This is usually enough to get them to relax and say, "Well, now that you mention it, don't you think it sounds a little muffled?"

There are others, of course, and we'll talk about them in a future column. But the most important thing to remember when dealing with any difficult client is that you're a warrior. And, as such, you live a life that the vast majority of humans can only vaguely imagine — a life in which people pay you to do something you'd do for free. So when the tension gets a little high, try to perform some internal kung fu. Remind yourself that your opponent is not your enemy. They, in fact, make your own mastery possible. (That's the grain of truth inside the cynical joke.) Honor them for this, and you'll find that even the toughest client is easier to handle.

Jim Bordner makes music, records audio, and studies the twisted anthropology of studio customers at Gravity Music.

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Living in an All Digital World

Get Connected

The most frequently asked questions on my Web forum are in regard to digital connectivity. For instance, "Mstibbs" writes, "I have many digital (lightpipe) optical devices in my studio and I want to be able to quickly route from one to another. Also, how do I make sure that the audio is distributed without clicks? Can you help?"

Yes, I can. First of all, simply stated, the digital optical connector and physical data path are the same regardless of what information is being sent. Light is light, digitally and optically speaking. Therefore, if you had a router that was capable of distributing and patching digital optical data, you could use it for ADAT (eight-channel) lightpipe, S/PDIF, and even multichannel (Dolby Digital, DTS, etc.). This means that any optical digital patchbay will work. Fostex and Z-Systems come to mind as two manufacturers of such devices; Hosa also recently announced a low-priced solution. The Z-Systems can be custom-configured with anywhere from eight to 32 I/O with a mix of AES, coaxial, or optical connections. I use a Z-Systems 16-in/16-out all-optical bay to route ADAT digital around my studio. It works great.

Now that we can route ADAT optical digitally, how do we get it to work? Usually it's not as simple as just connecting one digital device to others. The reason is that digital connectivity requires two critical elements in order for it to work properly:

1. **Data** — this is the information that stores the "content"

2. **Clock** — commonly referred to as word clock, this is the timing of the data

It's important to understand that these two elements can travel in the same cable. One example would be when you digitally copy a tape from one DAT machine to another. The first DAT machine should be set to "analog" input, which clocks the unit to its internal crystal for playback at the specified sample rate. The "slave" DAT machine is set to digital input; it looks for both the data stream (to be recorded) and the clock to which to synchronize. While the digital recording is in progress, the slave machine is running at the sample rate of the incoming digital data.

As soon as more than one device enters the signal path, however, things can get a bit tricky. Let's say that you want to digitally connect a Digidesign 888|24 Pro Tools interface and two Alesis ADAT M20's to a Yamaha 02R digital mixer. Although Pro Tools can output data in different digital formats, we're going to use AES since that's how the 888|24 sends and receives multichannel digital audio. The ADAT M20 can support multichannel AES, however, we'll connect the M20's via ADAT optical. Here's the setup:

- **M20 #1 ADAT optical — 02R tape channels 1–8 (ADAT optical card installed) slot 1**
- **M20 #2 ADAT optical — 02R tape channels 9–16 (ADAT optical card installed) slot 2**
- **PT 888|24 AES — 02R mic/line channels 1–8 (AES card installed) slot 3**

As previously stated, we know that one device needs to be designated as the master clock source in order for everything to work. Either one of the connected devices could serve as a clock master, or you could use a dedicated clock generator, typically known as a "word clock generator." Two popular units are the Rosendahl NanoSync and Aardvark AardSync II. (Either device makes an excellent choice for a master word clock generator.)

Here's an example of using a word clock (W/C) generator as the clock master:

- **W/C generator output — W/C input of M20 #1**
- **ADAT sync from M20 #1 — ADAT sync of M20 #2 (ADAT sync carries word clock from one ADAT to another)**
- **W/C generator SuperClock output — W/C input of 888|24 (requires the use of either a 256x W/C generator or a Universal Slave Driver)**
- **W/C generator output — W/C input of 02R console**

All devices would need to be set to "external W/C input." When this setting is selected, each device resolves its clock to the incoming W/C signal. Therefore it receives the sample rate via this connection (remember that the sample rate is the speed of the digital audio). The data travels through the AES or optical digital connection — however, the clock is derived from the dedicated word clock signal. Let's now add a TC Electronic M5000 with one AES card into the equation. We'll connect it to another card in the 02R using AES digital. When you try to connect the W/C generator to the M5000, you'll notice that there's no W/C input on it! Now what? In this case, the M5000 can receive clock from the incoming AES data because all AES signals contain word clock information (in addition to the actual data). Therefore, the M5000 would be set to resolve to digital input and it would work fine going digitally in and out of the 02R.

In the event that we didn't have a W/C generator, we could designate one of the devices in the system as the W/C master. Here's an example configuration using the 02R as the W/C master:

- **02R set to internal clock**
- **M20's set to digital input for clock source, connected to 02R via lightpipe**
- **PT 888I24 set to AES "digital" input for sync source, connected to 02R via AES**

The 02R resolves to its internal crystal while the M20 and 888I24 resolve to their respective channel 1/2 digital input. This is important to understand because if the channel 1/2 input of either device were to be disconnected, then neither the M20 nor the 888I24 would be able to properly sync to "digital" input. You would hear clicks and pops in the audio from the remaining ins and outs. Note that some devices allow you to choose which digital input should be used to resolve the clock; most require the use of input 1/2.

The key point of these examples is to help you develop an understanding of digital connectivity. To recap, W/C controls the timing (speed) of the digital signal. All signals must be tightly synchronized in order to avoid digital artifacts. If an M20 is connected to an 888I24 and sample 42,101 on the M20 arrives at a different time than sample 42,101 on the 888I24, the result will be poor sound quality and — even worse — clicks and pops in the audio. As soon as more than two devices are connected, you need to "centralize" your word clock source. The ultimate solution in today's all-digital studio is to use a master word clock generator. Plus, most W/C generators can resolve to video (more on that next column), so they work well as a clock source in mixed analog and digital environments.

If you have specific questions pertaining to your digital setup or just want to talk shop, be sure to log onto my Studio Tech Forum at www.eqmag.com. Thanks to all of you who participate there on a regular basis. Because of you, the forum has become a great resource for the music technologist. ■

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More on the brave new world of online music distribution rights

Hello Songwriters, Are You Listening? (Part III)

Welcome to Round Three of our dig into the current state of online music publishing. The last two months spent blundering about Web sites of two of the Big Three music performance collection societies (ASCAP and BMI) have taught us much about Frustration Management. This month, let's see what SESAC, the last of the Big Three has to offer us online. Then we'll pay a visit to the National Music Publisher's Association (NMPA) and its affiliate, The Harry Fox Agency (HFA).

SESAC

SESAC was originally called the "Society of European Stage Authors & Composers," but moved to Nashville and changed their name to the acronym. Like ASCAP and BMI, SESAC's (www.sesac.com) heavy-handed rhetoric lets us know right away that online music publishing should be left to the professionals. "The system required to compute compensation is based on many factors, including music trade publication chart activity, broadcast logs, computer database information, and state-of-the-art monitoring." God knows we little musicians could never handle it.

Calling itself the "most technologically adept" of the Big Three societies, the SESAC Web site turns out to be a woeful mix of "under construction" messages and suggestions on where to go in the site for information, but no direct links to these recommended areas. Ever heard the term "hypertext links," gang? There's no way to become a member online. By explanation, the Web site offers that: "SESAC has a selective process." That means you've got to apply. They don't want just anybody. But that's probably a good idea. Quality, not quantity, right? For songwriters who want someone else to take care of everything (so you don't need to worry your pretty little head about big, confusing numbers), this organization might be an appropriate choice.

If you're interested in being involved in managing your career, however, this company presents a few conundrums. If, as they say, their Web site represents "SESAC's continuing strategy to lead performing rights in technological development," then maybe SESAC shouldn't use white text on a black background. Really folks, your site-design and navigation is awful.

Helpfully, though, a link on the front page offers: "Simple, Comprehensive Online Licensing" and "Complete Online Works Registration." Clicking on either of these two links opens a message that links us straight back to the front page. Again, no direct link to the area they're talking about. So we start again at the top and click on "Licensing." This gives us several choices. Which one? Let's try "General." Yes, buried among the 30-odd examples of general licensing ("Country Club," "Cruise Ship," "Dance Studio," "Festival," etc.) is "Internet." We open the General License and try signing up. We see "Internet/New License." We click through to "complete an online license." This downloads a PDF to our desktop to fill out that gets sent automatically back to the SESAC site. An option to print and snail-mail the form is also available. After completing the form, we'll apparently be emailed in a "few days" confirming our license. Minimum annual fee is $75, but it's impossible to know just how real fees will be calculated.

THE HARRY FOX AGENCY (HFA) AND THE NATIONAL MUSIC PUBLISHERS' ASSOCIATION (NMPA, WWW.NMPA.ORG)

HFA isn't about licensing music for Internet use by end-users. It is, however, relevant to the question: "Who do I get permission from to record a cover of a song and put it online along with the lyrics?" HFA lets you file paperwork to pay standard rates for pressing a CD containing a cover song or songs. Here's how to go about it:

Go to the NMPA Web site and click on "HFA." From here, click "Licensing." From this page we choose "Make a Recording" from eight choices. This takes us to a page called "Mechanical Licensing." Getting permission to record and sell a cover of a song is about acquiring the above-mentioned mechanical license. HFA is the clearinghouse for just about every song ever recorded in America.

WEBLINK

www.music-law.com — A good resource with a practical Q&A series. http://lcweb.loc.gov/copyright — For all things government-related to the subject

If you have a credit card and are making less than 2,500 albums, you can use the HFA online registration service at: http://songfile.snap.com/nonpro_search.html. This will thread you through the HFA song file database (to make sure you and they agree on the song you're using). Let's search their database of "over 2 million songs!" Results are quick. There's the song we chose, along with a host of others that have similar titles. Next to the song result is a small disc that, when toggled, displays the seven available options: "Lyrics," "Listen," "CDs," "Sheet Music," "License," Info," and "Other Recordings." We click on "CDs." Surprise! Here we are at Amazon.com. Wonder what Amazon pays them for this link? No album match for our song, but we're not looking to buy anyway. Back to HFA and a click on "Sheet Music." This fires us over to www.jwpepper.com, the "Pepper Music Network," and a "no title found" message. But we don't want sheet music either, we want permission. Actually, we've always been proponents of the adage: "It's easier to get forgiveness than permission." We continue our search. "Other Recordings" leads us to just that. After fooling around with the song file database, it's back to the form. We're required to answer questions about the number of recordings we'll make, what countries they'll be manufactured and distributed in, and what type of organization we represent.

After answering these questions, we get asked for credit card info and, apparently, that's when we'll find out the fee. Sometime after this transaction is completed we'll apparently receive an actual paper license in the mail.

If you don't have a credit card, start at www.nmpa.org/hfa/mechanical.html#forms and download two documents: Mechanical License Request Form and Application for New Licensing Account. These documents must then be printed on "plain white paper" with "only typewritten or neatly printed, block-letter-style requests accepted." Then you must mail or fax the forms to HFA.

A separate license (and an additional fee) is required for reprinting lyrics of the cover song with your CD. HFA doesn't handle this sort of licensing, so they note that: "For these rights, you must contact the publisher(s) directly." It's great for artists to use a direct connection to arrange this kind of license, but we shudder at the thought of getting lost in a mega-bureaucracy negotiating a license for Led Zeppelin's "The Crunge" with Atlantic Records. Where's that confounded bridge, indeed!

The good news is the mechanical licenses aren't very expensive for low-run CD pressings, maybe the price of a tank or two of gas per song. The question is: How much of this money makes it into the pocket of the songwriter or rights holder? If only it were as straightforward as a biologist injecting a radioactive tag into the bloodstream to trace its path. In the meantime, the best we can offer is a link to HFA's "Royalty Payments Inquiry Form," which you presumably need to be a member in order to fill out: www.nmpa.org/forms/royalty_inq1.cfm. ■

ROGER NICHOLS

▶ continued from page 146

comes in. As soon as we do anything to the data, such as change the level, then the extra bits are filled with the results of the math. When it's time to go back to 24 bits, then the extra bits are truncated off and thrown away. This is where dithering comes in.

Dithering does not add any bits to the 24-bit (or 20-bit or 16-bit) audio signal. It adds low-level noise to the signal so that the LSB can detect a signal that's below its threshold. I won't go into the details now, but suffice it to say that a dithered 1-bit signal sounds better than a 2-bit converter, but not as good as a 3-bit converter.

Okay, now let's talk about 32-bit, 48-bit, floating-point, or what's-the-point. The errors caused by math only apply to the smallest bits. If we do the math using 48-bit internal word lengths, the remainder from the division of a 24-bit number will be below the 24-bit level, somewhere between bits 25 and 48. This means that there will be no round-off errors in our 24-bit signal. We can DSP or change levels all we want. There will only be one penalty at the end when the signal is changed back to 24 bits. If there's no dither or noise shaping, then the maximum penalty will be 1.25 dB at the –144 dB level.

Noise floors don't add up digitally as they do in analog, but the level can rise. If you add a full-level digital signal to another full-level digital signal, then the results will be one more bit than you have room for. This will put you 6 dB over the clipping limit. Oops. But you don't have to worry about the digital noise floor getting high by adding 32 or 48 tracks of stuff. You can mix together 1,000 tracks with nothing on them and have –144-dB noise level on the output.

Now, what you do have to worry about is the noise from each source. The noise from the synth, the room noise on the vocal, and the mic hiss on the horns…all of these noises are 100 times louder than the "noise floor" of the 24-bit converter.

Earlier we touched on the fact that the converter couldn't detect anything below the level of the LSB. Actually, the level of concern is about half the level of the LSB.

Have you ever noticed the way that your air conditioner thermostat works? The temperature is set at 75 degrees. When the temperature gets up to 76 degrees, the central air comes on and cools off the room until the temperature gets to 74 degrees and then shuts off. The central air stays off until the temperature gets back up to 76 degrees and the cycle repeats. This is called hysteresis.

The LSB in the converter works the same way. The level of the audio gets up to about 3/4 of the value of the bit, and the bit turns on. Even though the analog signal level is falling from this level, at every sample time (44,100 times-per-second) the LSB is turned on until the level of the signal reaches about 1/4 of the bit value. When you play back the digital audio signal, the LSB remains on, even though the signal level was falling. After a few samples, the bit turns off and stays off. This isn't exactly a reciprocal of the input signal. This is quantization error. If you use 24 bits instead of 16 bits, the quantization error is 256 times less.

Audio-One is hosting the second in a series of Roger Nichols Master Classes on mixing technique on May 19th in Miami. Check www.audio-one.com for more details.

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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 II- compressor, limiter, gate and expander/gate; Mod Delay - short, slap, medium, and long delays with modulation capabilities for chorus or flange effects and dither. AudioSuite plug-ins include Time Compression/Expansion, Pitch Shift, Normalize, Reverse.

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

MOTU AUDIO Hard Disk Recording Systems

MOTU · MOTU PCI-324 AUDIO

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

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

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

2408 mkII FEATURES-
- 7 banks of 8 channel I/O: 1 bank of analog, 3 banks of ADAT optical, 3 banks of Tascam TDIF, plus stereo S/PDIF
- Custom VLSI chip for amazing I/O capabilities
- Format conversion between ADAT and DA-88
- 8x 24-bit 1/4" balanced analog I/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

EzBus 24-Bit 96kHz Computer Audio Interface/ Control Surface & Digital Mixer

EVENT ELECTRONICS

The EZ Bus combines a PC and Mac compatible USB Computer Interface, Digital Mixer and Control Surface. The USB interface supports two channels of 24-bit/48kHz digital audio recording and playback and 32 channels of MIDI I/O. The Digital Mixer offers a with a wide range of high-quality 24-bit/96kHz analog and digital I/O, DSP and automation capabilities. While the control surface provides virtual mixing, routing and transport control for your favorite computer based DAW or MIDI Machine Control compatible multitrackvia dedicated faders, buttons and rotary encoders.

Computer Audio Recording Interface
- Two channel 24-bit/48kHz recording/playback via USB on Macintosh or PC computers
- 14 TRS balanced line inputs, 2 mic preamps with +48V phantom power and two instrument/line inputs)
- 24-bit/96kHz A/D/A conversion
- Optical I/O for ADAT Lightpipe and S/PDIF (automatically senses source type)
- Coaxial S/PDIF in and 2 assignable coaxial S/PDIF outs
- Analog outputs include Stereo Main Mix and Alt outs, 2 Aux Sends and a Headphone output
- Two independent MIDI ins and outs
- Word clock output
- Programmable footswitch jack
- Hot-swappable plug and play setup

Software Control Surface
- Fully programmable faders, switches, encoder knob, and transport controls for sending MIDI Controller data via USB and the MIDI outs
- Includes presets for Cubase 5, Logic Audio, Nuendo, and Cakewalk
- Transport controls with jog shuttle wheel for controlling sequencers as well as any MMC device
- Set and recall up to eight Locate points

Stand-Alone Digital Mixer
- 8 primary audio channels from any analog or digital input
- Mute and Solo on each primary audio channel
- 4-band EQ (sweepable high and low shelving plus two fully parametric bands) and Programmable dynamics (compressor/expander/gate) on each primary audio channel and Main Mix
- One stereo and two mono Returns
- 4 Sends per channel, assignable pre- or post-fader
- 4 virtual audio channels (EZbus Returns)
- 4 multi-input analog channels; accept up to three independent source signals per channel
- EZbus Audio Routing Matrix provides easy to use, flexible input/output routing capabilites
- Save and recall 32 internal snapshots of all mix and system parameters
- AudioAlert function notifies user of errors, such as overloading an analog input, digital dropouts, or clipping due to excessive EQ.
- ADAT Lightpipe provides 8 direct outputs for primary mixer channels— ideal for use as a front end for an ADAT or Lightpipe-equipped audio card
- Asynchronous sample-rate support via S/PDIF with high quality sample-rate conversion

TASCAM®

US-428 USB Digital Audio Workstation/ Controller

The US-428 is a 24-bit USB-based audio controller co-designed by TASCAM and Frontier Design Group. The control surface includes plenty of faders, transports and other dedicated controls compatible with the most-used functions in today's DAW applications. The US-428 supports a total of four channels of audio in and two outs simultaneously. The interface plugs right into a USB equipped PC or Mac computer— no opening your computer and no sound card to install. Musicians taking the leap from Portastudios to computer-based DAW programs will feel right at home with the tactile control surface.

FEATURES-
- PC and Mac compatible
- Works with most major DAW programs
- 24 bit D/A and A/D converters
- Bundled with Steinberg's Cubasis VST sequencing software for Windows (MacOS version shipping soon)

I/O-
- Total of four channels of audio in (analog or S/PDIF) and two out simultaneously via USB
- Two XLR mic inputs, two balanced 1/4" TRS inputs, two unbalanced 1/4" inputs (switchable to Hi-Z)
- S/PDIF digital I/O
- Two independent MIDI I/O (32 channels)

Controls-
- Unlimited banks of eight faders
- Transport, mute/solo and locate keys
- An EQ module supports control of up to four bands of fully parametric EQ
- Four aux sends and a panpot
- Can be customized to control everything from virtual synths to MIDI lighting panels

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JOHN PATERNO

▶ continued from page 40

going with Pete each morning, although half the time the approach changes dramatically once we've heard the song.

Which engineers do you respect and admire?

Tchad Blake and Bob Clearmountain are at the top of my list. Bob is one of the only guys out there who gets an amazing mix, and it still sounds like the record you put your life into for however many weeks, days, and hours. Joe Barresi gets great sounds and mixes, plus he gets some of the best guitar sounds I've ever heard. And then there's Geoff Emerick, Norman Smith, Andy Johns, and Mike Stone, who I'm still amazed by — what they did as engineers and mixers.

What was your most ridiculous experience in a recording studio?

Once, when I was assisting, a pretty well-known English heavy metal bass player was doing an overdub. He was playing along, and then all of the sudden he started playing all of these wrong chords. The producer stopped the tape and said, "Hey, the chords are F, Bb, and C." The bass player growled back, "I tried those and I didn't like them, so I'm trying some others!" ■

PRO TOOLS 5.1

▶ continued from page 103

mixing. Multichannel formats supported include LCR, Quad, LCRS, 5.0, 5.1, 6.0, 6.1, 7.0, and 7.1. In addition to multichannel tracks, plug-ins, and bouncing, there's also multichannel bussing, aux, and master channels, as well as support for surround panning.

The surround panner (see screenshot — also an example of Pro Tools' new Output window) provides two modes; "X/Y," which allows you to drag the pan position around the grid in "joystick" fashion; and "3-Knob," which uses the three Position knobs to set the pan position. The Divergence controls set the "width" of the panned signal in relation to adjacent speakers. I found Pro Tools' approach to divergence display to be somewhat hard to grasp; I prefer the "dish" or arc approach used by Kind of Loud on their Smart Pan Pro.

"Center %" and "LFE" control the amount of the signal that's fed to the center speaker and subwoofer channel, respectively. All multichannel pan controls can be automated. Unfortunately, you can't turn individual speakers on and off in the panner.

5. Stereo and Multichannel Tracks: With version 5.1, Pro Tools gains support for stereo and multichannel tracks. You can convert multichannel tracks to multiple mono tracks, but you can't just unlink the component tracks in a multichannel track. So if you want to edit just one track, say, cut out a click in just right channel of a stereo pair, you have to convert the multichannel track to multiple mono tracks. (Version 5.1 also now supports multichannel regions in the Audio Regions list.)

6. Auto-voicing (TDM systems only): In the past, you've had to assign each disk track in Pro Tools to a "voice" — somewhat analogous to a note of polyphony in a synthesizer. In 5.1, a new function called "Auto-Voice" can handle this for you. You can still "hard-assign" a voice to a track if need be.

7. Multiple output assignments: It's now possible for all channel outputs and sends to route to multiple busses and outputs. Need to feed a stereo mix to a DAT machine, cassette deck, and a CD recorder simultaneously? No problem. Want to mult an aux send to feed two different effects boxes? A snap.

OTHER NEW FEATURES

Pro Tools 5.1 also includes many new "convenience" and functionality enhancements such as Auto-Save, which can automatically save a copy of the current session at user-specified time intervals. You can also specify how many incremental backups are kept. Pro Tools now offers 16 levels of undo/redo, there's WAV file format support on the Mac, you can use a key command to toggle between waveform and volume automation views in the Edit window, and much more.

RUN, DON'T WALK

Pro Tools 5.1 is an excellent update to an already-powerful program. While we've covered many of the outstanding items in the new version, rest assured that there's a ton of stuff that we simply didn't have room to discuss. The combination of surround mixing, new capabilities, and function-enhancing features in Version 5.1 is right on the mark. A definite must-have for Pro Tools users. ■

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Hailed as a "Holy Grail of recording" by Recording Magazine, Auto-Tune is used daily by thousands of audio professionals around the world. Whether to save studio and editing time, ease the frustration of endless retakes, or save that once-in-a-lifetime performance, Auto-Tune has become the professional pitch correction tool of choice. Now Antares has introduced Auto-Tune 3. Preserving the great sound quality and ease of use of Auto-Tune, Version 3 adds significant new features and a snazzy new look. As a result of Antares research into the unique characteristics of various types of audio signals, Auto-Tune 3 offers a selection of optimized, "Source Specific" processing algorithms for the most common types of pitch-corrected audio material, resulting in even faster and more accurate pitch detection and correction. Choices include Soprano Voice, Alto/Tenor Voice, Low Male Voice, Instrument, and Bass Instrument. Other key new features include phase-coherent pitch correction of stereo tracks, and Bass Mode, which lets you easily apply pitch correction to fretless bass lines and other low bass range instruments. Auto-Tune 3 also lets you set target pitches in real-time via MIDI from a keyboard or sequencer track. For harmonically complex material, the "Make Scale From MIDI" lets you simply play the line from a MIDI keyboard or sequencer and then Auto-Tune 3 constructs a custom scale containing only those notes! Auto-Tune 3 also now supports high sample rates like 88.2kHz and 96kHz. Get Auto-Tune 3 today!

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MOTU Dream Studio 2

- **MOTU 2408mkII™** — 24-channel expandable PCI audio interface
- **Digital Performer™ 3.0** — award-winning workstation software with MIDI sequencing
- **SAC-2K™** — expandable, touch-sensitive, 8-fader automated control surface for Digital Performer
- **DigiMax™** — 8-channel mic pre-amplifier with 24-bit optical connection to the 2408mkII/computer
- **Bias Peak 2.5 VST™** — award-winning waveform editing software

PRECISION TOUCH-SENSITIVE CONTROL — SAC-2K™

The Radikal Technologies SAC-2K sets a new standard for hands-on control of Digital Performer. The SAC-2K's custom plug-in for Digital Performer gives you easy, one-touch access to every element of the recording process in Digital Performer with responsive, touch-sensitive automated controls. Within minutes, you'll achieve a whole new level of interaction and creativity that you never thought possible. Fader groups, mix automation, plug-in automation for up to 12 parameters at a time, window sets, transport control with jog/shuttle, I/O routing and assignments, solos, mutes, track-arming... it's all just one touch away. The SAC-2K is your all-access ticket to the world of Digital Performer-based recording, editing, mixing, processing and mastering.

RADIKAL TECHNOLOGIES

PRISTINE MIC PRE-AMPLIFICATION — DigiMax™

Why is the PreSonus DigiMax perfect for your MOTU rig? Because it's the purest path to digital. DigiMax combines 8 channels of award winning 24-bit mic pre-amplification with our unique simultaneous RMS/peak detection limiting and EQ enhancement, giving you maximum gain before clipping while maintaining the musical transparency of a compressor. The result? Fast, natural and versatile limiting on every channel. And DigiMax connects all 8 channels optically to your MOTU system in pristine, 24-bit digital glory.

PreSonus [audio electronics]

ADVANCED WAVEFORM EDITING AND MASTERING — Peak 2.5 VST™

BIAS Peak 2.5 VST is the ultimate editing and mastering companion for Digital Performer! Peak gives you lightning fast, nondestructive waveform editing with support for audio files up to 32 bits and 10 MHz, including 24-bit/96kHz files. Unlimited Undo/Redo with independent edit histories for each audio document gives you the freedom to work creatively. Select an audio region in Digital Performer, choose the "Use External Waveform Editor" command, and instantly switch into Peak! Peak's sophisticated options for on-the-fly marker, region and loop creation are simply unparalleled. Advanced looping tools include Loop Tuner™, Loop Surfer™, Loop It™ and Guess Tempo™. Process thousands of files—or just a few—using Peak's batch processor. Peak directly supports the 2408mkII and all other MOTU audio interfaces and includes Toast™ CD burning software for making your own redbook audio CDs directly from Peak's powerful playlists. Or create web or multimedia content using Peak's support for Shockwave, RealAudio, MP3 and more.

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MOTU Dream Studio 3

- **MOTU 1296™** — 12-channel expandable 24-bit / 96KHz PCI audio interface
- **Digital Performer™ 3.0** — award-winning workstation software with MIDI sequencing
- **HUI™** — automated, touch-sensitive control surface for Digital Performer
- **PowerCore™** — DSP accelerator for TC Works MAS plug-ins (and other advanced 3rd-party MAS plug-ins)
- **MAS STOR™** — reliable, high-performance SCSI storage and backup for MOTU Audio System based studios

FULLY AUTOMATED WORK SURFACE — HUI™

The Human User Interface (HUI) from Mackie is so tightly integrated with Digital Performer, it's like placing your hands on Digital Performer itself. Sculpt your mix with HUI's silky smooth motorized faders. Tweak effects parameters with firm, yet responsive V-Pot rotary encoders. Instantly locate to any position and track in your mix. You can even call up plug-ins on-screen directly from HUI. Keypad and transport controls let you locate Digital Performer's main counter instantly, just like the familiar keypad on your computer keyboard. HUI is an advanced hardware workstation console, complete with built-in monitoring and the user-friendly ergonomics that Mackie mixers are known for. Boost your productivity through direct hands-on control.

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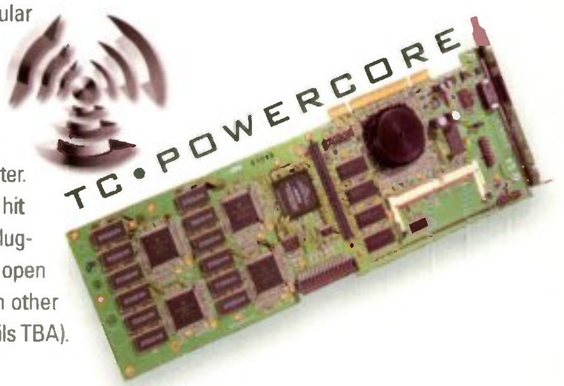
DSP TURBO™ FOR MAS•POWERCORE PLUG-INS — TC•PowerCore™

TC•PowerCore is a major breakthrough for Digital Performer's real-time MAS plug-in environment because it provides DSP-turbocharged plug-in processing. At last, the renowned TC TOOLS/96 studio-quality FX package (included), with TC MEGAVERB, TC Chorus/DELAY and TC EQ^{sat}, can be at your fingertips in Digital Performer, plus other TC | Works plug-ins such as TC MasterX (sold separately). These powerful TC plug-ins appear in DP's mixing board, just like regular native plug-ins, but they run on four powerful 56K DSP chips on the TC•PowerCore PCI card. It's like adding four G4 processors (equal to 2.8 gigahertz of extra processing power!) to your computer. Run 12 studio-quality TC plug-ins with no hit on your CPU power, and run other native plug-ins at the same time! TC•PowerCore is an open platform, so it will also run plug-ins from other respected 3rd party developers, too (details TBA).

TC | WORKS



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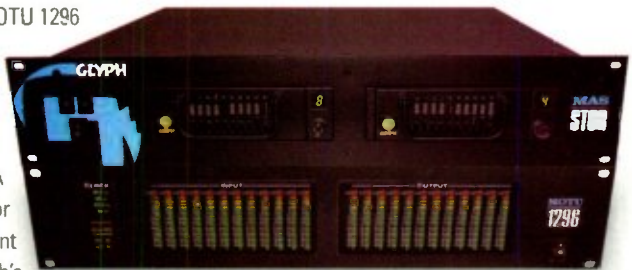

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Audio — one bit at a time

Let's Resolve This

Today we're going to talk about resolution, dither, noise floor, and cumulative errors in digital audio processing. Let's first assume that you're recording 24-bit, and all the internal processing is at 24 bits. Whenever you do anything to the signal, such as change level, or limit, or EQ, or anything, you add about 1.25 dB of noise to the signal down at the level of the smallest bit — this is assuming that the process adds no noise of its own. The math remainders cause the errors (noise) due to multiplications and divisions. We're dealing with binary math because digital audio is binary.

Now let's talk about noise floor. This is actually a misnomer in the digital world — it's more of a sensitivity limit. With 24 bits, nothing below –144 dB will be detected or recorded. Noise floor typically means the noise level when no signal is present. Based on this, the noise floor will be the noise of the resistor on the output, because there will be nothing...no signal present at all if all of the bits are zero. If you get your input level up above –144 dB, you'll start to record some information. With no noise shaping or dithering, such low-level recordings will be very distorted, but you can easily tell what's being recorded. At that level, you're basically recording with a one-bit converter.

Everyone focuses on this low noise floor and dithering to improve the quality of the signal at low levels, but that's not all there is to it. On a computer, when you zoom in to look at a waveform, you see a single smooth line running up and down as it crosses the screen. Change the vertical and horizontal magnification until you can see small vertical moves in the waveform. These are the high-frequency overtones of the signal. Now lower the vertical resolution one step at a time until the little wiggle goes away.

What you've done is change the magnification until the screen can no longer display the small movement of the waveform because of the size of the pixels on the monitor. This same thing happens when you don't have enough vertical resolution in an A/D converter. The little wiggle is there, but the size of the LSB (Least Significant Bit) is too small to capture it. This happens to the little wiggles in the loud parts, too.

Dithering adds about 1.5 bits to the apparent resolution of the recording. With dithering, you could detect and record a signal that was 9 dB lower than the –144 dB "noise floor."

There has been some misunderstanding of how 16-bit or 20-bit data is converted to 24-bit data when importing data from a lower resolution recorder. All digital devices start numbering the bits at the top, or Most Significant Bit. The MSB in PCM (Pulse Code Modulation) audio is the Sign bit. This bit determines whether the sample is negative (below the zero line) or positive. The next 15 bits (for 16-bit audio) set the level of the sample. Audio coding is binary, but let's just say that the level of this sample is 32,760 on a scale of one to 32,768. Just as the "3" in our number has the most worth (30,000), then the "2" (2,000), then the "7" (700), the "6" (60), and the "8" (8), the bits of our audio sample line up the same way, each one having more worth than the one to its right. In decimal math, each digit is worth ten times the one on its right. In binary arithmetic, each bit is worth two times the bit to its right.

Think of it like a 5-bit cash drawer and a 7-bit cash drawer. The 5-bit drawer has a place for $10,000 bills, $1,000 bills, $100 bills, $10 bills, and $1 bills. Our 7-bit drawer has the same setup plus a bin for dimes and a bin for pennies. If we move $11,111 from the 5-bit drawer to the 7-bit drawer, there is a place for every denomination. The dime bin and the penny bin remain empty. If we move $11,111.11 from the 7-bit drawer to the 5-bit drawer, we have a bin for everything except the dime and penny. Since there's no place for them, we must throw them away.

The exact same thing happens when we go back and forth between 16-bit and 24-bit machines. If we go from 16 bits to 24 bits, there's a bin for each bit and the bottom eight bits remain empty. If we go from 24 bits to 16 bits, the top 16 bits fit just fine, but the bottom eight get thrown away because there's no place to put them. They're truncated.

To carry this further, moving from 16-, 20-, or 24-bit into a 48-bit digital mixer works the same way. All of the extra bins are left empty (set to zero) when data

▶ continued on page 130

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