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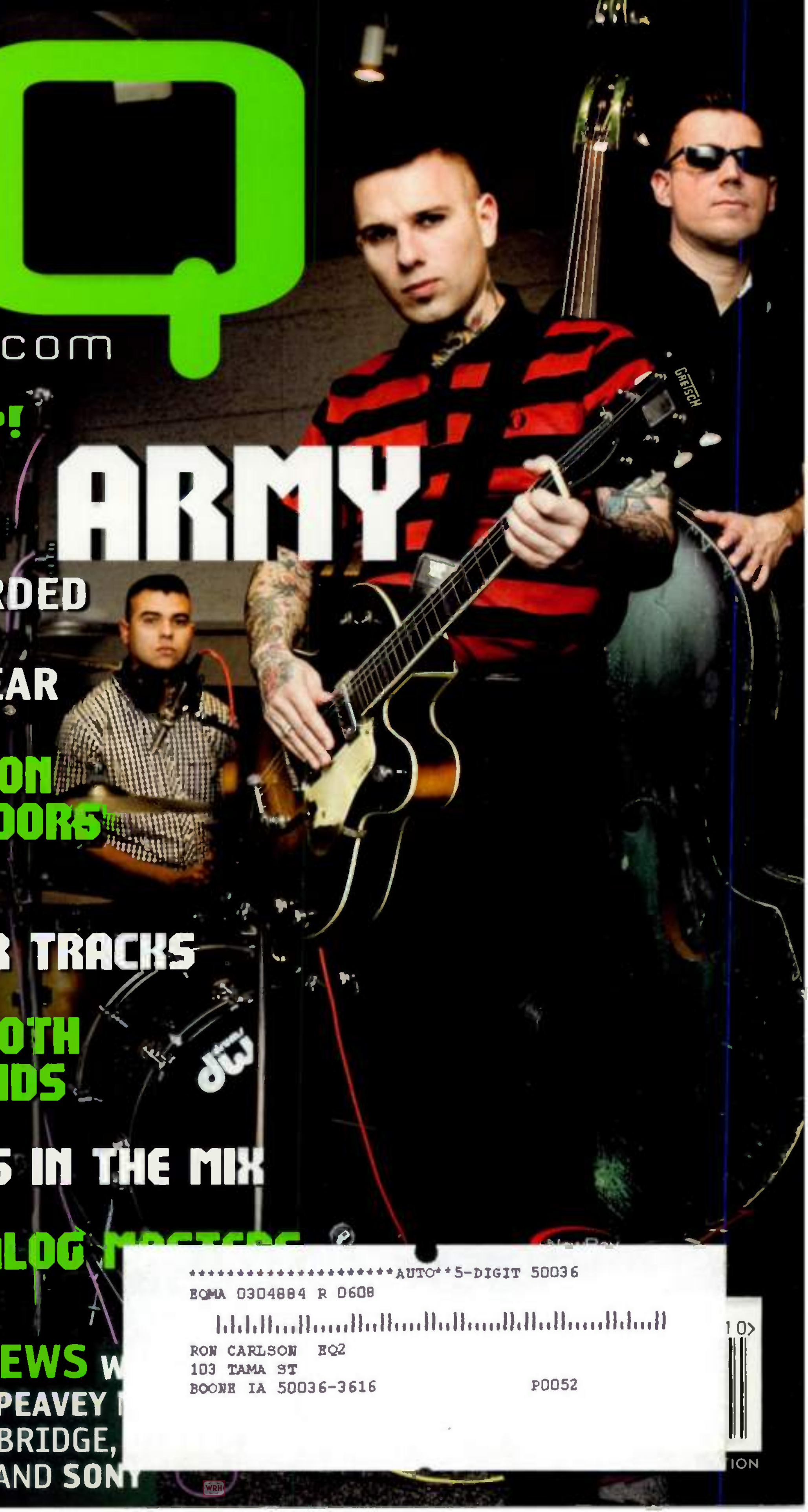

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FEATURES

20 BONGOS
Archaeology time! How team Bongo restored, remastered, and revitalized the Bongo's classic *Drums Along the Hudson*.

24 THE DOORS
Bruce Botnick reveals the techniques and philosophy behind recording, mixing, and remixing the Doors discography.

30 TIGER ARMY
Nick 13 and gang tell how they recorded the psychobilly album of the year, *Music From Regions Beyond*.

44 HOW TO CREATE AN EFFICIENT DAW SETUP
Get the upper hand on pops, heat, hum, latency, FireWire issues, and other gremlins with these practical tips.

48 ARCHIVING YOUR MUSICAL HISTORY
Don't let your musical legacy fade away. Here's how to preserve your past for the future.

30

OCTOBER 07

CONTENTS

PUNCH IN

8 Uberzone
10 Watching the Wheels Turn
12 Chris and Thomas

TECHNIQUES

52 GUITAR TRAX
- Andy Timmons Details the Studio Strategy for *Resolution*
- 6 Ways to Conquer Red Light Fever

54 BASS MANAGEMENT
- Get Your Signal Path Together
- 5 Quick and Easy Bass Sounds
- 3 Tips for Fighting a Bass-Lite Mix

56 KEY ISSUES
- How to Produce Bigger and Badder Synth Bass Sounds

58 DRUMHEADS
- 8 Steps to a Perfect Kick Sound

62 VOCAL CORDS
- Prepping a Vocal for the Mix

64 MIX BUS
- Beating Phase Issues in your DAW
- Does "Absolute Polarity" Matter?

66 CHEAT SHEET
- Windows XP Tweaks for Audio

REVIEWS

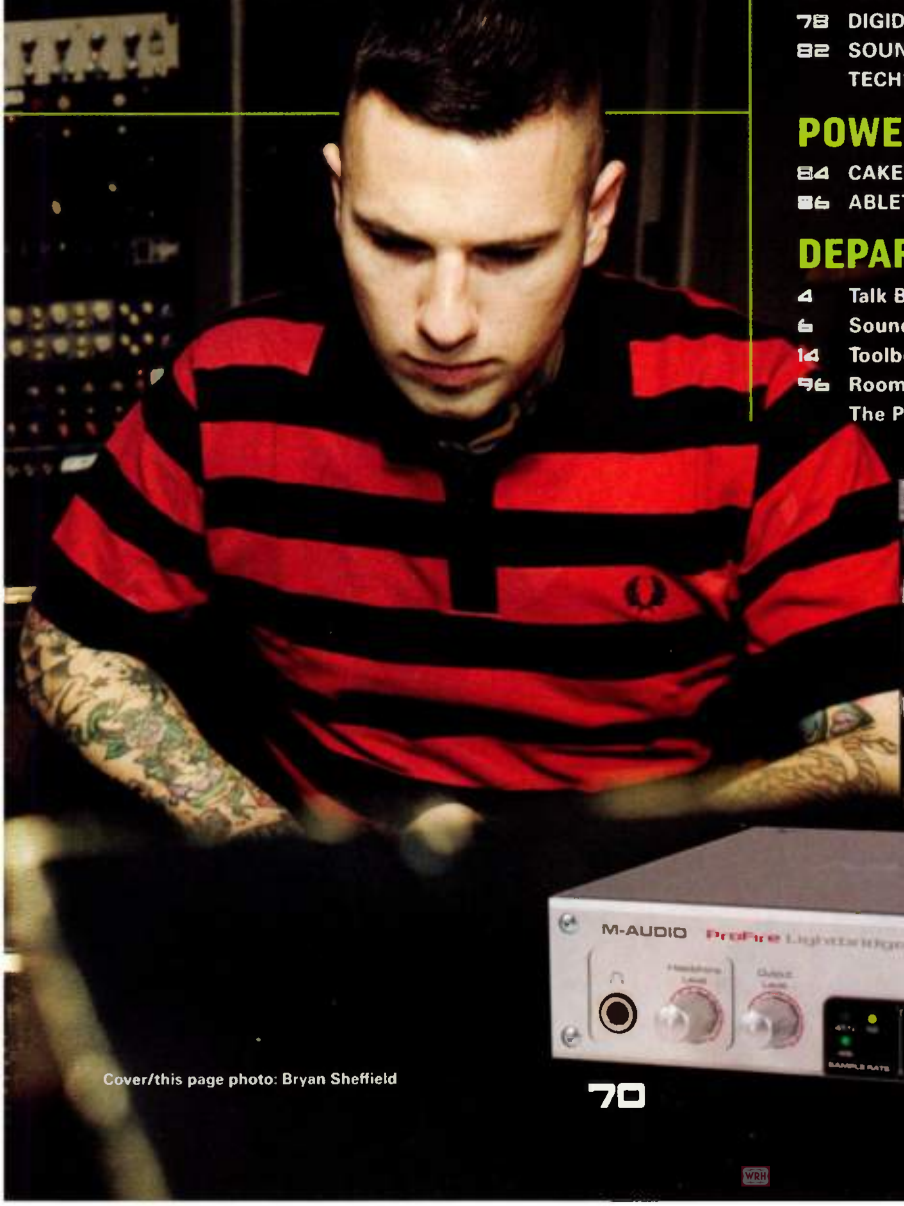
68 TC ELECTRONIC C400XL
70 M-AUDIO PROFIRE LIGHTBRIDGE
72 PEAVEY M1 + M2 STUDIO SERIES MICS
74 WAVES IR-1
78 DIGIDESIGN A.I.R. VELVET
82 SOUNDS: BIG FISH AUDIO ELECTRO-HOUSE, TECHNOCORE, AND SONY FLAMMABLE

POWER APP ALLEY

84 CAKEWALK SONAR 6
86 ABLETON LIVE 6

DEPARTMENTS

4 Talk Box
6 Sounding Board
14 Toolbox
96 Room with a VU: The Pleasure Machine, Brooklyn, NY



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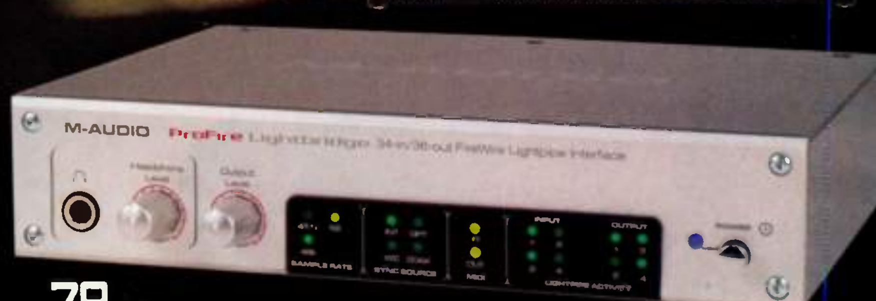

70

Cover/this page photo: Bryan Sheffield

Talk Box

PETTY ANNOYANCES

The high-tech world has given us wonderful goodies, but it has also brought a host of petty annoyances. I don't mean big things—like mastering engineers who ruin perfectly good recordings, or expensive hardware that becomes useless when a new card slot "standard" emerges — but those little things that drive you up the wall. Such as . . .

It's great that companies post up-to-date audio interface drivers on the Web, but can't they post up-to-date installation instructions, too — like whether you need to de-install existing drivers, or just install over them?

Computer brands are not a religious issue. At least the Apple/Windows factions aren't passing out leaflets at airports . . . well, at least not yet!

Why are software boxes that hold nothing more than a CD/DVD and a quick-start page so ludicrously huge? With companies fighting for shelf space and profitability, it makes sense to create a slimmer, less expensive, more environmentally-friendly package. Even the record industry figured this out when they nuked the CD "longbox."

Please, no more speaker reviews where someone states with awe that they heard sounds in their favorite recordings "they never heard before." All speakers have somewhat different frequency responses, so what would be truly unusual is if music playing through two different speakers sounded the same.

Three words for companies that print authorization codes in manuals: Use serif typefaces. A user shouldn't have to be psychic to tell the difference between capital "I," lower case "l," and "1."

Listen up, some of you sound library developers: Making a loop REX or Acid-compatible involves more than just bringing the file into ReCycle or Acid/Sonar, respectively, crossing your fingers, and hitting "save." Yes, it takes time to edit files properly for stretching, but that's what you're being paid for. And while I appreciate creativity, a sound named "CreepingBugFizz" doesn't tell me much. Include a few clues as to genre, style, or overall vibe.

On Windows, some software programs automatically create a system restore point upon installation. Why don't all of them? And is there any reason why ASIO latencies aren't reported correctly to a program from audio interfaces? Line 6 and some other companies' interfaces do, so we know it's possible.

Of course, I'm just scratching the surface here. To tell the world about the petty annoyances that tick you off (such as "editors who waste space in a magazine by bitching and moaning"), go to the Letters to the Editor forum at **www.eqmag.com** . . . and sound off!

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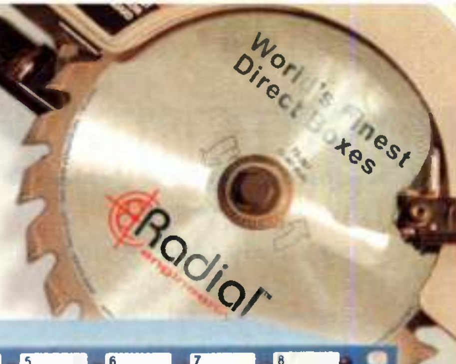

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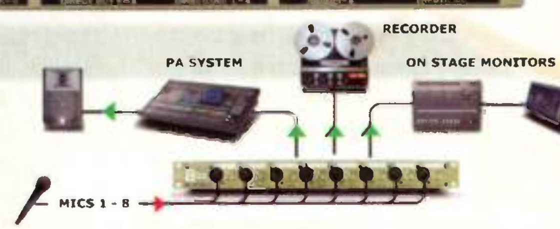

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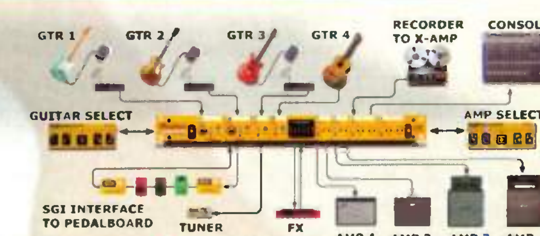

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SOUNDING BOARD

WHO DO YOU LOVE? TOWNSHEND!

I enjoyed the Pete Townshend and Jack Endino interviews [8/07]. Good stuff! Although being quite new to the recording world some of the gear talk was a bit over my head. Gives me something to work on!

Eric (via the forums)

Thank you so much for the wonderful Townshend interview. That was the best article I've ever read in your magazine.

Andy Cho (via email)

Five Words: Pete. Townshend. Best. Interview. Ever.

Dee (via email)

MOUTHING OFF

I enjoyed reading about Andrew Chaikin [Kid Beyond] in Punch In [8/07]. While I wasn't familiar with his work, I've now gotten an appreciation for what he's doing, and learning about his musical origins offered an interesting perspective as well.

As a mouth musician who also performs live, I can tell you it's not an easy act to do well. Not all mouth musicians would agree that the mic is the instrument though. Personally, I believe that my body is the source instrument. Still, your readers would probably be amazed at what equipment mouth musicians are using these days. Personally, I avoid MIDI, but use a wide range of effects from Moogs to Electro-Harmonix pedals to shape my sound . . . all starting with what my mouth can generate as a source controller.

Keep up keeping us informed (as usual),

Francesco Bonifazi aka "The Jazz Whistler" (via email)

SO SORRY

Chuck Zwicky called Doug Sax, who said that his last name is in fact spelled "S-a-x," not "S-a-c-h-s" as listed in the "Tracking Townshend" feature [8/07]. He also said that, indeed, he is the one who built the speakers in question.

Can you run a correction?

J.J. Blair (via email)

Craig Anderton responds:

I've worked on projects that were mastered by Doug Sax, so I'm well aware of his prodigious skills as a mastering engineer. So you might wonder, how do mistakes like this creep in, anyway? Well, I thought this might be a good opportunity to explain a bit about the proofing process.

First, it's usually good practice not to assume anything when proofing. Pete Townshend specified the last name as "Sachs," as did J.J. Blair. However, that seemed suspicious to me, so I Googled "Tannoy + Manley + Sachs" (the context in which the name was mentioned in the article).

The first entry that came up was from Outside Organization, whose clients include the Who, David Bowie, Prince, Chicane, P Diddy, Hilary Duff, Usher, Ronnie Wood, and many others — definitely not some fan site! On the page it said "TubeTech analogue summing and three band compression were used to prepare Pro Tools files for Mastering to CD via Sadie at Metropolis. Manley 400 Watt Triode Tube Monoblock amplifiers and Manley-Doug Sachs 10" Tannoy speakers were used for critical monitoring along with various Dynaudio and Genelec speakers."

There are many other references to Doug Sachs on the net in a capacity of mastering engineer, and there are references to Sax as well.

I also Googled "Tannoy + Manley + Sax" and got hits from that too. However, I figured Pete Townshend was on more intimate terms with Doug than I, and if Townshend, the author of the piece, and the artist's publicity company all say that the name is spelled "Sachs," then regardless of how I thought it should be spelled, I retained that spelling in the article. Ooops!

SO VERY, VERY SORRY

First off, I would like to thank you for your continued support and exposure through your feature "Gear Me Up Scotty" [8/07]. I would like to clarify a couple of things though. I have never produced Tori Amos. I was honored to assist on mixes for her live album, which was mixed at my studio [Ante Up Audio: Cleveland, Ohio]. While I was thrilled to be a part of that project, I don't want to take credit for work I didn't do.

One more thing: The artist I recently finished the record for is Will *Bowen*, not Will Bowman. Phone interviews are fun, but I think I mumble a lot. . . .

If Tori happens to call me to play produce her next record, I'll be sure to contact you immediately!

Thanks,

Michael Seifert (via email)

Matt Harper responds:

If Tori does call you, please tell her Matt from EQ says, "I can change."

SERIOUSLY, WE ARE WAY SORRY

I was glad to read "An Evening with Joe Chiccarelli" [8/07] and pleased to be in the photo with the Master Producer, James [Mercer] and Marty [Crandall].

In the text, however, you misspelled the name of *Bob Stark*. Bob is a great engineer with great big ears who's done fantastic work with Heatmiser, Everclear, Night Noise, Pink Martini, Dandy Warhols, and many more. He was my mentor in my intern days back in '92 and I owe him a great deal of gratitude.

Keep up the great work,

Sean Flora (via the forums)

Got something to say? Questions, comments, concerns? Head on over to www.eqmag.com and drop us a line on our Letters to the Editor forum, send us an email at eqeditor@musicplayer.com or snail mail c/o* EQ *magazine, 1111 Bayhill Dr., Suite 125, San Bruno, CA 94066 for inclusion on the Sounding Board.

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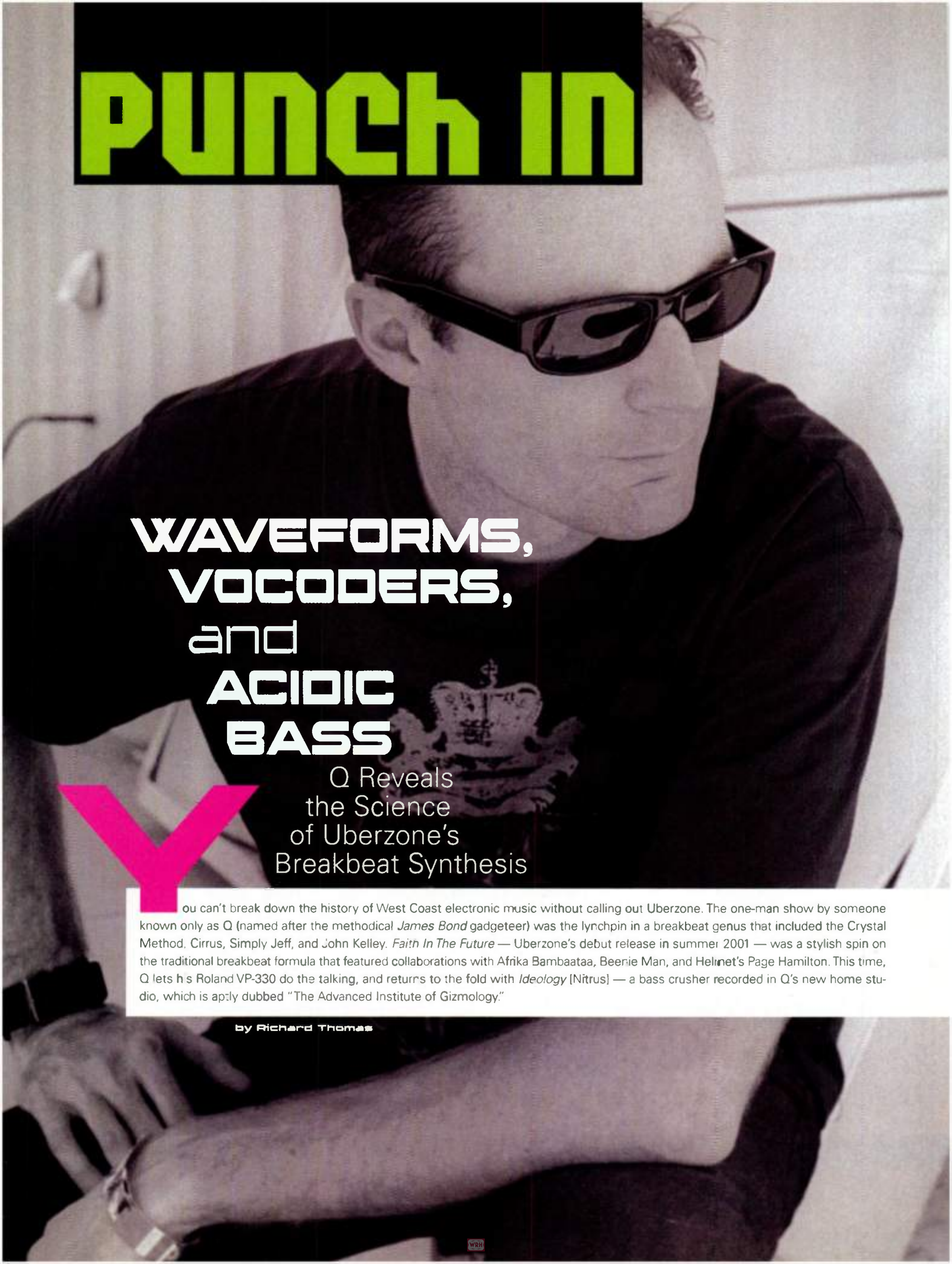


PUNCH IN

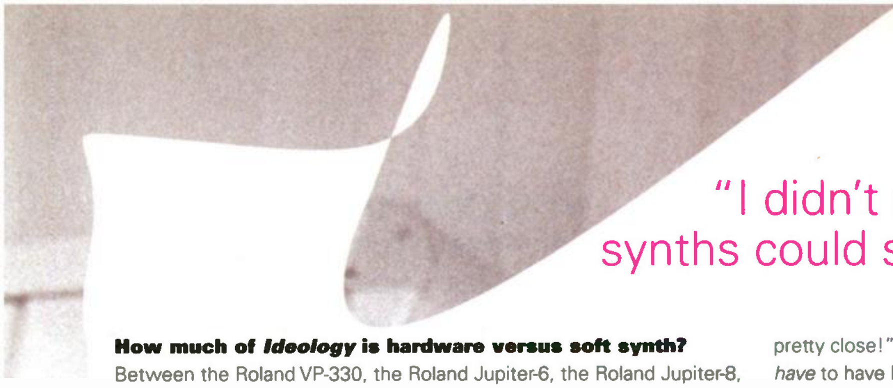
WAVEFORMS, VOCODERS, and ACIDIC BASS

Q Reveals the Science of Uberzone's Breakbeat Synthesis

You can't break down the history of West Coast electronic music without calling out Uberzone. The one-man show by someone known only as Q (named after the methodical *James Bond* gadgeteer) was the lynchpin in a breakbeat genus that included the Crystal Method, Cirrus, Simply Jeff, and John Kelley. *Faith In The Future* — Uberzone's debut release in summer 2001 — was a stylish spin on the traditional breakbeat formula that featured collaborations with Afrika Bambaataa, Beenie Man, and Helmet's Page Hamilton. This time, Q lets his Roland VP-330 do the talking, and returns to the fold with *Ideology* [Nitrus] — a bass crusher recorded in Q's new home studio, which is aptly dubbed "The Advanced Institute of Gizmology."

by Richard Thomas

WRH



"I didn't realize how amazing my synths could sound until I heard them through a decent DI."

How much of *Ideology* is hardware versus soft synth?
Between the Roland VP-330, the Roland Jupiter-6, the Roland Jupiter-8, and the Sequential Circuits Pro-One, I'd say a good 75 percent of the audio content on the record is analog. I also use soft synths such as M-Audio's GForce impOSCar and GForce Minimonsta:Melohman. But it's more about what I do to the sounds when I get them into the computer. I love the purity and programmability of the hardware, but when I get the basic tracks into the computer, I'm able to slice and dice them into micro and macro edits. Timing has always been a huge concern for me, and working with audio gives me the absolute ability to put things within the space and time I'm looking for. Then, of course, there's Native Instruments Reaktor, and all the other new-school plug-in effects that I can use to modulate parts. I almost always bounce the effects to a separate track so I can leave the original sound as pristine as possible, and also have the ability to further process the effects with EQ and compression as needed.

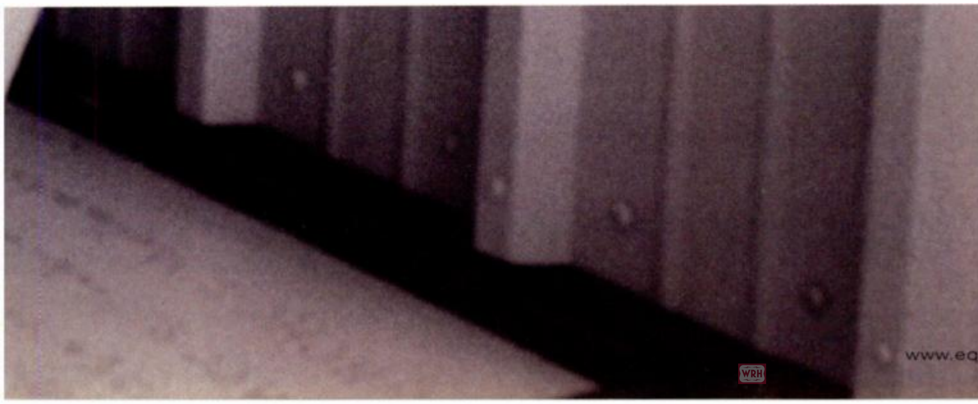
The VP-330 is such an important part of the record, but I think a lot of people underestimate the art of vocoding.
If you just plug a mic into the back of it and talk, you might not be able to get the sound you want. You definitely have to dress up the signal a bit, and the Shure SM70 is one of my new secret weapons. I'll run that mic through an Alan Smart C1 [*hardware stereo compressor*], and compress it a bit to get the tone a little more present before I go into the VP-330. Then, I boost the top end on the vocoder to accentuate the T, P, and S sounds. Once it gets into the computer, I use a high-pass filter to get the rumble off the bottom — the VP tends to add a huge bump in the 300Hz to 600Hz range. I notch 350Hz or 400Hz out because that frequency range can compromise intelligibility. That tweak can make the vocoder tracks sound a bit thin, however, so I run the entire vocoder subgroup — which consists of a dry, mono vocoder track; a stereo ensemble of the same part, using either the ensemble chorus on the VP-330 or the Universal Audio UAD-1's Roland CE-1 plug-in; and usually a stereo non-linear, or small room reverb — through the UAD-1's Neve 33609 compressor. The 33609 is more than a compressor — it has its own sound. It adds a little bite, and puts some body back into the vocoder group. It also helps marry everything together while pulling in the reverb and chorus effects to give the subgroup some nice dimension.

A lot of people rely on happy accidents for analog synthesis.
That's not me. When I was a kid, I'd sit in the bedroom with the Roland Juno-60 or the Pro-One and say, "I want *this* sound," and then I'd think of how I was going to get it. "I need a cutoff of about 30 percent, I want an envelope modulating the filter cutoff with a fast attack and a little decay, and then a sawtooth waveform, because it needs to be harmonically rich." I'd move different things, and then play the note and go, "Oh, that's pretty close!" I can totally appreciate the experimentation part, and you *have* to have happy accidents. But that craftsmanship thing, where you have an intent and a focus — *that's* what I love. I love the science and procedure that goes into electronic music.

The Jupiter-6 is such an integral part of that classic breakbeat sound — especially in terms of how it generates bass.
One of the really nice things about the Jupiter-6 that other people might say *isn't* so nice is that the oscillators start falling apart when you get into the lower registers. Certain oscillators behave completely different down there. For example, the Minimoog sounds immense when played in the lower octaves. The sound is full, fat, and tight. But the Curtis chips used for the oscillators and filter in the Jupiter-6 differ from the discrete circuit paths of the Minimoog. The Jupiter's oscillators have a sort of nasal quality, and you can really hear the pronounced cycling of the waveform in the lower octaves, and the bottom end is a bit wimpy in an interesting way. It's very acidy, but it has a nice bite — which allows the Jupiter-6's lower registers to poke through a dense mix. I think it's fair to say that part of my sound is that I use this perceived "weakness" to my advantage. I like to play the Jupiter-6 in the very low octaves into the computer, and then apply a 6db or 12db per octave highpass filter — usually around 250Hz–500Hz. Sometimes, I'll augment that waveform with an identical part played with a sine wave to give the part a stable, booming bottom end while still maintaining the acidic quality of the Jupiter-6's oscillators.

Given your history with vintage equipment, what signal chain have you found works best for your synths?
I have a pair of Avalon U5 preamps in my studio, and I didn't realize how amazing my synths could sound until I heard them through a decent direct box. Previously, I was relying on cheap op amps in crappy submixers and IC-based mixing desks for gain. I started bypassing the boards and submixers completely, and tracking directly into the A/D converter, using the U5s for gain and line balancing. It really did breathe new life into the basic tracks. It was an "oh, sh*t" moment — sort of like, "So *this* is why the big boys sound so good — class A signal paths with no signal loss in the low end, and pleasant non-linear distortion." Turntables and bass guitar also sound amazing through the U5s. I know the whole lo-fi thing is in vogue right now, but if you start hi-fi, you can always go lo-fi. The opposite isn't true.

Would you consider yourself "old school"?
Basically, I'm from whatever school that works. I have a little plaque over there that says "Innovate over Imitate." So I might go back and grab something from the past, or produce something I've never heard before. For me, it's always about trying to have an identity, and integrating my personality into the music. EQ



EQ tv.
See exclusive footage of Uberzone at www.eqmag.tv.

What Does the Universal/iTunes Standoff Mean for Indie Artists?

WATCHING THE WHEELS TURN

by Moses Avalon

Imagine you're a diehard fan of Amy Grant, Blink-182, Def Leppard, Elton John, or Ja Rule, and you go to iTunes to buy their new releases. But they're not there, and neither is the entire Universal Music catalog — and that's a huge chunk of music considering that Universal is the world's largest record company and all.

Apple is hoping that consumers will say to themselves, "I'll just buy from another artist that I can easily find here on iTunes. There are thousands to choose from." But Universal is banking on music buyers saying, "No. I'm loyal to my favorite artists. I'll click over to Rhapsody, Napster, or Yahoo, and get what I want — not what Apple wants to sell me."

So music creators and music buyers are bearing witness to an old-school standoff that started the day Universal boldly decided against renewing its bi-annual contract with Apple. Instead, Universal put Apple on a sort of month-to-month lease for the rights to sell the label's artists. Publicly, Apple CEO Steve Jobs seems very casual about this situation. But the truth is that this could be serious trouble for Apple unless, of course, music fans are more interested in the buying experience than they are in the actual music. This is literally the $200,000,000 question — as that's the annual revenue the music industry is gambling with if Universal's move inspires other big labels to follow suit. If push comes to shove, could the labels really afford to boycott the largest retailer of online music?

Probably not. So what is this really all about?

Many music insiders are attributing the Universal maneuver as a posturing strike to force Apple into allowing the label to dictate different pricing for different artists. Although album prices on iTunes vary, track costs are typically held to 99 cents whether the artist is on a major label or an independent. Universal would probably rather mimic the disparity in pricing seen at many brick-and-mortar retailers, where their releases would be more expensive than the typical indie album. Jobs has stated that he feels staggered pricing would alienate music consumers.

We must also throw in the possibility that the majors hate iTunes. Sure, Apple brings in a lot of revenue for the labels, but they would prefer people to have to pay subscription rates — such as those on Yahoo services — to purchase music online.

So, you might ask, "Do the major labels want *my* potential buyers to pay *them* for the right to buy *my* music?"

Kind of.

Universal wants all artists to note that they stand to make more with the subscription model than if the consumer simply buys a 128kbps file for a one-time fee of 99 cents that can play on only selected (networked) devices.

But what if your potential market doesn't want to pay that fee? Does that mean that you lose a viable means of online distribution simply because it doesn't keep major labels as fat as they like to be?

Jobs has said that he will *not* make iPods compatible for use with subscription services just to appease the majors. But that's not because Apple is a champion for independent musicians. The truth is it's because iPods would need to incorporate a program that generates royalties based off "play events" — a royalty generated from each play on an mp3 player or computer, payable to the label/artist and writer. By shunning digital-rights-management software, the iPod remains one of the only music players in the United States that silently, but proactively, endorses people experiencing music "shared" from unlicensed sites. Jobs has even gone so far as to state that labels — big and small — should remove the "lock" that disallows copies of music files to be shared. (Read: Labels should let everyone steal their inventory so Jobs cans sell more iPods and iPhones.)

If they could get away with it, every major label would probably team up and push until Jobs decided to play ball. But aside from the lost revenue, if all the majors were to conspire (in the legal sense of the word) to boycott iTunes, Apple would have a billion-dollar claim against the RIAA. The far smarter move is for a mega-major like Universal to break ranks, and pull out for personal reasons. There are no anti-trust issues, and yet it would seriously handicap iTunes, because Universal controls about one-third of the product in the marketplace.

So — will the majors win the upcoming standoff? And what's at stake here for independent artists?

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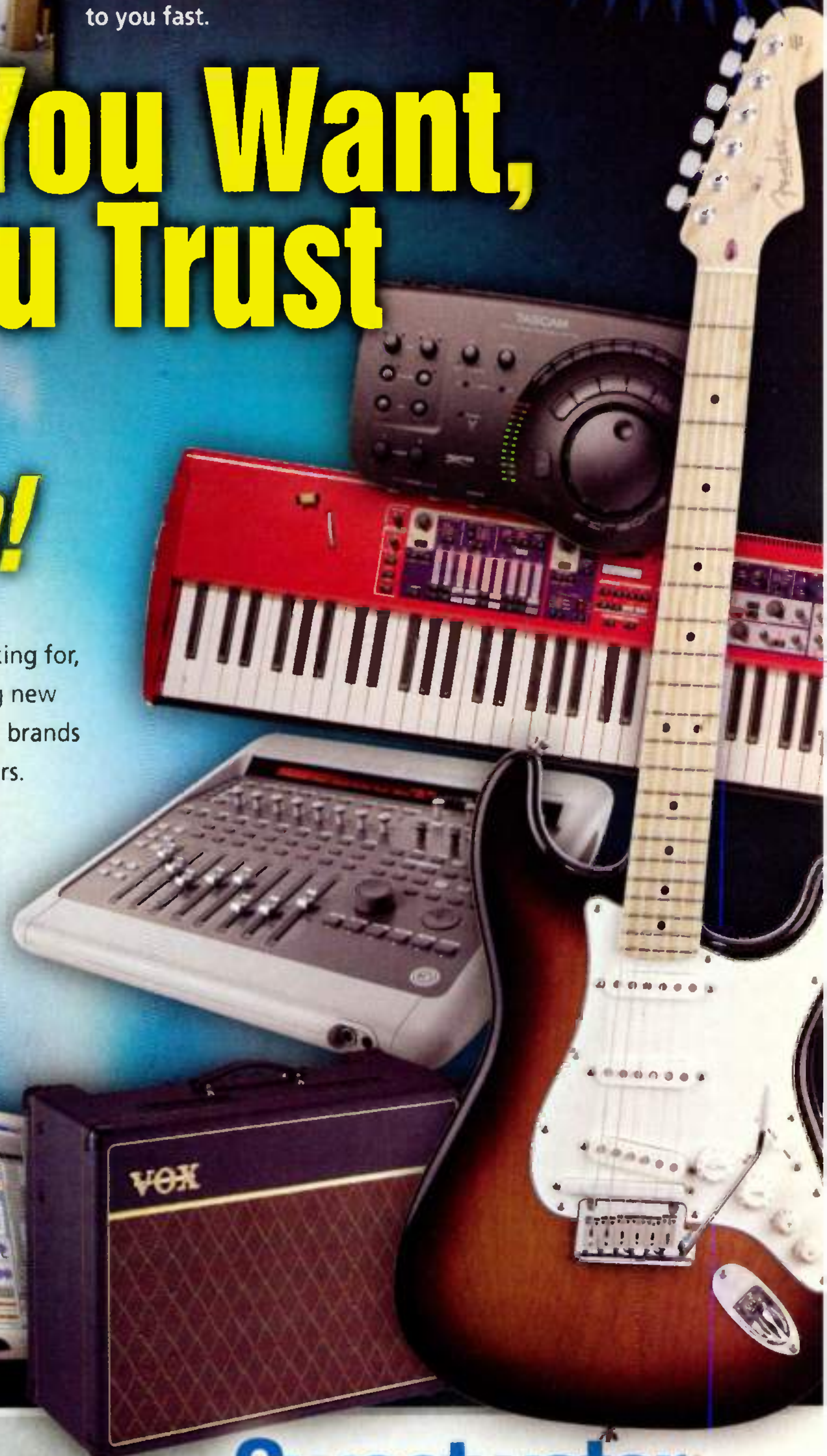

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by Shane Mehling

ETHEREAL SOJOURN

MARK HOWARD ON TRACKING THE SERENE HARMONIES OF CHRIS AND THOMAS

Alt-folk artists Chris and Thomas knew their voices had to be scrupulous recorded in order to realize their creative vision for *Land of Sea* [Defend/Boar]. So they put their trust in engineer/producer Mark Howard — the man behind Tom Waits' *Real Gone*, Bob Dylan's *Time Out of Mind*, Willie Nelson's *Teatro*, and many other vocal-oriented albums — to capture their haunting vocal harmonies. Here, Howard details how he devised a signature vocal sound for the duo.

"I do installations where I find interesting environments to record in," says Howard, "so I rented an old sound stage in North Hollywood — a three-story open space with a 40-foot high ceiling where the walls were padded like an old insane asylum. In a space that big, the sound dissipates. If you were talking to somebody face-to-face in there, and they turned around, you wouldn't be able to hear them anymore. That's amazing for isolation — especially when everything is being tracked in the same room.

"To record Chris' voice, I used an RCA-44 — Frank Sinatra's old ribbon mic — because he has a real low voice, and I've found that a ribbon mic will sometimes get a smoother, silkier sound from a vocalist with a lot of bottom end. Thomas, however, has a higher-midrange voice, so a Neumann U-47 worked better for him.

"I ran both mics through Neve 1066 preamps into IZ's RADAR 24 Digital. It's more like an analog tape machine, but in a digital format. It's great — real warm — and it never crashes. With Pro Tools, you're often stuck staring at a screen.

"As everyone is squashing the crap out of everything nowadays, I decided to go a more natural route — just light compression with a Teletronix LA-2A. You can always compress the vocals during the mix if you want more, but if you squish them too hard when you're recording, you're committed to the heavy compression. People tend to use compression to make up for bad sounds. They compress every track — as well as the stereo mix — and then the mastering engineer typically compresses everything again. You don't want to lose all your dynamic range, so focus on getting good sounds — and a good performance — and you won't feel the need to compress everything so heavily.

"However, I *am* a pretty big effects head. I use everything from the AMS DMX 15-80S delay harmony synthesizer to the Eventide 3500 to the TC Electronic Fireworx on vocals. For Chris and Thomas, I also used a Lexicon PrimeTime delay — which was Brian Eno's secret unit for Talking Heads' *Remain in Light*. You run your signal in, drop it a couple of octaves, and add however much delay you like, and you'll get a sound unlike any other.

"For *Land of Sea*, I used another Eno trick for the mix. I'd take a vocal track, send it through a bit of reverb, and then through a bit of delay to get a sound that's extremely wet, ambient, and huge. Then, I'll assign this sound to a separate fader from the dry vocal track, and add in just a little bit of the mammoth wash. It's a very subtle effect, but it helps set the mood for the track, and it becomes a very musical nuance that makes for a signature vocal sound." EQ

"PEOPLE USE COMPRESSION TO MAKE UP FOR BAD SOUNDS. THEY COMPRESS EVERY TRACK, THEY COMPRESS THE 2-TRACK MIX, AND THEN THEIR MASTERING ENGINEER COMPRESSES IT AGAIN, AND THEY LOSE ALL DYNAMIC RANGE."

Chris and Thomas tracking live in the studio with Mark Howard cohort/producer Hal Cragin.

Thomas (left) and Chris laying down banjo and guitar tracks for *Land of Sea*.

TOOL BOX

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■ NEUTRIK SCDX HINGED COVER

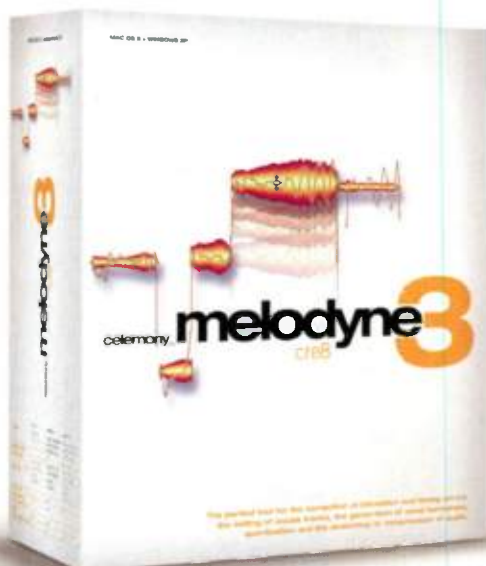
The SCDX Hinged Cover is compatible with all of the company's D-Sized chassis connectors and receptacles, and protects them from water, dust, and dirt in the unmated condition. Neutrik's SCDX also offers a hinged protection solution for its unmated XLR connectors: OpticalCon, SpeakOn, EtherCon, PowerCon, Phono and BNC, as well as USB and FireWire products. www.neutrik.com

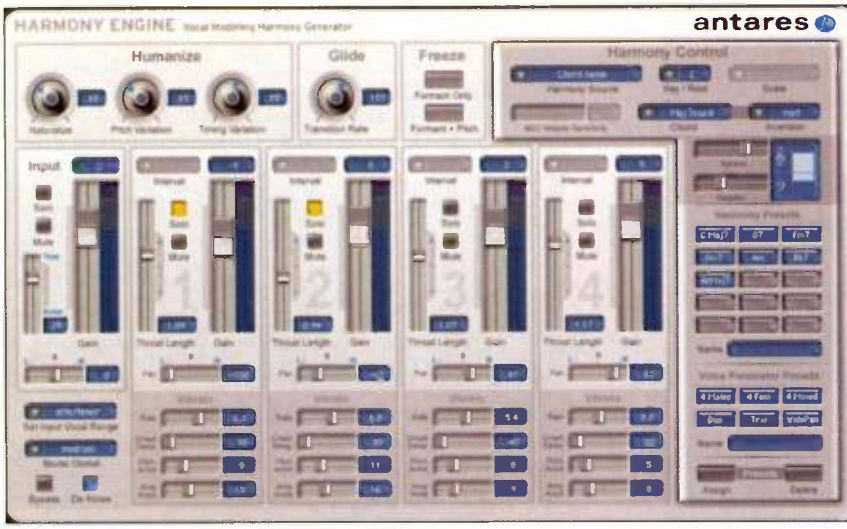
■ EAST WEST/ZERO-G CARNIVAL DRUMS SAMPLE LIBRARY

Zero-G's Carnival Drums ($159.95) features over 2,000 samples and 1.4GB of the sounds of the Brazilian carnival, and is available in the following formats: Multiformat, REX2, NN-XT, Kontakt, HALion, EXS 24, Apple Loops, AIFF, ACID, and WAV. www.soundsonline.com

CELEMONY ■ UPDATES MELODYNE FAMILY

Celemony's audio editing applications, the multi-track Melodyne cre8 and studio and the single-track Melodyne essential and uno, have been updated (free to registered users). The latest versions offer new playback algorithms, numerous improvements and bug fixes, and an easier method of program activation. www.celemony.com





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Designed for any musician or songwriter needing vocal or instrumental harmonies, Harmony Engine (for RTAS/VST on Mac OS X and Windows, as well as Audio Units) provides four independent harmony voices, a variety of harmony-generating modes, humanization features, and a flexible realtime preset system for harmony and vocal type. Mac versions are Universal Binaries, PC versions are Vista-compatible. www.antarestech.com

PROPELLERHEAD ADDS SIGNATURE PATCHES FOR THOR SYNTH

Signature patches from some of the world's leading artists and sound developers, from Richard Devine to Daniel Wang, will be available for Reason 4's new Thor polysonic synth. With six open "slots," Thor brings together six different oscillators, four different filter types, waveshaping, enveloping, and effects. www.propellerheads.se

NATIVE INSTRUMENTS GUITAR COMBOS 1.1.1

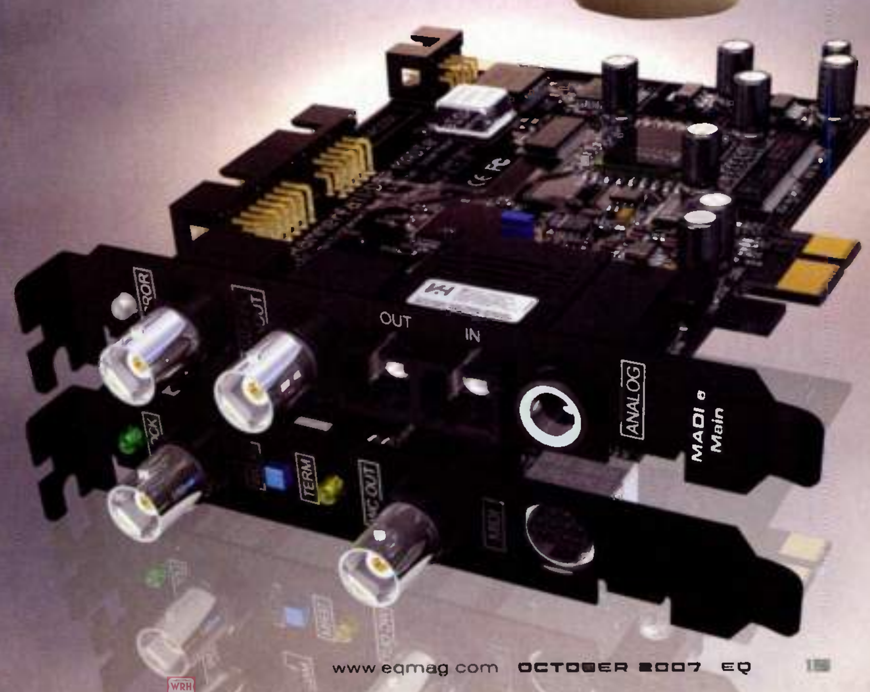
Guitar Combos is a series of three amp emulations (Twang Combo, AC Box Combo, and Plexi Combo). The 1.1.1 update (free for registered users) adds a "tapedeck" with time-stretching for convenient audio recording and playback from within the software, integrated metronome, and Universal Binary compatibility. www.native-instruments.com

MXL 990 USB STEREO CONDENSER MIC

With electronics based on the MXL 990, the MXL 990 USB Stereo Condenser Mic ($199.95) features two gold diaphragm capsules in an X/Y stereo configuration and connects directly to a Windows or Mac computer via USB, without the need for external mic preamps. www.mxlmics.com

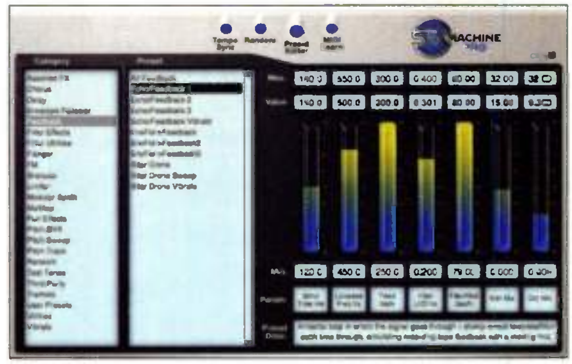
RME MADI FOR PCI EXPRESS

The HDSPe MADI card ($1,398) offers a 128 I/O-channel computer connection capable of routing and mixing all 64 input and 64 playback channels to 64 physical outputs. It supports 56- and 64-channel modes as well as Single and Double Wire formats; combine several MADI cards for even higher channel counts. HDSPe MADI comes with drivers for Windows XP (multi-client operation of MME, GSIF and ASIO 2.0), Vista, Vista 64, and Mac OS X (Core Audio and Core MIDI, for all Intel-based Mac Pro computers). www.rme-audio.com



TOOL BOX

■ **AUDIO DAMAGE DR. DEVICE PLUG-IN**
Dr. Device ($49), a multieffects plug-in for OS X AU/VST and Windows VST, includes an analog-style multimode filter, "tube"-style soft saturation and bit reduction, stereo analog BBD-style delay, and two-node XYZ control pad (hardwired for control from popular hardware XYZ pads). www.audiodamage.com



■ **SOUND GUY RESPATIALIZER PLUG-IN**
ReSpatializer (VST/AU; $99) is a panning, surround sound and spatialization plug-in that supports up to eight input channels, each of which can be individually positioned. In addition to mixing tracks for various surround formats, it's possible to create a binaural headphone mix or a mix for stereo speakers. www.sfxmachine.com

■ **RSO MUSIC MIXING VIDEO BUNDLE**
Recording School Online's The Music Mixing Bundle presents over 500 minutes of step-by-step details intended to give a deeper understanding into the process of mixing music.
www.recordingschoolonline.com

■ **MASTER CLASSES AT AES 123RD CONVENTION**
Master Classes include "The Art and Science of Making and Playing Great Recordings — The High Resolution Experience," "Op Amp Theory Refresher," "From Rough Mix to 'Yogi,'" "Vintage Microphone Mystique — How Sex Appeal Trumps Specs," and more. The AES 123rd Convention will be held at NY's Jacob Javits Center, 10/5–8/07. www.aes.org

■ **JOE BARRESI'S EVIL DRUMS BFD EXPANSION PACK**
Evil Drums ($249.95) from Platinum Samples features six full drum kits, six kicks, and six snares. Recorded to analog tape using compression and EQ, the drum sounds have as many as 250 velocity levels. Evil Drums also includes MIDI grooves performed by Pat Wilson (Weezer, The Special Goodness). www.platinumsamples.com

■ **YAMAHA MW SERIES II USB MIXING STUDIOS**
With added effects and expanded routing capabilities, these USB mixers are now bundled with a specially integrated version of Steinberg's Cubase AI4. The MW8CX ($329.99) provides eight input channels, including four low-noise preamps, as well as a digital multieffects unit. The MW10C ($249.99) adds an additional stereo input without built-in effects. The MW12C ($399.99) and MW12CX ($449.99) provide 12 ins, including four mono mic/line inputs and four stereo line inputs, and adds digital multieffects.
www.yamaha.com

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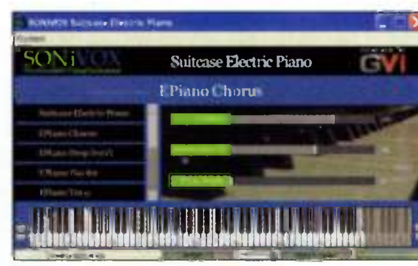
This cross-platform, audio/video production studio includes the FP10 10x10 FireWire recording interface, over $2,000 worth of music and video production software, five free uploads to iTunes from TuneCore, free BroadJam.com membership, over four hours of instructional videos, and a "Go Pro" guide on sharing, distributing, and selling your music and videos. www.presonus.com

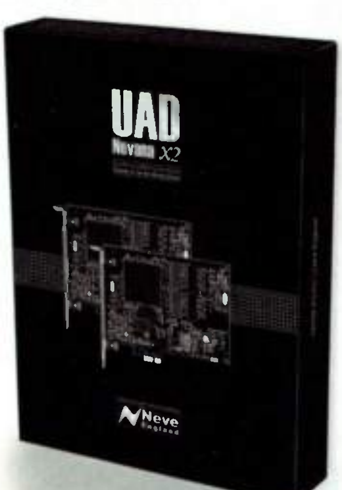
DIGITAL SOUND FACTORY VINTAGE E-MU LIBRARIES
DSF currently specializes in vintage E-mu libraries, converting classic sounds from the original Emulator, SP1200, Emax, and Proteus modules, into a huge library of downloadable SoundFont content. The site also provides detailed loading and usage instructions for all major DAWs. www.digitalsoundfactory.com

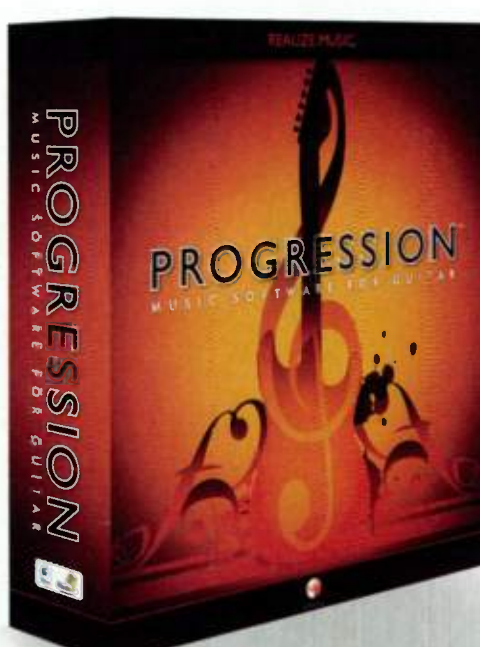
CRYSONIC SPECTRAPHY LE LIMITER/MAXIMIZER
Spectraphy LE is a look-ahead brickwall limiter and loudness maximizer based on its bigger brother SpectraPhy. Available for Mac OS X Universal Binary AU, VST, and PC formats, it features 64-bit internal precision and 24bit/96kHz support. www.crysonic.com



SONIVOX DVI SERIES VIRTUAL INSTRUMENTS
The DVI series of downloadable instruments (priced from $11.95 to $79.95) currently includes steel string and nylon string acoustic guitars, multiple drumkits, basses, electric guitars, harpsichord, electric pianos, vintage synths, harp, bagpipes, string ensembles, brass ensembles, and more; they're compatible with Windows standalone, VSTi, and RTAS formats, with Mac versions to follow. www.sonivoxmi.com

UAD-NEVANA X2 IN-THE-BOX MIXING SYSTEM
Universal Audio has teamed up with audio console/modular manufacturer AMS-Neve to create the UAD-Nevana X2 for Mac OS X & Windows X64/XP/Vista. It includes two PCIe DSP cards and all seven of the UA/Neve Classic Console plug-ins (1073/1073SE EQ, 1081/1081SE EQ, 33609/33609SE Bus Compressor, and 88RS Channel Strip) in AU, VST, and RTAS formats. www.uaudio.com





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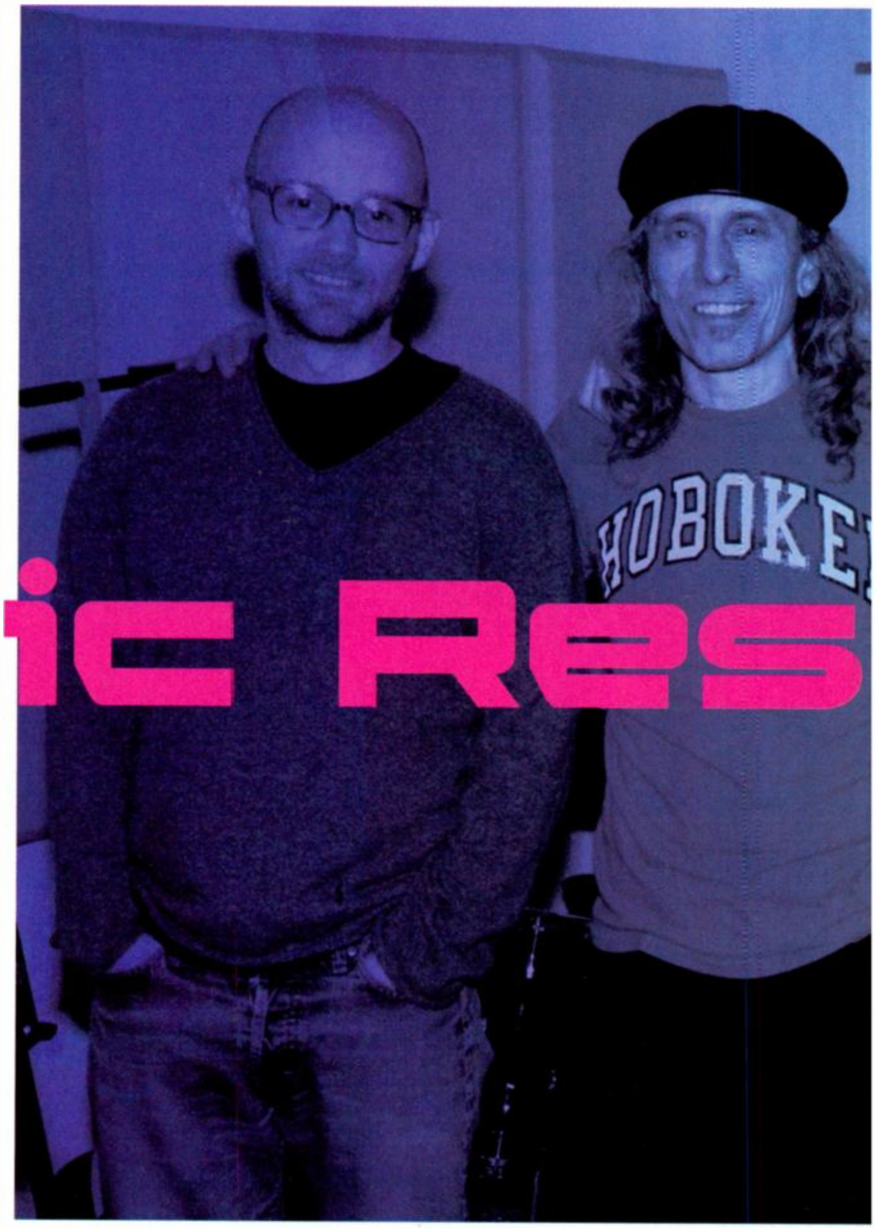
HOW RICHARD BARONE AND STEVE ADDABBO REMASTERED

Sonic Res

by Jeff Touzeau

Even at the so-called peak of the band's career, the Bongos never really attained the commercial success they deserved. But, like the foundations of Roman ruins upon which cathedrals were built, remnants of what the Bongos created are present in the music of REM, Marshall Crenshaw, and scores of other more celebrated post-punk pop groups. Magic was in the air when the Bongos were onstage and in their element, and their first album, 1982's *Drums Along the Hudson*, captures the band's unique vibe. Lead singer and guitarist Richard Barone — who was only 18 years old when Drums was recorded — was a wellspring of catchy pop songs, and each of the album's 15 tracks was as engaging as the last. But the sound just didn't do the music justice, and the album's unflattering recording quality has haunted the Bongos legacy ever since.

Nonetheless, many hold the album in high regard, so it wasn't a surprise that the Bongos were asked to re-release *Drums Along the Hudson* late last year. With the opportunity to record a bonus track with longtime fan Moby, the Bongos briefly reformed and entered the studio. But upon completion of the bonus track — where the group was in awe of actually sounding on record the way they heard themselves on stage — the boys decided to take it one step further. They called on producer Steve Addabbo, owner of NYC's Shelter Island Sound, and requested that he help restore and remaster the original mixes of *Drums Along the Hudson* — mixes that had been sitting in Richard's home since 1982.



Of course, many of us have great songs we cut years ago that we're almost ashamed of letting others listen to now. The songs are happening, but the production just sucks. This is how Barone felt about *Drums Along the Hudson* all these years. He was living with a master with which he was never fully satisfied. A master that needed tweaking — particularly on the low end.

"The drums had a much deeper sound on the mixes," he observes. "But when we mastered the album, we were told the grooves of vinyl records couldn't handle the low end we had on the toms. As a result, the drums ended up sounding much tinnier than I wanted. The sound we got at the mix never made it out of the mastering room."

Barone may have been disappointed with the results, but, back in the vinyl era, mastering engineers did have to deal with the fact that low frequencies required physically deeper and wider grooves. Wider grooves meant less space for cutting tracks into the vinyl, so low-frequency content was often sacrificed to accommodate putting more songs on an album. Thankfully, this problem doesn't exist within the digital mastering realm.

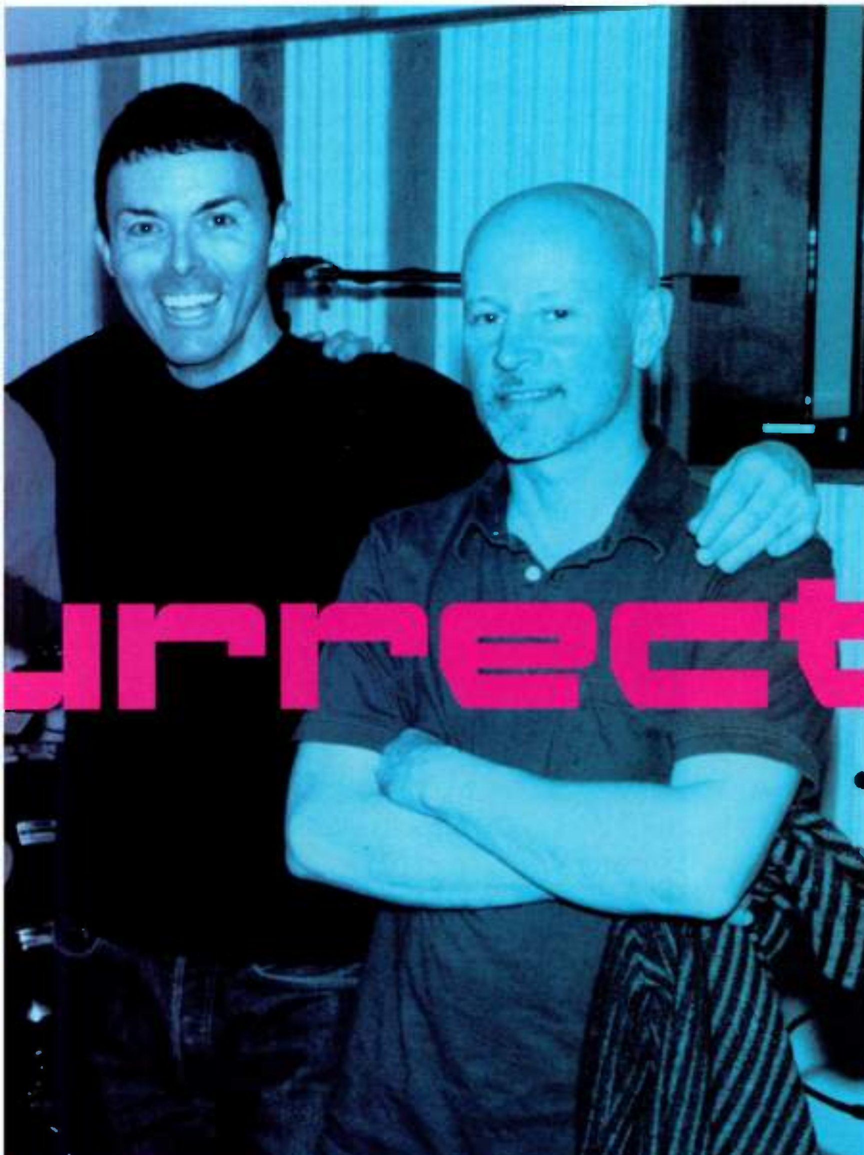
"A digital signal doesn't care if it's a tin whistle or a timpani, because it's saving the sounds as a set of numbers out of your A/D converters," says Addabbo. "Now we don't have to choose songs over production quality. When *Drums Along the Hudson* was originally mastered for vinyl in the '80s, you had between 18 and 20 minutes per side that you could work with, and more bottom end meant less playing time. In addition, mastering engineers faced another physical limitation with vinyl, which was the difficulty of maintaining hot levels as the cutter got closer to the inside grooves. Due to physics, the space is more cramped there, and there's less room for the needle to move. [*Editor's note: This led to a phenomenon called "inner groove distortion," and, to work around this, many vinyl albums were sequenced so that the end of a side had a ballad or other soft song.*] Back then, mastering was a real balancing act. However, digital audio doesn't have these physical limitations, so we can really crank the stuff, let the bottom be where it should be, and carry the desired frequency spectrum throughout the entire album."

But before any sonic challenges could be addressed for the album's remastering, the questionable state of the original 1/4" analog-tape reels had to be confronted.

"There was a major concern the tapes might not even play any more," Addabbo confides. "I've had pretty miserable results with tapes that old. You just don't know what you're going to get, because the adhesive that holds the magnetic bits to the backing of the tape can get gooey over time. You start to play the tape, and, within two or three minutes, it grinds to a halt. Then, you have this brown gunk on your tape machine's playback heads and rollers that looks like tar. This can clog your heads, and wreak havoc on the tape, because it's basically scraping off the oxides."

And that's not all. Over a period of years — usually a decade or more — magnetic tapes attract dust particles and other gunk that can sometimes render them unplayable. To counteract this and other issues, the practice of "baking" old tapes has become commonplace. In a nutshell, the heat from a convection oven uniformly degrades the unwanted gunk and excess particles so the tapes can be played once again without causing damage to your songs — or your tape machine.

"We absolutely had to bake the tapes," says Addabbo. "I have a convection oven in my home, and what I did was bake the six to eight 1/4" tapes at about 130 degrees for around two hours. It's unbelievable how well the tapes played after that. However, old tapes are still finicky, so it's good practice to transfer the audio to another medium as soon as possible. If you play the tapes too many times after baking, they probably won't hold up for very long. Having said that, the baked tapes were shedding so little, that I worked directly from them for a while. I could really listen to the sounds on the original tape masters, and basically treat the process like a 'new' analog mastering session because the tapes had held up so well."



Resurrecting the Bongos (left to right) — Moby, Rob Norris (bass), Richard Barone (vocals, guitar), and Frank Giannini (drums).

urrection!

The original leader tape used on the masters caused problems, as well. Someone had written vertical stripes with red ink as timing markers, and the ink touched the back of the oxide, which actually pulled the oxide right off the master. This compromised the sound of some song endings and beginnings. To fix these bits, material from CDs and safety masters had to be "flown" in once the audio was transferred to Pro Tools. Addabbo did the transfer at 88.2kHz, because the math the computer has to do to go from 88.2kHz to the conventional CD format of 44.1kHz is a "simple math" division by 2, as opposed to downsampling from 96kHz to 44.1kHz, which works out to division by around 2.1768707. When a computer has to do more complex math in these situations, the possibilities for round-off and quantization errors are greater.

Once in the digital domain, Addabbo and Barone tried to create a common sonic framework for the album. This was not an easy task, as the original tracks were recorded at three different studios by three different engineers, and on both sides of the Atlantic. They settled on the lead-off track, "In the Congo," as the benchmark for the rest of the album.

"It had the right bottom, and it just sounded exciting," says Addabbo. "The other mixes didn't sound as loud, and they required a lot more creative compression and EQ to get them sonically matched. Coming off the tape, I hit my Medici EQ — which is a fine-tuning mastering device. I put a very gentle bump in the low end at around 100Hz and 125Hz. I also tend to look at the 500Hz range to eliminate any muddiness that can build up in the toms and guitars. Cutting that frequency range a bit tends to really liven up a kick drum. If the mix is good, the changes can be very subtle — maybe only 1dB total of reduction."

One of Addabbo's favorite tools to help him achieve sonic balance is the Waves C4 multiband compressor, which can isolate and process the dynamics of a specific frequency, enabling the engineer to alter, for example, the thwack of a snare drum or the thump of a kick drum.

Photo by Brian T. Silak

Sonic Resurrection!

Home Cooking

Do you have some older tracks you recorded back in the days when you didn't know what you know now, and you'd like to resurrect them? Try your hand at remastering. Even if the tracks don't end up being released commercially, you'll learn a lot.

Start off with equalization. You may need to add some high end for sparkle, or trim some bass to lessen muddiness. But also try to reduce frequency "buildups" you might have missed the first time around. Sweep a midrange EQ set for a huge boost [Tip: turn down your monitors first] and a relatively narrow bandwidth — like an octave. Some frequency ranges will jump out compared to others. Find the range that jumps out the most, then set the gain to zero to re-acclimate your ears to the "normal" sound. Then, reduce the gain a dB or two at that frequency. This will often tighten up, and even out, the sound.

Add any dynamics control after the EQ is squared away. A few dB of loudness maximization or compression can make for a hotter, more consistent sound, but don't go overboard, or you could end up with the overcompression that mars so many recordings these days. —*Craig Anderton*

"With a multiband compressor, you can pick out things in the mix that don't sound very exciting, and really enhance any particular band," says Addabbo. "Let's say there's too much bass. If you create the right low-end setting on a multiband compressor, you can have it kick in exactly when a specific frequency band gets too loud, and pull back the bass at just the right moment. A multiband compressor is also handy for the midrange frequencies, where a vocal can often use a little more presence, as well as for the high end, where a cymbal can be overbearing."

Getting hot enough levels for broadcast was very important to Barone and Addabbo, as, like it or not, we are living in the age of competitive radio airplay.

"One of the things I was up against was making the CD loud enough so that it could compete with what's coming out today, but without making square waves out of it — which most commercial CDs these days are starting to sound like," says Addabbo. "To do this, I used a combination of analog and digital tools. I did some gentle overall compression on my SSL G384, using a very slow attack and a fast release. Generally, I don't use more than maybe 1dB of analog compression on the SSL, because it affects the overall mix too much. Then, I used Digidesign's Maxim and Waves' L1 and L2 digital plug-ins. The L1 was employed for a couple of more dB of leveling and limiting. On the Maxim—which sounds sweeter to me than the L1 and the L2 — I set its ceiling to −1dB. I'm telling the limiter it can't let the audio hit zero — that's the cutoff. The final stage in the mastering chain was my Universal Audio 2192 AD/DA converter. It's by far one of the sweetest two-track converters we have. Part of it has to do with its tube front end. Whenever I go into that box, it just seems to create a very pleasing master. It emulates the feel of 1/2" analog tape."

Addabbo made the 44.1kHz CD by creating a playlist in the Alesis MasterLink, then did the track spacing and IDs in Waveburner.

"In Waveburner, you have a timeline, and you import your 44.1kHz master files to it," he says. "It gives you a visual interface that shows you where your tracks are, and exactly how much time is between them, so you can make fine adjustments to ensure the right flow of your tracks."

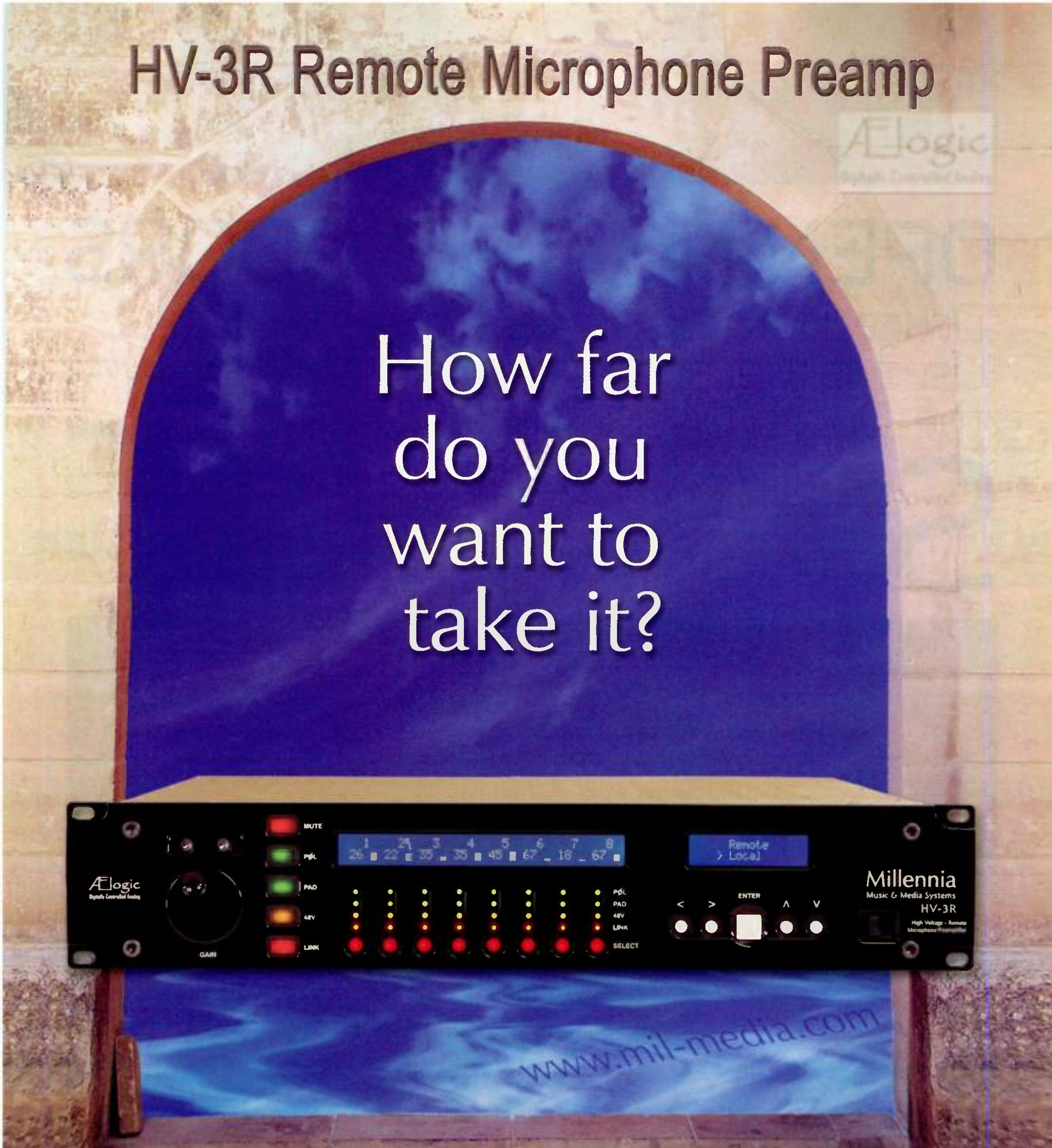
"The mastering process requires an entirely different frame of mind," adds Barone. "It's almost as if you have to become a 'doctor' for sound. As an artist, I've worked with some of the best mastering engineers — including Bob Ludwig — and I usually tell them, 'Here's my pain. What can you do to help me?'" EQ

Baked Goods

If your recording career pre-dates the digital age by a few minutes or so, you likely have some analog tape masters hiding out in your closet, storage bin, or garage. These don't have to be the forgotten jewels of your musical repertoire — especially as personal distribution channels such as MySpace are available for you to introduce past endeavors to new listeners. But, as Barone discovered with his *Drums Along the Hudson* masters, remixing or even playing old two-tracks can be problematic if the magnetic tape is decayed and gooey. Baking is a proven method for making analog masters playable again, and it can pave the way for you to transfer your old tracks to digital, dress 'em for today's audiences, and send them out into the public domain again. There's a wonderful article on the process by engineer Eddie Ciletti on composer Wendy Carlos' site at www.wendycarlos.com/ bake%20a%20tape/ baketape.html. You can also avail yourself of outside baking services, such as www.audio-restorations.com, which charges from $10 (cassette tape) to $50 (2" analog master) to resurrect old tapes. So, fire up the oven, and save your music! —*Michael Molenda*

Addabbo at work in the mastering lab.

Addabbo (left) and Barone (right) at NYC's Shelter Island Sound, fresh from the remastering session for *Drums Along the Hudson*.



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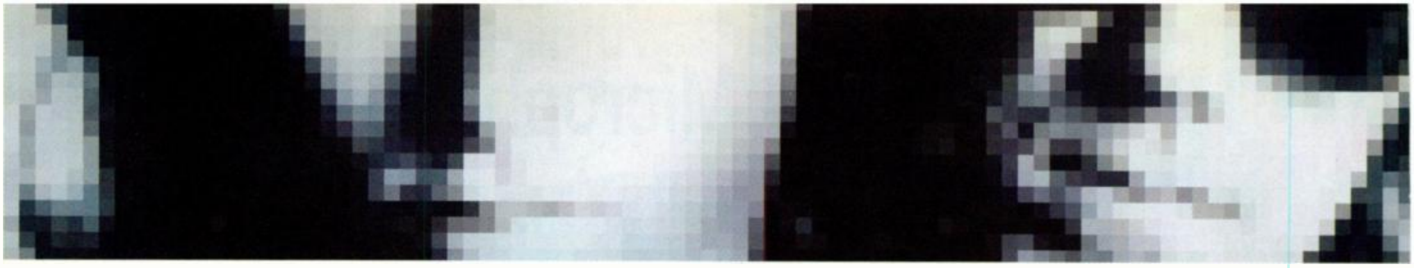


OPENING THE DOORS

Bruce Botnick Reveals the Techniques Used in Recording, Mixing, and Remixing the Doors Discography

Jeff Touzeau

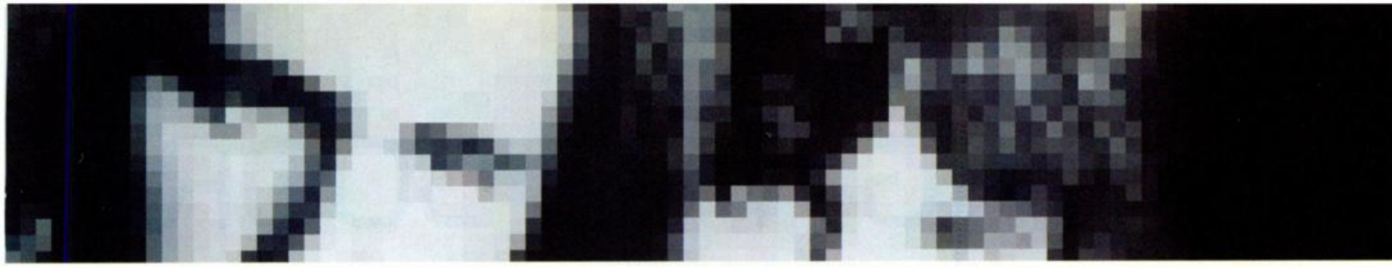
Summer, 1966, Los Angeles — a young Bruce Botnick is living his dream in a 10' x 20' room, jam-packed with recording equipment, and, at most hours of the night, slightly disheveled band members. Life is good, because life is all about recording music. The long hours don't really faze anyone.

A voice calls: "There's a band called the Doors coming in. They have a guitar, drums, vocals, and an organ, and the organ player plays a piano bass."

Forty-one years later, Botnick is back in the same place. Not literally, but certainly figuratively. The studio legend — who engineered *The Doors* (1967), *Strange Days* (1967), *Waiting For the Sun* (1968),

Photo courtesy of Bruce Botnick

Botnick recording the Doors at Sunset Sound's Studio 1.



The Soft Parade (1969), *Absolutely Live* (1970), and *Morrison Hotel* (1970), and co-produced *L.A. Woman* (1971) — is tasked with revisiting his classic Doors sessions, and using today's tools to usher the band's tracks into a new sonic era, as well as into the 5.1 format.

The ceremony is about to begin. Is everybody in?

So you're finishing up work on the remixed and remastered versions of all the Doors' studio albums. What will be the final medium for the masters?
Vinyl. Why? First, the fans have been clamoring for it. Second, the state of the art for vinyl today is much better than it once was. The technology is more effective, and the pressings are quieter and cleaner. They don't degenerate as fast. It's interesting, because we are going back to the original master tapes — from 1967 to '70 — and they are all in various states of disintegration. Some play perfectly, others don't play well at all.

What have you found in particular that's funky about the originals?
I retrieved the tapes — including the original stereo and mono masters, safety copies, EQ copies, and anything that was a complete master — from a climate-controlled vault in Hollywood. And, interestingly, I found that the first Doors album never ran at the right speed. When the Scully 4-track machine it was mixed on [at Elektra Studios, New York] neared the end of a full roll of tape, it would slow down because of the tension. As a consequence, the speed was pretty much normal at the beginning, but everything could be flat by the end. In fact, "Light My Fire" is a quarter-tone off by the end of the song. It's just unbelievable.

So, for these new high-resolution stereo remixes, I was finally able to get everything on pitch by playing.the 4-track master back at the proper speed when I transferred it into my Pro Tools 7.3 rig at 24-bit/96kHz. Some people complained we were not being true to these classic records

Botnick hanging out at 20th Century Fox Scoring.

by taking them into the digital world. But doing this saved us from the speed issues, and from the deterioration of the originals.

Would you say this is an instance where digital tools made a truly positive impact on the end result?
Absolutely. In the case of *Strange Days*, there is one song where somebody had taken the tape out of the vault, and there was a big stretch in it. I had to go to another safety copy to access and replace the 1-1/2 bars that were damaged. I couldn't have cut this in analog, because it wouldn't have matched. Now, I couldn't assume the tape was running at the same speed. Therefore, I had to A/B it against the original to see what would happen.

So is the first step to get everything to the right speed, and make high-quality transfers?
Yes.

What other elements needed sonic adjustments?
Over time, the tapes have lost some high end. Also, there are level changes that need to be addressed so that each song is the same volume as the ones that precede and follow it. And some of the songs are brighter than others — which is the nature of the beast — so there needs to be some global EQ.

Are you EQing in the digital domain?
Yeah. I'm really into George Massenberg's digital EQ, and the Sony Oxford EQ plug-ins. They impart a pleasing character. I'm more into doing subtractive equalization than additive. Something I heard from [mastering engineer] Bernie Grundman has been really sticking with me lately: If you listen, there is usually something "clouding up" your sound. If you have a really good microphone placed in the right spot running into a really good preamp, and so on down the chain, you're going to increase your chances of realizing a great sound from the start. But, sometimes, the room itself can introduce problems that you don't catch until the mix. And when you find them, your tendency may be to push something up. Where you might feel like you need to add some high end, you may just need to cut at 200Hz to clear the sound up. Then you can add a dB or two at the top end, and it will sound as open as adding 6dB or 8dB of high-frequency EQ, and never cutting at 200Hz.

These are your original recordings. You're not just some new-jack mixing engineer trying to spice up a classic album, so you have real perspective on these sessions. Given that, when you pulled the tapes, and started looking at them as soon-to-be 5.1 mixes, did you rediscover anything about the recordings that floored you?
Given the limited tools we had to work with, there were actually quite a few things that impressed me. And I was also incredibly depressed by other things. I'm proud these recordings still impart a certain beauty. To me, they sound pretty and sweet. It could get raw and rocking, but this sweetness still came across and held true. There were so many variables that made it all work, and you can never, ever go back. I'm not 19 anymore. I hear differently. I was eating different food, breathing different air, and recording differently.

In what ways were you recording differently?
One noteworthy technique was that I came up with a different way of tape delaying the Sunset Sound echo chamber. I had a three-track Ampex 200 machine that could handle 14" reels. It was able to provide separate record and playback equalization from the input to the output, so you could basically EQ the machine for your sound. There were NAB, CCIR, and AME equalization curves. It was a type of noise reduction, if you will. I remember using the AME EQ on the record

OPENING THE DOORS

side to put in more highs, then playing it back with the NAB curve to take out highs and reduce tape hiss. This would put a rise in the EQ curve in the chamber. Then, I would delay the output of the chamber — not the input. Nobody else was doing this that I know of, and it's a special sound.

Tell us how you recorded John Densmore's drums.

There were only three mics on the drums at any one time: a Sony C37 overhead at just about forehead level; another C37, flipped out of phase, underneath the snare drum; and an Altec Salt Shaker dynamic mic — the kind they used to use for announcements in airports — for the kick. And we didn't record with an outer drum head — which was why the kick had that real "pop" to it. That was it. Sometimes, I would put a Telefunken U47 about six to eight feet back from the kit in the room, and add some heavy compression to open up the sound. I also set up the drums against the far brick wall, because I liked the reflection of the drums coming back into the overhead mic — it added a real liveness to his sound.

HOME COOKING

When Bruce Botnick talks about using EQ before and after reverb — and changing the position of delay with respect to reverb — he's recalling the days when you had to physically move patch cords around and match levels (often with preamps and attenuators). But with today's software, you can try the same types of experiments in minutes — just by moving plug-ins around in track inserts.

Does pre-delay before reverb sound different than delaying the reverb signal? Try it yourself, but remember that, in those days, tape was providing the delay so it affected the sound. Also, plate and room reverbs had a very different character compared to modern digital reverbs.

Set up the following plug-in chain: EQ1 (high boost) > Delay > Reverb > Delay > EQ2 (high cut), and start by setting both delays to a specific delay time, like 70–100ms. Bypass them individually to hear how each one affects the sound, then enable both and vary the delay times — delay before and after reverb is a whole other sound entirely. Also try boosting the high frequencies going into the chain, and cutting the highs coming out to simulate the effect Botnick describes. Even though you won't be cutting tape hiss, the result will often be a "rounder" reverb sound. —*Craig Anderton*

What did you use for Jim Morrison's vocals?

A Neumann U47 — which has pretty much always been my favorite vocal mic. And yes, for *Strange Days*, we didn't use any pop filters. I hate them, because I can hear them. Jim was very controlled, so a light compression when he screamed into the mic was all it took to keep him sounding even. In those days, it was very common to have technical recording information on the back of the albums. They would list, "Trumpets: U47" and the like. So I would listen to the records, hear those sounds, and, then when I got back to the studio, I'd try out the mics. I'd say to myself, "Wow, it does have that sound character. That's cool. I'm going to use it!"

How did you mic Robby Krieger's guitar?

I used the U47 right up on the grille of his amp. I didn't know any better! Same for Ray Manzarek's organ — a Vox Continental — though we always ran the piano bass through a direct box. [*Editor's note: The bass was a Fender Rhodes keyboard bass, which, according to Manzarek, sounded too "blown out" on record. So, starting with* Strange Days, *the Doors used various electric bassists including Doug Lubahn, Ray Neapolitan, Harvey Brooks, Jerry Scheff, and guitarist Lonnie Mack.*] For the acoustic guitar tracks, we would just put one of the C37s back a foot or so from the body of his acoustic — which just sounded huge. He absolutely loved the sound of them.

What was the room like?

The room at Sunset Sound had concrete floors with asphalt tile, brick walls, and plasterboard ceilings that were three or four inches thick. That place was hard. It sounded hard. You could hurt yourself in there, and you can hear that come through in the recordings. The room influenced the sound enormously. I remember being at the old Record Plant, and doing a session in Studio B where Stevie Wonder had done "Living in the City." I set up the drums in there, set up my mics, and when the track came over the monitors, I said, "My God, that's the same sound!" Even though I was using my choice of mics, that studio had a very distinct sound to it. It blew me away.

What was it like mixing those albums the first time around?

You have to understand that most of these albums were done in a week! If you only have four tracks to work with — or eight tracks — there are only so many things you can do. Now, you have so many tracks to work with that you can virtually do anything, and you don't have to make a commitment at every step of the process. But having to make decisions quickly added to the feel of those records. We

Photo by Ed Colver

Where the wild remixing things are: Botnick's room in Ojai.

Control room view of Sunset Sound Studio 1, set up and ready to record.

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OPENING THE DOORS

recorded with reverb live, we compressed and limited live, and we added equalization live. What we heard when we were recording was the mix that you heard in the end.

But didn't the Doors' dramatic spatial elements lend themselves well to the 5.1 remixes of the albums?

Some of it really did. For example, when we did the original mix for "Riders on the Storm," and added the rain and thunder, I just had a tape machine running in the back, and, by serendipity, the thunder came in at the right place. It wasn't planned. That speaks to the notion that recording somewhat recklessly can end in moments of perfection. But when it came time to remix in 5.1 — which I did in Sonic Solutions — I rebuilt the mix, put the thunder in the place it had always lived, and then panned the rain, and added a delay to the thunder track to create a 360-degree environment. It ended up working really well.

What about "The Unknown Soldier"?

Well, there is that whole sequence where the prisoner is being marched to a stockade to be shot. So it marches all around the room. Jim is doing the cadence. The gun cocks and fires. Then, the crowd starts screaming, and the church bells are ringing. There's a great deal of movement going on. There were other places where I'd just put a nice piano in the rear, or maybe the background vocals. I would have Jim's vocal in the center speaker — just as we recorded it. In the beginning — up until *The Soft Parade* — we always had three Altec Lansing 604e monitors in front of us, because we recorded three tracks. It was normal to hear Jim in the center, so when I got a chance to remix these into surround, I got him back in his own environment. He's in his own place now, and it really makes a difference.

You clearly believe equally in classic techniques and modern tools. What have you held on to — despite the passing of the years — in terms of recording techniques?

Many things are the same as they ever were. You still decide what it is that you are going to record, and what your goals are. You still have to pick your instruments, your mics, and your room — partially on faith, and partially from past experience. You still have to set up your instruments. You still have to play well. You still have to really listen to what you've tracked before you start with the adjustments. When you look at it that way, not that much has changed. The reality of a good recording is still capturing good sources.

But if you're looking for advice, the best I can give you is this: Get everybody who is playing on the track in the same room, open up all the mics, assign them to their channels, and try to balance it all live. Learn to hear it all together, and make your decisions then. By focusing on one element at a time, you'll never get the perspective you need to tackle the recording of an album. The album is the whole thing — not just the sum of its parts. EQ

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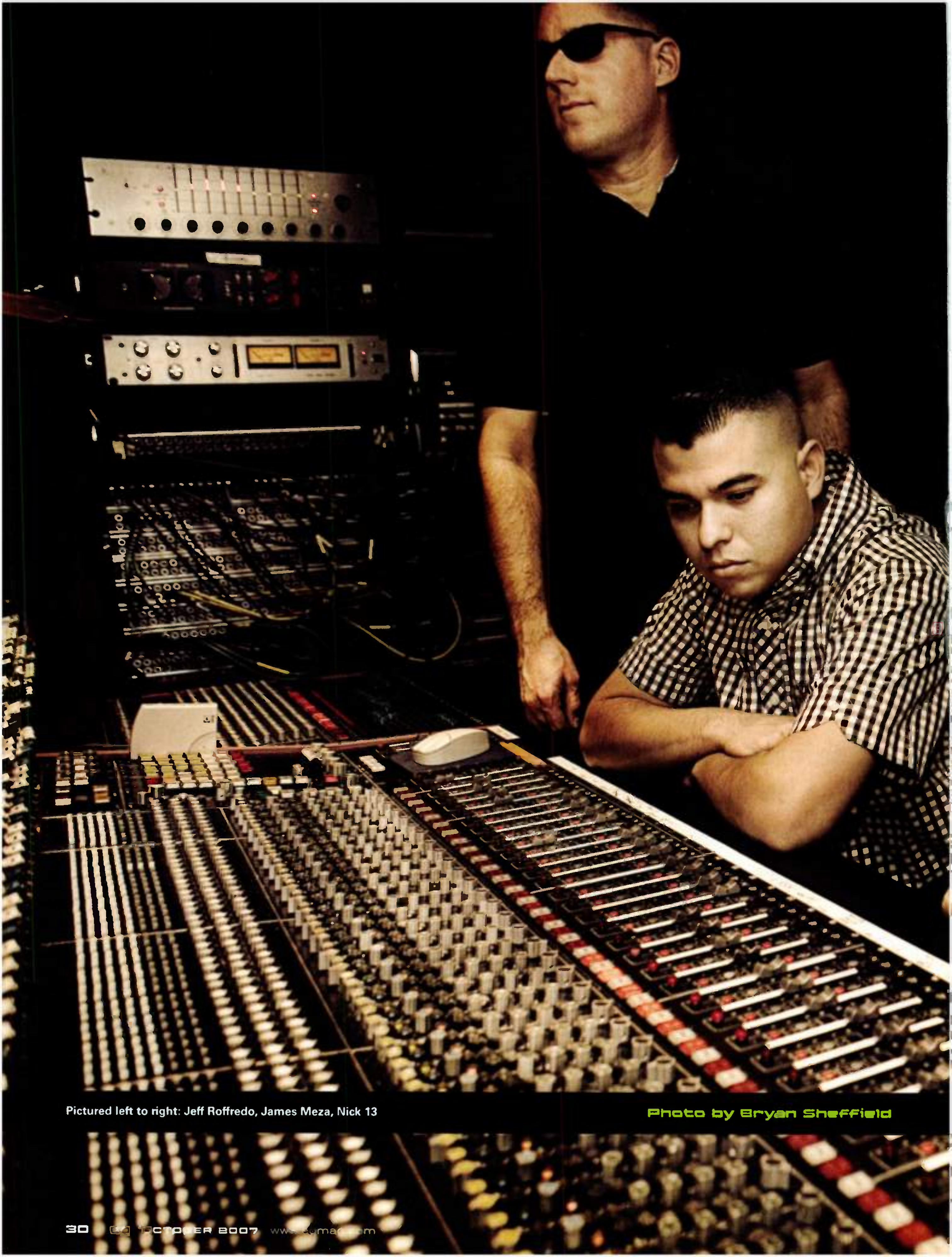
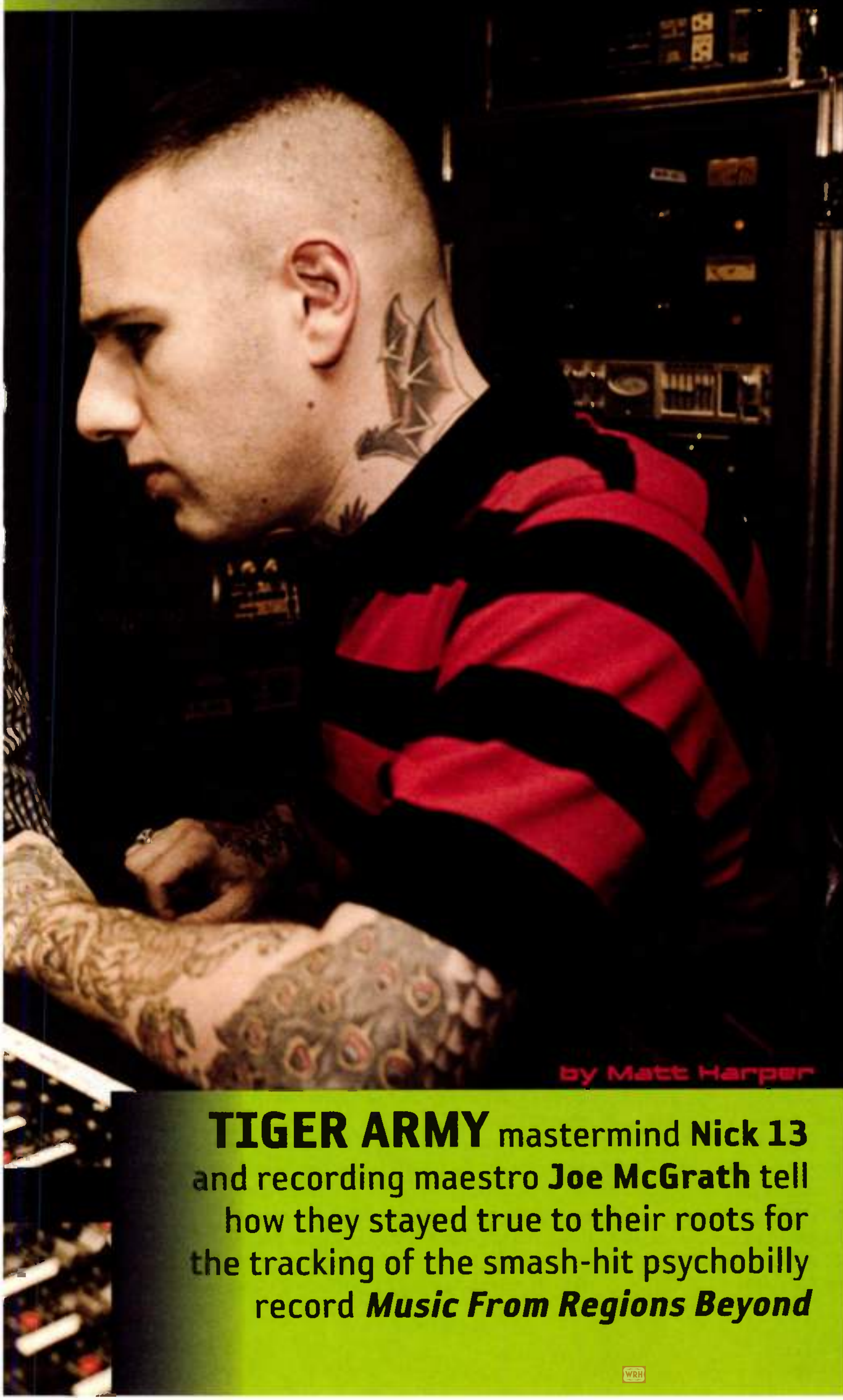


Pictured left to right: Jeff Roffredo, James Meza, Nick 13

Photo by Bryan Sheffield

Psycho Analyze



by Matt Harper

TIGER ARMY mastermind **Nick 13** and recording maestro **Joe McGrath** tell how they stayed true to their roots for the tracking of the smash-hit psychobilly record ***Music From Regions Beyond***

PSYCHO ANALYZE

"I'M anti demo," Tiger Army guitarist/vocalist/head honcho Nick 13 declares from back stage at this year's annual Vans Warped Tour. "In my experience, the first commitment of a song to tape always has a very special quality to it, something that, ultimately, is lacking in any later versions. I'm not into chasing that vibe. I'm not into spending all our time trying to recreate that magic of the first track. So I prefer the first cut to be in the studio . . . and for the record."

Beyond his general recording ethos, what Nick 13 is speaking to specifically is the process undertaken during the sessions for the band's latest album *Music From Regions Beyond*. Heralded by fans and critics alike as the band's most realized work yet, *Music From Regions Beyond* is a delectable arrangement of punk-infused rockabilly drenched in '80s pop sensibility — a play on the band's time-honored psychobilly standard of aggressive, yet catchy, punk laced with golden oldies rock and roll and spattered with honky-tonk. And it's a genre that Tiger Army has been largely credited with popularizing on American shores; a mixture of sympathetic musical styles that have won them legions of fanatic, adoring fans (one look at the band's Myspace page and the hordes of those bearing the tattooed mark of the group's mascot is proof-positive that Tiger Army is more of a lifestyle these days than anything else) and landed them the headlining slot at one of the summer's most popular package tour offerings.

But it's not that the world has suddenly caught on to what is by and large still a very underground form of music and just decided to rally behind Tiger Army. They aren't that lucky. To the contrary, when one first spins *Music From Regions Beyond* it becomes immediately obvious that the band had tasked themselves to take the psychobilly style and mold it into something more immediately appealing to the masses. That's not to say that the band has softened up or is simply turning out pop rubbish in hopes of striking it rich — Tiger Army have just successfully refined the style and released an album that satisfies the visceral urge of the punk rock congregation while simultaneously providing the kind of catchy tunes that morning commute radio fiends find themselves humming in the break room come mid-day.

"Our style these days can't be summed up in one word," Nick affirms. "But if you have to categorize it, it would still be called 'psychobilly'. Sure, it's morphed into a new thing, but the roots are still there — namely punk and 1950s rockabilly/rock and roll."

It's a clash of genres that, to the band, ultimately makes sense. "The first wave of both British and American punk bands were very directly influenced by '50s rock and roll," he continues. "Whether you were talking about the Ramones, the Sex Pistols, X, the Damned . . . the two genres were always connected. And I was always intrigued by those connections. Look at the Sex Pistols' *The Great Rock and Roll Swindle* — it had three Eddie Cochran covers!"

There's a lot more to the Tiger Army sound — especially on *Music from Regions Beyond* — than just the coalescence of punk and old time rock and roll. When asked about the country twang, juke joint rhythms, and *gasp* new-wave pop hooks rolling throughout the record, Nick responds: "Going from punk to Eddie Cochran, the Johnny Burnett Trio, Charlie Feathers — all that old Sun Records stuff — is easy. But if you move back another degree, you find that those artists were rooted in the hillbilly/bluegrass/honky-tonk traditions. Our influence follows the same path — Tiger Army starts with punk and then moves all the way to the pre-rock and roll acts. But you can also hear in us what punk gave way to: the new-wave/dark-wave/post-punk '80s acts like the Cure and Joy Division. If you listen to it, you can see where it makes sense. All those bands are kindred spirits, I believe."

"IT'S ALL ABOUT KEEPING THE ENERGY OF THE TRACK IN THE MIX – THAT'S HOW YOU MAKE A PUNK ALBUM." –NICK 13

For all their branching out stylistically, the punk approach to not only songwriting and performance but also *recording* is clearly evident in Tiger Army's sound. Though long gone are the days of the boys releasing seven-inches made entirely of super raw four-track basement recordings, the band still prefers to work in the fashion of their forefathers, largely ignoring modern means to the recorded end.

But when it came time to begin tracking *Music From Regions Beyond*, the band did decide to operate in one way that is decidedly non-punk: Hire a producer and forego the DIY method of self-producing.

This is all the more shocking considering that up until *Music From Regions Beyond*, Nick had produced all of Tiger Army's releases. "I was pleased with the way our last album — *III: Ghost Tigers Rise* — sounded. And because I was actually happy with the recording, I felt that I had taken things as far as I could as a self-producer," he humbly confesses. "I wouldn't have come to this conclusion after *II: Power of Moonlite*, because I felt we were pretty far off at that point. I wanted to learn more about the process — get a closer point of reference — before I collaborated with anyone else."

Enter punk rock super-producer Jerry Finn — he who has been behind the board for bands like AFI, Rancid, Green Day, Bad Religion and . . . *Morrissey*? "I was so extensively familiar with his catalogue," he says, "especially Morrissey. I could listen to the albums he produced and hear exactly what he would bring to the table."

As luck would have it, long-time Tiger Army pals AFI invited Nick to the studio in 2006 to lay down backing tracks for the *Decemberunderground* album, introducing Nick to Finn. Finn was exactly the type of producer Tiger Army needed — someone with the production chops to track a dyed-in-the-wool punk band such as Rancid and then turn right around and offer up the pop treatment necessary to make a mega-selling Morrissey album.

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PSYCHO ANALYZE

"By the end of the sessions, I had a good idea of Jerry's perspective as a producer. He became the only choice to produce this album. The way he works is rooted in punk. His punk background really shows [in the studio]," Nick enthuses. "It's all about keeping the energy of the track and not flogging it to death during the mix — that's how you make a punk album. That's how Jerry does it."

Being ballsy enough to bring a devil-may-care attitude into the tracking room is only the second step to cutting a album that sounds like a middle finger in the face of the status quo — a band must, of course, first have a place to flip the world off from. So Finn and cohort/engineer Joe McGrath took to streets of Los Angeles searching for a studio that was not only functional (*i.e.* stacked to the ceiling with what Tiger Army had specified as their must-haves: vintage Neve or API consoles, a tape machine or three, and a live room that would hold the trio comfortably so they could track their basics all together) but also had serious vibe. They wanted a studio that had housed the production of classic albums. They wanted a studio that, if possible, would hearken back to the famed Sun Studio — the birthplace of the rockabilly sound.

The team settled on Sunset Sound's Sound Factory Studio A. As McGrath recalls: "We knew Nick wanted to work in a room with a very specific feel. We walked in and it felt like the place had not been touched by time — it still had the original tiling on the walls! And it had a custom 36 x 16 x 38 custom API console! And a Studer A827 24-track! And an Ampex ATR 102 2-track! We knew we would be able to get a warm recording to match the vintage sound the band naturally has there."

"I BELIEVE IN SMASHING DRUMS TO TAPE, ESPECIALLY THE ROOM MICS." -JOE MCGRATH

Studio A was also a practical choice given the drum sound the band was aiming for. "The room there is not all that large," McGrath continues. "That would be a deal breaker for most people that think classic sound = huge drums = big room. But a huge drum sound isn't what psychobilly music is about. The drums are close and punchy, so we needed a smaller room to help keep the drum sound naturally tight."

Beyond this, the layout of the live room and its satellite rooms was perfect considering the band's intentions to cut their first-pass tracks live. With a main live room (26' x 23'), a "piano room" (14' x 20'), and small receptionist's office converted to an iso booth (9' x 14') laid out in an L-shaped pattern, sectionalized by sliding glass doors, Nick 13 and upright bassist Jeff Roffredo could join drummer James Meza in the main live room to lay down tracks while keeping their amps isolated in their respectively assigned rooms and minimizing amp bleed into the drum and room mics.

"They really didn't want to be crammed into booths. They are a very 'live' band, they needed that interaction, that eye contact, in order for them to perform the songs with the right dynamics," McGrath confirms.

Even with Studio A being more than well equipped to handle Tiger Army's needs, there were still a few foreign components that needed flown in before the band could get to work. "I'm not an analog purist," Nick says almost defensively. "I like to cut vocals in Pro Tools — it's a workflow thing if nothing else. But bass and drums in particular just sound better on tape. We had to have those instruments to tape. The natural compression of tape is really flattering to those sources . . . but that's not enough. We had to make sure the signal chain was up to par with what was going in, and what it was going to."

What was missing? The band's go-to mic pres — what McGrath calls "my desert island piece of gear" — the Chandler TG2.

They didn't need just one — they needed lots of them. Modeled after the EMI TG12428 (think: *Abbey Road* and *Dark Side of the Moon*), the band insisted on running almost every mic signal through their TG2s, ascertaining that, in the past, the naturally warm distortion the unit lends to the source is key to nailing the somewhat dark Tiger Army sound.

"The TG2 tends to color the sound a good bit," McGrath explains. "They aren't transparent by any stretch of the imagination. But colorful pres are good for this kind of music. I like the equipment to do some work — making an album is about adding to the natural sound of the band. We weren't recording classical music; it didn't need to sound 'true.' It needed to sound milky, vintage, warm. You can crank the output on the TG2 and run the input really hot and it won't break up. That's what makes this pre great. So Jerry and I brought in ten of them . . . and got ready to run every mic through them."

Assembling Nick, Jeff, and James in the main room of Studio A, Finn and McGrath armed the tape machine with 1/2" Quantegy 456 reels and began cutting the band live, with the intention of later transferring to Pro Tools and making slaves of the tracks. But the plan wasn't foolproof. Straight out of the gate the band began experiencing problems arising from the bass and the guitars not syncing up pitch-wise after the first few takes. Not a day into their sessions the decision had to be made to focus solely on getting great drum tracks first, leaving the guitar and bass parts as scratch tracks to later re-cut.

Sitting behind what McGrath calls a "Franken-kit" assembled by drum tech extraordinaire Mike Fasano, Meza's minimal set up consisted of a 20x22" DW kick, a 8x10" Pork Pie ride tom, a 12x14" Pork Pie floor tom, and 5x14" Ludwig Black Beauty snare — a brass drum long-held by purists as one of the most desirable snares for drummers looking for a crisp, sharp, and dry snare sound. McGrath, needing to capture a very present and cutting kit, was forced to custom-tailor his standard miking approach to bring the best out of a kit made up of somewhat disparate pieces.

"For the kick, I used an [Audio-Technica] ATM25, since it's a


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hypercardioid and that translates to extreme directionality," McGrath informs. "We needed some attack for the articulation, but also a good deal of 'woof'. Instead of setting a [Yamaha] NS10 outside of the kit on the outer head to get the 'woof' and just using the ATM25 to get the attack from the beater head — which I sometimes do — I just placed the ATM25 about half the depth of the drum, pointing a half-inch to the right of where the beater strikes the head. This way we got the attack of the kick, but also the 'boom' of the shell and the head. It sounded punchy, but solid."

When it came to Meza's Black Beauty snare, McGrath again went the hypercardioid route, employing a beyerdynamic M 201 pointed at a 45-degree angle at the sweet spot of the head ("dead-center of the head"). The polar pattern of the M 201 allowed for serious isolation, which gave Finn plenty of room to work with in placing the cutting snare sound prominently in the mix and getting a charged snare sound — a must for a proper punk production.

"B&K 4011s were each placed at 45-degree angles on the toms," McGrath shares, noting that he used them due to their ability to survive enormous SPL. "[They handle] around 130 dB," he continues, "so they can take drums. I prefer condensers to dynamics when it comes to toms. Condensers have a nicer top to them; they are smooth but pronounced. They really bring out the true tom sound. And the B&Ks just sounded *good*. They sound like they were made to record James' toms."

On the subject of overheads, McGrath shares: "I only used the overheads to pick up the cymbals. I've been doing this a lot lately: relying on my room mics to widen up the stereo image."

Many engineers are doing this now that we live in the age of (near) unlimited track counts, stating that it gives them more flexibility when it comes time to balance the drums out in the mix. McGrath further elucidates: "I put a matched pair of Royer R-121s, each two feet right and left, respectively, of where James sat, about one foot above the cymbals, centered over the kit."

Since the R-121 is a ribbon and thus has a figure 8 pattern, both sides equally pick up sound from the source. Turning the R-121 so the back is facing the source yields a brighter response, while the front is somewhat darker and thus better for already bright sources such as cymbals that need a mic to mellow them out.

"A matched pair of R-121s are also what I used for the room mics. I placed them at 180 degrees on each side of where James sat, about 12 feet from the kit," McGrath continues. "I always put room mics as far back from the kit as I can, so I can get as much of the room as I can, even when we aren't going for a real big, roomy sound. Later, you can just take those two mics and pan them hard left and right and really fatten up your stereo image."

"THE RULE WAS: NO GUITARS IN THE STUDIO THAT WERE MADE PAST 1960." -JOE MCGRATH

The room 'pumping' is apparent on *Music From Regions Beyond*, so obviously McGrath and Co. hit the compressors hard and often, right?

"I believe in smashing drums to tape, especially the room mics. So we ran the room signals into the Chandler TG1 compressor, which has a squishy, vintage sound. We set the attack mid to slow, the release mid to fast, and used the device as a limiter. This is a similar sound as you'll get with a [Universal Audio] 1176, but in that case you'll need to set the ratio to 8:1 to get the same results."

Compressing during the tracking process is brave enough — after all, it's not a commitment you can break later down the line — but McGrath didn't stop there: All of the Tiger Army drum tracks were EQ'd to tape as well. "We weren't afraid to commit to a sound right off the bat," Nick 13 adds. "That's part of the punk recording process — feeling confident enough in what you are doing to just do it."

McGrath wholeheartedly agrees. "Mixing should be about balancing what you've tracked, not overhauling your entire sound," he says. "So we'll compress and EQ certain instruments during the tracking. On the drums, for example, I used a Pultec EQP-1A on the overheads as a high pass filter, with a little boost at 10 and 12kHz to make the cymbals sparkle a little. When we heard the snare, we thought it needed some extra 'snap', so we used an API 550 to boost between 5 and 8kHz and add that extra 'crack.' The kick was sounding a little muddy, so we ran that through the EQP-1A, engaged at the 60Hz setting, and cranked the boost and attenuation knobs until the 200Hz range got scooped. It took the 'mud' out of the kick, and left some low-end room for another sound to live in.

"It was going to need it at some time or the other. Why not just do it now?"

"The rule was: No guitars in the studio that were made past 1960," McGrath says with a laugh. Nick 13 concurs: "That dry wood that you'll have in an old guitar is the key to the vintage guitar sound. So I brought in my collection, and Jerry brought in his, and we went to town."

With the drum tracks now laid to tape, the Tiger Army guys forged onward, cutting the guitar tracks before Roffredo's bass lines in an attempt to take advantage of the relative pitch regularity of the guitar as opposed to the upright bass, leaving Roffredo with a more concrete point of tonal reference from which to work instead of putting the onus on Nick to match the somewhat finicky nature of the upright bass.

"I used to play an early '60s Gretsch Anniversary," Nick tells, adding that the hollowbody guitar caused massive feedback problems in a heavily amplified environment and thus was retired from both the stage and the studio. "I went out and got a '57 Gretsch Duo Jet because it had a semi-hollow enclosed chamber in the body that gave me some of what I loved about the hollowbody tone without all the problems. Jerry also brought in a '59 Gretsch 6120 Chet Atkins model and those two guitars became the primary guitars for *Music From Regions Beyond*."

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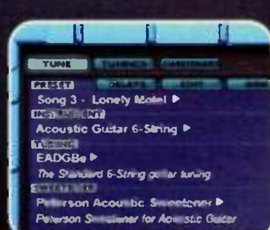

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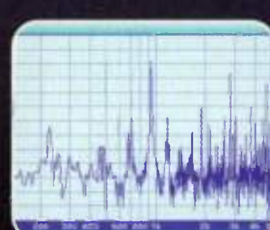

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PSYCHO ANALYZE

Needing a guitar that sounded more aggressive, more full, more bottom-heavy, Nick fell for Finn's '59 Les Paul Jr. which was used to track the album opener "Prelude: Signal Return", the almost-metal "Hotprowl", and the band's radio-friendly hit single "Forever Fades Away."

"I swear by Fender Tonemaster heads and the matching 4x12 cabs live, so we brought those in to the studio," Nick adds, hiding not his affection for what has long been regarded as the ultimate surf/rockabilly guitar amp. "We ended up using the Tonemaster in conjunction with a Fender Bassman, blending both signals and comping them down to one track to get a sound that was clear but appropriate for everything from the cleaner rockabilly parts to the punk guitar tones. And for everything clean, we just went through a silver-face Fender Super Reverb."

Having Nick's tone dialed in from the source ("all the distortion is from the amp," he says), McGrath threw his trusty R-121s up on the cabs and started getting to work. "I placed the mics slightly off-axis from the cone, about four or five inches from the grill — so about six inches total from the actual speaker itself," he explains. "This was great for the rhythm tracks . . . you'll usually get too 'washy' of a sound adding room mics in on rhythm tracks, and they wanted a focused sound. Plus, since the R-121 is a figure 8 pattern mic, you get a little kick back from the room as is and that's enough to fatten the sound up sufficiently."

Lead guitars were a different beast altogether. "We wanted some room sound for those," Nick says.

Moving to just the Tonemaster, McGrath says: "I put a Neumann U87 18 inches back from the R-121 and a Røde NT2 54 inches back from that to avoid phase issues. The mics were placed in a direct line from one another, with their diaphragms lined perfectly up. Since we weren't measuring, I could just pop the mics out of phase, move them until I got the thinnest sound possible, and then pop them back in phase and we would be good. We mixed both mics up just a bit under the R-121 track and then comped all three tracks down to one."

All mics were run into McGrath and Finn's massive TG2 racks, straight to tape, with no compression added. "The natural compression from the tape is enough for guitars," McGrath says matter-of-factly. "I don't want to slap the tape too hard by putting a compressor in the chain during tracking. Not for guitar at least. I wanted to leave Jerry some room to mess with that in the mix."

But this wasn't indicative of any "cold feet syndrome" on behalf of the band or the production team. Again, the troupe

HOME COOKING

Digital-audio workstations offer near unlimited track counts, and with such a bounty of sonic space available, it seems silly to commit to anything before the final mix. Go ahead — ecord gazillions of tracks, critically evaluate every single one, refine anything that's not perfect, and then comp the hippest performances from millions of little pieces of music. Be sure to spend eons doing all of this assessing and tweaking, because you certainly don't want one less-than-stellar moment spoiling the majesty of your track's total running time. Nope. You must be sure every element is absolutely flawless. It's like living with your significant other for 100 years to make sure their every utterance and personality quirk bring you nothing but bliss. If the century of scrutiny has failed to reveal one unseemly trait, you can safely schedule that wedding for Year 101. Until then, hold back on that commitment.

Sound romantic?

Well, some *will* find this scenario appealing. But to craft the ballsy psychobilly of Tiger Army, you're going to need to find love fast.

Part of the appeal of punk and its subgenres is that energy, passion, and mission are prized more than chops, technique, and sonic perfection. Some of the most influential '70s punk singles totally sound like crap — even compared to what a newbie with a home studio can produce today. But winning a Grammy or selling enough records to behave badly at Cannes was never the point. It was all about the fire of the performance — which is something the Tiger Army team understands.

Many of the original punk tracks were at the mercy of finance — meaning that the budgets were often so miniscule that they were recorded in small studios as quickly as possible. But as Nick 13 reveals, the concept of tracking live, tracking fast, and tracking in a vibey room also harkens back to the era of seminal rock and roll rebels such as Eddie Cochran. Heck, even the Brian Epstein-sanitized Beatles recorded their first record in less than ten hours.

The point is that a group of musicians who can really play fiercely — really shake every wisp of soul from their instruments and voices — are often best served by documenting the performance with as little fuss as possible. Multiple retakes, vast numbers of tracks, massive signal processing, and epic mix sessions will likely kill that messy and elusive spark of live fervor, leaving you instead with very well-recorded songs that are emotionally as flat as binder paper. To achieve the propulsive impact of live-in-the-studio passion, you must commit to sounds, EQ, and limitations. You must conceptually cripple your DAW, and most of the advances of the past 50 years.

So, if you're game, here are three suggestions for bringing the Tiger Army method to your own personal-recording scene.

It's Not About the Gear

Nick 13 and *Music From Regions Beyond* producer Jerry Finn had some very nice tools at their disposal, but the punk ethic is more about the process. So don't fret about your piddling microphone collection, your lack of vintage mic preamps, or any other gear-related snafus. Just start by rehearsing your band (or session musicians if you're a solo act) until everyone can play your hopefully well-crafted songs with confidence and style. Of course, don't over-rehearse everyone into a stupor. You want the songs to sound loose, but powerful.

Strangle Your Options

In order to ensure that I can't get paranoid and track a project into oblivion, I severely limit my tracking options. Drums are recorded to no more than four tracks (kick, snare, overhead left, overhead right). Guitars are confined to stereo rhythm tracks, and two solo/sweetening tracks. The bass gets one track, and keyboards no more than two. The basics — drums, bass, and rhythm guitar — must be recorded live, and vocals get one final track (although I allow myself up to three complete tracks of vocal performances from which to comp a master). Background vocals get a stereo pair of tracks. That works out to about 16 tracks — it's back to the '70s!

Commit to Sounds As They Go Down

The Tiger Army team tracked with EQ — a practice that cuts the crap and gets sounds down fast. When you limit your options to around 16 total tracks, it should force you to envision the overall sonic landscape early (as you pretty much know immediately what elements you'll be dealing with), making EQ choices less than scary. This is also where tracking basics all at once is key, because you can easily EQ the drums and bass to work with each other, rather than waiting to construct basics from various different performances. Tracking guitars with appropriate effects and EQ tweaks is usually a blast for the player, as it almost simulates the feel of a live gig.

If you've done everything right, you should be able to do a final mix by merely moving faders and selecting a vocal reverb and/or delay. Everything else should sound like a record. This is the true beauty of sonic commitment — you don't have to worry over details *after* the wedding, so to speak. Once you say "I do," you're done.
—*Michael Molenda*

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were daring — if not borderline indignant — when it came to tracking with effects and EQ.

"We're going for the vintage sound, but we're also following the vintage *modus operandi*," Nick says. "No plug-ins; no emulators of any kind. The guitar delays on the album are mostly from my Boss DM-2, set to approximately 150ms to get that slap-back, rockabilly sound."

"All the reverbs are from an Echoplex or an old Fulltone Tube Tape Echo. The settings are the same as they are when they are in the practice spot," McGrath adds. "We didn't reamp and then apply effects; we didn't save it for the mix. We weren't scared to commit to the sounds they came up with when writing the album. Plus, it's never the same when you add effects after the tracking. It doesn't sound like it does when you run a guitar into a pedal, through an amp, out of the speaker, through the air, into the mic, through the pre, and then to tape. It doesn't sound as real when you do it after the fact — it sounds *fake*, like the guitar sounds they used to get on '80s pop records.

"We needed to open up the top end on the guitars. And we needed to do it before the signal hit the tape," McGrath laughs. "So we gave the EQ a boost between 2 and 5kHz to let that top end breathe. Then all we had to do was pan the rhythm guitars hard left and right to get a real wide guitar sound. We sent the solos straight down the middle so they really hit you in the chest, with the levels matched to the lead vocal so they cut through the instrumentation . . . and then it was on to the bass."

"The upright bass in psychobilly is a direct atavistic link with early rock and roll," Nick remarks. "A lot of the rockabilly and honky-tonk in the earliest days didn't have drums. So there's a holdover in style from when the upright bass provided the rhythm for the song, when it served as the primary percussive device. This goes back to the Scotty [Moore] and Bill [Black] Sun Studios stuff in 1954.

"An upright bass interacts differently with a band when drums are factored into the equation, when you introduce cranked amplification into the picture and try to combine it in the way it was played in the Presley Sun Sessions."

Recording upright bass is notoriously difficult when tracking what is by all definitions — regardless of their diverse tastes and ambitious sonic palette — a loud rock band. Upright bass is a rather quiet instrument, and when laying down tracks live in the studio it almost always must be amplified. This brings in a host of other problems: It's a fretless instrument that reacts wildly depending on the player's attack and nuances that aren't meant to be amplified are made suddenly glaring when the bass is cranked. This results in a loss of perspective, and it's easy for the player to play less than dynamically when they are forced to compete with louder instruments. It's a delicate musical tool, and the Tiger Army guys were not of the mind to sacrifice clarity or low-end just to get the bass lines to tape. So with Roffredo bringing his vintage Kay upright into Studio A for the *Music From Regions Beyond* sessions, McGrath and Finn were forced to figure out a way to accommodate the rather "overparticular" instrument.

"We needed to get the sound of the body of the bass," McGrath explains. "So we set up a [Neumann] U47 and a Royer R-122V — which is a vacuum tube ribbon — pointed directly at the f-hole, since this is where you'll get all your low-end information. Personally, I felt the ribbon captured the natural sound of the bass better than the condenser. It may sound like a cliché, but the ribbon made it sound like your ear was right up on the body of the bass. We wanted that effect."

Capturing the percussive effects of Roffredo's left hand slapping against the fret board were of the utmost importance to the band, so McGrath placed a B&K 4011 approximately 18 inches dead-on from the midpoint of the neck. "It really brought out that 'tic-tac-tic-tac' sound of the fret board," McGrath says. "To me, the 4011 is just a great all-purpose mic. It's smooth top end really worked well in this application because it kept the percussive sound from being too harsh."

Roffredo's Kay, as Nick tells us, is also outfitted with two custom upright bass pickups — one fastened in a bridge position and another in the neck position. Both pickups have separate outputs and are actively engaged during the tracking process. "The bridge pickup brings out more of the body of the bass sound," McGrath explicates, "and the neck pickup gets more of the neck action and high-end." This produces an effect similar to what one would get if they decided to mic the upright where the neck and the body meet.

[redacted]

"Both signals run into a dual DI with separate outs, with the bridge pickup signal running into an [Ampeg] B15 and the neck pick-up into an old Comet 40B guitar amp, loaded with 2x12s," McGrath further elucidates. "The Comet has a great 'broken' sound when you run a bass through it. If you soloed the track, you could swear something was loose and rattling around inside — it sounds like two pieces of metal clacking together every time the bass hits the head. It was a great compliment to Jeff's style on the neck."

Placing another B&K 4011 on the Comet 40B and a DPA 4041-T2 omni-directional tube mic on the B15 (due to it's ability to handle up to 144 dB SPL and because the tube design adds a small amount of color to the source), the two signals were put through a two channel Tube-Tech LCA 2B compressor, used as two mono compressors.

"We compressed the R-122V and the 4011 from the physical bass itself as well," McGrath adds, describing the signal path as "through the TG-2s, of course, and into an EAR 660 compressor — which is modeled after the old Fairchild 660. The controls on the front panel are pretty arcane, and it's a really variable unit. You have a ton of control, but you are also stuck just twisting knobs until you find something that sounds right.

"I was very light with my compression, I can tell you that much," McGrath continues. "I don't suggest crushing a bass too much before it hits the tape — it's easy to lose valuable headroom if you do that. So a light compression and a cut in the 200–250Hz range for the R-122V, which was to clean up the murkiness you get when miking the f-hole, was all we needed to do. Then we just assigned each mic to a separate fader on the API, balanced the tones, then comped them down to one track for the mix."

"Working in Pro Tools, in my experience, is just plain faster," Nick 13 explains when asked why the tape was packed up and Sound Factory's Studio A vacated. "We knew the instruments needed to be put to tape, so we put a lot of our resources there, and then headed over to Paramount Studio B and cut the vocals in Pro Tools HD3."

While certainly not the most luxurious of the Paramount facilities, Studio B — with it's 40-input SSL 4000 E/G, was adequate for the band's needs. "It's a smaller studio," McGrath says, "but it had a large vocal booth — large enough that we didn't feel like we were tracking in a closet. And it was fine for the few overdubs we still had to do."

First things first: McGrath and Finn had to find a mic that suited Nick's dark, tenor vocals. Though the BLUE Bottle had been a go-to vocal mic on past sessions, the trio decided to shoot out an AKG C12, Neumann U47, and a Neumann U67 to see if — given the slight sonic change in direction for *Music From Regions Beyond* — perhaps an old standard vocal mic wouldn't be appropriate.

"We ended up going back to the BLUE," McGrath tells. "The mic has great clarity. It's true to the source, but it adds a lot of body . . . plenty of 'chest' sound. It's not nasally at all.

"Nick's voice is low enough that I had to shelve the low-end a tiny bit," he adds. "But the BLUE's sound is so crystal clear that I didn't need to boost any mids or highs. The color of his vocals is due to the TG2, which has a real Neve-like quality to it, especially when the source is vocals."

From the TG2, Nick's vocals ran into the EAR 660 before hitting Pro Tools. "Even though we were recording into Pro Tools, we wanted the front end to be all-tube," Nick discloses.

Expounding on this recording philosophy, McGrath comments: "Adding the 660 to the chain put one more colorful tube component in the line before we got into the digital world. It's the tubes and the circuitry of the components that makes Nick's vocals sound as warm and retro as the other instruments. We knew we could make this work in Pro Tools, and that it would make our lives easier. We could track more on the fly, comp easier, and play around with vocal effects without committing them to tape, which is too expensive as it is."

To make Nick's vocals more immediately manageable and dynamically consistent, McGrath compressed down 6 dB with the EAR 660. "I just got them sounding good; I didn't bother compressing too much during the tracking stage," he states. "Plus, Nick didn't really need it — he's a great singer and he knows how to control his voice. We didn't have to cut and paste everything, crush his vocals, or auto-tune all his parts. In fact, most every vocal part is a single take, a single track. We'd only double the vocals for the choruses. Otherwise we'd go with one vocal track. I find that doubling all your vocal tracks to get a fatter sound

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is a good way to keep the choruses from hitting as hard as they could. I definitely suggest keeping the verse vocals singular and only doubling for big choruses."

It's pretty bare bones, pretty punk rock. And that's important to the band given their roots. Taking it one step further in vocal department, and giving a huge nod to the old New York Hardcore and Oi movements, the band decided to bring in a group of friends to Studio B and add the icing to the proverbial cake: sing-along gang vocals.

"We just piled the band and however many of their friends we could fit into the vocal booth," McGrath laughs. "We put the BLUE about a foot above the tallest head, pointed down at a 90 degree angle to get that 'crowd to the stage' perspective. We wanted a more massive group sound than we could get with just the bodies we had, so we did three takes and then Jerry just mixed them all together so it sounded like a huge mass of people."

Classic guitars, vintage upright basses, small and punchy sounding drum kits, and minimal, crooning vocals — it's hard not to hear the psychobilly in the "new" Tiger Army sound even if there is a fair amount of New Order and Depeche Mode popping in now and again. There was one important element, however, that the band felt they needed to round out their sound on the record: pedal steel guitar.

Bringing in long-time Tiger Army contributor and world-renowned pedal steel player Greg Leisz to play on the country-inspired heartbreaker "Where the Moss Slowly Grows" was tradition, as Nick 13 explains: "We were put in contact with Greg back in 1999, and he's played on one song off of every album we've recorded since then. He's an incredible musician, very well regarded in the industry. And he's a staple of Tiger Army records now."

Wielding his trusty Fender Champ pedal steel and carrying a Fender Blackface Twin Reverb in tow, Leisz set up in Studio B and banged the section out in no time at all. "Greg is such a great player that there is nothing you can do to make him sound bad," McGrath laughs.

"We liked the sound we had achieved for Nick's clean tracks, so we kept the signal path the same," McGrath tells. "I put the R-121 five inches from the grill of Greg's Twin Reverb, slightly off-axis, and then ran that through the TG2. We had our sound pretty much right off the bat. With a pedal steel, the player is very much in control, dynamically speaking — he's controlling his sound with the volume pedal. There's not a lot popping up that you have to crush down with compression. Still, I put a TG1 in line set with a ratio of 2:1 with 3–4 dB of compression just to keep things nice and even. We sent him straight into Pro Tools from the compressor and let that be that. It was *real* easy."

"Easy" is perhaps the best adjective one could muster to describe the sessions for *Music From Regions Beyond*, at least relatively speaking. From day one, Tiger Army has experienced their fair share of hardships, from personal tragedies befalling band members (such as the shooting of ex-drummer Fred Hell), to regular turnover in the player department, to never quite realizing their true intentions in the recorded medium. This time, though, Nick 13 has little, if nothing, to lament.

"When I started this band back in 1995, almost nobody in the states was familiar with this style of music. There was no crowd for psychobilly here," he says with a sigh. "I simply couldn't find musicians who were interested in playing this style, let alone a large audience. It was frustrating — I wanted Tiger Army to be a full-time vocation. I was left in the position of recruiting every friend I could just to get through a recording session. There have been lots of people in this band that have fallen into the session category, and there were plenty — once the ball got rolling — that I intended to have stay as permanent members but just couldn't. But when I found Jeff and James, I was immediately struck by their chemistry as musicians. Everything quickly coalesced and I knew this was the strongest lineup I ever had. And I knew we were going to make the strongest album we ever had. If I could think it, they could play it. And if they could play it, Joe and Jerry could record it. It really helped push the music forward and now, for once, we are where we need to be." EQ

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by Merrick Angle

HOW TO CREATE AN EFFICIENT DAW SETUP

Pops? Heat? Hum? Follow these Tips for DAW Nirvana

For those who remember the days of the cassette-based 4-track, the shiny modern world of the Digital Audio Workstation is a godsend. Whether computer-based or standalone, DAWs have benefited from faster processing, cheaper memory, and competitive pricing — even the average home computer can handle serious recording.

While it's easy to get intoxicated by this brave new world, there are still problems that hinder even some of the fastest machines. Anyone who's used a DAW will likely have encountered glitches, clicks, and digital pops, as well as other problems like overheating and latency.

To obtain maximum performance, pros often dedicate a computer to audio only (and never, ever connect it to the Internet). It's a fine solution, but many folks have to buy the best computer they can afford, use it for everything, then hope for the best when it comes to recording. While I'm a fan of the "fingers crossed" approach, there are still some basic steps we can take to counteract the most common DAW problems. Before proceeding, though, I'd like to thank Des McKinney, a Toronto-based engineer/producer, musician, and the man behind the excellent home recording blog www.hometracked.com, who contributed numerous tips and techniques used in this article.

DON'T BE LATE!

Latency is the delay, caused by signal conversion and processing, between playing something into your DAW and hearing it come out of your monitors or headphones. Some level of latency is inevitable, but anything more than a few milliseconds is potentially noticeable and can throw a real monkey wrench into the works — especially with playback, overdubs, and intensive recording. To minimize your overall latency:

Fig. 1: In Cubase, you can adjust the number of sample buffers (being set here to 192 samples) as well as disk buffers; the latter affect audio streaming performance.

- **Update your audio interface drivers, whether ASIO, Core Audio, or WDM.** Improved drivers may not lower latency *per se*, but instead, provide more efficient operation that allows for lower latencies.
- **Choose the correct audio buffer size.** Giving the DAW more time to process audio (Figure 1) stops pops and clicks, but increases latency — so find the "sweet spot." Be realistic, though: There's a tradeoff between latency and CPU usage, so you may need to concede some latency to get acceptable performance from the rest of your system.
- **Reduce background interference.** A friend was vexed that his new, costly PC seemed to be underperforming. Then I found out he was running five other applications simultaneously, and even chatting on MSN messenger while trying to record! While most of us wouldn't dream of doing this, it illustrates a point: What is your DAW running as you record, mix, and master? Running one LAN (or DSL) connection can cause the same CPU hit as running two average plug-ins. Find out what's going on when you record, as some arcane background activity can push an already stressed DAW to the limit; to do this with Windows XP, right-click on the Taskbar and call up the Task Manager. This shows all programs that are running. If there's something you don't need ("ComputerCheckup.exe," anyone?), right-click on its name and select "End Process."
- **Disable any wireless connections.** Right-click on My Computer > Properties > Hardware tab, and select Device Manager. Identify and disable all wireless connections except for ones involved in recording (*e.g.*, Frontier Design Tranzport). As Des says, "Should your DAW be connected to the Internet anyway?"
- **If your machine was ever connected to the net, scan for viruses, spyware, and Trojans.** These can cause major slowdowns if present. Then, as you're no longer connected to the net (right?), shut down any software that scans for viruses, spyware, or Trojans. These are constantly working your hard drive and degrading performance.
- **Disconnect unneeded peripherals.** You probably don't need a web cam running in the background.
- **Manually switch off any background activity.** This includes automatic backup utilities like System Restore, scheduled updates, and Ethernet or Bluetooth networks.
- **Use an AGP graphics card.** If you're stuck with a PCI card, try different resolutions, color depths, and hardware acceleration rates.

You may encounter audio clicks when running at full acceleration.

■ **Freeze soft synths.** Software synths take a lot of CPU power, so use your host software's freeze function to "disconnect" them from your CPU.

TO PLUG-IN OR NOT TO PLUG-IN?

Plug-ins and soft synths are some of the most powerful tools in any DAW arsenal, but they take a toll on the CPU. Limit their usage until mixing, at which point you can increase latency; latency is most problematic when recording, because a few milliseconds of delay can be disconcerting. While mixing, though, a few milliseconds of delay when moving a fader is no big deal, and higher latencies leave more resources for plug-ins and soft synths. It's good to minimize plug-in use anyway while recording; faulty plug-ins are some of the biggest culprits in generating pops when laying down tracks. If problems persist, disable them all, and go through the plug-ins one by one until you find the guilty party. Then, check for any updates.

Also, be ruthless: How many plug-ins and synths do you actually *need*? For best results, use mono and simplified versions of plug-ins where appropriate. Waves plug-ins, for example, come in stereo and mono versions, and the equalizers have 2-, 4-, and 8- band versions. Using an 8-band stereo equalizer on a mono kick drum track that needs a single EQ cut is wasteful. Also, use sends and buses where possible to share a plug-in among multiple tracks, rather than inserting the same plug-in on multiple tracks. Lazy overuse of plug-ins could affect both DAW performance and your artistic vision, making your computer slow and your tracks flabby.

Fig. 2: The Radial Pro RMP is a direct box that also includes re-amping options.

FEELING HOT, HOT, HOT

Laptop recording is great in theory, but an older laptop might not have the requisite power for music; and while newer ones have become sleeker, slimmer beasts, this crams their components into even tinier spaces. The result can be overheating, so make sure the all-important ventilation slots are unobstructed, and the fan is free of dust or dirt. There are software-based ways to combat overheating (other than using minimal CPU power) like smcFanControl for Mac or Speedfan for PC (www.almico.com/speedfan.php). These allow you to monitor a laptop's temperature and adjust fan settings accordingly. Floating the laptop base so there's an air gap underneath it, using something like the Griffin technology Elevator (www.griffintechnology.com), is a good idea; some thriftier DAW users glue pencil erasers to the laptop's corners. Finally, when propping a laptop on your lap, make sure your legs don't block any of the vents — and never put a laptop on a rug or other soft surface.

GROUND CONTROL

A ground loop is electrical interference, typically hum, that occurs when one piece of gear can "see" two paths to ground. For example, suppose an amp that draws a fair amount of current sees ground through the third pin on its AC cord but also through the ground shield

FIREWIRE: "X," THE UNKNOWN

An increasing number of DAW users are forsaking traditional PCI-based cards and using FireWire or USB interfaces (even though cards offer a slight advantage in terms of speed). But FireWire doesn't always work perfectly; check out any recording forums, and you'll find a plethora of posts from people having problems with FireWire audio devices. Often this has to do with pilot error, but occasionally there are mysterious, intractable problems that no user or company can figure out. In fact, just because a FireWire device works without issues on the majority of systems doesn't mean it will work on yours. So, following are some tips to promote cordial relations with FireWire.

- **Follow installation instructions to the letter. Sometimes you need to install the software first, and sometimes you need to connect the device first.**
- **Connect FireWire devices while the computer and device are powered-down. In theory, you should be able to hot-swap FireWire devices; in practice, this isn't always true, with potentially disastrous results (*e.g.*, a fried motherboard).**
- **Power-up the FireWire device before turning on your computer.**
- **Use an external AC adapter instead of bus power.**
- **Check the manufacturer's website for recommended FireWire chip sets for a particular device. TI (Texas Instruments) is generally recommended, NEC usually is not.**
- **Avoid using combo FireWire/USB cards. Don't say you weren't warned.**
- **Mac owners, upgrade to OS X 10.4.10.**
- **If you own a PPC Mac, check out the power user software tips in last month's issue for details on using the CHUD utility. If you own an Intel Mac running an earlier version of OS X, try prayer.**
- **Dedicate a FireWire port to audio devices, and don't run other devices on the same port.**
- **Take forum posts about "This interface works perfectly" or "This interface is a piece of garbage" with a grain of salt. An interface can work perfectly on one computer yet not work on a seemingly identical computer. Try before you buy, or buy from a vendor with a liberal return policy.**
- **Use a high-quality cable. If the cable got crushed or stepped on, it might not work.**
- **If all else fails, there's always USB 2.0.**

Finally, if you do encounter problems, don't bang your head against the wall trying to troubleshoot without first checking the website of the device's manufacturer. There may be known issues that can be fixed with a simple driver download, or by disabling a conflicting device.
—Craig Anderton

of a cable connecting to a sensitive piece of gear, like a mic preamp. If a voltage differential exists between these two paths, some signal might be induced into the mic preamp's ground, and amplified to an audible level.

One potential solution is to plug all your sensitive gear into a single, properly rated AC outlet to minimize voltage differences. If that's not possible, at least try to use fewer power outlets, or connect everything to a single power source (via extension cords and power bars) so that they share a common ground. Don't think that two sockets in the same room are necessarily running on the same circuit; this often isn't the case.

As we are dealing with electricity, *be cautious*. Avoid the temptation to break off the ground pin from a three-conductor AC cable, as the ground provides a needed safety feature should a malfunction occur that makes the gear's chassis go "live."

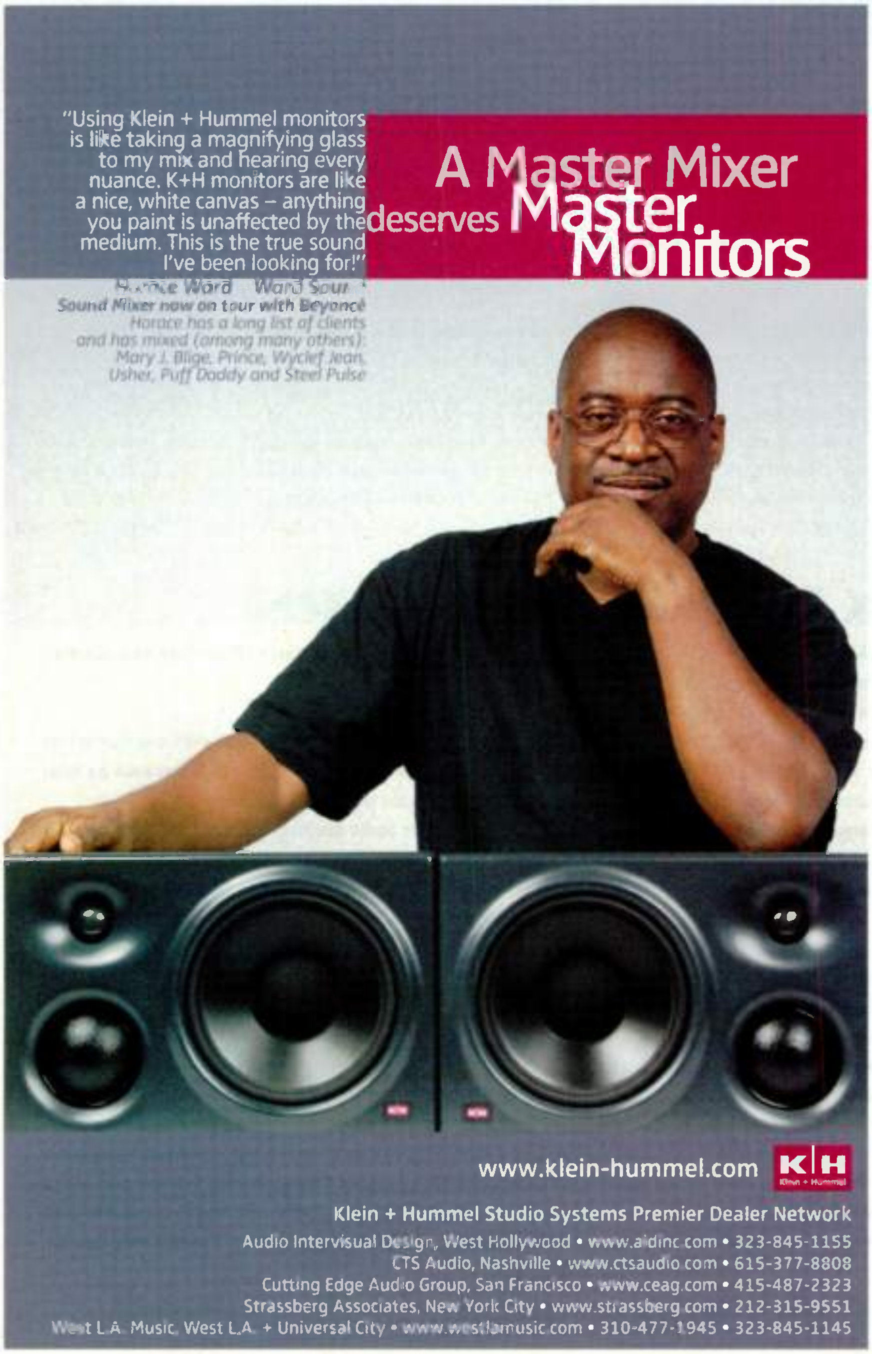
If the buzzing persists, isolate the equipment that's causing the noise. Disconnect everything, then reconnect the components one at a time until the noise re-appears. The last piece added is usually the one causing the problem. Try moving its power lead to another outlet.

RF interference is another potential source of buzzings. One of the most common reasons for picking up RF is oxidization on cable connections. Slight corrosion can actually create small crystals on the cable, turning your jack into a primitive — but functional — crystal radio. Plug and unplug connectors repeatedly to reduce corrosion, and use products like Caig's DeoxIT.

MAKE THE CONNECTION

Unbalanced connections could be another factor in adding noise or interference. Common sense dictates using the shortest cables possible, whether balanced or unbalanced; and don't drape them over devices like transformers. Balanced connections are more costly, so if you're on a budget, put them where they matter: mics and other low-level signals. A hot synth output going through a few feet of cable to a mixer will likely not cause problems. But if you can afford balanced lines, go for it — that's one less thing to worry about.

Still living with the hum from hell? Try DI boxes, which can help eliminate ground loops, give you a balanced input, and cut down on noise. For ground loops that occur between a keyboard and a mixer or DAW interface, try running the keyboard first through a DI box with ground-lift switch. Costs vary dramatically; a Behringer D120 will set you back about $30, whereas something robust like the BSS AR-133 retails for around $150. Radial Engineering makes a wide range of direct boxes, from "plain vanilla" types to models with routing and re-amping options (Figure 2). But if you find that you need to add lots of DI boxes, do the math: You may be better off getting one really high quality DI for when it's needed, and putting the rest of your money into a new mixer with balanced inputs and a few quality XLR leads.



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ARCHIVING YOUR MUSICAL HISTORY

Don't Let Your Legacy Slip Away . . .

by Roger Franz

You probably have some treasured music sitting around in "obsolete" formats — reel-to-reel tape, cassette, vinyl, or even early digital formats like Sony PCM-F1 or ADAT. You don't want that history to waste away. While you can still get machines for working with older media, their quality is on a downward spiral and prices are increasing. If you have unique collections that cannot be regenerated in new media, it's time to convert them to more modern formats.

We'll assume that you're a musician, not a professional archivist, and therefore have limited time and resources. Sure, you could import your material into a digital workstation and spend endless hours cleaning up, remixing, and EQing it to perfection. But you don't want to be a slave to the past, either. Here's the approach I took to converting a sizeable number of analog recordings while retaining the heart of the original sound, minimizing any negatives, and doing so in a reasonable amount of time.

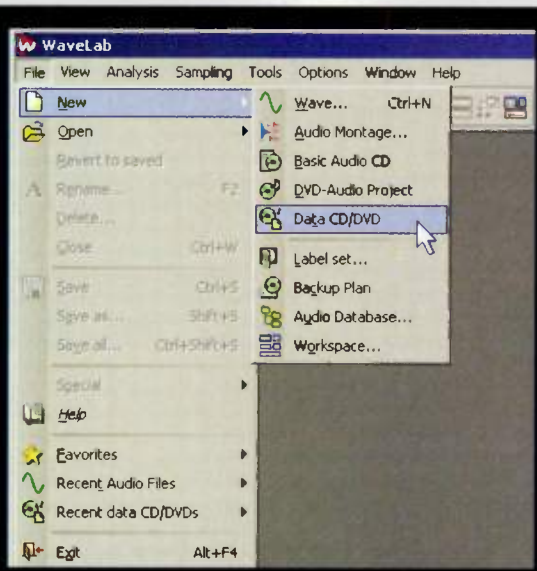

Fig. 1: Wavelab allows creating audio CDs and even DVD-Audio discs, but can also create archival data backups on CD-ROM and DVD-ROM.

PROJECT PLAN

Start with as clean a recording as possible; don't expect to fix it in the digitization process. If you have original multitrack tapes you can remix, so much the better, but you can only optimize what you have. This isn't about remixing legendary Jimi Hendrix lost sessions.

Be wary of consumer-oriented products that claim to take your old LPs or tapes and convert them to CD, as they offer little or no ability to use EQ or other editing options. For a more pro job, hard disk recorders are an affordable way to import analog media.

Connecting your analog gear to a digital recorder is still "wired technology," so use good cables. You'll need a phono preamp when running a turntable into your system; quality among these varies dramatically, so seek out the best possible. As you monitor the original recording, play with options like level, EQ, compression, limiting, and the like to see if you can enhance the original recording. Most digital recorders have presets with clever names that provide a clue to the overall sonic objective, but use these sparingly — you can always add more effects, but you can't take them away.

A HISS-TORY LESSON

When transferring analog tapes, the enemy is hiss. Pro reel-to-reel decks had decent specifications for frequency response and signal-to-noise ratio, but cassettes are a different matter. Most used a noise reduction system (Dolby B, Dolby C, or even dbx), so find a playback machine that offers the same noise reduction scheme used while recording. Turning off the noise reduction will give more perceived high frequencies, but the price will be more hiss.

Vinyl records will have pops and scratches but the turntable itself will likely contribute "rumble," where the motor sound gets into the audio, and hum (50/60Hz and its harmonics). Most turntables have a grounding post; creating a secure ground between your turntable and preamp will help reduce hum. If not, you may need to use a steep low cut (high-pass) filter to reduce low frequencies, particularly those below 50Hz in the case of rumble (this won't affect bass much, as mastering engineers often rolled off frequencies below 50Hz during mastering to make more needle-friendly grooves).

MAKING IT BETTER

Digital editing is a boon to those old recordings. Fade-ins and fade-outs can reduce noise preceding and following the music itself. You probably won't need to copy and paste if the core song is already in place, but if part of the song is seriously damaged, you may be able to find a similar section and replace the damaged part. Digital filters with steep slopes can remove subsonics from vinyl, or hiss from tapes. Many times highs were deliberately cut during mastering, so you can often cut above 10kHz and eliminate hiss or crackles without affecting the music.

Once the original analog sound is digitized, not only can you add effects, the adventurous can overdub new parts. However, if you make any significant changes, retain a copy of the original digitized material. That way you can live with any changes you make, but go back to the original if you decide later you went too far.

All the techniques involved with mastering apply here; even if the material was already mastered once, it may not have been under optimum conditions (for a useful review of the mastering process, see "Mastering in the Digital Age" by Dave Kutch, 2/07 *EQ*). Don't get carried away; theoretically, you're just changing the format. Still, the power to shape the final sound in ways not previously available to many musicians can yield many sonic benefits.

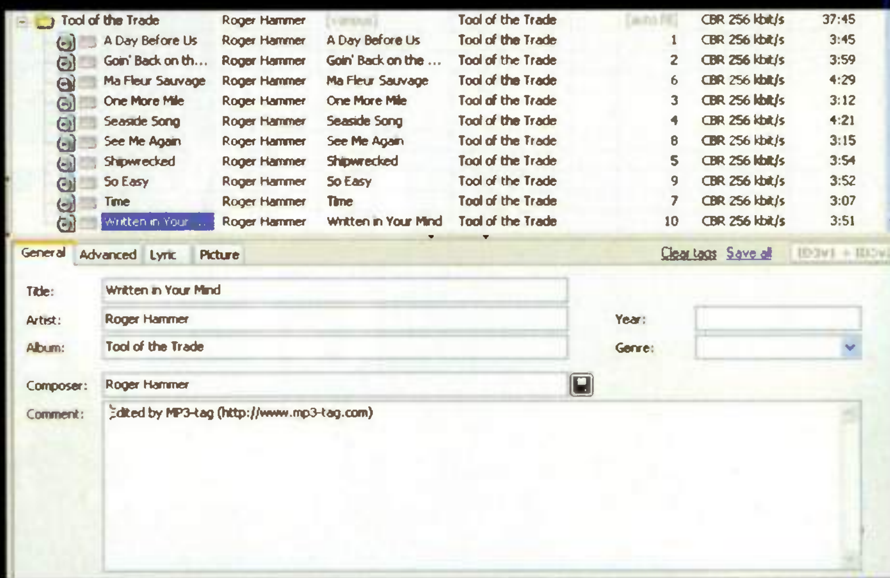

Fig. 2: MP3 Tag Editor allows adding information about MP3 files such as the title, artist, composer, various comments, and the like.

BURNING AND BEYOND

Next, transfer the file to a more permanent storage medium, such as CD, DVD-ROM, removable hard drive (but spin it up periodically or it may "freeze"), RAID array, etc. The key to this step is to watch your peak levels. While analog was famous for soft distortion, digital is not forgiving when overdriven. This is particularly problematic when trying to get a hot enough signal to be in at least the same ballpark as today's ear-burning, highly compressed levels. But err on the side of caution. Super-hot recordings can go out of fashion just as easily as they became the fashion; you don't want your recordings to sound "so 2005" in a few years.

To store the results as an audio CD, many programs — not just stereo editors, but also multitrack hosts — let you burn audio CDs. Some, like Sony CD Architect, allow sophisticated sequencing and processing options. Look for a program that handles CD Text, as you can store info on the title, artist, etc.

After archiving your material, you're still not done because you need to make a backup. As the old saying goes, "digital data isn't real until it exists in at least two places."

You may also want to go to create additional formats, like MP3, as they are a common way to exchange songs and don't take up much space. There are many options, ranging from free (Windows Media Player or iTunes) to sophisticated (two-track editors like Sound Forge, Wavelab, Peak, Audition, and the like; see Figure 1). I used Easy CD Creator.

Consider ripping the songs in two versions: the highest possible bit rate (which will sound close to CDs) and for MP3s, the "consumer standard" of 128kbps. Low rates are useful to post on the Web as streaming files because they download rapidly; I made these versions free, but asked for a donation from those who wanted to download the higher quality files.

Also consider creating a third version at 256kbps, which sounds fine in a normal listening environment. A typical song weighs in at about 5MB, which isn't much memory given the capacity of today's hard drives and flash RAM.

As I wrote all the original material, I also made sure that the song title and artist tags in the MP3 files were updated accordingly (although you can add these tags to any material you convert to MP3; other data compressed formats also allow for tagging). If your ripping software can't edit tags, MP3 Tag Editor is an inexpensive option (Figure 2).

LISTS AND LYRICS AND LISTS . . .

I already had my song lists in both Word and Excel, organized into albums with their own titles. But when they're all in one MP3 jukebox, it doesn't make any difference what album they came from! You can shuffle them any way you want. Surprisingly, alphabetical order flowed rather well (another reason to update the tags).

Lists are useful to make menus and hyperlinks on websites. One page can link to others and if you already have the menu lists, much of the work is already done. For example, an artist's works may begin with album titles, then go to individual songs, then further to lyrics or artwork in each album.

When printed out, your lyrics are useful cheat sheets for rehearsals, copyrights, or gigs. Once on your PC, you can burn lyrics along with the songs onto a multimedia CD. You could also include graphics or movies. I scanned all my artwork and graphics into JPEG files so they could be filed and copied like everything else; videos would be next.

For me, the next step beyond creating CDs and MP3s was "digital publishing" — posting everything I digitized on my website. Then you want to get into search engine optimization, so people can actually find you . . . but that's a whole other topic.

Regardless of your project's scope, keep working until it's done. When I ask myself what I've done with my life, I can answer that I have indeed done something: I've unwound my reels of rolling tape, and organized piles of words on paper, to put them in digital media. In my case, it was all my own material. But the same basic method works for anyone who wants to organize piles of their favorite stuff into compact and long-lasting formats that can be enjoyed for years to come.

It was amazing that everything I had worked on for years could be handled in the end like just so many files. I was even more amazed that all my stuff would fit on a single little memory stick or compressed onto one CD! That part was actually kind of depressing. But if nothing else, know that you can burn the best music of your whole life on your iPod, and probably everything else about yourself on a computer drive.

And now that I am fully unwound, I need to get back to work now to create more *original* digital material! EQ

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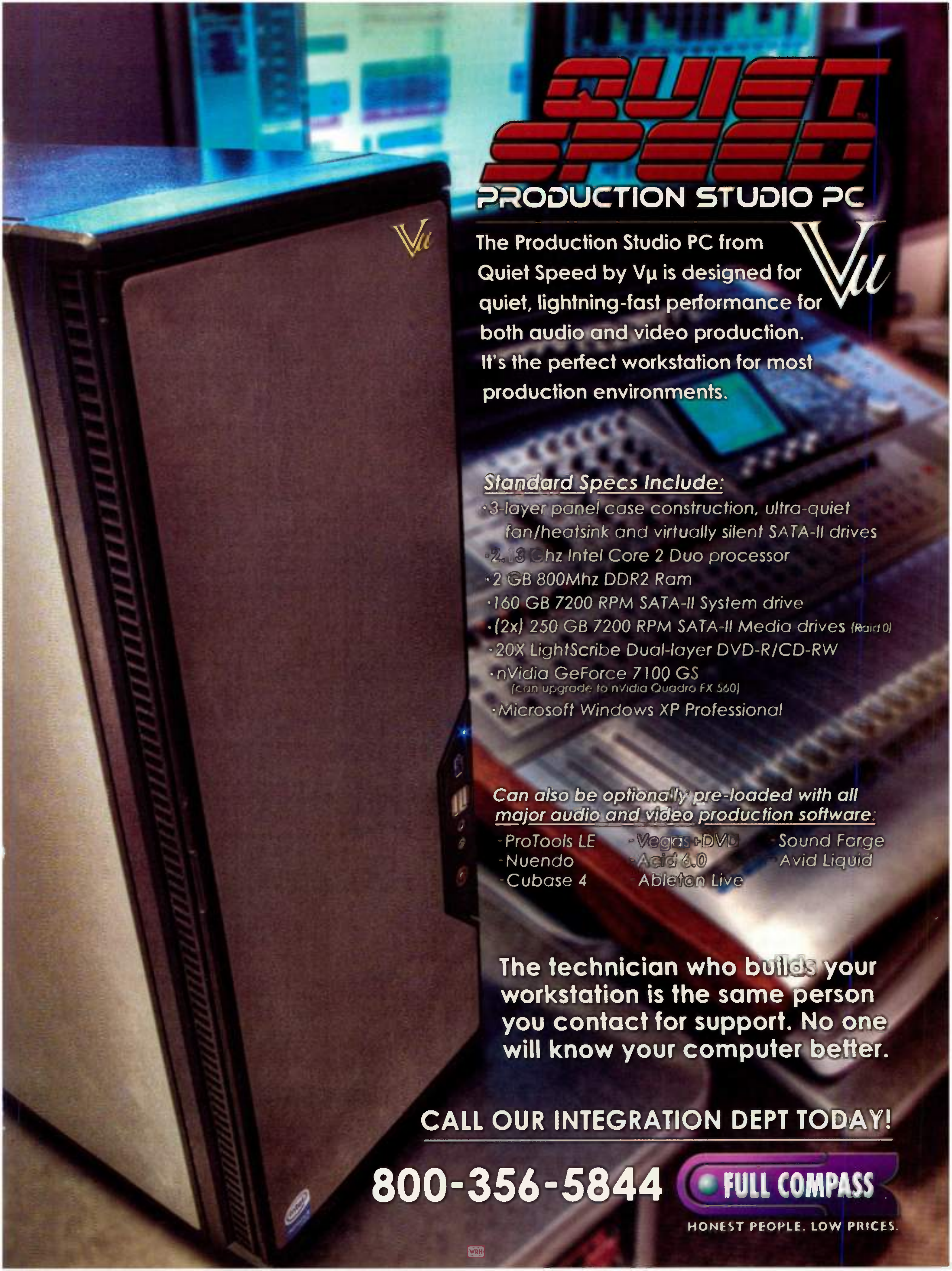
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ANDY TIMMONS DETAILS THE STUDIO STRATEGY FOR *RESOLUTION*

by Brian Tarquin

One of the great things about recording guitar is that every artist has a different way of approaching tone. For example, Andy Timmons broke with the current craze of layering guitar textures to infinity and beyond on his latest release, *Resolution* [Favored Nations]. Instead, he chose to record just one guitar track per song — a gutsy move.

"It was quite a process of experimentation and discovery," says Timmons, who, in addition to his careers as a solo artist and session player, is also Olivia Newton-John's music director and guitarist.

Timmons (left) raging through his dual Mesa Lone Star/Stiletto rig. Note the "MacGyver" appendages on his Memory Man delays (right) used to adjust delay levels with his foot.

"*Resolution* was the first record I've ever approached as a strictly trio record, with the guitar stripped down to a single performance. Steve Vai [who owns the Favored Nations label] inspired the approach when he was listening to some tracks on my previous record, *That Was Then, This Is Now*. Two of the songs had sections that were solely trio-based, and Steve said, 'I love hearing your fingers on the frets, and all your picking dynamics.' After he said that, I knew I wanted to try making an entire album that way. I also knew it would be a major challenge."

Of course, recording a tremendous guitar tone was key, as one sonically stark track would be what listeners' would hear. So here's the "Timmons Method" for crafting the stunning, stand-alone guitar tones that grace *Resolution*.

GO STEREO

Timmons chose a stereo signal path with a '68 plexi Marshall Super Lead panned hard to one side, and a '79 Marshall master-volume JMP panned opposite. The speaker cabinets were 2x12 and 4x12 Mesa/Boogie Recto models loaded with Celestion Vintage 30s.

"I'd typically run the amps clean," he says, "and get most of the distortion from an old Tube Works Tube Driver or an Ibanez Tube Screamer. I also used a Maestro EP3 tape echo, a Fulltone Tube Tape Echo, a Chandler Digital Echo, an old Octavia on "Hellipad," and, for "Resolution," I simultaneously recorded the Marshall and a direct line that I later re-amped through a Leslie 122 cabinet to get a bit of a Hendrix-y effect. We didn't change things up too much, and most of the songs were done with the same rig."

TRUST WHAT WORKS

Although many producers and engineers develop unique miking applications involving a plethora of microphones, it's not always the expensive tube condensers or arcane microphone placement that captures the heaviest guitar sounds.

"I used Shure SM57s for recording the guitar," he says. "We tried all kinds of different mics and miking techniques, but we always came back to the tried and true SM57."

JETTISON EQ

One of the most surprising aspects of the recording process for *Resolution* was that Timmons and co-producer (and bassist) Mike Daane did not use any EQ on the guitar during tracking and mixing. Daane wanted to go for a more natural and organic sound, and the duo employed mic-placement techniques to refine the guitar tones.

"Mike was a major component on this record," says Timmons. "He has a great ear for music, and we are able to push and pull each other to get the best result. Once the basic guitar tone was agreed upon, we simply moved the mics until we discovered the position that produced the perfect sound for whatever we were going for on a particular track."

LOVE YOUR BABY

Timmons didn't assemble an armory of vintage and varied guitars to record Resolution. He mostly relied on his Ibanez signature model, the AT300, which is armed with jumbo frets, a Wilkinson tremolo, DiMarzio Cruiser neck and middle single-coils, and a DiMarzio AT Custom bridge humbucker. This was the perfect choice for realizing the performances on *Resolution*, as he knew the instrument inside and out, he designed it to accommodate his individual approach, and he has played it on stage and in the studio for years. The tip here is that an intimate familiarity with your main guitar can help you focus on your performance chops — a good thing when one guitar track must communicate everything you wish to say.

WORK AT THE PACE THAT FIRES YOUR CREATIVITY

Timmons and Daane employed technology as needed to produce the coolest sounds and optimize the process. The basic drum and bass tracks were recorded on 2" analog tape through an old Neve console to produce some funky tape coloration and a vintage vibe. These tracks were bounced to Pro Tools, and then Logic to allow Timmons the freedom to develop his parts at a pace that accommodated his work schedule and creativity. (In fact, some of the basics on *Resolution* are a year or more old.) When all the guitars were tracked, the audio files were bounced to Pro Tools for the final mix. EQ

6 WAYS TO CONQUER RED LIGHT FEVER

by Michael Molenda

Red light fever is an affliction that bedevils every guitarist from time to time. It doesn't matter how skilled you are, how much experience you have, or how scrupulously you've rehearsed — at some point in your life, you're going to hit Record, and you will totally screw the pooch (to borrow a phrase for "failure" from Tom Wolfe's *The Right Stuff*). Here are some hopefully inspirational suggestions from guitar stars who have faced the terror of not tracking a unique and brilliant part, but who have busted through the malaise to get something down.

REVIEW HISTORY

"Sometimes, you want to dream up a new way of playing. You're sitting down, just hearing notes, and trying to find them. You're not really thinking about where other people put their fingers to get a particular sound. But, at some point, something in your memory says, 'Lightning Hopkins did something like that,' and you seek out exactly how he did it. All of a sudden, you take that data as a new building block, and you say, 'Okay, I can move forward on this.' And then you wind up with a new style of riff, and, if it's a good one, it becomes the new building block in that particular area." *—Joe Satriani (Guitar God)*

STUDY OTHER STYLES

"Most things that are modern are just combinations of pre-existing things. I made some discoveries at one point as a player that a lot of the techniques used by country guitarists were not really used by rock guitarists unless they were playing country rock. Take, for example, the technique of internal string bending, where you're holding a chord, but one of your fingers is bending a note within it. If you're a country player, you do that with a Telecaster and a clean sound. But I realized you could integrate that approach with other guitar styles, and with distortion — which makes the pitch of the bent strings beat against each other in very cool ways. There are so many things that put the power in your hands, as opposed to using effects that can derail the personality of the player. Listen to Les Paul — he uses every part of the string, and he's always changing pickups and fooling with his tone knobs. In the course of one line, he'll play very close to the bridge on the low-*E* string on the treble pickup with all the tone on, and, three notes later, he'll play the same line four octaves up on the neck pickup with all the tone down." *—Jon Brion (Grammy-nominated soundtrack composer, session guitarist, and producer)*

Bruce Kulick tracking his upcoming solo album.

CHANNEL YOUR HEROES

"When I want to stretch my limits, I definitely need to find something that sparks me, pushes me, and inspires me to come up with a great solo or guitar part. In these situations, I put myself in the mind frame of one of the guitarists who have influenced my playing, such as George Harrison, Jimmy Page, or Jeff Beck. For example, Page had a huge vocabulary, so I'll ask myself, 'What would Pagey do here?' I think of my heroes almost as creative muses who can point me in the right direction." *—Bruce Kulick (Kiss, Grand Funk tour guitarist, solo artist)*

SCREW EXPECTATIONS

"Your ego wants your parts to be perfect, and this can kill you. In some way or another, you might be somewhat unsure of yourself, and afraid of criticism — afraid somebody might say you were a little out of time, or a little out of tune, or you bent that note too much or too little, or whatever. I don't give a sh*t. I figure it's like this: I'm going to do what I'm going to do, and some people are going to like it, and some people aren't going to like it. A perfect guitar solo never got me a platinum album." *—Earl Slick (David Bowie, solo artist)*

LET YOUR BAND BE YOUR GUIDE

The way I played was very much shaped by the Doors. Being the only guitar player, and not having a bass player, made me fill up certain holes that I normally wouldn't have filled. For example, I used my thumb to hit bass notes, and I'd play rhythm and lead at the same time." *—Robby Krieger (Doors)*

LISTEN TO THE MUSIC

"A lot of guitarists have an ironic edge to their playing — it's almost like they're commenting on the music as they're writing it. You need to have the ability to enter into the moment. You can't think about what you're doing — you just short-circuit that element of consciousness. Now you may hit a clam if you take this route, but I feel it's critical that you listen really deep inside of the music that you want to come out of your fingers. The way to do that is to just observe the music, and not try to be smarter than it. For example, if you're playing a solo, and you think, 'That was a nice riff,' you can be sure that doom is awaiting you around the next bar." *—Lenny Kaye (Patti Smith)* EQ

GET YOUR SIGNAL PATH TOGETHER!

by Chris Mara

Most people believe that tracking bass, especially in "standard" forms of pop music, is simple — "DI it." But those people are only partially right. As with all instruments, there are a ton of variables — and possibilities — involved in recording bass, depending upon your artistic intent. Sure, if you want a straightforward track, cutting bass can be super-easy. But if you want to experiment with how a bass *can* sound, the map everyone tends to use isn't always going to help you navigate the terrain.

That may sound vague, but you don't need to drive cross-country in the Mystery Mobile to unmask the secret of getting cool bass tones. You just need to consider what makes a bass track suck, and what makes a bass track awesome, and then apply that knowledge to your own sessions.

The first step to recording a good sound is to make a good sound. Set your rig up right, change those crusty old strings, put a tuner in line (and use it after every take), and write a good song. Do all those things, and you're 75 percent of the way there.

But you have to look at signal paths, too, and one basic setup I use is bass > direct box > mic preamp > compressor > DAW. This simple chain provides a fast and accurate idea of what kind of bass sound you're bringing to the table.

Now, you may wonder, "Why does he route the DI into a mic pre?" Well, direct boxes operate at mic level, so a preamp is often necessary to boost the signal to the line level your compressor likes to see.

Another question: "Why a DI first? Why not use your amp sound?"

Using a DI is good way to gain forward momentum when tracking. It's a lot simpler than miking an amp, and it also makes it easy to suss out any potential issues — such as grounding — with the bass itself. The DI offers a less-complicated signal path than what you'll have when recording through an amp, so if any ground hum or noise issues arise when you add a cabinet signal, you'll know the source of the problem.

Use a clean, honest, and quiet preamp (I swear by APIs) so that you have a transparent sound without that pesky 60Hz hum. Then, compress the signal so there's about 3dB to 5dB of gain reduction. Don't use too much compression. Apply just a little to trap peaks and ensure consistency — you can always compress more in the mix. Listen carefully to the results you obtain from using this setup, and if you don't like the sound, look back to the instrument. Adjust your tone knob, think about throwing that pick out and playing with your fingers if you are getting too much attack, and set your action a bit higher if you hear any fret noise.

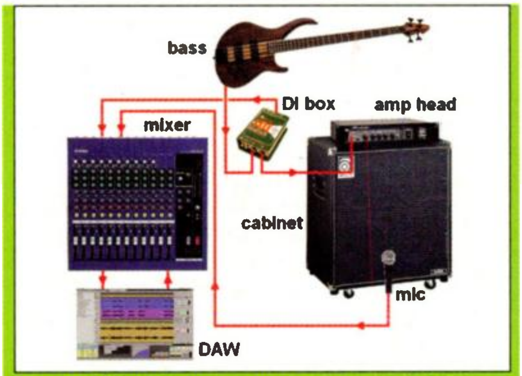

Fig. 1: The complete bass signal path.

You should now have a workable tone, but if you're not 100 percent happy with the results so far, start swapping out preamps, and making other changes in search of a better sound. This is trial and error, but it's better to get a good bass sound first, than to try to fix it in the mix. For example, a few weeks ago, I was using my trusty API 512 on a session, and I found its sound was a bit too aggressive for the song. Switching to a Drawmer 1960 mellowed the tone out perfectly, as it had a built-in tube compressor that slowed the slew rate down a good bit. (A fast slew rate — meaning the maximum amount of change imposed on a signal — produces a punchy sound, while a slow slew rate is associated with a warm sound.) So, play around a bit, and see what you can get out of your DI-to-preamp signal before you go off in a different direction.

You could stop right here, but getting a good bass-amp sound to complement your DI track can often be the secret ingredient to success. So, break out that bad boy Ampeg SVT, set your EQ on the amp head, and mic the cabinet. Most DIs have an instrument input, as well as a send for your amp, so plug the bass output into the instrument input, and then patch the amp send into the bass head's input. Next, run the bass head's speaker output into the cabinet, and mic that puppy up with a mic that can handle a lot of sound-pressure levels and low frequency information. I like using an AKG D112, an AKG D12, or a Sennheiser MD421. These mics have natural EQ curves that just work well for bass. Figure 1 illustrates this, if you need a visual evidence.

If you run into grounding issues that result in hum or other problems, flip the DI's ground switch. If there's still some buzz, you may have to use a 1/4" Y-adaptor (one male to two female 1/4") to split the bass signal to the DI and amp simultaneously).

Now, record your parts, but assign the DI and amp signal to different tracks, as there will be some inherent phase issues. (You'll have to nudge the amp signal ahead in time to match your DI signal, because the amp signal takes longer to reach your DAW.) For best results, match compression settings for both the amp and DI signals, as drastically different settings may exacerbate any existing phasing issues.

Regarding compression settings, I try to keep the release time "in sync" with the tempo of the bass line, so that the processor has stopped compressing by the time the next note hits. To achieve this, use a semi-fast attack, set a ratio of 1.5:1 to 3:1, play with the threshold to get 3dB to 5dB of gain reduction, and then tweak the release until it meshes just right with the bass. If this doesn't work, try the opposite approach of cranking down, so the compressor never stops compressing fully from note to note. You can do this by increasing the ratio up to 4:1, then dropping the threshold for more compression (about 5dB to 10dB of gain reduction). Adjust the release time to maintain a consistent level.

I usually end up using more DI signal in the mix than amp signal, as the amp signal is most useful to round out the bass tone from the DI. It adds a healthy, low, pillow-y veneer to your track. But don't take my word for it — experiment to see what works best for you. Just keep in mind that it all starts with the instrument and you — the player. EQ

5 QUICK & EASY BASS SOUNDS

by Michael Molenda

Experimentation is essential in the home studio. You have to commit to bending and shaping signals until they bow to your will, and produce just the right sound for the creative opus you're crafting. That's the drill sergeant, "take no prisoners" line, and it's mostly right and true and holy. But you won't always have the time or available brain matter (hey, we're human — we get tired) to refine every signal until it's perfectly formed, and unique to your vision. Sometimes, ya just wanna crank a knob or two, and get on with it. So, for those times when you're fatigued, frustrated, or cranky, here's a menu of bass recipes that will deliver good, foundational tones with minimal hassle.

THE STRANGLERS SNAP

Jean Jacques Burnel's spikey, aggro bass lines — along with Dave Greenfield's Ray Manzarek-inspired organ washes — have propelled the Stranglers' punk bombast since 1974. (Listen to 1977's "Heroes," or 2006's "The Spectre of Love" for a taste of Burnel's low-end violence.) Set compression at a ratio of 10:1, and a threshold of –15dB with relatively fast attack and release times to produce a vicious squash. You really want the bass in your face. Then, cut 100Hz by 3dB (to toughen the lows), boost 800Hz by 3dB (to pump up the pluck), and boost 3kHz by 6dB (to crank up pick attack).

RUBBER SOUL-ERA MCCARTNEY

For the warm, blossoming punch of songs such as "The Word," "You Won't See Me," and "Wait," set a relatively light compression ratio of 4:1 and a threshold of –10dB. You want to ensure the bass line is front and center, but a little dynamic interest is cool. Then, go for that gloriously fat McCartney tone by boosting 100Hz by 3dB or more (for a bit more boom, also try boosting around 60Hz to 80Hz by 3dB), sneak in a slight boost of 2dB at 1.5kHz for some thud-y pluck, and diminish highs from 10kHz on to evoke some vintage warmth.

MOTOR CITY GROOVE

Capture the silky, muted pulse of Motown tracks such as the Supremes' "Stop in the Name of Love" by boosting 100Hz by 2dB, boosting 250Hz by 3dB, cutting 80Hz by 6dB, and cutting around 1kHz and 3kHz by 6dB. Compression should be solid, but not oppressive — try a 2:1 ratio and a –10dB threshold.

HIP-HOP BOOM-BOOM

Emulate the car-door-rattlin' low end of rap and dance tracks — which often utilize keyboard bass and/or machine kick drums for the wallop — by boosting 40Hz by 3dB, 80Hz by 6dB, and 100Hz by 6dB. Then, add a slight attack by boosting 2kHz by 3dB, and cut highs by dumping every frequency above 8kHz by 6dB or more. This sound works better on minimalist, accent-oriented parts. Now, use some light limiting, and rock the house!

BUZZ BOMB

Guitar players aren't the only cats who can deploy distortion. If your song could benefit from a little punch and sizzle, plug your bass into a fuzz or overdrive pedal. Compress heavily to help accentuate the distortion — try a 10:1 ratio at a threshold of –10dB. Tweak the EQ to push the overdriven midrange attack — a 6dB boost at 3kHz should help — and control any low-end mud by cutting 100Hz by 6dB. EQ

3 TIPS FOR FIGHTING A BASS-LITE MIX

It's simply not fair. You've done everything right during the tracking phase, and your bass tone is tight, punchy, and fat. But here you are at the mixdown session, and your band mates are dropping your track so low in the mix that you're fearing for your job. (And all those comments about how cool the White Stripes sound *without* a bass player aren't helping!) Well, keep your cool. Here are a few ways to overturn three common reasons why engineers tend to bury the bass.

ARGUMENT 1: THE BASS IS FIGHTING WITH THE KICK DRUM.

RESPONSE: Don't get into a brawl over which instrument is more important, simply reach over and pan the bass track slightly to the left (say, around two o'clock), and pan the kick drum track slightly to the right. Voilà! Changing the mix positions should not only clarify both tracks, but also add some dimension to the low-end thud. Musicians tend to freak out when you move kick drums off center, but listeners seldom notice such slight adjustments, and your bass will be slammin'. If you're really brave, suggest that the kick provide the snap (cut 80Hz to 100Hz by 6dB, and boost 1kHz by 3dB), and the bass deliver the wallop (don't change a thing on the bass EQ). There's no need to have two or more mix elements throwing identical body punches to the listener.

ARGUMENT 2: THE BASS IS MUDDYING UP THE GUITARS.

RESPONSE: It's more likely that your guitarist, in the pursuit of a ballsy tone, added some unnecessary low end to the guitar tracks. Snip the boom out of the guitars by cutting 3dB to 6dB at around 200Hz, and again at around 500Hz. Let the bass handle the lows and low-mids, and let the guitars address the mids, and no listener will be the wiser. The mix should still sound tough, propulsive, and ballsy as all hell. You can also try the kick drum trick, and move the bass and guitars slightly off center to further differentiate the tones.

ARGUMENT 3: THERE'S TOO MUCH LOW-END CONTENT ALREADY. WE HAVE TO THIN OUT THE BASS.

RESPONSE: What madness is this? It's so easy to scapegoat the bass track, but, let's face it, there's probably too much low end pumping on everything else. Solo common muddy-low-end offenders such as the kick drum, toms, guitars, and keyboards, but also ensure that the vocals aren't slaughtered by plosives that add undesirable booms and pops to the overall mix. Don't forget to solo effects tracks, as well. Rampant bass frequencies can compromise, say, a room reverb or a delay as much as anything else. Then, cut low end from every mix element that doesn't absolutely have to live in the bass track's sonic space. *—Michael Molenda*

HOW TO PRODUCE BIGGER & BADDER SYNTH BASS SOUNDS

by Craig Anderton

There's a saying that "You can never be too thin or too rich." That may work for Nicole Ritchie, but it's only half-right for keyboard synth bass: Rich is good, but thin isn't. Looking for a truly corpulent bass sound that's designed to dominate your mix? These techniques will take you there.

LAYERS GOOD, LAYERS BAD

A common approach to crafting a bigger sound is to layer slightly detuned oscillators. However, that can actually create a thinner sound because slight detunings cause volume peaks but also, volume valleys. This not only diffuses the sound, but often makes it hard for a bass to sit solidly in the mix because of the constant sonic shapeshifting. Following are some layering approaches that *do* work.

Three oscillators with two pitch detunings. Pan your main oscillator to center, and set it to be the loudest of the three by a few dB. Pan the other two oscillators somewhat left and right of center, and detune both of them four cents sharp. Yes, this will skew the overall sound a tiny bit sharp; think of it as the synth bass equivalent of "stretch" tuning.

Dual oscillators with detuning. You can get away with detuning more easily if there are only two oscillators, as the volume peaks and valleys are more predictable. Pan the two oscillators left and right of center, set them to the same approximate level, tune one oscillator four cents sharp, and tune the other four cents flat. If that's still too diffused, pan them both to center, tune one to pitch, detune the other one eight cents sharp, and reduce the level of the detuned oscillator by –3 to –6dB.

Three oscillators with multiple detunings. If you must shift one oscillator sharp and one flat in a three-oscillator setup, consider mixing the two shifted oscillators somewhat lower (*e.g.*, –3 to –6dB) than the one-pitch oscillator panned to center. This will still give an animated sound, but reduce any diffusion.

Three oscillators with layered octaves. This is one of the most common Minimoog bass patches (Figure 1), and yes, it sounds very big. Adding a slight amount of detuning to the lowest and highest oscillators thickens the sound even more, as this simulates the drift of a typical analog synthesizer.

Two oscillators with layered octaves. While this doesn't sound quite as huge as three oscillators with layered octaves, removing the third oscillator creates a tighter, more "compact" sound that will

Fig. 1: This Arturia Minimoog V shot shows an archetypal Minimoog patch, with three oscillators set an octave apart via the Range controls. Note that the lowest and highest oscillators are tuned a bit off-pitch to add more sonic animation.

Fig. 2: The upper envelope generator picture from Cakewalk's Rapture shows a quick percussive decay that adds punch. The lower envelope setting achieves a more sustained punch effect by kicking the envelope full on for a couple dozen milliseconds.

Fig. 3: Feeding a lowpass filter with a reasonable amount of resonance through distortion can create a sound that resembles hard sync.

Fig. 4: Native Instruments' Pro-53 is one of many soft synths that provides pulse waveforms. Better yet, a pulse width control determines whether the pulse is narrow or wide.

Fig. 5: Cubase 4's Monologue synth makes it easy to assign the mod wheel to filter cutoff as one of the filter modulation sources (circled in red).

cede some low-end territory to other instruments (*e.g.*, kick drum).

Sub-bass layer. Drum 'n' bass fans, this one's for you! Layer a triangle wave one octave below any other waveforms you're using. (You can also try a sine wave, but at that low a frequency, a little harmonic content helps the bass cut through a mix better.) For a *really* low bass end, layer three triangle waves with two tuned to the same octave (offset one by +10 cents), and the third tuned one octave lower and offset by –10 cents. Sub-bass patches also are excellent candidates for added "punch," which provides the perfect segue to . . .

PUNCH!

There are two main ways to add punch to a synth sound.

Percussive punch. This requires adding a rapid amplitude decay from maximum level to about 66% of maximum level over a period of 20–25ms (Figure 2, top). To emphasize the percussiveness even further, if a lowpass filter is in play, give its cutoff a similarly rapid decay. However, for the filter, bring the envelope down from maximum to about 50% of maximum over about 20–25ms.

Sustained punch. This emulates the characteristics of the Minimoog's famous "punchy" sound. (Interestingly, the amplitude envelopes in Peavey's DPM-3 produced the same kind of punch; after I described why this phenomenon occurred in *Keyboard* magazine, Kurzweil added a "punch" switch to their keyboards to create this effect.) Sustained punch is simple to create with most envelope generators: Program an amplitude envelope curve that stays at maximum for about 20–25ms (Figure 2, bottom). This is too short for your ear to perceive as a "sustained sound," but instead comes across as "punch."

PSEUDO-HARD SYNC

If your soft synth doesn't do hard sync, there's a nifty trick that gives a very similar sound—providing you can add distortion following the filter section.

Figure 3 shows Rapture set up for a hard sync sound on one of its elements. Note the setting of the Cutoff, Reso(nance), and BitRed controls (Bit Reduction is set for a Tube distortion effect, as shown in the label above the control). The envelope shown toward the bottom sweeps the filter cutoff from high to low. As it sweeps, the filter's resonant frequency distorts, producing a hard-sync like sound. The crucial parameter here is resonance; too little and the effect disappears, two much and the effect becomes overbearing . . . not that there's necessarily anything wrong with that. . . .

CHOOSING THE RIGHT WAVEFORM

For most big synth bass sounds, a sawtooth wave passed through low-pass filtering (to tame excessive brightness) is the waveform of choice. As a bonus, if you kick the lowpass filter up a tad more, it brings in higher harmonics that add a "brash" quality.

For a "rounder" sound that's more P-Bass than Synth Bass, try a pulse waveform instead (Figure 4). I prefer narrow pulses (around 10–15% duty cycle), but wider pulse widths can also be effective. The same layering techniques mentioned earlier work well with pulse waves, but also experiment with layering a combination of pulse and sawtooth waves. This produces a timbre somewhere between "tough" and "round."

Triangle and sine waves have a hard time cutting though a mix because they contain so few harmonics. If you want a very muted bass sound, use a waveform with more harmonics like sawtooth or pulse, then close a lowpass filter way down to reduce the harmonic content. This provides a rougher, grittier sound due to the residual harmonics that remain despite the filtering. However, while triangle waves aren't necessarily great solo performers, they're excellent for layering with pulse and sawtooth waveforms to provide more low-end girth.

THE ALL-IMPORTANT MOD WHEEL

Just because you're playing in the lower registers doesn't let you off the hook to add as much expressiveness as possible. Some programmers get lazy and do the default move of programming the mod wheel to add vibrato, but that's of limited use with bass. If you want vibrato, tie it to aftertouch, and reserve the mod wheel for parameters where you need more control over the sound.

Creative use of modulation could take up an article in itself, but these quick tips on useful modulation targets will help you get started.

Filter cutoff. This lets you control the timbre easily. If the filter is being modulated by an envelope, assigning the mod wheel to filter cutoff (Figure 5) can also create a more percussive effect when you lower the cutoff frequency. Try negative modulation, so that rotating the mod wheel forward *reduces* highs.

Volume envelope attack. To transform a sound's character from percussive and punchy to something "mellower," edit the mod wheel to increase attack time as you rotate it forward.

Layer level. Assign the mod wheel to bring in the octave-lower layer of a sub-bass patch. This pumps up the level and really fills out the bottom end.

Distortion. Yeah, baby! Kick up the distortion for a bass that cuts through the mix like a buzzsaw, then pull back when the sound needs to be more polite.

Resonance. I'm not a fan of highly resonant synth bass sounds (they sound too "quacky" to me), but tying resonance to mod wheel provides enough control to make resonance more useable. EQ

8 STEPS TO A PERFECT KICK SOUND

by Garrett Haines

As a drummer and a studio owner, I can attest to the fact that many drummers do not know how to properly tune their kick drums. And their records suffer for it. So unless you are triggering your kick (and keeping a 100 percent wet kick mix), or planning on spending hours with SoundReplacer fixing a screwed-up kick track, you need to start from the source and get the best-sounding kick possible. The way to do that, my friends, is to just tune the damn thing properly. Here are eight simple steps to a glorious kick drum sound.

STEP 1

Fig. 1. Run your finger or palm along the bearing edge to feel for imperfections.

Remove all lugs, the hoop, and the head, and wipe away any dirt or dust buildup. Run your finger around the bearing edge of the drum (Figure 1) to feel for abnormalities such as dents, cuts, or grooves. If you find anything, simply fill it in with ski or candle wax. (If the damage is severe, you may need the edge re-cut.) Having good contact between the head and the bearing edge is one of the single most important aspects of a good drum sound, so make sure everything is nice, clean, smooth, and even. And while you're at it, check for loose lug and spur screws, and replace them if necessary, as they can vibrate and scratch your shell — or even allow a whole lug to break off.

STEP 2

Resonant heads are very important to a drum's sound. With rare exceptions, factory heads are cheap, and they're either difficult to tune, or do not hold their tuning. So if you have a new kick, replace the factory head before recording.

Place the new head on the drum, and spin it around the rim a few times to make sure the head is "in-round," and not a factory defect (Figure 2). Align the head so the logo and/or vent sits where you prefer, and reseat the hoop over the head. Some drummers prefer to keep the hoop in the same place as before, contending that the hoop mates with the shell over time. I like to rotate the hoop each time I change a head.

STEP 3

Before replacing the tension rods, make sure each rod has a washer. Apply a small amount of lubricant to the end threads (Figure 3), as this will help maintain tuning over time. Finger-tighten all of the rods. Some wrinkles may still be visible on the head, but stop once you can no longer easily move the rods with simple figure pressure.

STEP 4

Now it's time to seat the head. There are a few ways to do this, but I prefer the CPR method. Lean over the drum, place your palm in the middle of the head, reinforce with your other arm, and apply constant, but firm, pressure until you hear a long crackling noise (Figure 4). This breaks up the excess epoxy that holds the head into its rim all at once so the epoxy doesn't break up over time and cause unexpected tuning shifts.

STEP 5

Fig. 5. Using two keys simultaneously helps apply a more even tension across the head.

Using two drum keys, tune any given lug along with the lug directly opposite of it (Figure 5). Start by doing only a one-half turn before moving to the next lug pair. Why two keys? Because it's time-efficient, provides more even pressure to the head, and avoids the traditional versus modern tuning-sequence

Fig. 2. Once the head is on the shell, spin it around like a wheel to check for defective heads or an "out-of-round" shell.

Fig. 3. Using a lint-free swab, apply a tiny amount of lubricant to each tuning rod.

Fig. 4. One way to seat a head is to apply firm, but steady pressure for a few seconds. You should hear a series of cracks and pops as the epoxy settles.

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DRUM HEADS

debate. [*Note: Traditional tuning methods dictate that lugs are tightened in the following order: first lug, opposite lug, next lug, opposite lug, and so on. This method evolved in the days when drumheads were made of skin and other variable materials. Modern tuning starts with a given lug, then proceeds in a clockwise or counterclockwise fashion. Proponents of this method contend that modern head and drum construction make the traditional method obsolete.*] As you increase the tension around the head, the overall surface will start to smoothen out. Continue until the head is smooth of all wrinkles.

STEP 6

The next step is establishing a preliminary pitch. First, isolate the head you're tuning by either placing a towel on the floor and setting the drum on it, or resting the kick on your foot while you tune the face-up head (Figure 6). Using a felt or percussion mallet (sticks have undesirable attack on kicks, and can damage the head), move around the drum, hitting about two inches inside of each lug. At this point, you should be listening for each lug's pitch relative to the other lugs. If the pitch is too high, back off a half to three quarters of a turn, and tune back up. Stop when the pitches are about the same. As with guitar tuning, to lower the pitch, loosen the tension beyond your target tone, and then re-tighten until the desired tension is achieved.

STEP 7

Keeping the opposite head muted, strike the head in the center of the drum. Listen to the sound of the decay of the hit. If it fades out in a smooth, even way, the head is in tune with itself. If you hear "beats" (similar to playing harmonics when tuning a guitar), you know that one or more lugs are out of tune. If you have an overhead light, you can watch the reflection of the light after you strike the head (Figure 7). While it's not good to tune by eye alone, this trick can be helpful in a noisy environment. If the reflection is "warbled," and the sound decays with an unpleasant or beating nature, the head is out of tune. If the drum fades out in a smooth manner, the reflection cast in the head should come back into focus quickly and evenly.

If the drum is still out of tune, it's probably just one or two lugs that are the culprits. Go back to striking about two inches inside each lug. At this point, use your free hand and barely touch the center of the head as you strike each lug. This should help isolate the decay of each hit. Once you find the lug that's too high or low, adjust it. Keep in mind that this is an imperfect science. Sometimes, the "responsible" lug is the one right next to, or opposite of, the "bad sounding" lug. Once you find the last troublemaker, tune it up and give the whole head a big hit. Regardless of pitch, the head should be in tune and decay evenly. Unmute the second head, and evaluate the drum as a whole. If you like what you hear, now is the time to fine-tune the kick.

STEP 8

Tossing a blanket or a pillow in the kick is lazy, negates shell overtones, and makes heads sound like wet cardboard on your recording. If you want a tight, focused sound, start with a batter head that has built-in control devices, such as the Aquarian Super Kick II, or the Evans EMAD System. You can also tape a fabric square

Fig. 6. If you can't mute the bottom head with a towel, rest the drum on your foot.

Fig. 7. Use a bright overhead light as a visual guide when tuning.

on the inside of the batter head, placing the tape along the top. When the head is struck, the fabric will flap away from the head, and then return to rest, effectively muffling unwanted overtones.

Fig. 8. The Kik Brik is an aftermarket muffler/damping device that helps bring the most out of inexpensive, or hard-to-tune heads.

If you like how a pillow limits shell overtones, try an aftermarket device such as the Kik Brik (Figure 8) These are made for varying shell depths; they can be mounted to touch one, both, or neither heads; and they come with Velcro strips to keep in place if the drum moves.

If you want a sharp attack, avoid the old "coin taped to the beater head" trick. This will damage your head. Instead, use a store-bought patch, or just tape an old credit card to the head.

Finally, if you notice that after your rough tracks, the kick and the bass are conflicting on certain notes, try changing the tuning of your head, instead of EQing your way out of the situation. Your album will love you for this. EQ

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PREPPING A VOCAL FOR THE MIX

by Craig Anderton

As far as I'm concerned, the vocal is the most important part of a song. It's the conversation that forms a bond between performer and listener, and the focus to which other instruments give support.

And that's why you must handle vocals with kid gloves. Too much pitch correction removes the humanity from a vocal, and getting overly aggressive with composite recording (the art of piecing together a cohesive part from multiple takes, and the subject of a future Vocal Cords) can destroy the continuity that tells a good story. Even too much reverb or EQ can mean more than bad sonic decisions, as these can affect the vocal's emotional dynamics. But you also want to apply enough processing to make sure you have the finest, cleanest vocal foundation possible — without degrading what makes a vocal really work. And that's why we're here.

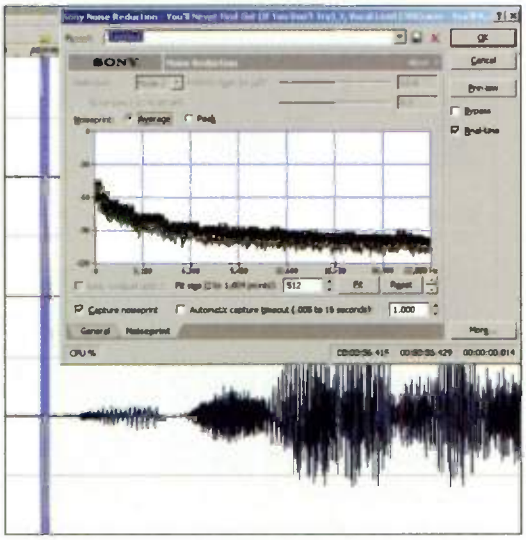


Fig. 1: A good noise reduction algorithm will not only reduce mic preamp hiss, but create a more "transparent" overall sound. This shot shows the "noiseprint" taken by Sound Forge 9, which will then be subtracted from the entire signal.

Vocals are inherently noisy. You have mic preamps, low-level signals, and significant amounts of amplification. Furthermore, you want the vocalist to feel comfortable, and that can lead to problems, as well. For example, I prefer not to sing into a mic on a stand unless I'm playing guitar at the same time. I want to hold the mic, which means mic-handling noise is a possibility. Pop filters are also an issue — as some engineers don't like to use them — but they may be necessary to cut out low-frequency plosives. In general, I think you're better off placing fewer restrictions on the vocalist, and having to fix things in the mix, rather than having the vocalist think too hard about, say, mic handling. A great vocal performance with a small pop or tick trumps a boring, but perfect, vocal.

Okay, now let's prep that vocal for the mix.

REMOVE HISS

The first thing I do with a vocal is turn it into one long track that lasts from the start of the song to the end, then bounce it to disk for bringing into a digital audio editing program. Despite the sophistication of host software, with a few exceptions (Adobe Audition and Samplitude come to mind), we're not quite at the point where a multitrack host can always replace a solid digital-audio editor.

Once the track is in the editor, the first stop is generally noise reduction. Sound Forge, Adobe Audition, and Wavelab have excellent built-in noise reduction algorithms, but you can also use stand-alone programs such as Diamond Cut 6. Choose a noise reduction algorithm that takes a "noiseprint" of the noise, and then subtracts it from the signal. Using this simply involves finding a portion of the vocal that consists only of hiss, saving that as a

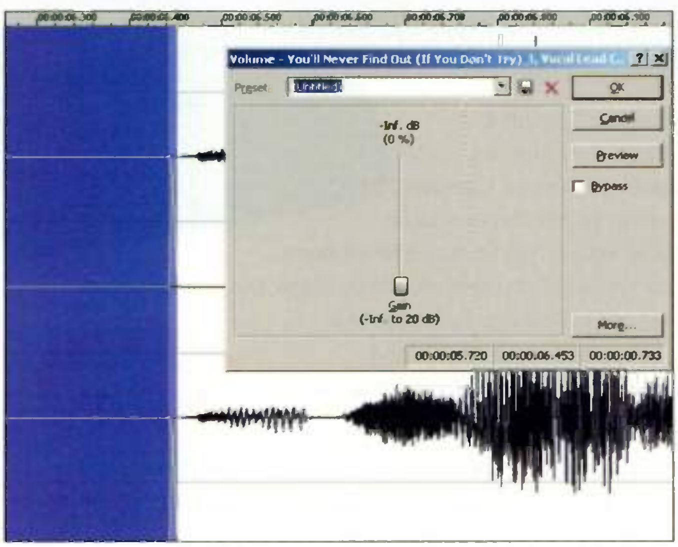


Fig. 2: Cutting out all sound between vocal passages will help clean up the vocal track. Note that with Sound Forge, an optional automatic crossfade can help reduce any abrupt transition between the processed and unprocessed sections.

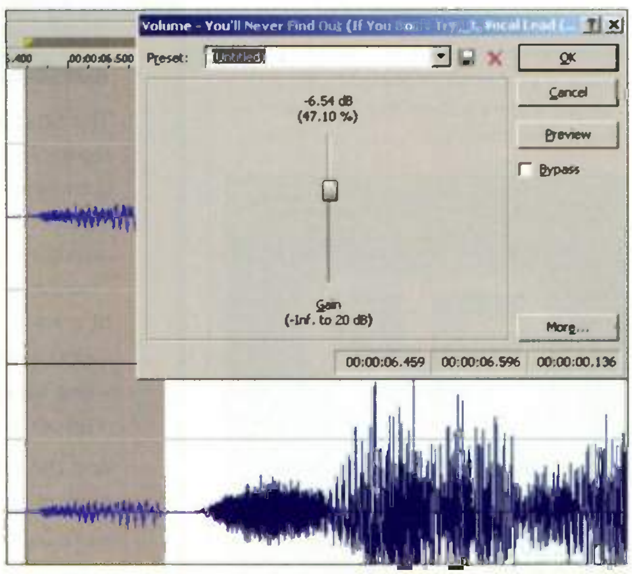


Fig. 3: The highlighted section is an inhale, which is about to be reduced by about –7dB.

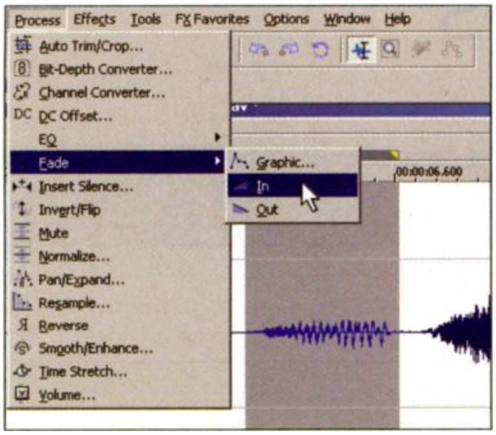


Fig. 4: Imposing a fade-in over an artifact is another way to control a sound without killing it entirely.

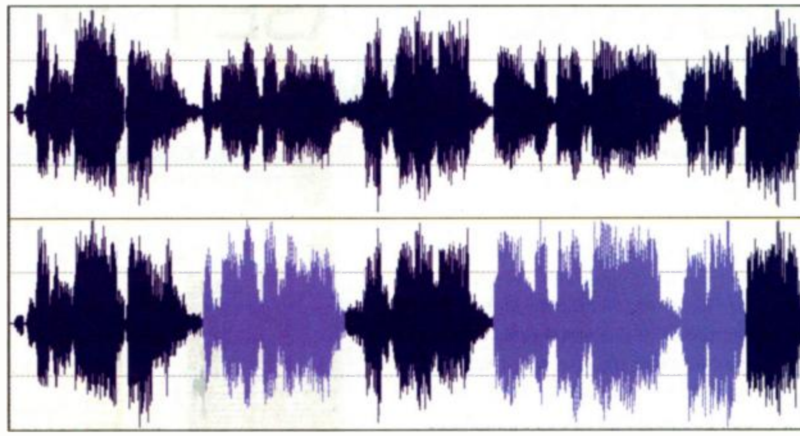


Fig. 5: In the lower waveform, the sections in lighter blue have been normalized. Note that these sections have a higher peak level than the equivalent sections in the upper waveform.

reference sample, then instructing the program to subtract anything with the sample's characteristics from the vocal (Figure 1).

There are two cautions, though. First, make sure you sample the hiss only. You'll need only a hundred milliseconds or so. Second, don't apply too much noise reduction. About 6dB to10dB should be enough — for reasons that will become especially obvious in the next section. Otherwise, you may remove parts of the vocal itself, or add artifacts, both of which contribute to artificiality. Removing hiss makes for a much more open vocal sound that also prevents "clouding" the other instruments.

DELETE SILENCES

Now that we've reduced the overall hiss level, it's time to delete all the silent sections between vocal passages. If you do this, the voice will mask hiss when it's present, and when there's no voice, there will be no hiss at all (also see the Power App Alley in this issue on Sonar 6, which describes how to reclaim disk space when removing silence).

With all programs, you start by defining the region you want to remove. From there, different programs handle creating silence differently. Some will have a "silence" command that reduces the level of the selected region to zero. Others will require you to alter level, like reducing the volume by "-Infinity" (Figure 2). Furthermore, the program may introduce a crossfade between the processed and unprocessed section, thus creating a less abrupt transition. If it doesn't, you'll probably need to add a fade-in from the silent section to the next section, and a fade-out when going from the vocal into a silent section.

REDUCE BREATHS AND ARTIFACTS

I feel that breath inhales are a natural part of the vocal process, and it's a mistake to use hard disk recording to get rid of these entirely. For example, an obvious inhale cues the listener that the subsequent vocal section is going to "take some work."

That said, applying any compression later on will bring up the levels of any vocal artifacts, possibly to the point of being objectionable. I use one of two processes to reduce the level of artifacts.

The first option is to simply define the region with the artifact, and reduce the gain by 3dB to 6dB (Figure 3). This will be enough to retain the essential character of an artifact, but make it less obvious compared to the vocal.

The second option is to again define the region, but this time, apply a fade-in (Figure 4). This also may provide the benefit of fading up from silence if silence precedes the artifact.

Mouth noises can be problematic, as these are sometimes short, "clicky" transients. In this case, you can sometimes cut just the transient, and paste some of the adjoining signal on top of it (choose an option that mixes the signal with the area you removed; overwriting might produce a discontinuity at the start or end of the pasted region).

PHRASE-BY-PHRASE NORMALIZATION

A lot of people rely on compression to even out a vocal's peaks. That certainly has its place, but there's something you need to do first: Phrase-by-phrase normalization.

Unless you have the mic technique of a k.d. lang, the odds are excellent that some phrases will be softer than others. If you apply compression, the lower-level passages might not be affected very much, whereas the high-level ones will sound squashed. It's better to get the entire vocal to a consistent level first, *before* applying any compression. This will retain more overall dynamics. If you need to add an element of expressiveness later on (*e.g.*, the song gets softer in a particular place, so you need to make the vocal softer), you can do this with judicious use of automation.

Referring to Figure 5, the upper waveform is the unprocessed vocal, and the lower waveform shows the results of phrase-by-phrase normalization. Note how the level is far more consistent in the lower waveform.

However, be very careful to normalize entire phrases. You don't want to get so involved in this process that you start normalizing, say, individual words. Within any given phrase there will be a certain internal dynamics, and you definitely want to retain them.

ARE WE PREPPED YET?

DSP is a beautiful thing. Now our vocal is cleaner, of a more consistent level, and it has any annoying artifacts tamed — all without reducing any natural qualities the vocal may have. At this point, you can start doing more elaborate processes, such as pitch correction (but please, apply it sparingly and rarely!), EQ, dynamics control, and reverb. But, as you add these, you'll be doing so on a much firmer foundation. EQ

BEATING PHASE ISSUES IN YOUR DAW

by Jeff Anderson and Dr. P. T. Gilham

Phase cancellation occurs when two identical sound waves are equal and opposite to each other; in other words, as one waveform increases in amplitude, the other decreases at the same rate. If you add the instantaneous values of these two waveforms, the result is zero. As a result, the two sounds cancel out, and assuming they truly are identical, you will not be able to hear anything.

At this point, some of you (you know who you are!) are rushing to the www.eqmag.com forums to tell us that what we're really talking about is *polarity*, not *phase*. And by and large, you're right; polarity is a total "flip" of a signal that affects all frequencies, whereas phase changes can sometimes be associated only with specific frequency ranges (*e.g.*, phase variations in a speaker crossover). But everyone knows what we mean by "phase," so that's the term we'll use.

To avoid phase issues, the first step is to record everything in phase. This occurs through proper mic placement, and adhering to the "Three to One Rule": When using multiple mics, for every unit of distance from the sound source, the mics should be at least three units apart from one another. This means that if one mic is three inches from a guitar cab, the second mic should be at least nine inches from that mic. But what if there are still phase issues, we're not into re-cutting the track, and are stuck with fixing it in the mix?

DEMONSTRATING PHASE CANCELLATION IN YOUR DAW

First, let's learn more about phasing. Here's a way to demonstrate phase cancellation and learn the proper techniques for aligning your tracks in your DAW, as well as get a feel for the process of correcting phase.

1. Make two new audio tracks in your DAW. Assign both outputs to the same speaker or channel, and pan both to center.
2. Record a sine wave onto each of the two tracks. To do this with Pro Tools, record your Signal Generator plug-in's output into each track. You can also use a standalone signal generator found with common studio maintenance equipment, a plug-in similar to Signal Generator, a digital audio editing program that can generate sine waves (Sound Forge, Wavelab, Audition, Peak, etc.; see Tech Bench, 11/06 issue), or even an oscillator found on a mixing console or tape machine. The frequency of the sine wave doesn't matter, but a 300Hz tone is a good place to start.
3. Put your DAW in play mode, and listen to the outputs from each track. If you turn track one on and off, and assuming that both sine waves start at the same point in their cycle, you'll notice that the combined amplitude of both waves is louder than a single track's amplitude.
4. Zoom all the way in so that you can analyze the peaks and valleys of each of the waveforms (Figure 1).
5. Leaving track one alone, select track two and physically move the waveform to the right or left. As you move track two around in time, listen to the combined outputs from both tracks. Note that once the two sine waves are opposite of each other (*i.e.*, the crest of the wave on channel one aligns with the valley of channel two), both tracks will become inaudible, or close to it if the channels aren't perfectly matched.
6. Experiment further with moving one of the waveforms around in time while

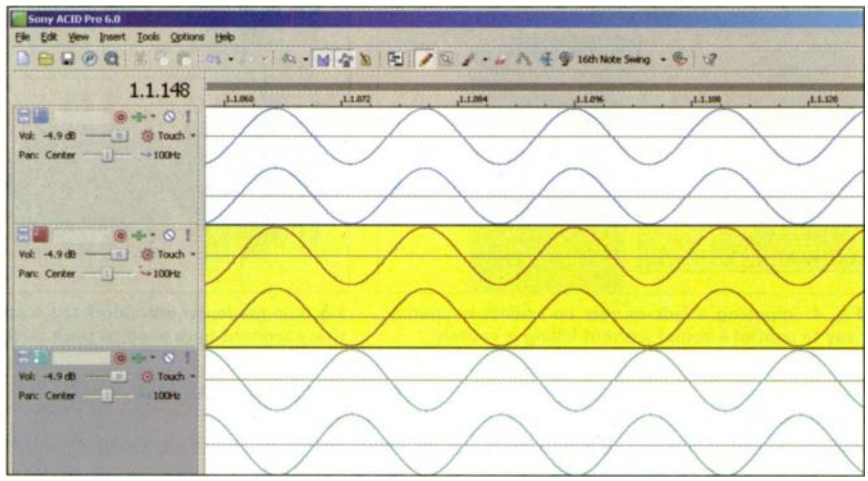


Fig. 1: Three stereo sine waves have been loaded into Sony Acid Pro 6. Taking the middle (yellow) waveform as a reference, the upper waveform is in-phase with the reference — the crests and valleys for both signals match. The lower waveform is out-of-phase, as its crests and valleys are equal and opposite to the reference waveform.

DOES "ABSOLUTE POLARITY" MATTER?

Consider a kick drum: When it's hit with a beater, a rush of air pushes forward from the front of the drum and to the listener. Now consider that same kick drum playing back through a speaker. Does the hit cause the speaker to move out, thus pushing air, or is there reversed polarity somewhere in the picture, so that the speaker "sucks in" air?

If the speaker moves out, it's considered to match the absolute polarity of the kick drum. Now, most texts will tell you that a sound heard in isolation (*i.e.*, not compared to any other sound) will sound the same whether its polarity is absolute or reversed. And in theory, this makes sense. Yet some people can perceive when something like a kick drum is "sucking" instead of "pushing," thus making a case that absolute polarity does matter, and can make a difference to the overall sound. —*Craig Anderton*

listening to the sum of both of them. Notice how the volume changes based on the location of each waveform; the amplitude will double if the peaks and valleys match up, but if they're out of phase, the sound will be attenuated greatly — or even inaudible.

REAL WORLD PHASE TECHNIQUES

Instead of using a sine wave, let's now apply the same concept to instruments that we find in our recordings. Let's say you're mixing a song with two snare tracks (*i.e.*, top and bottom mics were used to record the drum). Having one source with more than one mic may mean phase issues; if you find your snare tracks are out of phase, simply invert the polarity of the bottom snare track (flip the signal upside down so the crests of the waveform are now in phase with each other), and you're probably set.

But not always. While this works for some tracks, other tracks may be more problematic. Say you have two guitar tracks (one track is a Shure M57 directly on the grille of the cabinet while the second track is a Neumann U47 a few feet back from the amp to capture some of the room), or two bass tracks (a DI signal as well as an amp signal). In the case of the guitar, the amp output had to travel further to the room mic than it did to hit the close mic. The bass signal, likewise, had to go further to hit the amp than it did to hit the DI. Temporarily mix the two signals together in mono; if the sound becomes weaker instead of stronger, you have phase issues that need to be addressed.

In this case you should follow steps 4–6, moving the second track so that it aligns with the first. The two tracks may not look identical (especially for the guitar tracks, as they are from different mics that have different frequency responses, positioned at different distances from the source) — but the similarities will be clear enough to allow you to position the tracks accordingly. *[Note: One good way to give yourself a point of reference when recording your bass is to track yourself plugging in at the beginning of the take. The "pop" of the cable will give you a nice spike on your waveform that you can then use to help you align all of your tracks.]*

By moving your tracks around in time, you can greatly alter the sound of the finished product — so it's a good idea to experiment with some phase tricks even when your tracks aren't canceling out totally. For example, I've noticed that flipping the polarity of a single kick drum track can make the drum sound more punchy and direct. Similarly, inverting the phase on a hi-hat track can make the hats really cut through the mix.

However, when checking phase between two tracks, always monitor them in mono until you've determined that the phase relationship is correct. Two out-of-phase signals can give a gloriously wide sound when panned in stereo (this was the basis of "stereo simulation" in many effects boxes), but they'll disappear completely when played back in mono. EQ

WINDOWS XP TWEAKS FOR AUDIO

BY CRAIG ANDERTON

Cheat Sheet delivers concise, explicit, step-by-step information on how to do specific recording/audio-related tasks. This installment describes useful tweaks to Windows XP for pro-audio applications.

PRIORITIZE BACKGROUND SERVICES

ASIO works "in the background" compared to other computer tasks. So, it's important to prioritize Windows for Background Services to minimize audio dropouts and related problems.

- Right-click on **My Computer** and choose **Properties**. Click on the **Advanced** tab. Click on **Performance Settings**. Click on **Advanced**. Under **Processor Scheduling**, click on **Background Services**. Click on **OK**.

TURN OFF INDEXING

Indexing (the Mac has a similar function) builds a catalog of files in order to allow speedier searches, as the computer need only search this catalog to find particular files, rather than scan the entire hard drive. While convenient, indexing can degrade hard drive performance while the indexing occurs. One sign that indexing is occurring is if your hard drive shows lots of activity that doesn't seem to relate to what you're doing. Turning off indexing is simple — if you know where to look.

- Double-click on **My Computer** to show the disk drives in your computer. Right-click on a disk drive icon, and select **Properties**. Click on the **General** tab. Toward the bottom of the window, uncheck **Allow Indexing Service to index this disk for fast file searching**. Click on **OK**.

CHOOSE HIGHEST PERFORMANCE GRAPHICS

Windows XP's eye candy is fun, but it requires some CPU power. To maximize CPU efficiency for audio, set graphics for high performance, instead of best appearance.

- Right-click on **My Computer** and choose **Properties**. Click on the **Advanced** tab. Click on **Performance Settings**. Click on the **Visual Effects** tab. Click on **Adjust for Best Performance**. Click on **OK**.

SET GRAPHICS FOR CLASSIC MODE

This is another graphics-oriented tweak. In case you're wondering whether it's worth it (these days, graphics performance tends to be offloaded to graphics cards, requiring less CPU), it does make a difference, and graphics performance can influence audio performance.

- Right-click on the desktop. Click on **Properties**. Click on **Appearance**. Under **Windows and Buttons**, choose **Windows Classic Style**, instead of Windows XP Style. Click on **OK**.

NUKE UNNEEDED STARTUP ITEMS

When you boot your computer, many programs install code that instructs Windows to load that code on startup. But do you really want, say, RealPlayer or iTunes Helper to be loaded into RAM and demand your computer's attention? We thought not, so here's how to prevent these from loading (and reduce boot time, as well).

- Click on **Start** and click on **Run**. In the Run box, type **msconfig**. Click on **OK**. Click on the **Startup** tab. Uncheck anything you don't think you need, including performance degraders like **Microsoft Fast Find**. After unchecking what's not needed, click on **OK**.

TAKE CONTROL OVER SYSTEM RESTORE

System Restore has saved me more than once. It's a cool feature, but you don't need to have it on all the time, as it uses system resources. I simply set a restore point manually as needed (like before installing a new piece of software or swapping a driver). Here's how to turn off System Restore. (Note that you can't turn off system restore on your C: drive without turning it off on all drives.)

- Right-click on **My Computer** and select **Properties**. Click on the **System Restore** tab. Check the **Turn off System Restore on all drives** box. Click on **OK**.

ELIMINATE THE BACKGROUND SCREEN

If you have a pretty desktop picture, it's loaded into memory. If the picture is a large bitmap, you could be losing a few megabytes of RAM (this may sound insignificant, but every byte helps). You can reclaim this memory by setting a simple background color, like black.

- Right-click on the **desktop**. Click on **Properties**. Click on **Desktop**. Under **Background**, scroll to the top of the list and select **None**. In the **Color** drop-down menu, choose the color of your choice. Click on **OK**.

REDUCE HARDWARE ACCELERATION

This is *not* a recommended tweak unless you have corruption with bit-mapped graphics (for example, dragging a window causes parts of it to disappear or smear). Also try this tweak if using the mouse causes freezes.

- Right-click on the **desktop**. Click on **Properties**. Click on **Settings**. Click on **Advanced**. Click on the **Troubleshoot** tab. Move the hardware acceleration slider **one notch to the left** of Full. Click **OK**, then click on **OK** again. (If this doesn't solve the problem, repeat the procedure, but try moving the slider **two notches to the left** of Full. If this doesn't solve the problem, look elsewhere for the solution — perhaps a new graphics card driver.)

INCREASE MONITOR REFRESH RATE

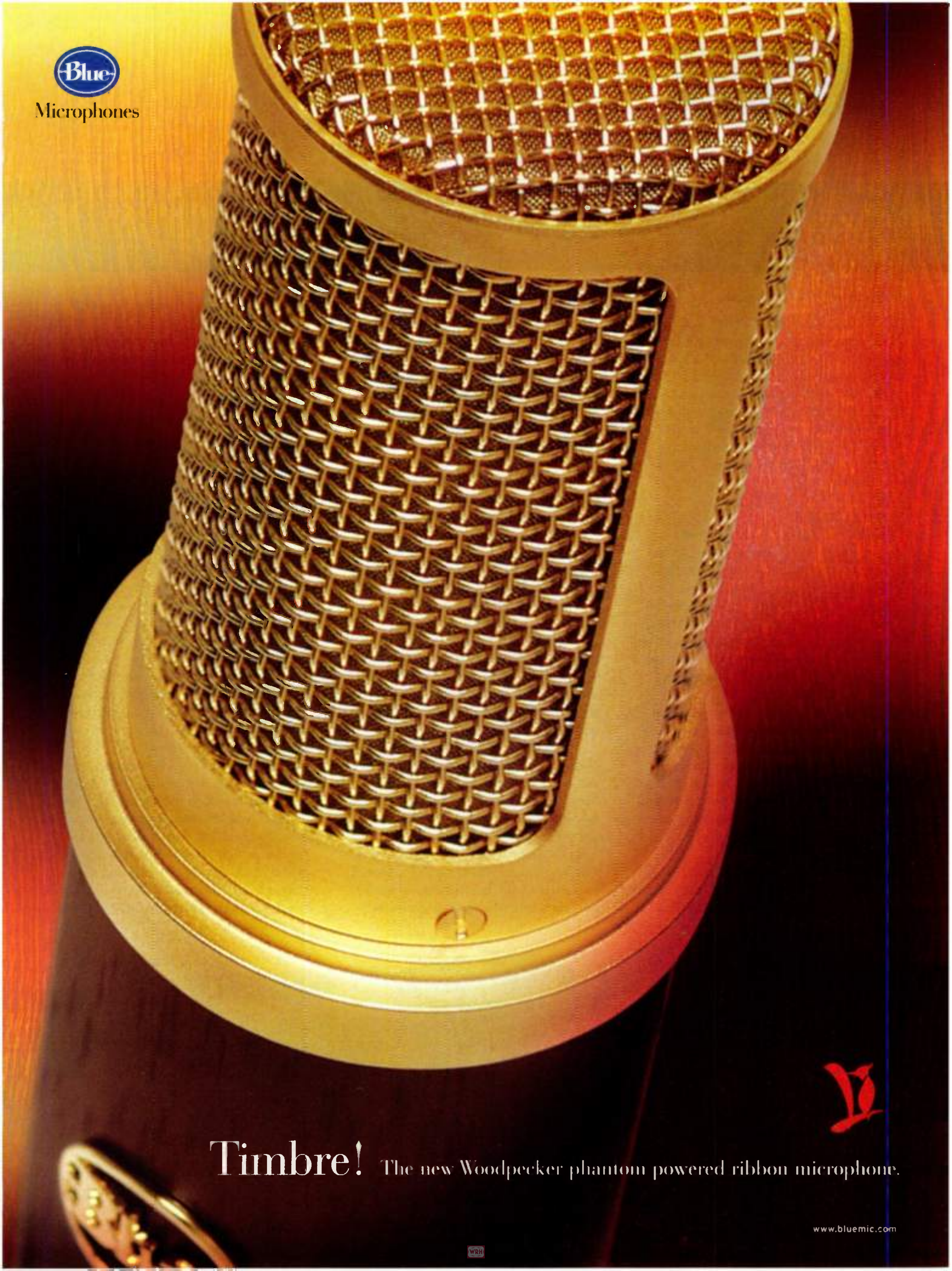
This doesn't improve the computer's performance, but it will make staring at the screen for long periods of time much more bearable if you use a CRT monitor (this tweak is not relevant with LCD monitors). The default refresh rate for most monitors is 60Hz, which can cause noticeable flickering. Raising this gives a steadier, less fatiguing image.

- Right-click on the **desktop**. Click on **Properties**. Click on **Settings**. Click on **Advanced**. Click on the **Monitor** tab. Check the box that says **Hide modes that the monitor cannot display**. Under **Screen Refresh Rate**, choose a higher refresh rate, like 70Hz. Click on **OK**, then click again on **OK**.

GET NORTON OFF YOUR COMPUTER

I've known several people who bought a computer with a trial version of Norton Utilities installed, but didn't subscribe to it, choosing to use some other kind of protection instead. This can lead to conflicts that bring your computer to a crawl, and using the standard uninstall option in Windows won't clean out all Norton remnants.

- Instead, go to http://service1.symantec.com/Support/tsgeninfo.nsf/docid/2005033108162039?Open&src=short_instuninst&docid=2004101207033236&nsf=nip.nsf&view=docid&pid=2004101207033236&pkb=nip and download the Norton removal tool. Follow the instructions to remove all Norton components from your computer, but ignore the part where it wants you to reinstall the software.



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TC ELECTRONIC C400XL

Dual Stereo Gate/Compressor for Studio and Stage

by Jeff Anderson

I bet you're expecting what I was expecting when I first heard about this product — another two channel box with the standard ratio, threshold, attack, and release controls, and maybe a stereo link. But there's a lot more going on in this unit than you might think: tons of presets, multi-band dynamic compression, and parallel processing . . . let's check it out.

OVERVIEW

You can configure the two channels as two independent mono engines (which can be effective as two inserts on two separate tracks), as a stereo unit, or run them in "serial" mode, where the signal passes internally from the output of channel one into the input of channel two. This is perfect for when you want to use both a gate and a compressor on one track. Very cool.

Eighteen presets cover most instruments, like kick drum and bass guitar, as well as de-essing and de-hissing applications. These presets set the multiband compression, limiting, expansion, and gating parameters, and are especially helpful in "on-the-fly" live scenarios (for which the C400XL is ideal, in addition to its studio capabilities).

Each compressor has controls for threshold, ratio, makeup gain, and mix of the processed and straight levels. The gates offer control over threshold, ratio, release time, and processed amount. There are also input meters, as well as a "damp" meter that indicates the amount of attenuation from –1dB to –24dB. In addition, a threshold LED can notify you either of levels exceeding the preset threshold in the multiband compressor, or the input signal triggering the peak limiter. Around the back, you'll find XLR stereo connections, AES digital in and out jacks (also XLR), and MIDI ports for uploading software.

IN USE

I've been finishing up a four-song EP, so I decided to put the C400XL to test on the mix, working from the kick drum up. Putting the unit in dual mono mode, I inserted the C400XL into my console, setting the preset to Bass Dm. After a few tweaks (setting the threshold at –40 and the ratio at 3), and applying a total of about –6dB of compression, I had a pretty good sound.

I then tried out the gate. For those not familiar with gates, they're designed for reducing noise and act like an audio "door" that opens and closes, letting the signal through or blocking it. The threshold control determines when the door will open based on the input signal level (*i.e.*, low-level signals like hiss aren't sufficient to open the gate, but higher-level signals are), ratio sets how long the gate stays open, and release determines how fast the gate closes.

The gate worked wonderfully, but I noticed that I couldn't retain the "click" of the kick drum with the gate engaged. Usually I'll set the threshold very high to get a click from the gate opening really quickly, but presumably there's a very short attack time because I couldn't achieve this effect with the C400XL.

Still, the kick sounded good, so I moved to the snare. As I wanted to use both the gate and the compressor, I set the unit into dual mono mode. The drummer was playing his hats louder than his snare, so I wanted to use engine one as the gate to help get all that hi-hat out of the snare track. Hitting the gate hard (with a high threshold setting) solved this problem nicely, perhaps better than some of my other gates would have.

Engine two was the compressor, so I set the preset to Snare Dm and started tweaking the threshold. I understand now why this unit is so good for live apps: The threshold knob has a ton of play, and you can affect the level of compression greatly by slight knob adjustments. But this brings me to my main gripe with the C400XL. Typically when adjusting compression levels, I'll watch the meters as I get the desired amount of gain reduction, then do a "reality check" by hitting a bypass button to compare the processed and dry sounds. But on the C400XL there

TC Electronic C400XL

is no bypass button — the compressor is bypassed only when the mix knob is turned all the way to the left while, generally, I will have the knob turned all the way to the right. This is annoying, and I hope TC adds a bypass button on their next generation of this box.

But it sounds good, so I can't complain much. Adjusting the makeup knob to set my levels, I recorded the C400XL's output into Pro Tools so I could use the processed snare track in my mix, and free up the unit for the bass track.

Or should I say, "bass tracks." I had three: a DI track, a signal from a sweet Ampeg SVT 3 Pro, and a SansAmp track. Usually, I just run to my Universal Audio 1176 to compress bass. Old habits die hard, but it was fun to work with something else for a change. Inserting the C400XL into the DI channel of my console, I turned the preset to Bass Guitar, set the ratio to 4, and cranked the threshold until I got about –12dB of compression. I was beating the hell out of the track, and I noticed that the C400XL performed similarly to the old dbx 160X, as it added a good amount of "gloss" and "polish" to the DI track. After mixing in a bit of the amp and the SansAmp signals, we were good to go.

The vocalist I was working with, Malachi Jagger, has an extremely smooth and well-controlled voice, so I wasn't looking for a compressor to keep his volume under control; I needed something to help his voice stand out in front, and cut through the mix. So I inserted the C400XL, set the preset to Male Vocals, used a ratio of 5, and achieved a good sound that was fairly transparent. Still, there was a bit of sibilance to his track, so I used the De-esser preset at about 6kHz, and really cranked it up. Perfect — just what I wanted.

CONCLUSIONS

This is not at all your typical two-channel gate/compressor combo; it has some unique features that are usually found only in much more expensive units. The presets are incredibly helpful, and they make this box useful in both live sound and studio recording environments. So if you're looking for something that can pull double-duty, want multiband compression with parallel processing capabilities, and excellent sound quality, for the price the C400XL really can't be beat. Now if they would just add a bypass button! EQ

PRODUCT TYPE: Stereo/dual mono/serial compressor with gate.
STRENGTHS: Sounds great. Short learning curve. Useful choice of presets. Good metering capabilities. High bang-for-the-buck ratio.
LIMITATIONS: Needs a bypass button, not just a mix knob. Weird jacking system around back (all are upside down). Tough to get a "click" out of the gate if you want it.
LIST PRICE: $349
CONTACT: www.tcelectronic.com

M-AUDIO PROFIRE LIGHTBRIDGE

The Fine Art of Excellent Boredom

by Craig Anderton

Boring can be bad — like synthesizers that use the same architecture Bob Moog hit upon 35 years ago, a guitar that's the 9,475th knock-off of a Fender Stratocaster, or movies featuring pop stars. But boring can be good — *very* good — when you're dealing with Things That Connect to Computers. Concerning an audio interface, you want to set it up, plug it in, then forget that it exists.

Which brings us to the delightfully boring ProFire LightBridge (Figure 1), a FireWire 400-to/from-ADAT lightpipe bridge that handles 32 channels at 44.1/48kHz or 16 channels at 88.2/96kHz. It's cross-platform, and incorporates:

- ASIO, Core Audio, WDM, and GSIF II drivers
- A headphone out with dedicated level control
- Balanced 2-channel 1/4" outs with level control
- Four ADAT in optical connectors and four ADAT outs
- Basic front panel indicators for lightpipe and MIDI activity, sample rate, and sync
- A multipin connector with an appropriate breakout cable that accommodates MIDI (including MIDI Time Code and MIDI machine control over two DIN 5-pin jacks), word clock on dual BNC connectors, and S/PDIF I/O with dual phono jacks.

If you want more specs, that's why websites were invented — so check out www.m-audio.com. What they don't tell you is that this is a sturdy box with a substantial feel, and is sufficiently compact that it will sit politely in a corner of your desktop as it dutifully shuttles audio back and forth between your computer and lightpipe-compatible devices.

WHAT'S A LIGHTPIPE?

When Alesis introduced the ADAT digital multitrack recorder, it came with an unusual digital audio interface: a thin, somewhat sexy fiber-optic cable capable of handling 8 channels of 48kHz/24-bit audio. This provided an elegant, inexpensive solution for transferring audio, as it didn't require a multiconductor cable with esoteric (read: expensive) connectors.

As the ADAT juggernaut marched across the world of recording, other devices incorporated ADAT interfaces to hitch a ride on its popularity. The digital mixer was a logical candidate, as you could pipe the output from the mixer directly into the recorder for tracking, or send signals from the ADAT to the mixer while mixing. Then came audio interfaces, which usually incorporated a lightpipe interface to accept and/or send the signal from ADATs; AD/DA converters also tended to favor lightpipe interfacing.

By the late '90s, the ADAT was well on the road to extinction, having been bested by cheap computing power and world-class digital recording software. But the ADAT interface refused to die. It adapted to the 88.2/96kHz world by using multiplexing (SMUX) technology to send four channels of high-resolution audio over the standard 8-channel hardware, and new product categories elected to go lightpipe — such as octal mic preamps that fed into audio interfaces.

What's more, the ADAT interface doesn't seem destined to go away any time soon. Its main competitor at the time, TASCAM's TDIF interface (which was based on multiconductor cable) is now an historical footnote, and MADI (a promising 56-channel interface) hasn't really gotten any traction other than in high-end applications.

APPLYING THE PROFIRE LIGHTBRIDGE

First up: installation. I elected to subject a dual G5 Mac to this process, because frankly, it's a magnet for FireWire issues and I figured if the LightBridge worked on this, it would work on anything. I went to the M-Audio site to download the latest drivers (always a good idea; Windows fans, note there are drivers for Vista 32-bit) and installed them. Kudos to M-Audio for warning you to turn off your computer before connecting the device.

Fig. 2: The control panel applet lets you determine sync, active ports, and other crucial hardware details.

Fig. 1: The ProFire LightBridge's front panel.
It takes less time to learn than it does to say the product's full name.

Upon power-up with the LightBridge connected, the computer emitted an intermittent, high-pitched sound whenever the main hard drive did disk access. This had happened with almost all other FireWire devices I've connected, so I turned off "Allow Nap" under *Apple Menu > System Preferences > Processor* (this requires downloading and installing the CHUD utility from the Apple website; for details, see "The Software Power User Guide" in the 9/07 *EQ*) and as with the other devices, the problem went away.

Then things started to get boring. I went to the Mac's Audio MIDI Setup, and yup, the interface was there. Part of the driver installation process deposits a control panel applet (Figure 2) where you set things like sync source, active ins/outs, and the like; it also worked as expected. So I hooked up my venerable (and still wonderful) Panasonic DA7 mixer, then booted up MOTU's Digital Performer 5 to see if all was well in audio-land.

I tried syncing the DA7 to the LightBridge: Check. Vice-versa: Check. Send audio to Digital Performer and record it: Check. Route the recorded audio from Digital Performer back to the DA7: Check. Sync LightBridge to DA7 word clock: Check. Sync DA7 to LightBridge word clock: Check. Try S/PDIF: Check. Monitor output: Check (and there's a decent amount of headphone level). Observe lights on front panel: Check.

By this time, I was getting pretty bored so I thought I'd liven things up by transferring eight tracks of audio (courtesy of Sonar) in my Windows machine through my Creamware SCOPE card's ADAT out to the LightBridge's ADAT in, and thence into Digital Performer. Check. As a last resort to try to find *something* to complain about, I lowered the sample buffer down to a CPU-straining 64 samples. That worked fine, too.

CONCLUSIONS

The ProFire LightBridge does exactly what it promises to do, and does so with zero fuss. It's easy to install, easy to use, and I don't even have to make a subjective value judgment about converters because it's all digital — and if there's any jitter, I sure can't hear it.

So the bottom line is simple: File under "boring can be good," because — thankfully — the LightBridge is both. EQ

PRODUCT TYPE: ADAT lightpipe to/from FireWire interface converter.
TARGET MARKET: Those needing to interface ADAT lightpipe-compatible devices (mixers, converters, mic pres, etc.) with computer-based systems.
STRENGTHS: Accepts/outputs 32 channels of ADAT lightpipe I/O and two channels of coax S/PDIF I/O. Additional stereo analog line out and headphone out. Cross-platform. Includes 4-pin to 6-pin and dual 6-pin FireWire cables (each six feet long). Good documentation.
LIMITATIONS: Nothing significant.
LIST PRICE: $499.95
CONTACT: www.m-audio.com

Hard Lines

PEAVEY M1 AND M2 STUDIO PRO SERIES MICS

Meet a Pair of Condenser Contenders

by Jeff Anderson

Admittedly, in the recording world, Peavey is not the first manufacturer that comes to my mind when discussing studio mics — to me, the company is synonymous with guitars, basses, amps, live sound gear, and the like. Why? Because while they've produced some great recording tools over the years (some of which were truly innovative, like the StudioMix), many of their products have flown "under the radar."

Now here we are with two (relatively) new mics in hand, and they say Peavey right on the box. Will they help focus more attention on what Peavey can bring to the studio?

OVERVIEW

First off, both mics are condensers. Cosmetically they are very similar, but there are some important differences. For example, the M1 has a cardioid pattern only, while the M2 allows choosing among cardioid, omni, and figure eight patterns. And diaphragms? The M1 has one and (surprise!) the M2 has two.

Now to the similarities: Both mics are based around large capsules with gold membranes. They each have a –10dB pad (which, when engaged, allows either mic to handle 140dB without distorting). They also both sport a low frequency rolloff feature, offer an overall response of 30Hz to 20kHz, and include a swivel mount and case.

Now that we've covered the basics, let's listen to how they perform.

APPLICATION

First impression from the requisite "plug them in, sing through them, and see how they sound" test: dark. Now, that's not a bad thing — there are many applications in which you'll find that character incredibly useful. But they are definitely not on the "bright" or "shrill" end of the scale.

So I try them as drum overheads. Why? Because nowadays I use overheads primarily for cymbals. Fifteen years ago, I

Peavey Studio Pro M1

Peavey Studio Pro M2

would have been looking for my overheads to pick up pieces of the entire kit, minus the kick. But now that I have unlimited tracks (thanks to digital!) and can use multiple mics to pick up each drum (as well as the room), I find myself lowering overheads and placing them strategically towards the cymbals so that the mics' primary task is capturing the cymbals.

This application is where darker mics come in extremely handy — most cymbals can use a bit of mellowing, and dark mics perform that function well. This was certainly the case with these mics. I set the M2 to cardioid so that it matched the M1, engaged the –10dB pads on both, and sent them through the preamps of my DDA DVM224v console, directly to Pro Tools. Starting at about 28" above the cymbals, I noticed the mics' low-end frequency response combated the harsh, "shrilly" quality of the cymbals — but I thought it would be appropriate to add a little more brightness. Lowering the mics to around 15" above the cymbals, about one foot apart from each other and cocked outwards away from one another, added just enough brightness to the sound. The positioning of the mics (facing away from each other) gave a good stereo image of the cymbals, and I was very happy with the resulting sound.

Next I tried the M1 and the M2 paired with a Shure SM57 (running into a Nice Pair preamp with a moderate gain stage, and into Pro Tools) on a Mesa Boogie Triple Rectifier (the source guitar was a '60s Gibson Les Paul). With the 57 around two fingers from the grill, off-axis from the speaker, and the M1 and M2 both about 20" from the amp, I had to engage the pads to cool down the cranked signal. As I expected, the sound was a bit dark due to the low frequency response, so I flipped the rolloff switch to see how it would sound.

It wasn't bad, but I decided we would be better off focusing on room sounds. So I took the M1 away, flipped the M2 into the omni pattern, and moved the mic a few inches closer to the amp. Bingo: The result was a really balanced, even sound — part room and part amp.

CONCLUSIONS

For the price, these are some pretty good mics. They have a unique sound, and their relative "darkness" makes them very practical in certain situations. Both mics can handle some serious SPL, and they have all the necessary options to make them useful on many sources, from drum overheads to loud guitar cabinets to general room duties. If you already have mics that provide high-end sizzle, the M1/M2 could be the ideal complement to round out your mic locker. EQ

PRODUCT TYPE: Large diaphragm condenser microphones.
TARGET AUDIENCE: Budget recordists, or those looking to add a good, low-cost mic to their arsenal.
STRENGTHS: Fair price. M2 has multiple polar patterns. Particularly good for drum overheads.
LIMITATIONS: You have to purchase the shockmount separately.
LIST PRICES: M1 $349, M2: $499
CONTACT: www.peavey.com


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SAMPLING SPACES AND EFFECTS WITH WAVES IR-1

It's a Cool Reverb, But How Easy Is It to Create Your Own Reverb Effects?

by Bill Ring

Alhough Waves' IR-1 Native has earned a place as my main reverb plug-in, I'd never tried to create my own reverb effects because the program comes with so many great sounds. But then I decided to sell some of my older outboard gear, as well as move out of the refurbished barn where I've lived and recorded for several years. The barn has a fantastic ambient sound; what a pity I couldn't take the place with me . . . or could I?

YOU SAY YOU WANT A CONVOLUTION

A convolution reverb like the IR-1 can analyze what happens when a signal with known characteristics, such as a wide-range frequency sweep, plays through speakers in a room (or through an effects device or other processor). It then records the resulting spectrum to one or more audio files. Waves calls these "sweep response files" because the IR-1's impulse is created by a logarithmic frequency range sweep; other programs may use a pistol shot, or other sound that contains a wide spectrum of frequencies. The convolution reverb can then load the recorded file, and "impress" its characteristics on another signal. The process resembles sampling in the sense that it captures the characteristics of something that actually exists, rather than synthesizing a particular sonic characteristic.

(When sampling hardware devices, remember that the IR-1 is primarily intended to capture real room responses for reverb. When you sample processors that include effects like pitch shift and modulation, the results may be unpredictable — although possibly interesting, as I learned by sampling Eventide H3000 effects. On the other hand, sampled delays and reverbs should sound very close to the originals.)

STEP BY STEP, SLOWLY I CONVOLUTED . . .

Determined to have my sound, I took the plunge into impulse response recording. The instructions in the Waves documentation were sparse, so I thought others would find my method for stereo or mono sampling with Waves' IR-1 helpful. Once you've mastered this procedure, you can likely extend it to surround sampling with the IR-360.

To use IR-1 or IR-360 you must have a computer-based digital audio system for a host, so we'll assume you can use the same system to record your sweep responses.

1. Create a session. Start a new session (or "project," "song," etc., depending on your software) at the highest available sample rate and bit depth, up to 96kHz/24-bit. Create six mono tracks. Set the input to "monitor" mode rather than any auto-monitor option (the usual default).

2. Choose a sweep file. IR-1 includes a disc labeled "Sweep Files and Audio CD." Open it as a data disc in your computer, then locate the sweep file that matches your session's sample rate and bit depth. If you don't have the disc available, download sweep files from www.waves.com (the choices range from 44.1kHz/16-bit to 96kHz/24-bit).

3. Import the sweep file. Import the desired sweep file into tracks 1 and 2, and position both to start at *exactly* one second. Route track 1 to the left main output and track 2 to the right. To avoid feedback, tracks 3 thru 6 should *not* be routed to the main L/R output. Now "save as" a copy of this session as a template for future use, and continue working with the original.

4. Make a destination folder. During installation, IR-1 creates a folder named "IR-1 Impulses V2" that contains all of the Version 2 program's impulse response files. On a Windows computer, this folder's default location is C:\Program Files\Waves\Plug-Ins\IR-1 Impulses V2. Find this folder, then create your own folder within it to store your effects.

5. Prepare to record. We'll consider three possible modes: *mono* (mono in/out),

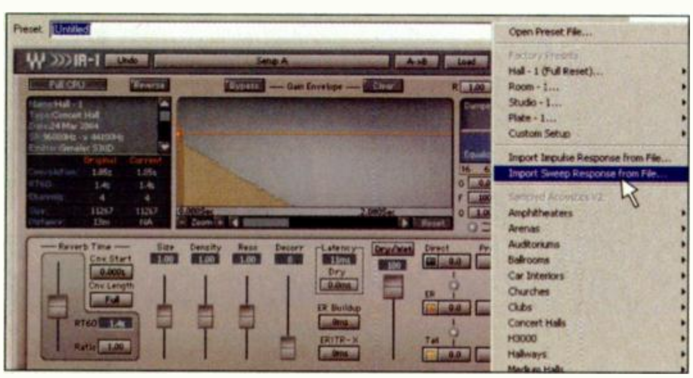

Fig. 1: After sampling an acoustic space or piece of hardware, the next step is to load the resulting sweep response file.

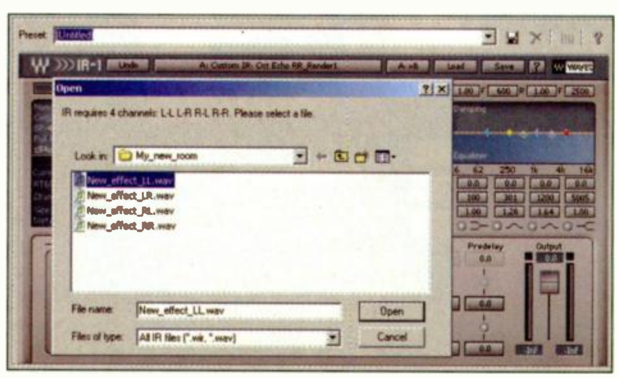

Fig. 2: Before you can create a preset, it's necessary to load your .WAV files in the order required by the program.

m/s (mono in/stereo out), and *full stereo* (stereo in/out). We'll cover each as if you're sampling a hardware device, but the method for sampling a real space is identical (see the sidebar "Sampling a Space with Mics and Speakers" for information on suitable speaker and mic setups). When sampling a room, the "device input" is the input of the power amp or powered speaker(s) that play back the sweep tone, and the "device output" is the output of the mic preamps that pick up the impulse. (Note: When sampling a space, begin playing the sweep at a very low level to avoid damaging the speakers, then raise the level to produce an adequate sound pressure level at the microphone position. Somewhere around 80 to 85dB SPL should be good, but the exact level is not critical.)

- **Mono:** This is primarily for mono hardware effects. Connect one of your computer system's main outputs to the device input, connect the device output to a record input on the computer system, route it to track 3, arm the track, and play the sweep. See that the signal flows properly to the device and back again, and adjust your levels at each stage for a strong signal with no clipping. Let the sweep play all the way through, because with some effects the output level will be much louder toward the end of the sweep.
- **M/S (mono/stereo):** Follow the above general procedure, but send the left device output signal to track 3, the right to track 4, and arm both tracks. Adjust gain as needed, making sure that the L and R levels are the same at each point in the signal path.
- **Full Stereo:** This requires two passes, and creates four response files. Connect the left and right out from the computer to the left and right in of the sampled device. Send the left device output to tracks 3 and 5, and right out to tracks 4 and 6. Arm tracks 3 to 6, play the sweep, and set your levels. Be sure to set identical input gain on all four tracks, and do not change levels between record passes. (Note that while you can create a stereo effect by recording L and R responses in a single pass, then using IR-1 "efficient" mode to process the files, there's no real need to do this. The two-pass method takes little extra time, and once you have a full stereo effect, you can always load it into IR-1 "efficient" to conserve processor power if needed.)

SAMPLING A SPACE WITH MICS AND SPEAKERS

In addition to playback and recording equipment, you will need one or two monitor speakers, two identical preamps, and a matched pair of mics. Either cardioid or omni-directional mics will work, depending on how you intend to set them up. Every element in the signal chain should have a frequency response as close to perfectly flat as possible. For suggestions about what equipment to use, see the detailed descriptions of the effects supplied by Waves.

For mono/stereo recording, center a single speaker at the performance position. This is considered best for a reverb you intend to apply to monaural sources such as a lead vocal, or single instruments. To capture a full stereo image, place two speakers symmetrically left and right of center. I know of no "rules" about how far apart the speakers should be; probably the best approach is to record a number of different responses with the speakers set at different, carefully documented spacings.

There appears to be more of a consensus on mic placement, although there's certainly no harm in experimenting. The orthodox methods are either to position two omnidirectional mics roughly 1.5 meters apart, or a pair of cardioid mics in an X-Y configuration at an angle of 110°, with the capsules exactly 17cm (6.7") apart. Distance from the speakers will depend on the characteristics of the space you are sampling and the type of sound you want to create. Again, multiple recordings at various distances is a sensible approach.

To sample my barn, I chose a single speaker (a Mackie HR824) and a pair of Beyer M201 microphones in an X-Y pattern. As the space is fairly small, my main purpose was to capture an ambience that would be useful on vocals and solo instruments. I'm pleased with the results so far, but I intend to try again soon with a pair of the new Neumann KMD series digital microphones (which I'm scheduled to review in an upcoming issue of EQ — stay tuned).

The barn ambience to be sampled; note the two mics and the speaker.

Photo: Marlis Momber

6. Record sweep responses.

- **Mono or M/S:** Start recording from zero time and be sure to run past 26 seconds. The actual sweep lasts only 15 seconds, but the program expects to see a full 10 seconds after the sweep ends to allow for a long reverb tail.
- **Full Stereo:** For the first pass, mute the right output from the computer system and disarm tracks 5 and 6. Record as described above. For the second pass, disarm tracks 3 and 4 and arm 5 and 6. Mute the computer system's left output and unmute the right. Record as before; on all record passes, be sure to begin at zero and continue past 26 seconds.

7. Trim the sweep response files. Trim the recorded tracks to discard any signal before 1 second and after 26 seconds. This ensures your effect will have the correct predelay. (After you're proficient with recording room samples, try taking the process to the next level by changing the predelay time, thus making the apparent listening position sound further from, or closer to, the stage. Do this by trimming the response to begin a little earlier or later than I second; but avoid cutting off the response's start, and remember each region must be exactly 25 seconds long.)

8. Save each trimmed sweep response file as a .WAV file. The destination should be the folder created in step 4. In Pro Tools, use the "export selected as files" command in the Audio Regions List Menu. With other software, you may need to "render" the selected regions to save them as new .WAV files. Name each .WAV file for the particular effect you're sampling, with a suffix to indicate the type of sweep response it contains — when making sever-

SAMPLING SPACES AND EFFECTS WITH WAVES IR-1

al recordings of the same effect with different parameter settings, give each variant a unique effect name. With some programs it's easiest to let the software name the files as they are saved, then rename them manually. Name the files as follows:

- **Mono:** [effect]_M.wav
- **M/S:** [effect]_L.wav and [effect]_R.wav
- **Full stereo:** [effect]_LL.wav, [effect]_LR.wav, [effect]_RL.wav, [effect]_RR.wav (in that order when exported from tracks 3 to 6).

9. Create the impulse response file. Insert IR-1 into a bus in your session. Choose the version that matches your sampling mode ("mono," "m/s," or "full"). Open the IR-1 window, click on Load, then select "import sweep response from file" from the Load menu (Figure 1). In the "open" window that pops up (Figure 2), navigate to the folder containing your .WAV files and load them in the order required by the program: L, R for m/s, and LL, LR, RL, RR for full stereo. After selecting all the required .WAV files, IR-1 will crunch the numbers for awhile and store the result as a .WIR (Waves impulse response) file in your effects folder. The program will automatically name this file based on the first .WAV file you loaded.

10. Save a preset. In the main IR-1 window, click Load and select "Import impulse response from file" from the menu. Select the .WIR file created in step 9. Next, back in the main IR-1 window, click on Save and select "save to new file." This creates an .XPS file; name it as desired. Next, you will be asked to enter the preset name, which will appear in the load menu. Give both the .XPS file and the preset the effect name you chose in step 8.

11. Check it out! Remove the IR-1 insert, reinsert it, and load your new effect. If everything went correctly you should see your effect folder in the load menu and find your new preset inside). Import some music into the session, and give a listen. When you're satisfied with your effect, you can delete the .WAV files or move them to another location for safekeeping. *Be sure to leave the .WIR and .XPS files as they are.* If in the future you want to rename or move the .WIR file for some reason, you can do so, but afterwards you will need to perform step 10 to create a new preset.

Congratulations! You are now a member of the convolution reverb sampler's club. EQ

PRODUCT TYPE: Convolution reverb plug-in for creating ultra-relatistic ambiences.
TARGET MARKET: Studios that want a convolution reverb effect, but who also want to create new and original effects.
STRENGTHS: Excellent sound quality. Ships with over 120 useful impulses. Relatively low CPU requirements. Can use your own impulses to create novel effects. Many adjustable parameters.
LIMITATIONS: Documentation on creating your own impulses is sketchy. Not cheap.
PRICE: $800 (list)
CONTACT: www.waves.com

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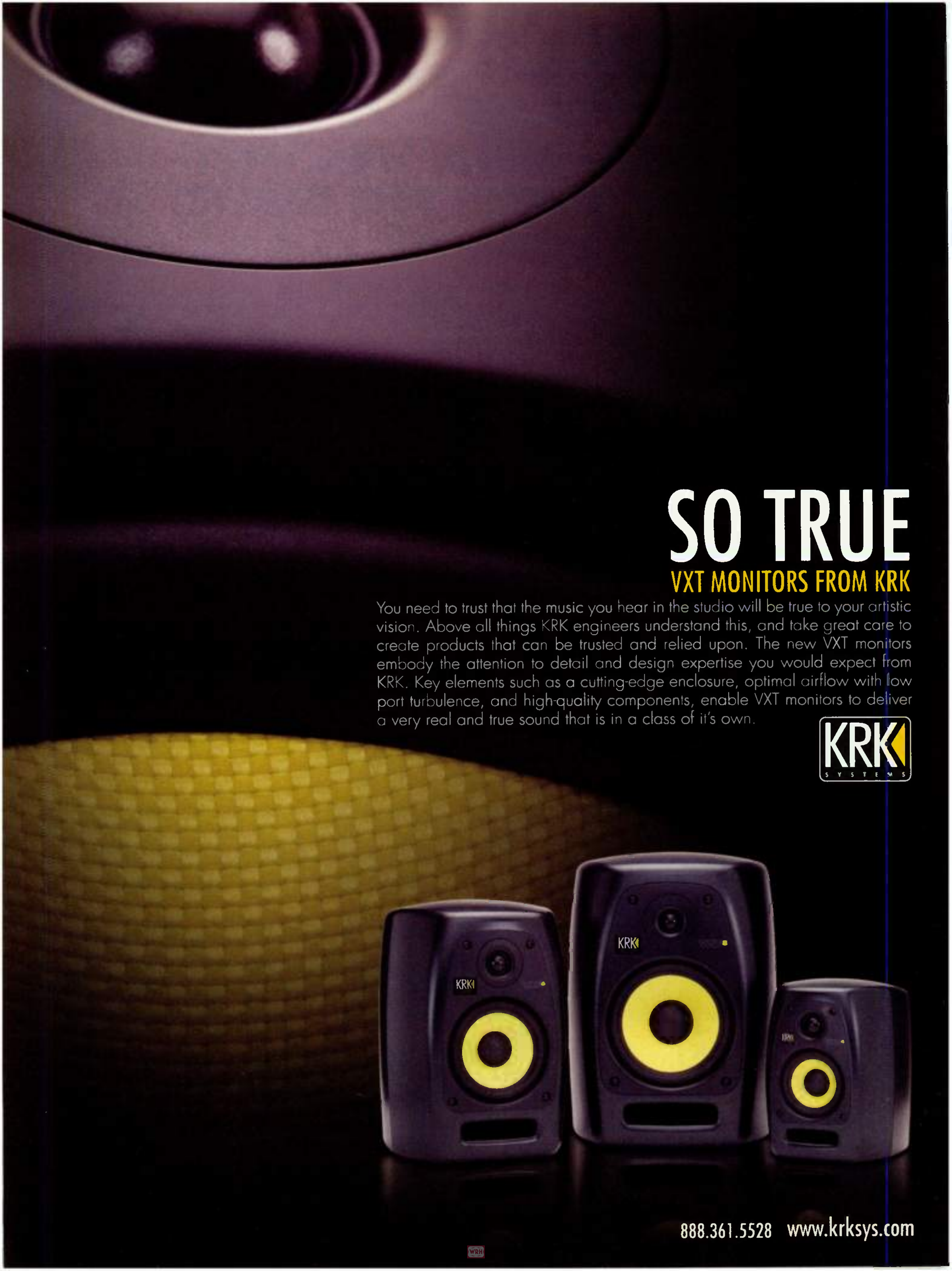
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DIGIDESIGN A.I.R. VELVET VIRTUAL ELECTRIC PIANO

It's the Piano, Man

by Phil O'Keefe

Remember when you first played a Rhodes or Wurlitzer electric piano? I was fascinated by the range of sounds — from bell-like tones when playing soft, to the meaty growl when you played it hard. And that exquisite stereo tremolo! I even loved to play electric pianos "unplugged," and hear them acoustically.

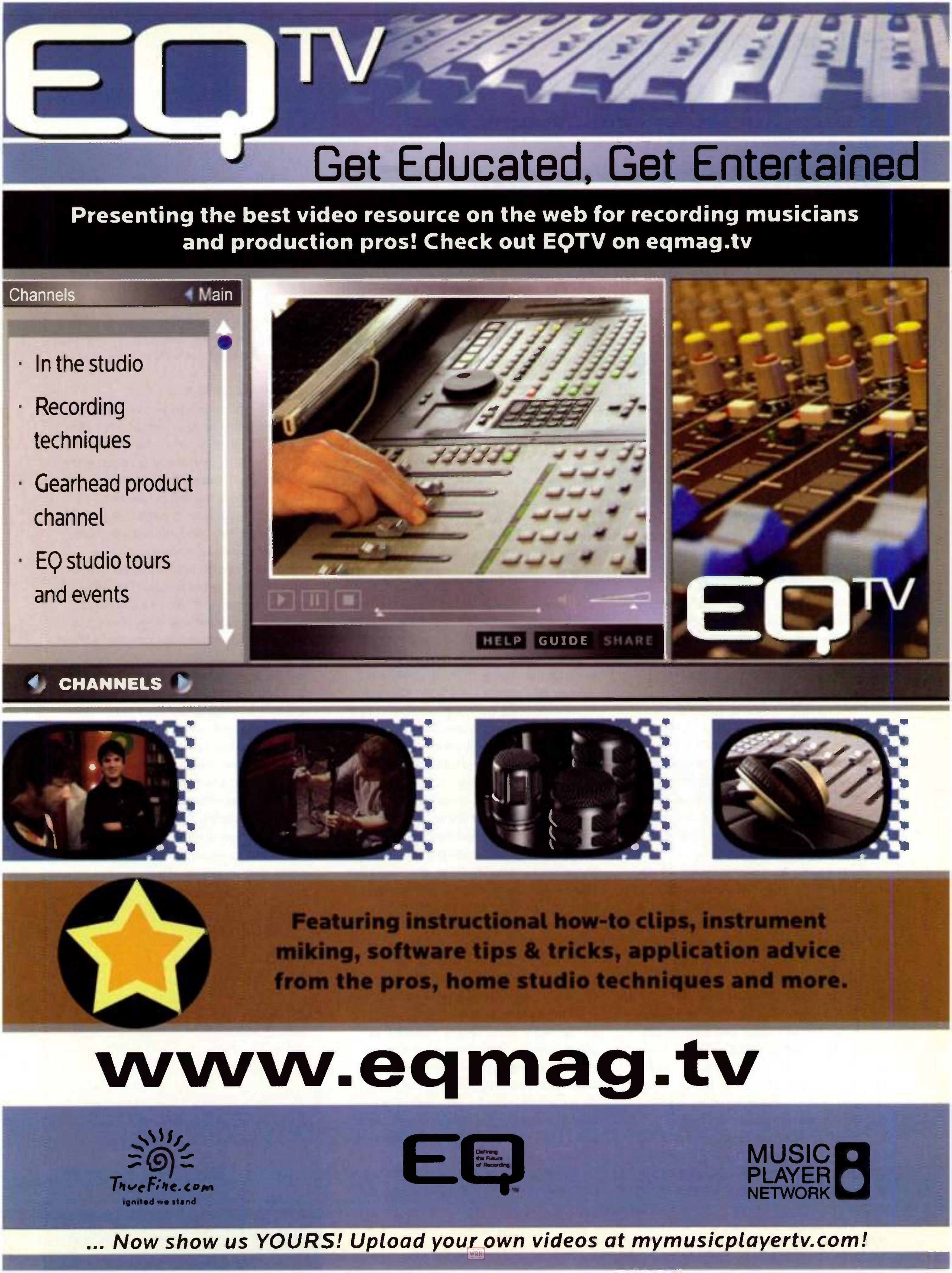
The electric piano remains the same magical instrument as always. I've recorded countless of these lovely monsters over the years, but electric pianos in pristine condition are getting more expensive and rare; besides, they take up a lot of space. And given the tuning and maintenance issues that are part of a complex electro-mechanical instrument, I often turn to virtual instruments and sample playback synths instead — even though the sounds often leave me feeling I'm settling for "second best." So I couldn't help but wonder whether Velvet — Digidesign's RTAS plug-in for Pro Tools — hits or misses the realism mark. Let's find out.

USING VELVET

The brainchild of Digidesign's Advanced Instrument Research group, Velvet is available on CD or via an online Digistore download purchase (323MB Windows XP, 351MB OS X Universal Binary) and once installed, takes about 700MB of hard drive space. The installation is painless, but requires an iLok to handle copy protection.

Velvet includes four vintage electric piano sounds: Rhodes Mark I, II, and Suitcase 73, and a Wurlitzer A200, with user-editable parameters that allow for considerable variations. While I wish that other electric pianos (*e.g.*, Hohner Pianet, RMI, and Yamaha CP70) were also available, in fairness no other virtual instrument I've met offers all these sounds. Besides, the chosen models will likely satisfy the majority of users — especially due to the wide-ranging editability.

Inserting a Velvet plug-in into a Pro Tools session loads the basic samples into your computer's RAM, so changing from one piano type to another requires a few seconds of load time. Of the many presets, Mk I is a classic Rhodes tone with a fat bottom, clear top and great "splat" when played hard; the Mk II generally leans towards a Dyna-Rhodes type tone — thinner and brighter, with not as much bottom and a



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DIGIDESIGN A.I.R. VELVET VIRTUAL ELECTRIC PIANO

SO HOW *DID* THEY DO IT?

As mentioned in the article, we asked Digidesign to comment on how they got Velvet to sound so good. Though Peter Gorges (Director of the Advanced Instrument Research Group) wouldn't "tell all," he was kind enough to offer some insights into Velvet's technology.

"The exact sound generation method is protected know-how, but I can say the 'sound' is based on samples — so the actual notes you hear are sampled and not modeled, as we wanted to make sure Velvet sounded absolutely authentic, and modeling techniques didn't come close enough for our needs.

"As you've noticed, there is no audible velocity switching in Velvet — you can play any note with velocity increasing in increments of one, and won't hear any switching in the XXL version. I can't say what we *are* using, but we're *not* using velocity switching. This is why we can offer a Timbre parameter that changes the entire velocity-to-softness/hardness response, which would be impossible with switching.

"The dynamic modeling is a set of multiple algorithms that take the static input from the sample and alter it by modeling the behavior of the piano depending on your playing, other notes still sounding, the analyzed behavior of the original piano, and of course the current parameter settings. They pretty much create everything you can't capture with samples. To hear what I'm talking about, load the Wurlitzer and set the Key Off parameter to staccato. Now play long and short notes with varying velocity and notice the changes. You'll hear changes in length, pitch variation, and the 'bouncing' of the virtual tines. Those factors make a decisive difference to the user's perception of authenticity and playability, and we spent a lot of time to get them right. In fact the parameter set for just the staccato behavior comprises 60 parameters, all interacting with each other and requiring different settings for each piano."

more defined attack. The Suitcase is similar to the Mk I, but with a touch more midrange and a somewhat jazzy tone. The A200 Wurlitzer is the electric piano that powered so many Supertramp and Ray Charles songs.

The on-screen virtual keyboard simulates velocity by where you click on the keys with your mouse — useful for auditioning sounds over a range of velocities. However, this won't generate MIDI data; you have to play Velvet via a MIDI keyboard. But playability is one of Velvet's strong suits . . . with a good keyboard controller, it *feels* like a vintage electric piano.

As to editing, the preamp section offers a three-band EQ with a fully parametric midrange band, as well as compression and tube drive controls. Key Off controls the note release response and works well for simulating an electric piano's reaction to quick staccato notes, while the Pedal Noise and Condition (subtle tuning and velocity response changes that simulate "age") parameters add to the realism and tonal flexibility. Four sliders adjust Velvet's velocity curve, while two separate velocity response controls — volume and timbre — adjust the volume and timbral variations that are introduced with different playing dynamics.

The MEM switch chooses among three sample size settings: ECO, MID, and XXL. While running Velvet with a Digi 002 on an Athlon 64 4200 dual core with 2GB RAM, CPU usage differs by only a few percentage points when running the XXL instead of ECO version, and RAM consumption changes by only about 69MB. Unless you're really on the edge with RAM, use the XXL version. It's not that the ECO or MID versions sound bad, but it's *impossible* to hear any velocity switching with the XXL version. I asked Digidesign to explain how they coaxed a sample-based instrument to respond so impressively . . . but citing "trade secrets," they weren't telling! Digidesign calls the process "dynamic modeling," so I surmise modeling technology fills in any gaps in the sample set. Regardless of how it's done, the results are excellent, and well worth the modestly increased system resource demands.

Several of the preset sounds take advantage of a fairly extensive set of effects. In addition to mono and stereo tremolo (which can go either before or after the rest of the effects), you'll find fuzz, wah (insertable pre or post fuzz), ring modulator, bit crusher, chorus, flanger, two different phase shifters, amp/speaker cab simulations, and delays (mono, stereo, and tape). Most sound at least decent, while the delays and tremolo are very tasty. They're a useful bonus, but not really the main attraction as you can process Velvet's output with other plug-ins anyway.

Velvet is the first electric piano plug-in I've heard that can simulate an *unplugged* sound. The Mechanics controls simulate the "clunk, ring and thump" you hear when playing an un-amplified electric piano; lid on and off options are available. While not normally captured by an electric piano's pickups, adding a bit of this noise can increase the illusion that you're sitting at a "real" instrument. Mechanical noise and pickup volume are adjustable separately; you can even bypass the mechanics from the effects section and send it directly to the output.

CONCLUSIONS

In addition to no audible velocity switching, there is also no obvious sample pitch shifting or split points — very, *very* impressive. While plenty of synths do a good job at the thinner, David Foster "Dyna-Rhodes" type sound, Velvet is the first plug-in that knocked me out with its heftier electric piano tones. The thickness and weight of the splat and growl when you dig in is remarkably realistic in terms of both dynamics and tone. I've been impressed with some previous A.I.R. virtual instruments, and they've worked well in my mixes, but Velvet is the first software instrument from any company that has brought me back time and again just for the sheer joy of *playing* it.

I was going to try to talk our band's apartment-dwelling keyboard player (who was also impressed by Velvet) into "storing" his vintage Suitcase at my studio, but I no longer feel the need. Velvet gives me "that sound," and many more, with none of the hassles of the real thing. For realism, sound quality, modest system resource use and exceptional playability, Velvet has no equal. It's the new king of virtual electric pianos. EQ

PRODUCT TYPE: RTAS electric piano virtual instrument.
TARGET MARKET: Pro Tools users hungering for realistic, yet highly user-configurable, electric piano sounds.
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LIST PRICE: $249.00
CONTACT: www.digidesign.com

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Vanguard Records		www.vanguardrecords.com	85

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BIG FISH AUDIO ELECTRO-HOUSE

This isn't just about loops, but sampler patches and one-shots with fx, kicks, percussion, guitar chugs, etc. The WAV files are the stars, containing the 816MB of sounds from which the other loops and files are derived — although I wish the WAVs were Acidized. Loops come in three tempos (125, 127, and 130 bpm), and the filenames indicate tempo and key.

But what you want to know is: Will these loops get people to dance until dawn, exude sweat from every pore, and offer gratuitous sexual overtures? Sadly, I didn't get to test the latter — but the first two are a given. I started with the bass loops; there's not a loser in the bunch, from acidey stuff to muscular riffs. The drums are great too, melding a tight electro feel with some of the looseness of house. Loops with synths, guitars, top (percussive), and multitracked loops round out the collection.

Despite a house undercurrent, the sound is tighter, more aggressive, and has a raw, techno/electro sort of edge that drills straight to the underground. And there are no throwaways. Even the fx have some real class, sliding in effortlessly for breakdowns and transitions.

These sounds cross the line from good to inspiring by adding a fresh, creative twist to a genre that sometimes verges on cliché. The example at www.eqmag.com took about 20 minutes to create; yes, the loops play together that well. Highly recommended. —*Craig Anderton*

CONTACT: Big Fish Audio, www.bigfishaudio.com
FORMAT: DVD-ROM with loops (duplicated as WAV/Apple Loops/REX) and sample presets (NN-XT/EXS24/HALion/Kontakt2); 24-bit/44.1kHz; audio CD for auditioning
LIST PRICE: $99.95

SONY FLAMMABLE

This CD provides 15 construction kits, each with an associated demo song (playable with the included Acid Xpress software for Windows); as the demos are royalty-free, you can use them in your own tunes — check out some of the excerpts at www.eqmag.com.

Flammable: Club Joints & Street Anthems enters the land of beats, blunts, and bling, emphasizing short phrases with a big sound. Sometimes menacing, sometimes somber, and always intensely direct, the demo song titles hint at what to expect: "Hands Up," "Razor Edge," "Ruff Side" . . . you get the idea.

Interestingly, though, the production quality is pristine and sophisticated — there's no lo-fi dirt here. This is seriously cool, because you can always screw things up, but you can't clean up something that wasn't clean initially. Far from wimping out the sound, this approach gives *greater* intensity, because there's nothing between you and the tone. The kick isn't that flabby thing that shakes car doors, but a huge sound that claims the low end as its own; and the occasional vocal phrases have the power that good production brings to the party. The only vinyl noise you'll find is a sample of same, in case you want to add it.

The Acidization shows Sony's usual expertise, making it easy to mix and match the loops (most of which live in the 90–100 bpm range). This is a specialized disc, but within its specified genre, delivers with precision and power. —*Craig Anderton*

CONTACT: Sony, www.sonycreativesoftware.com
FORMAT: CD-ROM with 291 Acidized WAV files and 29 one-shot drum hits; 16-bit/44.1kHz
LIST PRICE: $39.95

BIG FISH AUDIO TECHNOCORE

'Cause tonight we're going to party like it's 1989 . . . if you missed the rave scene the first time around, follow the signs to the abandoned warehouse with the monster sound system, look for the Happy Faces, expect your body to vibrate from the four-on-the-floor kick — and keep an eye out for the police.

However, it's not 1989, and the techno here is a shade softer. This isn't Belgian hardcore techno, the disco-tinged Italian variant, or the relentless German strain, but the more flowing version heard in French clubs of that era. The 25 construction kits stick to minor keys, and average around 145 bpm (total range is 120 to 180 bpm); each kit also includes isolated drum loops and various hits, affording a degree of customization.

Quite a few of the loops have reverb, which contributes to the softer vibe but also limits their aggressiveness. I would have preferred drier tracks, but if you're into the type of sound embodied in these loops, the reverb makes the loops more plug-and-play.

Those who want the really hard stuff may be put off by ambience. But if you want techno that could also fit in a soundtrack or video game, provide beats for other musical adventures, or form the basis of a genre that's perhaps more 2009 instead of 1989, *Technocore* lays a suitable foundation. —*Craig Anderton*

CONTACT: Big Fish Audio, www.bigfishaudio.com
FORMAT: DVD-ROM with 990MB of WAV files loops (loops also duplicated in Apple Loops/REX formats) and demo version of Ableton Live 6; 24-bit/44.1kHz
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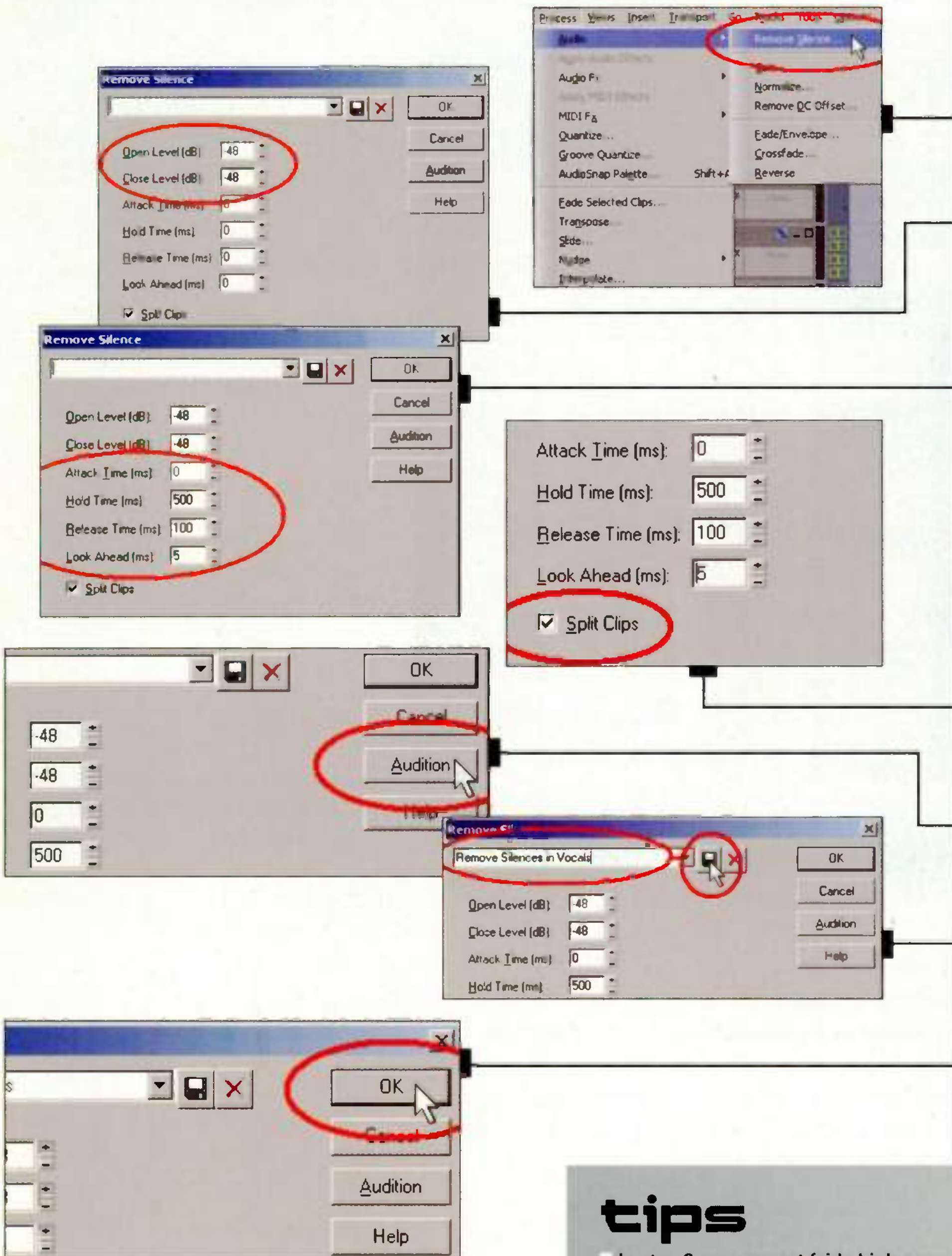
BY CRAIG ANDERTON

CAKEWALK SONAR 6

Use Sonar's DSP to clean up your tracks

OBJECTIVE: Remove sections of audio tracks with low-level noise, such as hiss or hum.

BACKGROUND: Sonar's "Remove Silence" DSP function may seem like "just a noise gate," but it can also reclaim hard drive space because if you define passages as silence, they won't be stored within a track as audio. This function can also physically remove those parts of a long clip containing silence, thus creating a series of smaller, separated clips.



steps

1. Select a Track or Clip, then go Process > Audio > Remove Silence.

2. Signals louder than the Open Level parameter value are not considered silence, while signals lower than the Close Level parameter are. Adjust these parameters so that unwanted low-level signals, or actual silence, fall into the silent range.

3. Attack determines how long it takes for the gate to open after being told to open, while Decay specifies the amount of time for the signal to continue after the gate closes. A non-zero Hold value guarantees that the gate will stay open for a specific amount of time, regardless of the signal level, while the Lookahead parameter opens the gate before the signal reaches the Open Level, according to the time specified by this parameter. Set Lookahead to a fairly low value to avoid cutting off part of a rising transient.

4. To physically remove the silent parts, thus creating multiple clips separated by space instead of silence, check the Split Clips box.

5. Check out the results of your parameter settings by clicking on the Audition button. Sonar will playback and repeat a short section of audio.

6. If you think you'll want to use these settings (or a variation on these settings) in the future, create a preset: Enter a name in the name field, then click on the Save (floppy disk icon) button.

7. When all is set as desired, click on OK to apply the DSP to the selected Track or Clip.

tips

- In step 2, you can set fairly high open and close levels to create effects like gated drum decays.
- In step 3, Hold is very handy if the gate "chatters" when a signal crosses back and forth over the open/close threshold.
- When stripping lots of small sections, be conservative with your settings (low Open and Close values, a couple hundred milliseconds of hold time, no attack, a dozen milliseconds of lookahead time, and about 250 ms of decay) to make sure you don't remove anything you want to keep.
- If you remove silence in an entire track, listen to the track to make sure all is well before it's too late to hit undo.

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POWER APP ALLEY

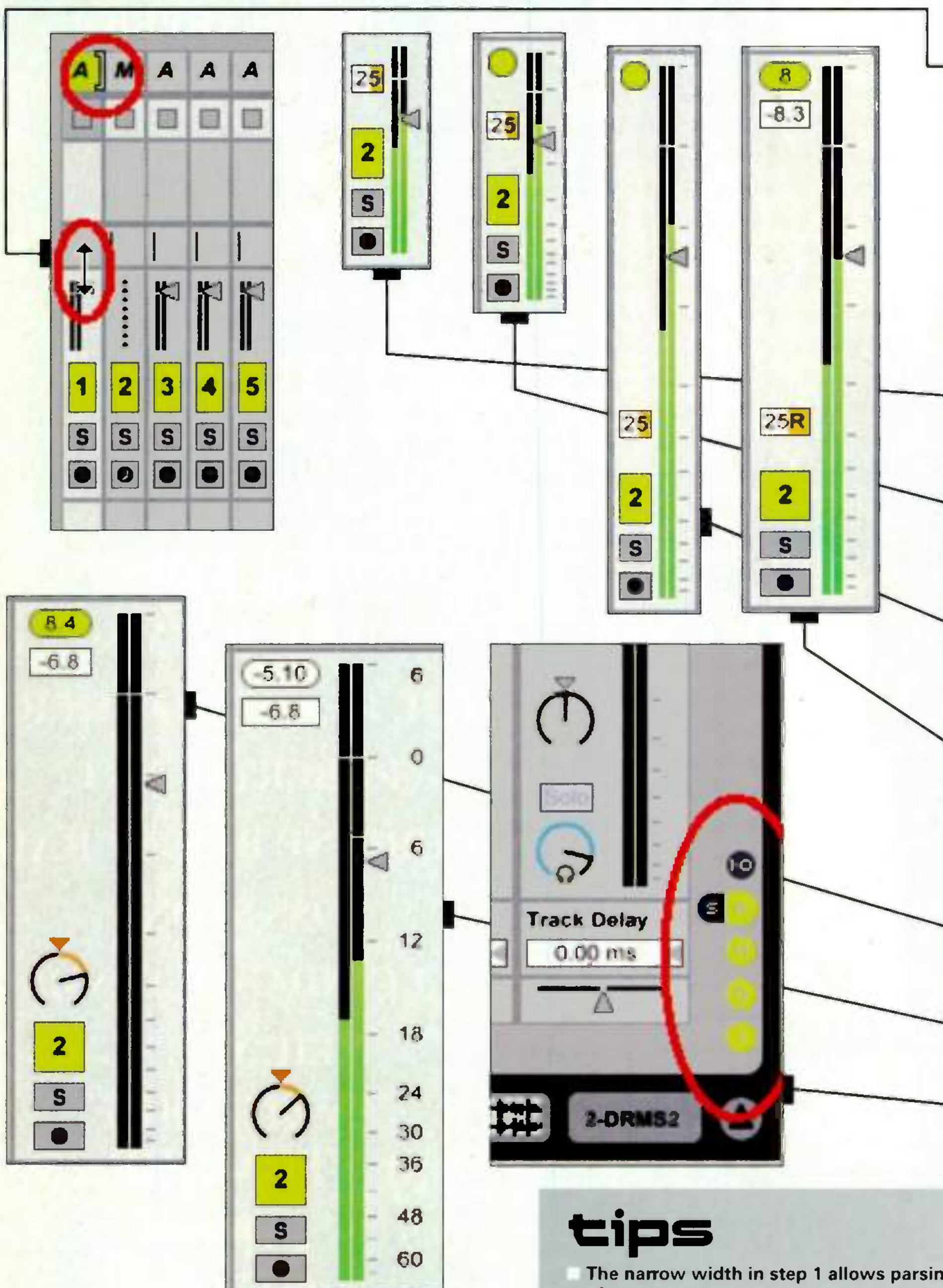
BY CRAIG ANDERTON

ABLETON LIVE 6

Have it your way with Live's mixer

OBJECTIVE: Configure Live 6's Session View mixer to your specific needs.

BACKGROUND: In version 6, Ableton made the Session View mixer far more configurable, as well as capable of showing more information about a channel. As this configurability can take up more screen space if used to the fullest, Live includes several customization options.



steps

1. To change channel width, click on the line to the right of the track name; the cursor turns into a bracket. Drag left to decrease width, right to increase. To change channel height, click on the splitter line above an audio channel's fader (or a MIDI channel's meter). The cursor turns into a double arrow. Drag up to raise the height, drag down to lower. When set to its narrowest width and height (as shown), all you see is a short fader and three buttons (channel active/inactive, solo, and record).

2. Making the channel strip one step "wider" reveals the pan control (the numeric field to the fader's left).

3. Widening the channel strip further shows a channel peak indicator, which turns chartreuse to indicate when an overload has occurred. Click on the overload indicator to reset it.

4. Dragging the channel to its maximum height extends the height of the fader and meter — good for precise level adjustments.

5. Increasing a channel's width after increasing its height shows a numeric value in the peak indicator. This indicates how much the track's highest peak is above, or below, 0. Also, a rectangular field opens up to show a numeric value for the fader setting.

6. Widening the channel even more turns the pan field into a knob; rotating the knob displays its numeric value in the lower left part of the status bar.

7. Extending the width further shows meter/fader calibrations.

8. Use the buttons toward the mixer's lower right to show/hide various sections (sends, delay, crossfader, etc.).

tips

- The narrow width in step 1 allows parsing level/pan settings quickly, and lets you fit lots of tracks on the screen.
- Changing width also affects the send controls. With a strip at its narrowest, the send controls aren't visible. Increasing the width shows the sends as fields with numericals, and increasing further turns the fields into controls.
- When the overload indicator circle is white, the signal has not gone above 0 for that channel since playback began.

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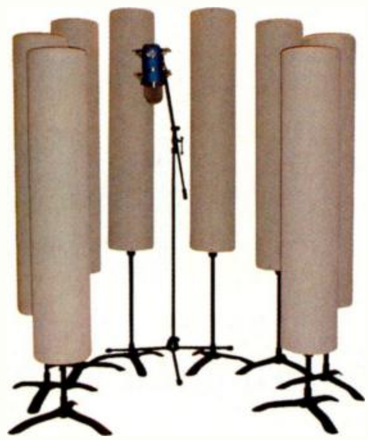

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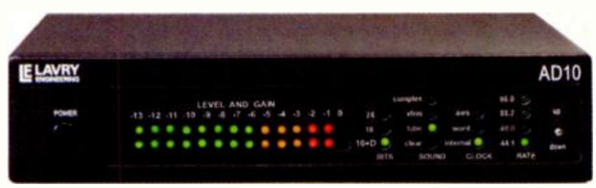

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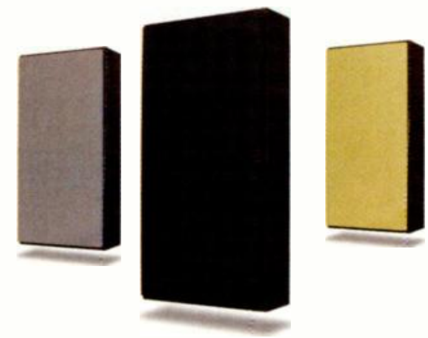

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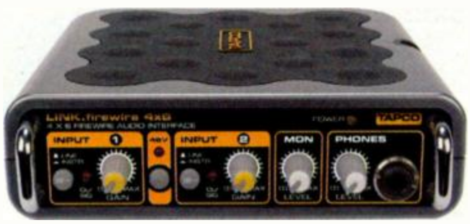

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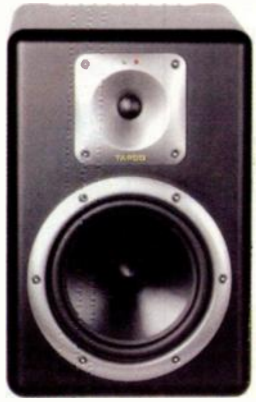

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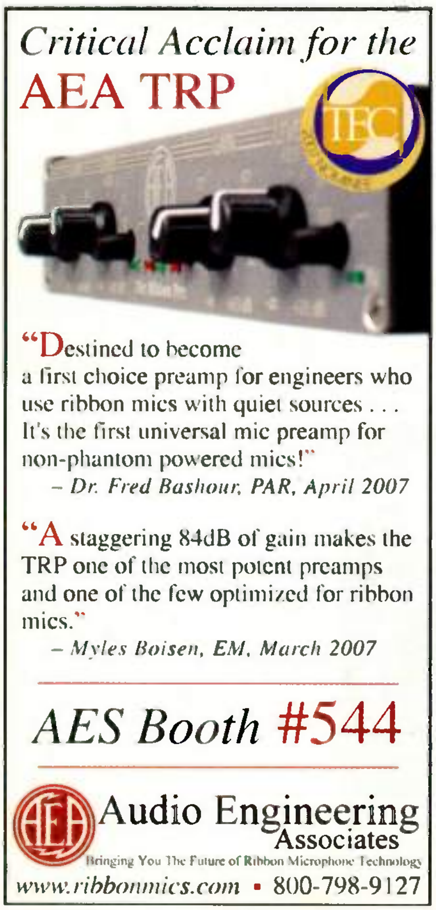

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by Sal Russo

STUDIO NAME: The Pleasure Machine
LOCATION: Brooklyn, NY
CONTACT: chrisink@excite.com
KEY CREW: Christopher Walsh
CONSOLES: Digidesign 002; Soundcraft Spirit E6
COMPUTERS: Apple iBook G4, iMac G4, Intel Core 2 Duo
RECORDING HARDWARE: Alesis MasterLink; Digidesign Mbox
RECORDING SOFTWARE: Digidesign Pro Tools 7.3 LE
STORAGE: Glyph Companion; LaCie d2 Quadra; Maxtor One Touch
SOUND TREATMENT: Auralex Acoustics AudioTile Shockwave, Q'Fusor Diffusors
MONITORS: Hafler M5 Reference; Roland DS-7 Bi-Amp Monitors
AMPS: Hafler TA 1100
PRES: SSL XLogic Alpha Channel; Studio Projects VTB-1
MICS: AKG D2000S (2); Audio-Technica 3060, Sennheiser e835; Shure SM57 (4), SM58 (4); Studio Projects C1
OUTBOARD: Alesis NanoCompressor, NanoVerb; Aphex Aural Exciter; Furman RR-15Plus Rack Rider
INSTRUMENTS: Epiphone John Lennon Casino; Fender GDC-200, Jazz Bass, Rhodes, Stratocaster, Telecaster; Gibson SG, Songwriter Deluxe; Tama Starclassic Maple drums
AMPLIFICATION: Fender Acoustasonic 30; Hartke HA 3500; Reverend Hellhound 40/60; Vox AC15
NOTES: The transition from "home studio" to simply "studio" is one of the most liberating experiences a musician/recordist can have. In the Pleasure Machine, four musicians comprising numerous bands and projects have finally taken that plunge, claiming a room in the Northside Music Complex, in the musician-heavy Williamsburg section of Brooklyn — an area that's home to the likes of the Yeah, Yeah, Yeahs, They Might be Giants, and many, many others.

It's true: With such a live, vibrant and ever-changing music scene, Williamsburg was ripe and ready for a studio to record the sudden influx of talent now foregoing the trip across the East River and staying put in The Burg. When asked about the ethos he abided by when he and his partners-in-crime opened the part rehearsal space/part recording studio affectionately dubbed The Pleasure Machine, guitarist/producer Christopher Walsh said, "We are from Brooklyn, we play in Brooklyn, we're damn sure going to record in Brooklyn."

Given the need for the room to have multiple uses as not only a rehearsal space but a room to record tracks, the partnership searched for a sound treatment solution that would optimize the 234 square feet of rectangular space. In particular, mid and high frequencies endlessly bouncing off bare drywall were troublesome, producing an indecipherable, constant wall of noise — hardly what's needed when you're trying to write, let alone record.

Upon contacting Auralex Acoustics with the room's dimensions and receiving an aptly dubbed "Personalized Room Analysis," the guys armed their walls with broadband absorbers arranged in an array (check out the before and after photos above). These were positioned along the rear wall to break up the flat wall reflections and scatter mid-high frequency energy evenly throughout the room, giving a more open and natural sound.

After rolling in the gear (a limited, but effective, group of equipment), all was good to go. From demos to indie recordings — such as Kieran McGee's "Narrow Mind" — the Pleasure Machine has become a one-stop shop for local artists who need a place to work out their songs, and achieve a respectable recording before moving on to the big time.

Pretty amazing how a few choice pieces of gear and a well-treated room can turn a space the size of a small studio apartment into a functional practice space/studio, eh? EQ

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