THE RECORDING MUSICIAN'S GUIDE TO VIRTUAL INSTRUMENTS

EQ

www.eqmag.com

SERJ TANKIAN

THE SYSTEM OF A DOWN FRONTMAN REVEALS HIS HOME STUDIO SECRETS

Exclusive!
FIRST LOOK AT SONAR 7
plus: ARTURIA JUPITER-8V, BLUE WOODPECKER, KRK VXT8, MACKIE 1604-VLZ3

RICHARD DASHUT ON MAKING FLEETWOOD MAC'S *RUMOURS*

RICHIE KOTZEN'S SIMPLE METHOD FOR KILLER GUITAR TONES

6 STEPS TO A SLAMMIN' SNARE

STANLEY CLARKE ON RECORDING HIS ALEMBIC

3 WAYS TO BIGGER BASS SOUNDS

TAMING VOCAL DYNAMICS

A MIC SO SENSITIVE, IT EVEN PICKS UP COMPLIMENTS.

"The KSM32 is becoming my mic of choice for many applications including vocals, overhead drums and bass cabinet. The accessories and hardware are brilliantly executed."

— Eddie Kramer

"In the price/performance ratings, I would give the KSM32 a solid 10. For what you were planning on spending for a single microphone you can have a pair of KSM32s."

— Roger Nichols

"In my 35-year recording career, I've never come across a mic that works so well on so many instruments. If you are looking for a mic that is accurate, uncolored, pristine, dead quiet and doesn't cost much, you may want to own several KSM32s. I just ordered a dozen."

— Tom Jung

Experiencing the performance of our KSM32 cardioid condenser microphone is, quite frankly, beyond description. But to be fair, we've let some discerning professionals try anyway.

The Shure KSM32. It will capture more than your voice.
For more information, visit www.shure.com.

www.shure.com

SHURE®
LEGENDARY PERFORMANCE™

I can

:: record
:: arrange
:: edit
:: process
:: mix

Be heard.

Version 2.0

NOW FEATURING!

steinberg | CUBASE LE4

WINDOWS VISTA™ COMPATIBLE

lexiconpro.com

Lexicon's Desktop Recording Studios help thousands of musicians record and produce their own music. Now we've upgraded these products with Version 2.0 software featuring Steinberg® Cubase LE 4™ and Windows® Vista™ compatibility.

Cubase LE 4 offers many of the same features as Steinberg's award-winning Cubase 4, such as integrated virtual instruments and all-new VST3 plug-ins. Combined with Lexicon's own Pantheon™ reverb plug-in, this powerful new workstation makes it easy to turn your demo samples into polished attention-getting productions.

Visit Lexiconpro.com to find out how you can BE HEARD.

Version 2.0 ALPHA LAMBDA OMEGA

Lexicon

A Harman International Company

PUNCH IN

FEATURES

NEW! TECHNIQUES

74

78

NOVEMBER 07

CONTENTS

Cover/this page photo: Greg Watermann

REVIEWS

POWER APP ALLEY

DEPARTMENTS

Talk Box

DO YOU VIDEO?

You're all familiar with the recording revolution, or you wouldn't be reading *EQ*. You'd still be dreaming of the day your band would be big enough to rent time at a studio with a 24-track MCI 2" tape machine. (Yes, I know some of you would rather use a 24-track tape recorder than a computer, but you get the point.)

Even a $100 "lite" version of a sequencer, coupled with a $1,000 computer, runs rings around something like an ADAT-based 8-track studio—which was such a breakthrough only a little over a decade ago.

But if you've been wrapped up in recording, you might have missed the parallel video revolution. Video lets you present your music in a new and creative context, as well as engage your listeners more fully. Sites like YouTube have provided the venue for those seeking a competitive edge to make the jump from audio-only media to a true multimedia package.

I became involved in video in the '80s, when you needed a bulky (and expensive) camera to record tapes you loaded into at least two (expensive) video tape decks, feeding into an (expensive) switcher so you could transfer the good bits from one deck to another—with analog signal degradation thrown in for free. Later on, when computers came on the scene, I remember using Mac IIci machines that took *overnight* to render a single frame of animation. Yes, those were the days . . . *not!*

Today, the same computer that's powerful enough for your music is powerful enough for video editing. Thanks to inexpensive hard drives, the formerly daunting storage requirements of video are no longer an issue. What's more, a quality hand-held camcorder costs well under $1,000—less than four-track tape recorders when they first appeared—and serious video editing software can cost as little as $100 or less.

It's not hard to edit videos: You already know how to cut, copy, and paste. And, video processing is very much like audio; we even use the same terms (well, you make audio tracks "brighter," right?).

It may seem incongruent to talk about video in a recording magazine, but it's the next step in presenting your music to the world—and audio is a crucial, essential part of the audio-for-video equation. What are you waiting for?

EQ

Vol. 18 No. 11
November 2007

www.eqmag.com

Please direct all advertising and editorial inquiries to:
EQ, 1111 Bayhill Dr., Ste. 125, San Bruno, CA 94066
(650) 238-0300, Fax (650) 238-0262, eq@musicplayer.com

Please direct all subscription orders, inquiries, and address changes to:
800-289-9919, outside the U.S. 978-667-0364, eqmag@computerfulfillment.com

Back Issues: Back Issues are available for $10 each by emailing backissuesEQMag@nbmedia.com

Periodicals postage paid at San Bruno, CA, and at additional mailing offices.

Faith is One Thing
Control is Another

studiokonnekt 48

Studio Control Audio Interface

Imagine there was a professional audio interface...
that put all essential recording tools in one box,
that had extensive digital and analog I/O,
that easily managed your speaker setups,
that was equipped with TC DSP effects,
that offered a 24/8 channel digital mixer,
and included a remote to control it all.

...well, you'd want one.

included software

Fabric R Studio | *Fabric C Studio* | *Integrator* | *ResFilter* | *Assimilator Konnekt* | *24/8 Channel Digital Mixer* | *Speaker Management* | *Konnekt Tuner*

t.c. electronic®
ULTIMATE SOUND MACHINES

WWW.TCELECTRONIC.COM

SOUNDING BOARD

WORSHIP THY MASTER . . .

The Rush *Snakes and Arrows* [9/07] article was very informative. I was really interested in the way it was done in the studio with Nick [Raskulinecz]. He's a master. I wanted to add to the end of the "Mixing, Mastering and Digging the Results" section. Rich Chycki made a comment that ". . . we did take nuclear weapons to make sure they didn't crush it to death at the mastering facility" but the article made no mention of the actual mastering engineer!

Mastering is the last creative process in making an album. The final touches are not done at the mixdown. An album doesn't go straight from the mixing session to duplication. Now I know Nick and Brian Gardner [mastering engineer for *Snakes and Arrows*] are great fans of one another, and I'm sure he didn't leave it out intentionally to mask Brian's talent . . . it was only that *EQ* wanted to focus on the recording and mixing, which is cool. Still, I see "mastering" and I get interested—mastering is in my blood and I *love* Rush. I'm honored that they finally came to Bernie Grundman Mastering and that they got results they'd never had on any of their other albums.

Brian Gardner *a.k.a.* "Big Bass Brian" at times of the year will have mastered up to 40% of the Top 10 charting albums. Even though it has been since "Tom Sawyer" that Rush had a Top 10 hit, they're always in my Top 10. Great album and great engineers all around! I'm just dropping you a letter to do what mastering engineers do: add the topping to the bottom line.

Paul Grundman (via email)

. . . OR MAYBE NOT!

I read the Rush *Snakes and Arrows* [9/07] article with mounting excitement. As a long-time Rush fan, and with seeing quotes from producer Nick Raskulinecz such as "I challenged them to write the most screwed up, complicated instrumental that they had ever written," how could I not be excited?

So, I "rushed" out to buy the new CD. Unfortunately, I never got to the aforementioned instrumental. In fact, I could only listen for a few seconds before I began to question my hearing. So I quickly put on *Moving Pictures* and was met with blessed relief—my hearing was fine.

Analysis on my PC confirmed my suspicions: This record had been ruined. Presumably this was done in mastering, but who knows? The album is another damn square wave all the way through! I took the CD back to the store, wondering if the 180-gram vinyl version or the SACD might be any better, but then I decided that I wouldn't be finding that out with *my* hard earned bucks.

To Nick and Rush: You guys should get your money back from Brian "Big Bass" Gardener if this is his doing. Of course, for all I know, you delivered it to him in the same sorry state—but if you all insisted he "make it louder," you should pay *him* a penalty for damaging his excellent reputation. It's not his fault he couldn't refuse working on a new Rush record—who could?

Man am I upset. I thought you guys knew better!

One more thing: Having all those mics on the drum kit may have provided some ambience and attack, but the sound is thin. You can use a tape measure on the snare all day long to keep it in phase, but what about the phase relationships of the toms? Sheesh. It takes *some* doing to make Neil Peart's drums sound that thin.

To *EQ*: I really enjoyed the article, though. Your new long-form interviews, which I first noticed with the Pete Townshend interview [8/07], are very welcome . . . and something I haven't seen since *Studio Sound* magazine went defunct.

Keep it up,

Jack Huggins (via the forums)

LOOKING GOOD

I was shocked when I opened the September issue. Talk about an overhaul! The mag looks great! And the instrument sections y'all added are really cool. Please tell me that these are going to be in every issue.

Sincerely,

Marcus (via email)

Matt Harper responds:

Our new Techniques section? You would do well to get used to it, and get used to it fast, sir.

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.

Note: Letters may be edited for length and/or clarity. Direct correspondence by EQ editorial is not guaranteed. All submissions become the property of EQ magazine and can be published in any medium.

Get Real

SONAR™ 7

- SONAR is made for musicians by musicians
- SONAR offers the best audio quality in the industry: 64-bit end to end
- SONAR places no artificial limits on your production: unlimited tracks, busses, & inserts
- SONAR has better music creation features, our specialty for over 20 years
- SONAR offers superior instrument integration & control
- SONAR is backed up by Cakewalk's unparalleled customer service

Find out more at www.cakewalk.com/sonar

PUNCH IN

by Lily Moayeri

TRACKING ROONEY'S *CALLING THE WORLD*

Rooney guitarist Taylor Locke's head spins right around every time a female walks past his outdoor table in the busy Hollywood eatery he is having an afternoon drink in, but it snaps right back to center—as does his attention—when the matter at hand becomes the recording of Rooney's second album, *Calling The World*. Although Rooney's self-titled debut—a throwback to the poppier side of '60s and '70s rock—was simple and infectious, the follow-up was recorded, in its entirety, multiple times.

The first run at *Calling The World* was with producer Tony Hoffer at Sunset Sound. Rooney's idea was to make the record like it was 1970—recording in a room all together, live to tape.

"We were more excited by *how* we were making the record than the actual record we were making," says Locke. "I don't dislike the way it sounds. It sounds retro, old, lo-fi, and grainy. It has its charms, but it just wasn't a competitive-sounding record."

Starting over, this time with the god of modern-rock producers, Howard Benson, the meticulous approach of the second *Calling The World* sessions was the exact opposite of the previous experience.

"Benson has this very elaborate guitar-layering strategy where all the guitar parts are broken down into different sounds," explains Locke.

VINTAGE VIBES/

"You play a part for only four bars. Then, you retune and double the part. You play the next four bars, and then retune and double that. When you get to the end of the song, you dial in a new guitar sound, and do the same thing—again and again—until there are anywhere from six to 20 perfectly-tuned guitar layers per song."

But this very mechanical approach ultimately went against the playful and carefree nature of Rooney. So, next up to the production podium was John Fields (Switchfoot, Soul Asylum, Mandy Moore), who approached the band as a fan, and offered to track a song for free. Documenting ideas quickly in GarageBand—starting only with a programmed MIDI drum track, and then adding guitar, bass, and synth lines—the band moved the demo tracks from Locke's and Rooney vocalist/guitarist Robert Schwartzman's makeshift bedroom studios to Fields' Pro Tools-equipped home facility to cut vocals. The vocal sessions were so successful that, although the instruments were deemed scratch tracks, the Rooney guys decided to use the vocal tracks as "finals." To do this, click tracks had to be laid down to match the vocal performances so that the rest of the instruments could be recorded. Ultimately, the band shot over to the late Jeff Porcaro's Seedy Underbelly studios to track drums, and then to the Dust Brother's studio—The Boat—to cut the guitars, bass, and keys.

It was at The Boat where Rooney discovered the signature of *Calling The World*—Locke's and Schwartzman's killer '70s-styled guitar tones. Locke, wielding a veritable arsenal of Gibsons (a Les Paul Special, Junior, and Standard; a vintage SG; and an ES-330 and ES-335) ran simultaneously through a Vox AC30 and a plexi Marshall half-stack. Schwartzman—a strict Fender proponent—ran his vintage Jazzmaster and a Telecaster through a '57 tweed Twin and a '52 Deluxe. Both sets of amps were miked with a Shure SM57 right on the grille, and a Royer R-121 ribbon mic placed approximately 12 inches back, and pointed directly at the cone.

"When you get to the end of the song, you dial in a new guitar sound, and do the same thing—again and again—until there are anywhere from six to 20 perfectly-tuned guitar layers per song."

"Fields pulled up the two faders, and we immediately noticed the ribbon was very warm and thick, while the SM57 had a bright midrange that was very in-your-face," says Locke. "When we submixed those two tracks down to one, we ended up using about an 80/20 split in favor of the SM57s. We didn't want too much low end—just enough to round out the tone, and produce that warm '70s guitar sound."

Moving out of The Boat and into their respective home studios, all the members of Rooney were still able to communicate with Fields for the duration of the sessions—regardless of whether they were physically in the same place or not. The band would record last-minute additions to their compositions on GarageBand, and email the tracks to Fields, who would print a new mix, and send it back to the group for approval.

The vocal sessions were so successful that, although the instruments were deemed scratch tracks, the Rooney guys decided to use the vocal tracks as "finals."

"Fields is attached to iChat nearly 24/7," says Locke. "He lives in a computer. We cut the new tracks in three weeks, lived with his mixes for another couple of weeks, made our notes, and then we booked two days at Encore's Studio A in Burbank—the big SSL J 9000 room—for the final mixes. Well, we were basically using the SSL for monitoring from Pro Tools, because Fields output a stereo mix from his Power Mac G5. We mixed the whole record in two days. The entire process was a much better way to work than how we were doing things before. Fields really showed us there's a very savvy, modern, and efficient way of making an album, and making it sound *good*." EQ

"We didn't want too much low end—just enough to round out the tone, and produce that warm '70s guitar sound."

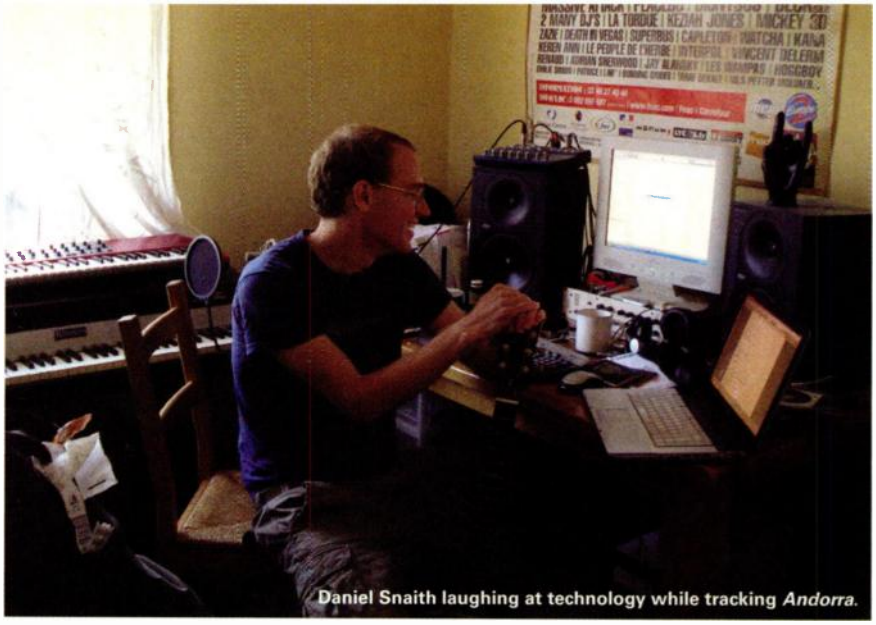
MODERN METHODS

Rooney (left to right): drummer Ned Brower, vocalist/guitarist Robert Schwartzman, bassist Matthew Winter, keyboardist Louie Stephens, and guitarist Taylor Locke.

True Grit

Caribou's Dan Snaith and the Artfulness of Muck

by John Payne



Daniel Snaith laughing at technology while tracking *Andorra*.

Electro-music superpower Caribou's new album, *Andorra*, is mastermind Dan Snaith's latest salvo in a series of works that start somewhere along chilled ambient (and even chillier IDM), and stroll down a path cobbled with pure beat, dense texture, and the tricky tonal turns of '60s pop and psychedelia. Jaw-droppingly sumptuous, *Andorra* brings together Snaith's love of gritty electronic timbres and heavy, visceral beats with intricate matrices of vocal harmonies and rock instrumentation. It's complex, but what's to be expected from a musician who, in his spare time, has managed to obtain a PhD in mathematics?

What is a bit shocking is that the album was recorded entirely in his bedroom with a motley and minimal setup that hasn't been upgraded much since he became Dr. Snaith.

"It's okay," says the Canada-born, London-based musician. "I don't have a very clean recording aesthetic anyway. I like things to sound a bit sloppy. And when I pile everything up, I like the instruments to all mash together, rather than sound distinct from each other. That actually *helps* my sound. It doesn't detract from it."

Even applying compression "properly" is a side issue for Snaith.

"There are some tracks on *Andorra* where compression is a large part of the sound of the track," he says, "Some people say that when compression is used right, the listener shouldn't be able to discern it. I beg to differ. On 'Irene,' it's the sound of the drum machine and a Rhodes being really compressed together, and then put slightly out of phase with each other that produces the really whoozy, pulsing kind of sound that I love. I never really read too much about recording, and maybe that's why I do this. I just don't know much about recording in the traditional sense, and any gear I've had I've always just messed around with, and said, 'Oh, I like the way that sounds.'"

Snaith, a pianist since the age of five, started recording in his bedroom as a teenager.

"I got a drum set, a computer, and a MIDI keyboard," he says. "That was a real eye-opener for me, because I realized I didn't need to drive into town, get all my friends together, have a band practice, and then record a 4-track demo. I just spent all my time writing and sequencing music on a cheap keyboard and a computer, and learning how to arrange things."

Inspired by jazz pianists and prog-rock keyboardists such as Rick Wakeman and Keith Emerson, Snaith began composing music that showcased his technical wizardry. But the world of keyboard virtuosity was effectively disrupted when he was introduced to Britain's ambient techno scene in the early '90s.

"I was struck by the fact that it was all made in a not-too-distant fashion from the simple setup I had," he says. "And the music was not about technique. It was about programming things, rather than playing them, and it was a real challenge for me to wrap my head around that."

Although music software and computers constantly improve, Snaith's recording setup has remained the same. He tracks on an outdated version of Sound Forge on a cobbled-together computer, and incorporates samples off a vast collection of old records that he rips straight onto his hard drive.

"I still use Acid 4.0," he says. "It's loop-based, and it also works perfectly well with a multitrack sequencer. It doesn't matter much, because I don't do any effects processing inside the computer—everything happens *before* it goes in there. For most any instrument I'm recording, I use an Oktava MK012. It's a really bright mic, and it naturally adds a bit of the '60s and '70s quality that's on a few of the tracks on the new album."

While nearly every drum part on *Andorra* is a looped sample from his extensive vinyl collection, Snaith did whack the tubs for a few tracks, recording a single floor tom by positioning the MK012 above the top skin, and a Shure SM58 under the bottom head. Most of the other "natural" sounds on *Andorra* come courtesy of a Tanglewood Violin bass ("a knock-off of Paul McCartney's famous Hofner") and a Gibson SG, with a Boss SP-303 Dr. Sample acting as a preamp for both instruments.

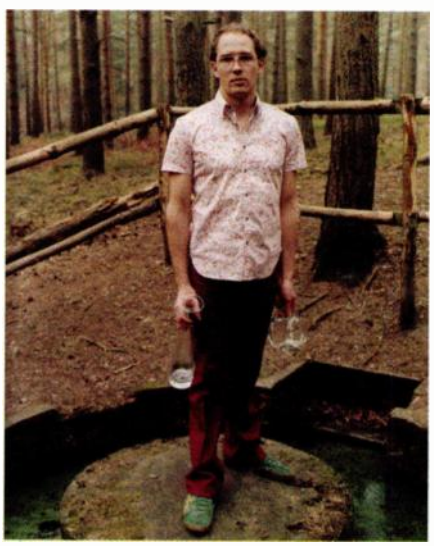
Still, the trademark Caribou sound is that of intriguingly blended and distorted keyboard sources. Never straying too far from his trusty old Fender Rhodes, Snaith nonetheless embraced the sounds of a group of beloved, antique synths, such as the Jen SX1000 and the Yamaha PSF 480.

"The Jen was the first synth sold in department stores in England," he says. "It's a mass-produced kind of synth with just one mono voice, and a few knobs you can twiddle around to get different sounds. The Yamaha was my first keyboard thing. It boasts a wicked bunch of crappy bossa nova and cha-cha beats, and it allows the user to change the envelope and basic parameters of the sound."

In another example of Snaith's somewhat casual approach to gear, *Andorra* became the first Caribou album made with the aid of working studio monitors.

"On the last album, one of the speakers had a torn tweeter," Snaith chuckles. "This time, I set up a decent monitoring system—a pair of Mackie HR 824s—which, amazingly, helped gauge the final sound with far more precision and speed [*laughs*]."

Although he's not immune from a random case of GAS (Gear Acquisition Syndrome), Snaith emphasizes that each and every purchase is made only to facilitate his creative vision—which isn't very dependant on the newest or greatest technology. In his mind, it's the vision and the technique that ultimately matters, not the tools. EQ


Daniel Snaith a.k.a. Caribou.

Sweetwater's shipping team works late into the evening to get your gear to you fast.

No sales tax collected on orders placed outside of Indiana

The Gear You Want, Brands You Trust

In Stock and Ready to Ship!

A lot of retailers talk about having the gear you're looking for, but do you get it when you really need it? Our amazing new state-of-the-art distribution center is stocked full of the brands you want, ready to go out our door and straight to yours.

- **Located in central U.S. for even faster delivery service**
- **Most orders ship the same day**
- **We use FedEx for dependable, quick delivery and safe handling**

FREE SHIPPING* NO MINIMUM PURCHASE

*ON MOST ORDERS. SEE WEBSITE FOR DETAILS.

Sweetwater

Music Instruments & Pro Audio

sweetwater.com • 1-800-222-4700

The Call of the

Jason Holstrom Goes Organic for his Hawaiian-themed Thieves of Kailua

by Shane Mehling

Guitarist, producer, and United State of Electronica contributor Jason Holstrom was mesmerized enough by a trip to Hawaii a few years back that the island's mystique inspired his solo debut, Thieves of Kailua. In fact, the island vibe permeated his Seattle bedroom studio so aggressively, that Holstrom completely dropped all reliance on electronic instruments, samples, and loops. As a result, the album is a seductive mix of naturalistic elements, from vocal harmonies, slide and classical guitar, horn parts, and even a ukulele. As far from the propulsive chatter of U.S.E.'s brand of electronica as a steel factory is from a babbling brook, *Kailua*'s beautiful and organic soundscapes evoke all the comfort and bliss of a tropical paradise.

You handled all the instrumentation on the new album. Can you walk me through the recording process?

The palette of sounds on the album was decided by what was available to me in the form of physical instruments at the time. I didn't want to use any samples or virtual instruments. I wanted the album to be totally organic. So I stuck with what was in my room: ukuleles, nylon-string guitars, bass, drums, sax, and my voice. Each track began with either ukulele or nylon-string guitar. I would essentially track some rough ukulele chords through an MBox into Pro Tools LE 5.2 until I got these huge, chiming loops, and then I'd lay guitar, bass, sax, percussion, and, finally, vocals as I developed the uke pieces into proper songs.

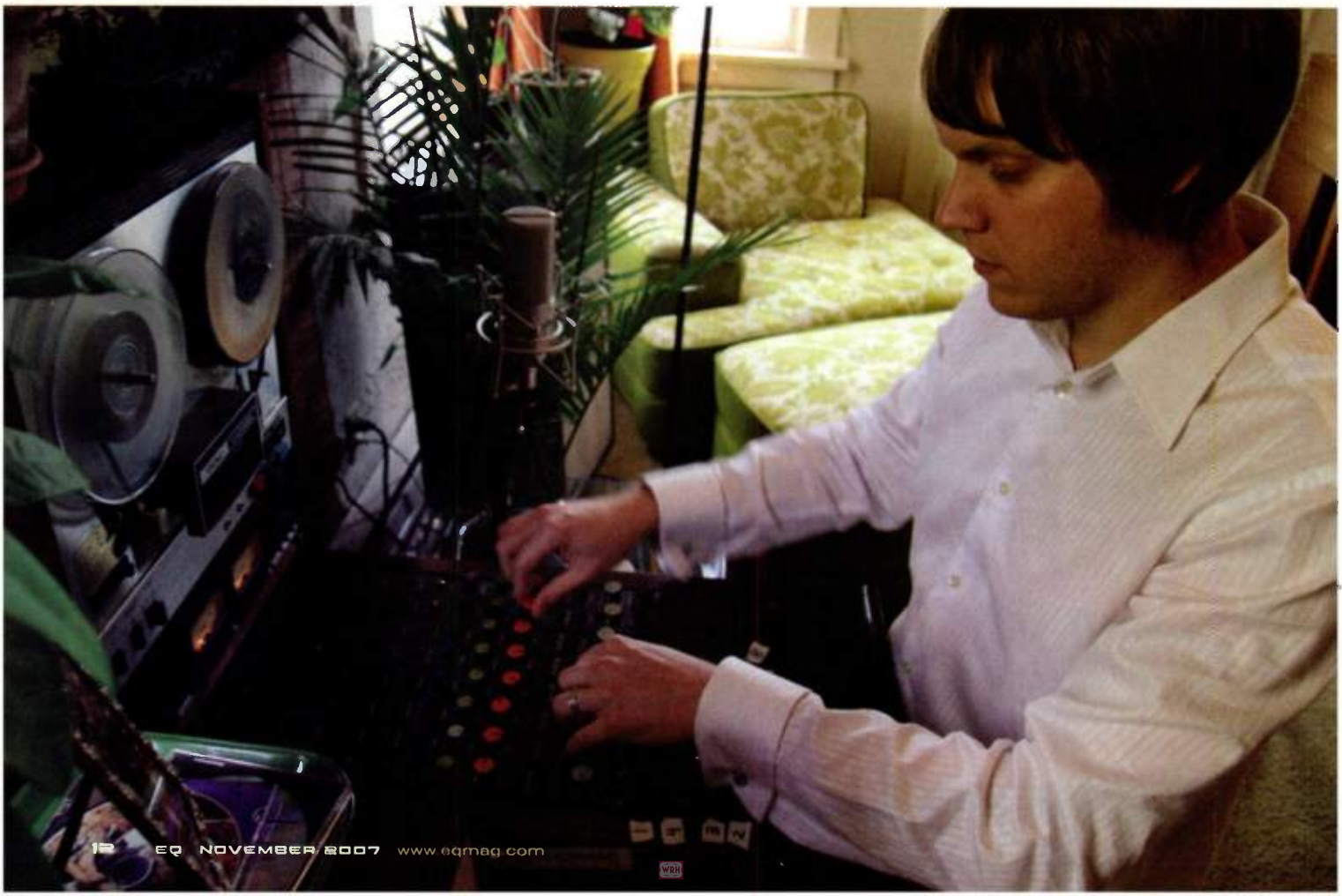
How did you record the ukulele and nylon-string guitars?

I used an Oktava MK012 for all the strings. The uke was a little mahogany Martin that has a really woody midrange. I found it sounded best if I positioned the mic away from the soundhole at around the tenth fret. This helped mellow out the low mids. For the guitars, I found that setting the MK012 about six inches back from the body of the instrument, and pointing directly at the soundhole, produced the fullest sound.

This was all done in your bedroom, right?

Yes. It was a very much "roll out of bed and hit record" kind of studio

Jason Holstrom tweaking the mix in his bedroom studio.



A world of sound at his fingertips: Holstrom in his revamped recording digs.

[*laughs*]. In fact, my bedroom window served as what I call "the nature booth." I'd open it when I would record vocals, and just let the natural outside sounds of the birds—which you can really hear on "Hula-Bye"—leak onto the tracks. When I needed some rain sounds to pepper a track, I'd just open that window and record the rain!

The room the record was made in, while limited, really added to the sound of the record. The queen-sized bed helped deaden the space, but recording surrounded by glass and hardwood floors made for a really bright, fast reverb.

Tell me about the percussive elements of the album. I don't hear a single snare.

I didn't want rock drums. I wanted a tribal feel to the percussion, so there's no snare on the album. I just played the kick as if it was a snare. I'd position it head-size up, and play it with two mallets—which is why a lot of the bass-drum parts are pretty fast-moving. Due to the speed, I needed a real tight kick-drum sound, so I would set the MK012 about a foot off the head. That provided a tight attack that really cut through in the mixes.

I'd often track the kick for four bars, and then do some looping on the grid to give the percussion a really tight foundation. After that, I would play a floor tom by striking the head with maracas. Playing a pounding floor-tom beat with maracas makes it sound like there are three people drumming in the room, and it also makes sure your shakers and floor toms are in time. These were miked the same as the kick. I would usually double track the floor-tom parts to give that stereo "super-percussion" sound, and pan each track hard left and hard right.

You used the MK012 mic on quite a few sources.

I only used two mics on the entire album. All the vocals were tracked through the large-diaphragm Oktava MK319 condenser, and everything else was recorded with the small-diaphragm Oktava MK012 condenser. The bass was sent directly into the MBox.

Elaborate on your EQ tricks.

I used to be in the habit of slicing the entire low end out of most sources. I found the MBox preamps tend to need a little low-end clean up, and cutting the lows left room for the other tracks without killing clarity. With such busy and rich mixes—and such limited outboard gear—I really had to be creative with my EQ to get good-sounding vocals. I'd track the higher vocal parts really close to the mic, and that would leave me with tons of mud to be cleared out—as well as tons of breathy hisses—so I got pretty aggressive with the EQ on some of the tracks. Plus, the Digidesign Tel-Ray Variable Delay plug-in that I was using on the vocals really brought up the lower-mid frequencies, so I had to cut there, and then brighten them up afterwards.

As there are track limitations to Pro Tools LE, how did you approach the mix?

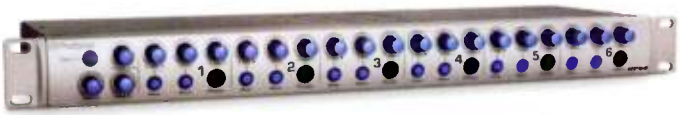
I only had 24 tracks, so I'd max out one session with all the instrumentation, and then bounce that into a separate session for all of the vocals. Then, I'd do the final mix from the vocal session. That was a great way to max-out the track count, and also focus on recording vocals as the mix played back. And if I really needed to turn the instruments up, I could go back into the instrumentation session, and bounce it right in again. It was pretty easy, actually.

You're not making the analog purist crowd very happy right now.

If the mic is placed well next to a good instrument that's in the hands of a good musician, you can EQ your way through the rest [*laughs*]. EQ

Photos: Par Andreasson

TOOL BOX



PRESONUS HP60 6-CHANNEL HEADPHONE MIXING SYSTEM

Incorporating six independent headphone amps, the HP60 features dual stereo inputs as well as external input (for each channel) so you can mix among three stereo audio streams (mix A, mix B, and external input). Each channel features headphone level, mix control between A and B inputs, external input volume, mute, and mono. www.presonus.com

ARTURIA ANALOG FACTORY V2.0

Analog Factory 2.0 ($249, free update for registered owners) features 3,500 sounds, presets from the Jupiter-8V soft synth, improved effects management, and VST 2.4/Cubase 4 as well as Universal Binary compatibility. www.arturia.com

AUDIOFILE ENGINEERING RAX 2 SOFTWARE

Rax ($50), designed for onstage use, is Universal Binary virtual rack software for Mac OS X that enables musicians to organize and control AU plug-ins via a 16-channel virtual mixer and 25 virtual effects racks. Build set lists, then control them via MIDI and the Apple Remote. www.audiofile-engineering.com

MCDSP NATIVE UPGRADE PROGRAM

Having discontinued sales of individual LE products (however, the Project Studio Bundle containing all LE products is still available for $495), McDSP is offering existing LE customers the chance to upgrade individual LE plug-ins to Native for $195. www.mcdsp.com

APPLE LOGIC STUDIO

Logic Studio includes Logic Pro 8 ($499, $149 upgrade from Logic Pro 7), a major upgrade with an intuitive new interface; MainStage (live performance application that turns the Mac into a streamlined live rig); Soundtrack Pro 2 audio post production software; 40 Studio Instruments; 80 Studio Effects plug-ins; and a vast Studio Sound Library. www.apple.com

STUDIO PROJECTS CS5 MIC

The CS5 ($1,149.99) is an externally-polarized, pressure-gradient large-capsule transducer mic with FET impedance converter. It features five selectable polar patterns, four selectable pads, four high pass filters, and four low pass filters for added control. A shockmount and foam wind screen are included. www.studioprojects.com

NATIVE INSTRUMENTS KONTAKT 3, GUITAR RIG 3

Along with its high-resolution audio engine, sound-shaping capabilities, unique KSP script programming, and universal sample format compatibility, Kontakt 3 ($449, $149 upgrade from Kontakt 2) also includes a new instrument library with six themed collections, based on over 33GB of source material. Other improvements include new effect algorithms, additional supported sample formats, and faster loading times. Guitar Rig 3 ($559 with Rig Kontrol audio interface/foot control, $339 software only) now offers a total of 12 amps and 44 effects, a reworked tone library with improved preset handling, and a redesigned Rig Kontrol pedal board. The Live View feature displays a large-scale, full-screen view of the most vital parameters for live use. www.native-instruments.com/afharmony01.info

DIGIDESIGN STRUCTURE LE AND STRUCTURE FREE

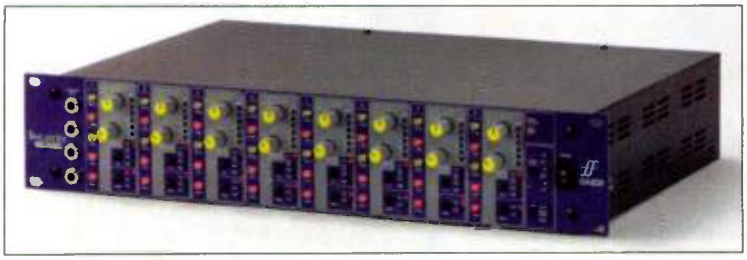
Optimized for Pro Tools software, Structure LE provides many of the same features as Structure for a fraction of the price, while Structure Free provides Structure sample playback at no charge. Both support playback of unencrypted SampleCell (1 and 2), Kontakt (1 and 2), and EXS24 sound libraries, and feature an integrated REX player that plays REX 1 and 2 files. www.digidesign.com

WHISPERROOM EXTERIOR ACCENT PACKAGE

The Exterior Accent Package consists of wood grain accent panels inlaid into the exterior of WhisperRoom wall components. This cosmetic feature does not change the overall design or noise reduction capability, and is available with all WhisperRoom models and configurations. www.whisperroom.com

LACHAPELL AUDIO MODEL 583S VACUUM TUBE PREAMP

The input stage of the 583s Vacuum Tube Preamp ($1,249) is based on LaChapell's 992EG platform in single-ended form, coupled with Burr-Brown and THAT Corporation output drivers, and delivers >70dB of gain. Equipped with a 12AX7/ECC83 tube, the 583s utilizes the Cinemag CMMI-10 or optional Jensen JT-115k input transformer. www.lachapellaudio.com



FOCUSRITE ISA 828 MIC PREAMP

The ISA 828 features eight ISA transformer-based preamps in a 2U chassis, offering the Focusrite signature sound at the lowest cost per channel to date. Thanks to its 25-pin D-Type connectors, the ISA828 also integrates with ProTools HD, as well as other popular hard disk recorders and mixing desks. www.focusrite.com

TOOL BOX

GENELEC SE DSP MONITORING SYSTEM

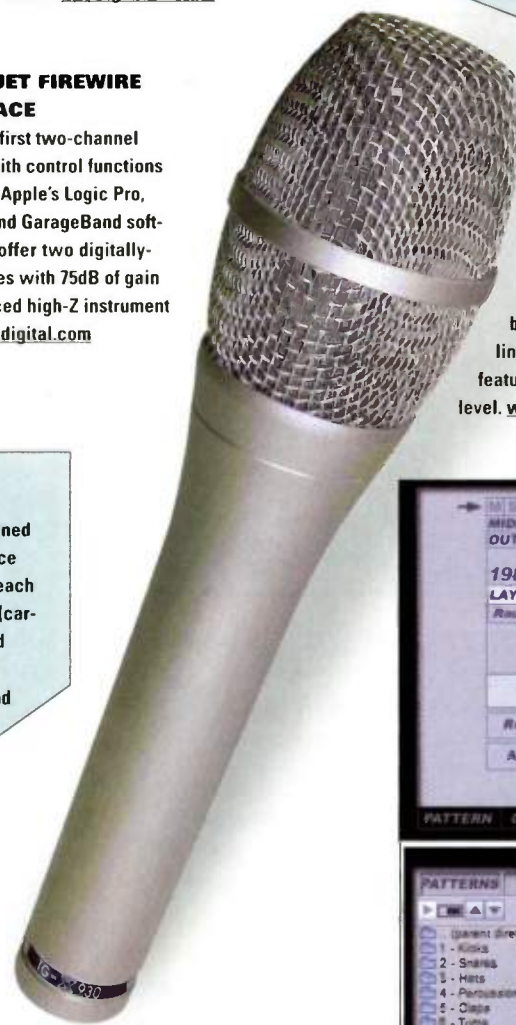
The SE DSP Monitoring System combines the SE7261A 10" DSP subwoofer with 8130A digital input active monitors (5" bass drivers and 3/4" tweeters), for easy configuration into sub-assisted stereo or multi-channel reference systems. Software control is simple, as the SE7261A DSP subwoofer provides all necessary network connections to the host computer and supplies DSP resources through its high-pass AES digital outputs to the 8130A. www.genelec.com

IK MULTIMEDIA SAMPLETANK 2.5

SampleTank 2.5 XL ($499; free update to L and XL owners) comes with 2 DVDs with over 1,800 sounds, over 6.5GB of samples, plus a bonus sound disk. The update features five send insert effects and five master insert effects, master loop sync, new reverb derived from Classik Studio Reverb, and Universal Binary/Vista support. www.ikmultimedia.com

APOGEE DUET FIREWIRE MAC INTERFACE

Duet ($495) is the first two-channel audio interface with control functions built directly into Apple's Logic Pro, Soundtrack Pro and GarageBand software. The inputs offer two digitally-controlled mic pres with 75dB of gain and two unbalanced high-Z instrument ins. www.apogeedigital.com



BEYERDYNAMIC TG-X 930 CONDENSER VOCAL MIC

Based on the MC 930 studio condenser mic's capsule construction, the TG-X 930 provides a condenser's sensitivity in combination with the TG-X touring line's durability. The TG-X 930 also features a very low internal noise level. www.beyerdynamic.com

AUDIX M1250 AND M1280 IN MICROS SERIES

The M1250 and M1280 feature newly designed circuitry with immunity from RF interference caused by cell phones and GSM devices; each model has four interchangeable capsules (cardioid, hypercardioid, omni-directional, and supercardioid. The Micros are available individually, or as part of the MicroPod and MicroBoom Series.
www.audixusa.com

FXPANSION GURU V1.5

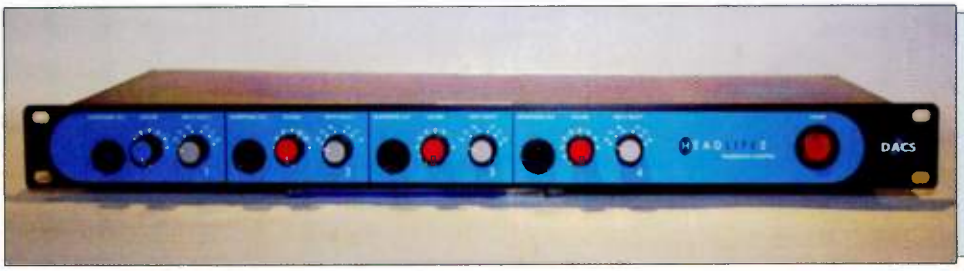
V1.5 of GURU ($249; free update to existing owners), the creative beat box workstation, has over 50 new features and an additional 3GB of original audio content. New features include audio rendering and export, creative randomization, sample reverse, and a new high-accuracy beat slicer with sensitivity and resolution controls. www.fxpansion.com

ROGER LINN DESIGN ADRENALINN III EFFECTS PROCESSOR

AdrenaLinn III combines a guitar modeler with a wide variety of tempo-syncable (internal drum machine or MIDI) modulation, delay, and looped filter sequence effects. The new version adds reverb, compression, tuner, enhanced modulation effects, and offers 40 amp models (including 4 bass amps) with improved tube emulation. www.rogerlinn.com

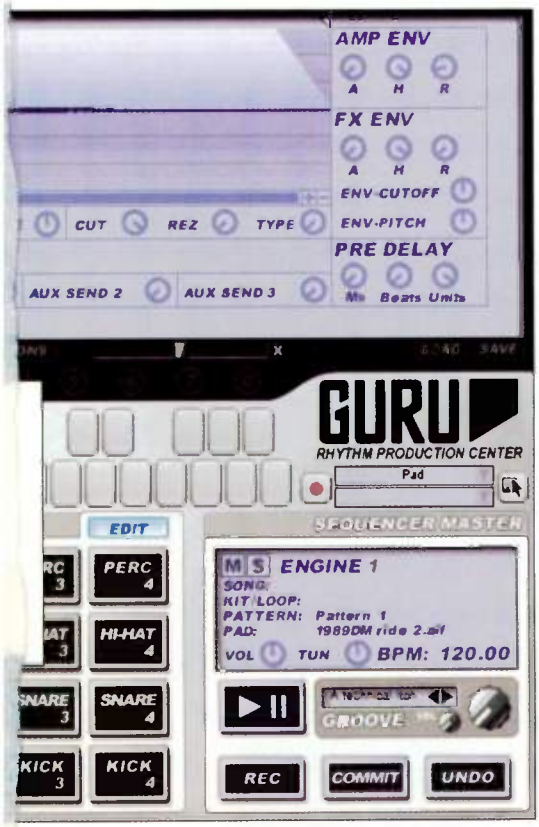
PCAUDIOLABS ECLIPSE 14-TERABYTE SERVER

Eclipse is an AMD-based dual processor system featuring 14 terabytes of storage housed in a 3U rack using Windows XP x64 software. Every drive bay in Eclipse is removable—change drives quickly and easily for different projects. www.pcaudiolabs.com



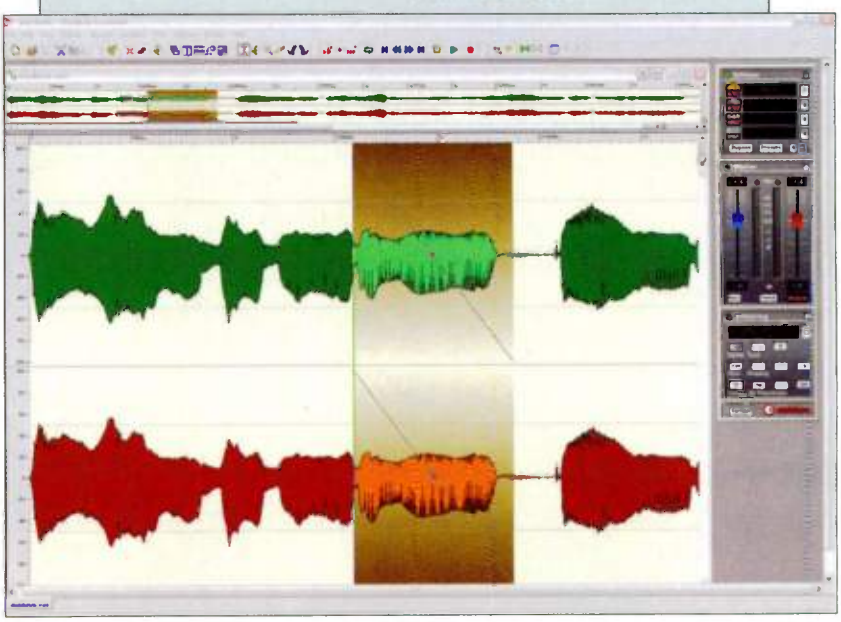
DACS CLARITY HEADLITE 2 HEADPHONE AMP

HeadLite 2 is a 4-input, 4-output amplifier with input selection for each amp. This upgraded version features new ICs that improve the unit's overall performance, particularly into low impedance headphones. www.dacs-audio.com



STEINBERG WAVELAB ESSENTIAL 6

WaveLab Essential 6 ($149.99) combines audio editing, mastering, and CD burning with an integrated podcasting module. The integrated video track allows creating audio montages and soundtracks for home videos, while the podcasting components add an Internet publishing toolset including audio upload, FTP client, and support for the RSS 2.0 standard. www.steinberg.net



All prices are manufacturers' suggested retail price. Toolbox material is provided courtesy of Harmony Central, Inc., and is used with the express written permission of the publisher.

Why Choose Full Compass?

WE WILL EXCEED YOUR EXPECTATIONS

GREAT SERVICES!

- ONLINE ORDERING
  FULLCOMPASS.COM
- PAYMENT OPTIONS
  6 MONTHS NO PAYMENT AND NO INTEREST
- FREE SHIPPING
  FOR SELECTED BRANDS AND MODELS. GROUND SHIPPING ONLY.
- GEARCAST
  NOW AVAILABLE ON iTUNES

Most competitive pricing. Our goal is to provide you with the very best value on the equipment you purchase from us. While we would love to offer you the convenience of comparison shopping, our sale prices are usually much lower than what we are permitted to publish. The majority of our vendors enforce MAP or "Minimum Advertised Price" policies on their products, which dictates the minimum sale price any dealer may publish. By talking with a sales pro at **800-356-5844**, you can save a significant amount of money.

Your personal consultant. At Full Compass we employ the most knowledgeable sales professionals in the audio and video industry. They offer valuable insight into the products we sell. The first time you place an order with us, you'll receive your own sales professional who will work with you on an ongoing basis to come up with the right product or system to fit your needs.

Great products. Full Compass is proud to offer an extensive selection of products and services. We have your favorite brands!

- AUTHORIZED SERVICE & PARTS
- BILINGUAL SALES PROFESSIONALS
  (Tenemos expertos que hablan Español disponibles en x1178 y x1164.)

Call or click for your FREE 400 page catalog!

FULL COMPASS

PRO AUDIO | VIDEO | AV | LIGHTING

FULLCOMPASS.COM | 800.356.5844

30 YEARS OF SERVICE

Call your sales pro today.

WE WILL EXCEED YOUR EXPECTATIONS

Call today.

800-356-5844

FULLCOMPASS.COM

30 YEARS OF SERVICE

First Hand News

FLEETWOOD MAC CO-CONSPIRATOR RICHARD DASHUT REFLECTS ON RECORDING AND MIXING *RUMOURS*

by Heather Johnson

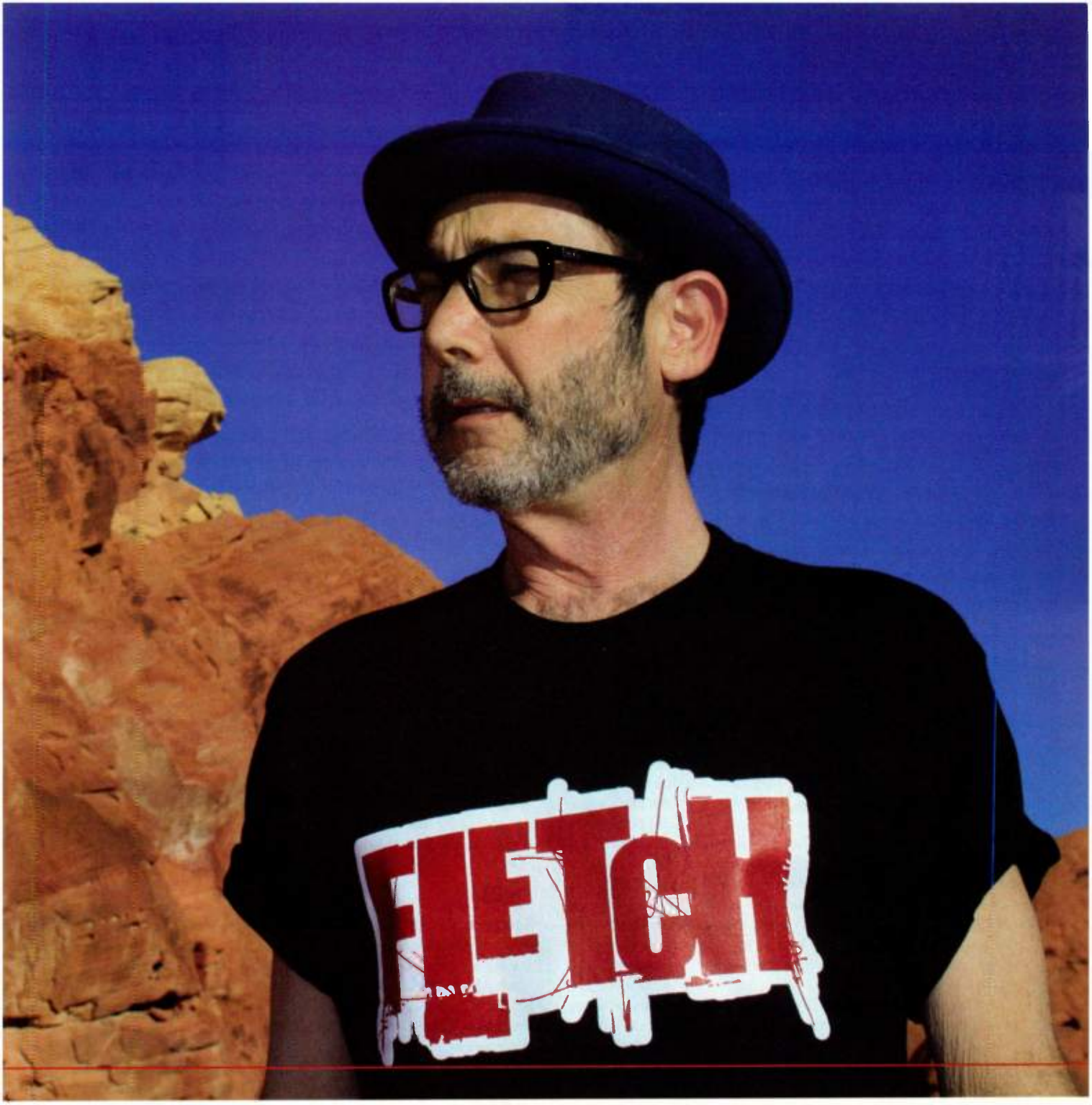
Fleetwood Mac had soldiered through years of mid-level success as a powerful '60s blues outfit lead by guitarist Peter Green and his "magical" Gibson Les Paul, until hitting ups and downs with various line-ups and a tsunami of drinking, drugs, and mental illness. Although the band did chart a few bona fide hits and radio-play favorites ("Black Magic Woman," "Albatross," "The Green Manalishi," "Hynoptized"), its constant career stalls and infighting had made it sort of a commercial, ahem, *albatross* by 1975. Few would have expected that a combination of good luck, fortuitous meetings, and renewed creative energies would not only change the veteran band's fortunes, but also transform it into one of the most successful rock acts of all time.

While the big bang was drummer Mick Fleetwood discovering Stevie Nicks and Lindsey Buckingham by sheer chance—and having the insight to invite them to join his band—the event also brought the duo's friend, engineer Richard Dashut, into the fold. Dashut went from mixing live sound for the band's 1975 *Fleetwood Mac* tour to co-producing 1977's *Rumours*—a record that spent six months atop the U.S. record charts, won the Grammy for Album of the Year, and went on to sell 30 million copies worldwide.

You started in this business from ground zero. How did that happen?

I got my first studio job around 1971, at Crystal Sound in Hollywood. I was a janitor, basically. At the time, people like Jackson Browne, Joni Mitchell, Carole King—a lot of really huge stars—were recording there. So here I am—this little punk kid, 20 years old with his eyes wide open—more than happy to answer phones and sweep floors just to be around it. I really wanted to be in the movie business, but after hanging out there, I decided *this* was for me.

I was let go from Crystal—although they tried to hire me back two weeks later. When Dave Devore and Keith Olsen—who I met when they were mastering a record at Crystal—found out I'd been let go, they got me a job at Sound City, which is where they did all of their recording. My first job there was as an assistant maintenance man. But when the head maintenance engineer asked me for a resistor, and I said I didn't burn my draft card, he quickly realized I wasn't suited for electronics. So Keith made me his second engineer, and I got to go into the control room and operate the tape machines. We had Ampex MM1100 and MM1200 tape machines that had to be aligned every day. I had to have the tape heads cleaned, the board



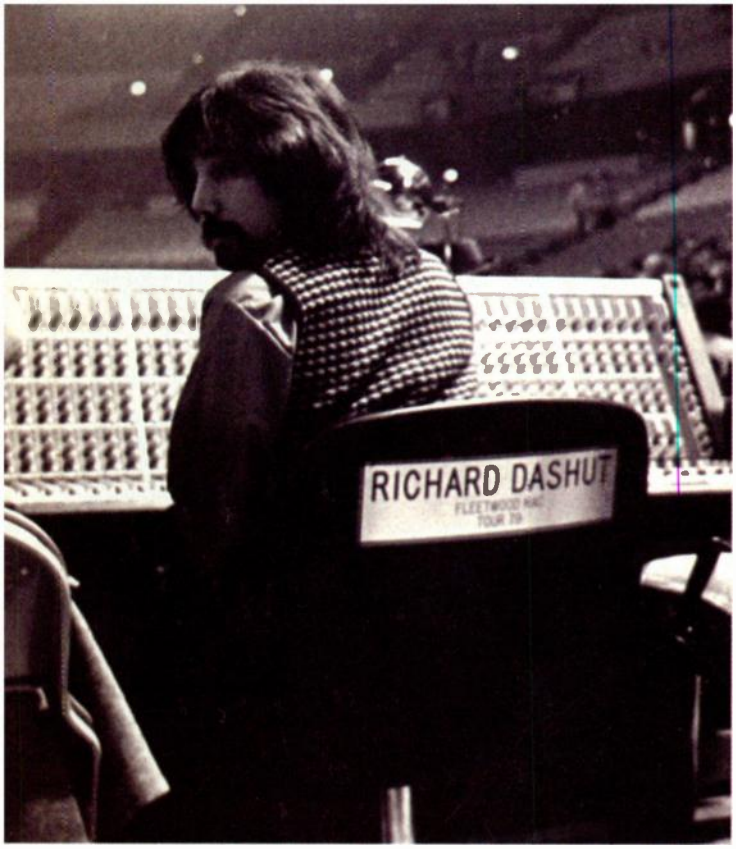
cleaned, and the room cleaned and prepared. We were working 18-hour days, doing a lot of commercials—string sessions where I had to set up all the seats and headphone boxes—but, inevitably, one of the string players would plug in their crystal headset and short out the whole system.

We had some great sessions. People like Jerry Wexler worked there. We did Elton John sessions, and we overdubbed the Tower of Power horn section. You could absorb so much. I had to keep myself occupied in those long sessions, so I would listen intently to the music, and think like a producer would. If someone made a mistake, would I stop the machine? What would I do? What ideas would I have? I learned that way—as well as by paying attention to what the engineers and producers were doing.

Didn't you meet Lindsey Buckingham at Sound City?

Yes. On my second or third day, I was relegated to painting the control room ceiling in Studio A. There were a few other people helping me—in particular a gentleman and a young lady who was trying to paint the ceiling, but kept getting more paint on her hair. I grabbed the paint roller out of her hand and showed her how to do it. That lady turned out to be Stevie Nicks, and the gentleman was her boyfriend, Lindsey Buckingham. Within two hours, we had already decided we were going to get a place together.

They were living with Keith Olsen at the time, and were in a band called Fritz. Fritz had broken up, and they had gone on their own as Buckingham-Nicks. Lindsey took me into the maintenance room and played me his demos, and the first time I heard them, I fell in love with the music. "Monday Morning"



Richard Dashut, mixing live sound during Fleetwood Mac's 1979 tour.

was on there, as was "I'm So Afraid," "Frozen Love"—a bunch of stuff. That's where my real music education started—with Lindsey Buckingham.

When you met Buckingham and Nicks, you were on your way to a good career as a studio engineer, but then you tossed it aside to go on the road with Fleetwood Mac. Why?

The Buckingham-Nicks album didn't do so well, and they got dropped from the label. We decided to take matters into our own hands, and [Sound City owners] Joe Godfried and Tom Skeeter offered us studio time to produce our own record—which was very generous of them. We were in the middle of that when Mick Fleetwood came along. Keith was demoing the studio for him as a place to record, and he was using the Buckingham-Nicks album to show Mick how great the room sounded. Mick took one listen, and basically asked them to join. That was the good news! The bad news was that we had to stop doing the second Buckingham-Nicks record. But "I'm So Afraid," "Monday Morning," and "Blue Letter"—which we had started developing—went on the first Fleetwood Mac record they did together.

I had parted with Keith Olsen, and had left Sound City by the time they started recording *Fleetwood Mac*. But they were getting ready to go on the road, so Lindsey called and asked if I'd mix the live sound for the tour. Because I was young, and had a tremendous sense of adventure and curiosity at the time—and also because of the girls and the money—I decided to say "yes" [*laughs*].

Was going from studio to stage a difficult transition?

It was not easy. In the studio, you have everything under control, and you can spend time getting things perfect. But, on the road, you have to deal with the elements—the audience, the room acoustics, the capacity of the room. We didn't have sophisticated equipment back then, either. You just had to work by the seat of your pants, and strictly by ear. It was great, because it taught me the basic sensibilities of layering and mixing. And when you have to get things together in a timely manner—like by the end of the first song—you get very adept at mixing on the fly.

***Roadie* magazine voted you "Best Live Mixer"—or something to that effect. What would you attribute that to?**

My mixing style was different. A lot of people in a live situation would just mix and hold—just get it under control, and then leave it. For me, every song was different. I was a dynamic mixer—but not a proper one. If you want to hear the absolute perfect mix, you can go home and listen to the record. To me, mixing live was an emotional thing. It was about a show—about getting an emotional response out of an audience. I would ride lead guitars and drum parts when I wanted to make a point, and then to drive home the point, I'd mix them a lot louder than most people are used to hearing them. It was a way to get people off their feet, and create an emotional experience.

I took a lot of that experience with me into the studio—especially before automation, when the mixing itself was a performance. You had to remember how you had it the time before, you're relying on two or three other people—"hands across the board" as we used to call it—and you would all be riding levels, and what one person did affected the other. That's why we had grease markers. We'd draw a line and didn't dare go above it! But mistakes often made the best mixes. For instance, at the end of "Go Your Own Way," the kick drum was way too loud—to the point where it would start hitting the compression on the radio. If you listen to the guitar solo at the end, the compressor would pump the guitar in rhythm to the kick, and it added to the whole drive of the song. That was a lucky mistake. It wouldn't have happened if we were using automation. The mix was half emotional/half technical, and when you got to the end of a great mix, it was like the ending of a great show.

When the band asked you to produce *Rumours* you brought out engineer Ken Caillat. How did you meet, and what made you two such a good team?

I had just been on the road with the band for about a year, and we went in after the tour to do a remix at Wally Heider Studios in Los Angeles, where Ken was working. We probably smoked a joint together—who knows [*laughs*]. Things went so well with that remix, that when it came time to do *Rumours*, I asked him to come work with us. He ended up doing most of the engineering, and I worked more with Lindsey in developing the band's music.

Someone else was supposed to produce *Rumours*, and they wanted to put strings on the record. That turned the band off, so they decided against using him. When we were remixing "Rhiannon," Mick brought me out to the parking lot, and said, "Dashut. You're co-producing the next record with us." I was an engineer, not a producer! Who wanted to be *responsible*? I just wanted to get my sounds [*laughs*].

What made Ken and I such a great team was that we had great communication between us, and not much overlap. We both had our specialties. Ken loved fooling with the knobs and tweaking things, and I loved working with the band and the music, and being the interface between the technical and the creative sides. But I did a lot of engineering, too. I even tuned all the drums. But somebody had to be behind the talkback, and somebody had to sit behind the board, and I found myself mostly behind the talkback.

I was way over my head. But living with Lindsey was educational. We'd play Motown, the Beach Boys, The Beatles, the Stones, and other records, and he'd show me what frequencies to listen for, and how to layer instruments. We would have a session in front of the record player, and then put what we learned into practice at the studio.

Critics praised your—and Ken's—"attention to acoustical detail." What methods did you employ to accomplish such great sounds?

It was the endless pursuit for the perfect sound. Once, we spent ten hours getting kick drums sounds in The Record Plant's Studio B, and then ended up moving to Studio A, and building a special drum platform to get what we wanted. Mick had a very light foot, so we had to try especially hard to get

M-AUDIO®

Meet the only mobile recorder that sounds better than the MicroTrack.

palm sized

MicroTrack II Professional 2-Channel Mobile Digital Recorder

Sound matters—that's why the MicroTrack™ recorder has become the only choice for audio professionals worldwide. Now M-Audio has designed the MicroTrack II, featuring even higher audio fidelity, an extended input gain range, workflow enhancements including faster file transfer rate, and more. A new standard from the people who set the standard, the MicroTrack II delivers what's most important—superior-quality recordings.

- 2-channel WAV (BWF) and MP3 recording and playback
- new features include extended input gain range, analog input limiter and BWF file marking ability
- balanced ¼" TRS inputs with line inputs and 48V phantom-powered mic preamps
- drag-and-drop file transfer to PC and Mac via high-speed USB 2.0
- storage via CompactFlash or Microdrives (not included)

Includes stereo electret microphone, earbuds, headphone extension cable with lapel clip, power supply, USB cable (A to Mini B) and protective carrying case with mic pouch

GET M-POWERED

www.m-audio.com

PERFORMANCE MIXING

Dashut reveals that his experience working the board for Fleetwood Mac's live shows inspired the "performance mixing" approach that he and co-producer/engineer Ken Caillat employed in the studio to energize the sound of *Rumours*. As Dashut details, back in the days before reliable automation, it was the engineer—and as many able hands as he or she could recruit—that moved faders, fiddled with outboard effects, twisted pan knobs, and assembled the mix landscape in real time as the tape was running. The head engineer would often have to command the tangle of hands hovering over the console like a field marshall under fire, yelling things such as "mute channel 16 *now*" and "fade in the second rhythm guitar on the next downbeat."

It was, as Dashut describes, a process of emotion and technique, and it was often as frustrating as it was exhilarating, because there was no "undo" command. If you screwed the last fade, you'd have to start the mix all over again from the top, or mix the fade separately, then cut tape and edit in the desired section.

However, at its best, performance mixing captured sonic and spectral arrangements that were just as inspired and impassioned as a guitarist hitting a transcendent solo, or a vocalist locking into the perfect blend of tone and phrasing. Like all the best musical moments of the pre-digital era, the mix *was* a real-time performance, rather than a DAW operation that can be edited, refined, and saved as countless recallable versions for eons forward. Think about that.

If you feel this type of energy and vitality is missing in your home-studio productions, consider cutting loose the safety nets of the digital age for a mix or two. (Of course, you can still return to DAW mixing if you don't dig the results, so taking a chance isn't really a risk at all.) Tank automation. Completely. Pretend it doesn't exist. And that means for everything—effects, panning, bus assignments, and so on. From now on, every mix move will be done by your own hands in real time—win, lose, or draw.

When you're done, compare your performance mix with a conventional DAW mix that you tweaked, edited, and worried over for days (or weeks). Determine if the "p-mix" delivers a sense of impact, drive, and drama that the "d-mix" lacks. If not, then technological advances have clearly enhanced your production style. But if your p-mix *does* possess more vibe, take the lesson to heart. In the end—as Dashut warns—it's not the method, it's the music. —*Michael Molenda*

the right kick-drum sounds. We nicknamed him "featherfoot" [*laughs*]. He was also known as the shuffle king, because he had one of the most amazing right hands in the business. His talent was in his right hand and snare feel, and the bass drum would follow that.

Anyway, he used his road kit on some songs—which had a very large kick drum—and it was hard to get a good, tight sound out of it. We shoved a Shure SM57 about four inches from where the beater struck the drum head to pick up the attack of the drum head, and then put an Electro-Voice RE20 a foot away from the outside head to get the sound of the kick. Because Mick didn't hit hard, we weren't getting the bottom end from his kick drum. The bottom end on that album came from John McVie's bass. So we opted for the combination of midrange presence and low-midrange punch that those two mics provided.

Still, we had no set technique. We just tried to match the sound of the song. On some songs, we wanted a more ambient-sounding kick—which the RE20 was good for—and other songs needed the drier, deader sound that the SM57 captured. We would record the kick in stereo, and then combine the two tracks, mixing one mic louder than the other, depending on the song. We definitely came up with a bigger sound than we would have gotten with just one mic.

Did you apply this recording philosophy to other instruments on *Rumours*?

Oh, yeah. We recorded most of the instruments in stereo, and then blended the tracks to get huge sounds. For example, the electric guitar on "Dreams" was composed of four signals: a direct signal, a miked signal, a signal running from the output of the amp head, and another direct signal from a volume pedal. Then, we'd pan it all in stereo so parts would sweep from left to right in a very majestic way. It was always about choosing the right mics with the right space for the right song, and that has to be done with one's ear, and with a high degree of sensitivity to the music.

It sounds fun, but it was probably a bit nerve wracking approaching the mix for each song individually, instead of just settling on a basic sonic blueprint for the album.

I remember going through *nine* pianos at the Record Plant in Sausalito. Of course, we were probably more whacked out than the pianos [*laughs*]. *Rumours* took a year to record, and *Tusk* took about a year and a half. Fleetwood Mac didn't believe in pre-production, and that's good and bad. The bad news is that the album cost a million dollars—which was a ridiculous amount of money back then. The good news is that because we went in with a totally open mind—not preconceived in any way—we came up with things where the sum of the parts were far greater than they could have been if we had worked everything out beforehand. A lot of the songs were written in the studio. A lot of the lyrics were written in the studio. It was a trying, but very exhilarating process.

I understand there were some serious issues with the 24-track tape that resulted in many more hours being spent in the studio.

When you drag analog tape across the tape heads, you're basically wearing it out. After about six or eight months, you can really wear the tape out, and it's the high end that goes first. It's subtle, so you don't notice it until the day you listen to your tracks, and realize "My God—these drums are dull. Where are the cymbals?" This is what happened to us, and we were freaking out because we didn't know how to get our high end back. We were starting to see daylight where the tracks were! And we were scared, because Warner Bros. was waiting for the album. Thank God, we had done a set of safety copies after we cut the master track. Even though the safeties were second-generation backups, they still had all the high end, and we overlaid all of the overdubs to those tapes. This was before SMPTE synchronization, so we had to get two 24-track machines, and sync the machines by ear. Ken would put on headphones, and we'd pull up the drum overhead tracks on both the original master and the safety master, and then we'd listen to the phasing between the cymbals. Ken would have to turn the VSO control until the phasing would start to go out to time sync the machines. And, of course, it would go out too far, and then you would have to stop and punch in sections. It was an 18-hour process to do three or four songs. But we pulled it off, and when you listen to *Rumours*, those are the safeties you're hearing!

When you were recording basic tracks, did the band record together?

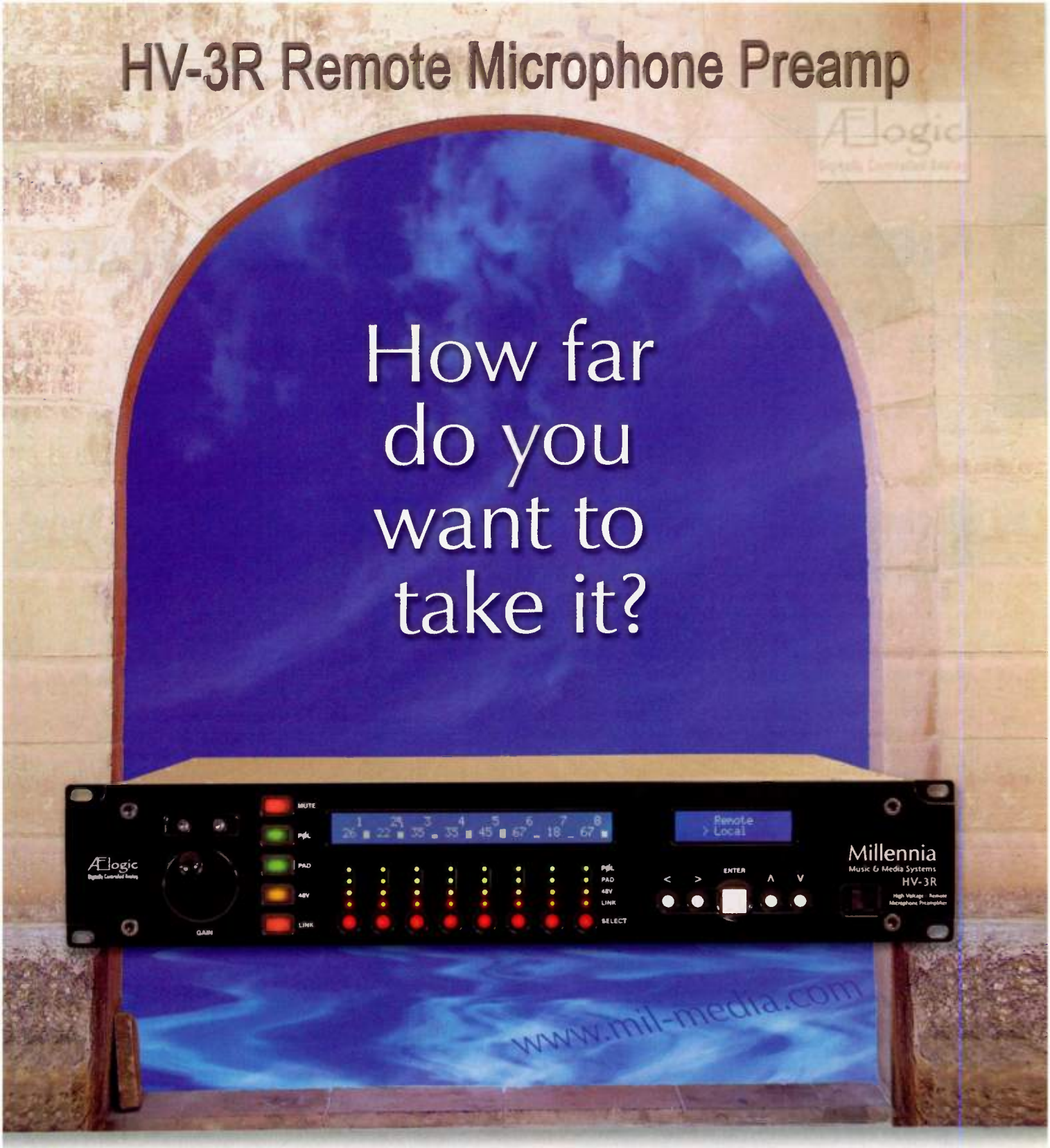
The guitar solos and drums on "The Chain" were played together. Other than that, I don't think any of the instruments were actually played together. Everything was overdubbed. It took an intense amount of work to get everything to sound natural, but when the parts are right, it's going to sound like they were never overdubbed. When you're working with a group of people as talented as Fleetwood Mac—and you have that kind of time and budget—you have the freedom to experiment and work by your gut instincts.

Considering that you learned your craft before the digital age, how have the new tools changed your approach to recording?

Pro Tools and other computer-based recording systems have changed everything. These systems have really given us the ability to *change* songs after the fact—not just correct them. I find myself relying more on that. You try not to, but you do.

At the same time, you have to be very conscious about getting most of what you need out of the performance. Take the auto tuning of vocals. It used to require 40 takes to get a performance that right. The singer would be so tired, they would just give it up [*laughs*]. I feel we miss a lot of that these days. There's no question that technology vastly improves your ability to do things, but, in the '70s, we really had to rely on the song. The reason you're recording in the first place is because of the song. Today, the method has become almost as important as the reason for doing it, and that can be dangerous. EQ

HV-3R Remote Microphone Preamp



Millennia
Music & Media Systems

www.mil-media.com

530-647-0750

Eight Channels of HV-3 preamps
Remote control via Ethernet or MIDI
Pro Tools HD Compatible
ÆLogic Remote Control PC Software
HROE option for three outs/channel

NEW YORK GROOVES

The Big Apple's premier producers and engineers share their secrets for putting some urban grit into your mixes.

by Dan Daley

Is the immediacy of the New York sound—from old doo-wop records to modern rap and hip-hop hits—a byproduct of the city's vibe seeping from its subway tracks to an artist's basic tracks? Nile Rodgers rode the group Chic to stardom and a fertile production career that has included hit records for Madonna and Duran Duran. But he also rode the subway a lot, and he says the rhythmic groove of the cars made its way onto more than a few records.

"If you listen to the very first Chic single, 'Everybody Dance'—that's where the groove for that song comes from," he says. "The subway connected us. I even lived on it when I was a runaway. We called it the '20-cent hotel'—a reference to the cost of a token in the 1970s."

The subway was also Rodgers' introduction to ambience.

"The public restroom at the West 4th Street station was large, and it had a wonderful reverb," he recalls. "My acoustic guitar would sound so amazing in there. So when we did Chic at Power Station Studios, we would use the ladies bathroom as the reverb chamber. At the end of 'Le Freak,' when you hear the words 'Ahhh, freak out,' you also hear [engineer] Bob Clearmountain putting a ton of plate reverb on the kick drum. That became a signature of the New York sound."

"When I first came to Los Angeles in the 1970s, I could really compare how the two places worked," says transplanted Brooklyn native, and multiple Grammy-winning producer/engineer Al Schmitt. "You could see the things that were special to New York. For instance, not a lot of guys in L.A. were using tube microphones back then. But in New York, everything was tube. And there was also the attitude. There's nothing like it anywhere else. Guys like Tom Dowd and Phil Ramone were *aggressive* mixers.

Bob Power, famed hip-hop producer/engineer known for his work with A Tribe Called Quest, The Roots, and Nas, at Chung King Studios. Photo by Louis Myrie.

Ari Raskin, producer/engineer for the Brand New Heavies, Talib Kweli, Will.I.Am, and others.

They had a lot of input into sessions. They weren't just setting up microphones and getting balances. They were giving you their opinions—whether you asked for them or not! Maybe it's because you have to fight your way on the subway every day, but we never held back in New York."

"If you listen to records made in Los Angeles, and what we did in New York, the L.A. tracks tended to acquire a glossier pop sheen," says Elliot Scheiner, a Grammy-winning producer/engineer who is well known for his work on classic Steely Dan albums, such as 1976's *The Royal Scam*. "That was fine, but it's not New York. Here, the kick was tighter, and there was more space between the instruments. They're pop records, but they're jazzier-sounding, and New York is jazz central. There was also a lot of emphasis on the low frequencies in New York. For example, we'd put a saxophone through a rotating speaker cabinet to get more textured lows onto it. If you want to hear the contrasts between the two sounds, listen to artists who recorded in both cities. For example, compare Van Morrison's *Wavelength* and *Astral Weeks*—which were made at New York's A&R Studios—with his later work in Los Angeles."

Nile Rodgers, producer for Chic, B-52s, and Diana Ross.

Michael Brauer—a native New Yorker who has worked with Aretha Franklin, Luther Vandross, and Hall & Oates—describes New York sounds as having a prevalent snarl.

"At Media Sound where I worked, if people didn't have a 'snarly' look on their face, it meant you probably weren't giving them what they wanted," he explains.

Media Sound is long gone, but the snarl lives on. Brauer added some New York-style grit to the mix for John Mayer's *Continuum*—a "throwback" to the Media Sound days, he

Home Cooking

In the current era of home studios, mobile-recording strategies, digital modeling, and easily replicated timbral landscapes, it's arguable whether regional sounds truly exist. For example, a Los Angeles producer can certainly evoke the mythical toughness of a New York producer's sounds—even if the recordings are being made in a wood shed in New Zealand.

However, environments can certainly impose themselves on the artistic temperament, as they did most prominently from the '50s up until the mid '90s, when big studios often possessed unique and magical sonic resonances, and were run by idiosyncratic staffs (the Sun Sound, the Motown Sound, the Muscle Shoals Sound, the Trident Sound, the Sigma Studios Sound, the Criteria Studios Sound, and so on). Of course, the culture and vibe of a city and its musicians also enter into the equation. So if you want to immerse your homegrown recordings in the stereotypical angst, energy, and immediacy of classic tracks birthed in legendary New York studios, you'll need to get yourself into a New York state of mind.

DO IT NOW, PUTZ! **Sure, you *could* overthink everything, and spend days dialing in every minute detail of your recording. But then, you'd be what Sinatra would call a "Harve." Practice your piece so you can strut in and get it recorded quickly, while passion, excitement, and the soul-shaking shock of delivering something fresh and untested are pulsing through your body. Don't forget that "immediacy" is derived from "immediate."**

GROOVE. **Joe Blaney mentioned the influence of New York's club culture while recording the Clash, and the city's diverse, vibrant, and sweaty dance scene has absolutely funked up much of the music made within its borders. Make sure *your* tracks punch, stutter, and pulse. If you can't move to your music, it's probably too white bread to evoke a NYC vibe.**

SNARL. **Michael Brauer talks about the snarl in New York sounds. Clean, calm, and polite tones are only groovy if you're committed to bringing back the Andy Williams production style. Get belligerent. Tough. Compress signals until they snarl and spit like a cornered raccoon. Hammer the mids until they're sharp enough to rend flesh. Pummel the lows until they're as powerful as a body punch from Chuck Liddell. Distort everything—at least a little bit—and be daring enough to leave the ragged edges alone. If you can't handle the aggressive snap and roar, it might be time for you to forget about dialing in the New York sound and consider moving to Boca Raton. —*Michael Molenda***

says—by running the vocals, snare, kick, and bass through a Thermionic Culture Vulture stereo tube distortion unit. To get just the right blend of snarl and source sound, Brauer routed the Culture Vulture to an aux send, and then brought the effect return up on another channel. This routing strategy also ensured that the original sound retained its impact.

Independent engineer/producer Joe Blaney mixed the Clash's "This Is Radio Clash" single at New York's Electric Lady Studios in 1981—as well as recording the band's *Combat Rock* with producer Glyn Johns—and he says his studio approach was heavily influenced by the city's urban club culture.

"You'd hear these huge, raspy kick drums coming out of the hip-hop clubs from early rap records and 12-inch B-sides, and you would incorporate those sounds into your own records—maybe even subconsciously," recalls Blaney. "You wanted to make the kick drum sound dangerous, and we did that by saturating the tape and distorting it. You can hear this kind of kick drum all over 'This Is Radio Clash.'"

To get a tough, Clash-like kick today, Blaney suggests sending the kick signal through a Tech 21 SansAmp, or running the track out to a guitar amp and miking the speaker. Back in the day, Blaney often saturated the analog tape by boosting subsonics between 30Hz and 50Hz. In the DAW environment, you can let the amp or processor deliver the grit and saturation, and boost a more audible frequency range of around 80Hz to 100Hz for the low-end boom.

Bob Power's engineering career spanned numerous eras in New York's R&B and urban genres, including work with Michel N'degecello, Chaka Khan, A

8 WAYS TO ADD A NEW YORK VIBE TO YOUR HOME STUDIO

1 Raise your own rent.
2 Acquire a rodent problem by purchasing lab rats at a bio supply house.
3 Arrange for an aspiring rapper to get capped in your foyer.
4 Instead of saying "Testing 1, 2, 3" for a mic check, shout "F**k me? F**k me? F**k *you*!"
5 Answer your cell phone with an irony-laced "Home of the hits!"
6 Keep on hand a large assortment of grease-stained menus from delicatessens and Chinese restaurants that closed at least a year ago.
7 Install a rickety, noisy generator in the next room to get that subway-running-under-the-building sensation during tracking sessions.
8 Shut your studio down. —*Dan Daley*

PRODUCTION | **THE PROFESSIONAL'S SOURCE**

real world solutions from industry professionals!

www.bhproaudio.com

800-947-5518 | 420 Ninth Ave. New York 10001 | We Ship Worldwide

B&H PHOTO · VIDEO · PRO AUDIO

Tribe Called Quest, and De La Soul.

"A component of the New York sound is the fact that the music is loop-based," he says. "Everything tends to sound close-in, and, as a result, the sound is a bit tighter and less ambient on a New York record."

Noise is also a factor. Power remembers A Tribe Called Quest's *The Low End Theory* LP—which, upon its release in 1991 was credited with establishing alternative rap as a definable genre, and putting Busta Rhymes on the map—used audio artifacts as an element of its gritty sound.

"I was asked *not* to clean it up," Power recalls. "Noise became part of its authenticity."

Purposefully going for noisy artifacts today can be as simple as capturing sounds with cheap microphones, suggests Power. "You can also apply both high and low bandpass filters to a signal, and then crank the 'ugly' area of midrange between 900Hz and 1.5kHz. Another technique is simulating the days of low-resolution, 8-bit samplers by taking the input signal way down on a modern sampler as you capture sounds. You lose one bit for every 6dB you're down on the conversion. When you're done, slam the sound with compression. And if you want that telephone vocal sound, don't just turn the mids way up. A cooler option is to take a Sennheiser MD451, place it just above the speaker of an actual speakerphone, and add a lot of compression to that."

"I like running vocals through headphones, miking them, and then putting them back in the mix as an effect," says Pat Dillett, engineer for They Might Be Giants. "It makes a kind of eerie, scratchy thing that's hard to describe, but very New York."

Engineer Ari Raskin, who has engineered for Talib Kweli and the Brand New Heavies, suggests that having the guts to do things that go against conventional recording wisdom is also a component of the New York sound.

"I was doing the Brand New Heavies *Get Used to It* album at Chung King Studios," he says, "and we were recording what we thought would be a scratch drum track for the song 'Sex God.' The situation didn't really allow me time to get a full-blown drum sound. Instead, I opened the vocal chain—a Sony C-37A mic and a Universal Audio 1176 preamp—as well as a Royer ribbon mic positioned on a nearby guitar amp that wasn't being played through. We all looked at each other, and thought, 'Whoa—that's an aggressive and gritty drum sound, and it's all mic leakage from a typically small New York City live room!' When I mixed the song, I sampled that scratch kit's kick drum—also cranking up the Neve VR EQ gains to accent the anger of the drum room—and I blended it with the final drum track to give the kick more up-front punch. It's a sound you probably wouldn't expect to hear coming out of Los Angeles, London, Miami, or Nashville. It's very New York. EQ

BASS PLAYER Magazine presents:

BASS PLAYER LIVE!

October 27 & 28
New York City

Millennium Broadway Hotel
145 W. 44th Street at Broadway

Learn!

Clinician Lineup: Steve Bailey, Oskar Cartaya, Stanley Clarke, J.D. DeServio, Jimmy Earl, David Ellefson, Matt Garrison, Andrew Gouché, Stuart Hamm, Jimmy Haslip, Jonas Hellborg, Bakithi Kumalo, Tony Levin, Dave Pomeroy, Chuck Rainey, Leland Sklar, Gerald Veasley, Gary Willis, and Doug Wimbish.

See!

Top artists, live clinics and performances, and the exclusive EQ Recording Lounge. The BPL concert: Presenting bass greats Tony Levin and Leland Sklar with the *Bass Player* Lifetime Achievement Award, and performances by Matt Garrison and headliner Me'shell Ndegeocello.

Play!

The latest basses, amps, effects and more at the hands-on exhibit hall!

Visit BassPlayerLIVE.com to sign up for the BPL enewsletter and receive up-to-the-minute details about the weekend's events and appearances!

NewBay Media

REGISTER TODAY AT: www.bassplayerlive.com

SPONSORED BY

Alleva-Coppolo · Ampeg · Ashdown Engineering · D'Addario · DR Strings · Elixir Strings · Epifani · ESP · GK Gallien-Krueger · Hartke · Lace · mtd Kingston Series · Mesa/Boogie The Spirit of Art in Technology · Peavey · SWR · Trace Elliot

Dead RECKONING

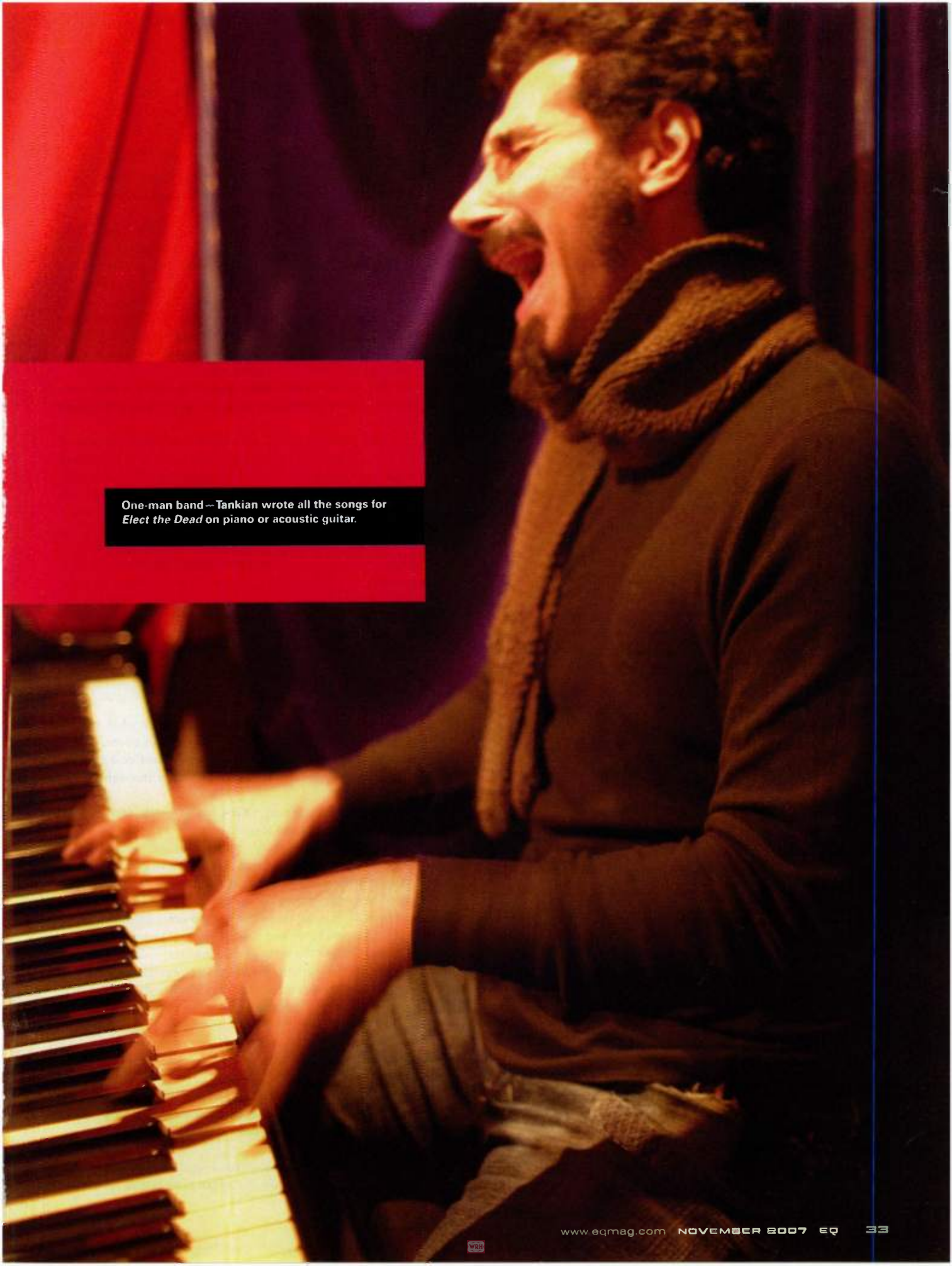
SERJ TANKIAN TAKES TOTAL CONTROL OF HIS HOME-STUDIO PRODUCED SOLO ALBUM, *ELECT THE DEAD*.

"I've never considered artists more socially responsible than plumbers, or gardeners, or politicians," says System of a Down vocalist, poet, activist, multimedia performer, and now solo artist, Serj Tankian. "I think everyone has a voice, a vision, and a reason to exist here. It's each of our duty to connect with that vision, and to follow it full heartedly. If someone's vision is to be socially or politically active, so be it. If someone's vision is to write an amazing love song, then so be it."

Tankian's vision seems to be following his immense creative curiosity along every uncurling tendril of anything having to do with art—a direction typified by his tackling the songwriter, artist, producer, engineer, and record company roles for his new solo album, *Elect the Dead* [Serjical Strike/Reprise]. Working in his home studio—Serjical Strike Dungeons—assisted by only a few friends (most notably, operatic soprano Ani Maldjian, SOAD drummer John Dolmayan and Praxis, Buckethead, and Guns N' Roses drummer "Brain" Mantia), Tankian marshaled his varied influences to craft an expansive hard-rock album that echoes System's signature rhythmic twists and dynamic shifts, while embracing a near-cinematic architecture of tones and moods.

by Michael Molenda

Photography by Greg Watermann



One-man band—Tankian wrote all the songs for *Elect the Dead* on piano or acoustic guitar.

Dead RECKONING

Somehow, Tankian is able to strike balances between seemingly disparate elements (vicious guitars and tender strings, near-poetic lyrics and silly banter, etc.) to conjure songs that initially hit hard, and then reveal vast soundscapes of adventure and intrique. As a result, *Elect the Dead* is not only a kick-ass conceptual-rock album, but also a fitting tribute to a man whose artistic force is divided between communicating deep truths and simply having fun.

Your songs are often more like novellas, than little ditties about love. Does that conceptual approach affect how you develop sonic landscapes for your work?

That's interesting. But I think a great love song—even if it's a dorky love song—will transcend the personal and become universal. If people will associate with what you feel, then that's a good song. When you listen to a Beatles song, for example, you get that transcendence. You listen to it, and you say, "I feel that. I'm not them, but I feel that." That's good work, and it comes from the fact that music doesn't belong to us as artists. Music comes from the universe, and, at best, we're just great presenters.

But even if we're all antennas for the muse, we still have to document what's in our heads onto tape, so to speak. Is your process to leave the inspiration raw, or refine it?

When it comes to making the music, I follow my gut, and that's that. But I do have the expanded experience over the years of working with [producer] Rick Rubin and Daron [Makakian, SOAD guitarist]—who is an amazing arranger. The System experience has taught me a lot about how to make things sound great. For example, when I did the vocals during a System session, Rick would always tell me, "Pronounce that word better. I want to hear exactly what you're saying." All of that stuff sticks in your head, so when you're doing your own project, you can't help but apply what you've learned to make things better.

> "AS SOON AS YOU FINISH WRITING, THE SONGWRITER HAS TO STEP ASIDE."

Let's say I'm listening to a song in the control room; the songwriter and producer parts of me agree that the bridge isn't as powerful as it should be. So what do I do? Well, I'll go in, and try something else. I've experimented a lot with this record to get things just right. I've re-recorded guitars, changed tunings, and changed tempos. I've even taken parts from songs and inserted them into other songs. On the song "Feed Us", for example, the breakdown is from a whole different song that didn't even make the record. The part is a different tempo, too, but it works perfectly. If I showed you the exact edit point, you might be able to hear the tempo difference, but it wouldn't bug you very much. It's done in a way where you wouldn't guess it's not a part of the same song.

Now, while you don't want technical things to get in the way of a listener's experience, I don't think a recording should be perfect. I think it should be perfectly *impassioned*, but not technically perfect. For example, there are vocal lines in songs such as "Saving Us" and "Elect the Dead" that I left from the original demos because the vibe was so *there*. I couldn't replicate it. I mean, I could replicate the words. I could sing it better. My voice could sound better. But that original organic and intuitive feeling that comes from saying something for the first time could not be recaptured or improved upon. After you're forced to sing or play something three or four times, you're going to lose some of that impassioned strength—and that's as much a part of the quality of a record as the sonic side.

Given your desire to document those moments of discovery and passion, I'm assuming you save just about everything you record?

Thanks to digital hard drives, we can save whatever we want—and, yes, that's everything [*laughs*]. However, I did lose four drives in the process of making this record. To this day, we don't know what happened. At one point, we were losing data every two weeks. We didn't know whether we needed to upgrade Pro Tools, or whether it was the computer. We didn't know where to go. In three out of the four cases, we had other backups, but that fourth drive had more than 100 songs on it. I had to pay someone to recover the data. In fact, the original version of "The Unthinking Majority"—the first song from the album that we released as a teaser—was on that drive. The truth is, I even had a safety of that drive, but the data was backed up so long ago, that I didn't know how many tracks had been changed or added since the backup. So the songs wouldn't have been lost completely, but I figured I would have lost about 20 hours of new performances if I didn't pay to have the data recovered.

Is this one of those "note to self" scenarios?

[*Laughs.*] Yeah, I don't know if I was out of town, or what, but the backups clearly weren't done when they should have been.

How did the songwriter, artist, producer, and co-engineer parts of your brain communicate with each other throughout the recording process?

As soon as you finish writing, I think the songwriter has to step aside, and the producer has to step in. I was doing both jobs, so, in my case, the producer was very in tune with the whole project [*laughs*]. But when I wrote the songs, I already knew how I wanted to hear things. Until the actual track was close to what I had envisioned in my mind—or *better*—I didn't give up. I struggled. I recorded the album, and sat with it for three months before I started mixing. Then, I went back and forth, and tried all kinds of techniques. I took a lot of time mixing. As the artist and songwriter, I was happy with the songs, and I didn't want to f**k them up as a producer. So I took my time, and I got some good advice from friends—musicians, producers, and many other people—and I got the tracks where I really wanted them. I wanted to have the confidence in my ears and mind to be able to say, "I know what the right thing is, and this is what I want." When I got that confidence, I got the sound I wanted for the record.

But you also draw your musical experience from a lot of areas—rock, world music, film scoring, electronica, and so on. How difficult was it for you to focus your varied influences to produce a cohesive solo album?

I've always wanted to make as diverse a record as I could, but also somehow make it work as a whole. You've got to be able to tell a story

SONY

Find the right music content every time.

Fast, easy, and 100% royalty-free.

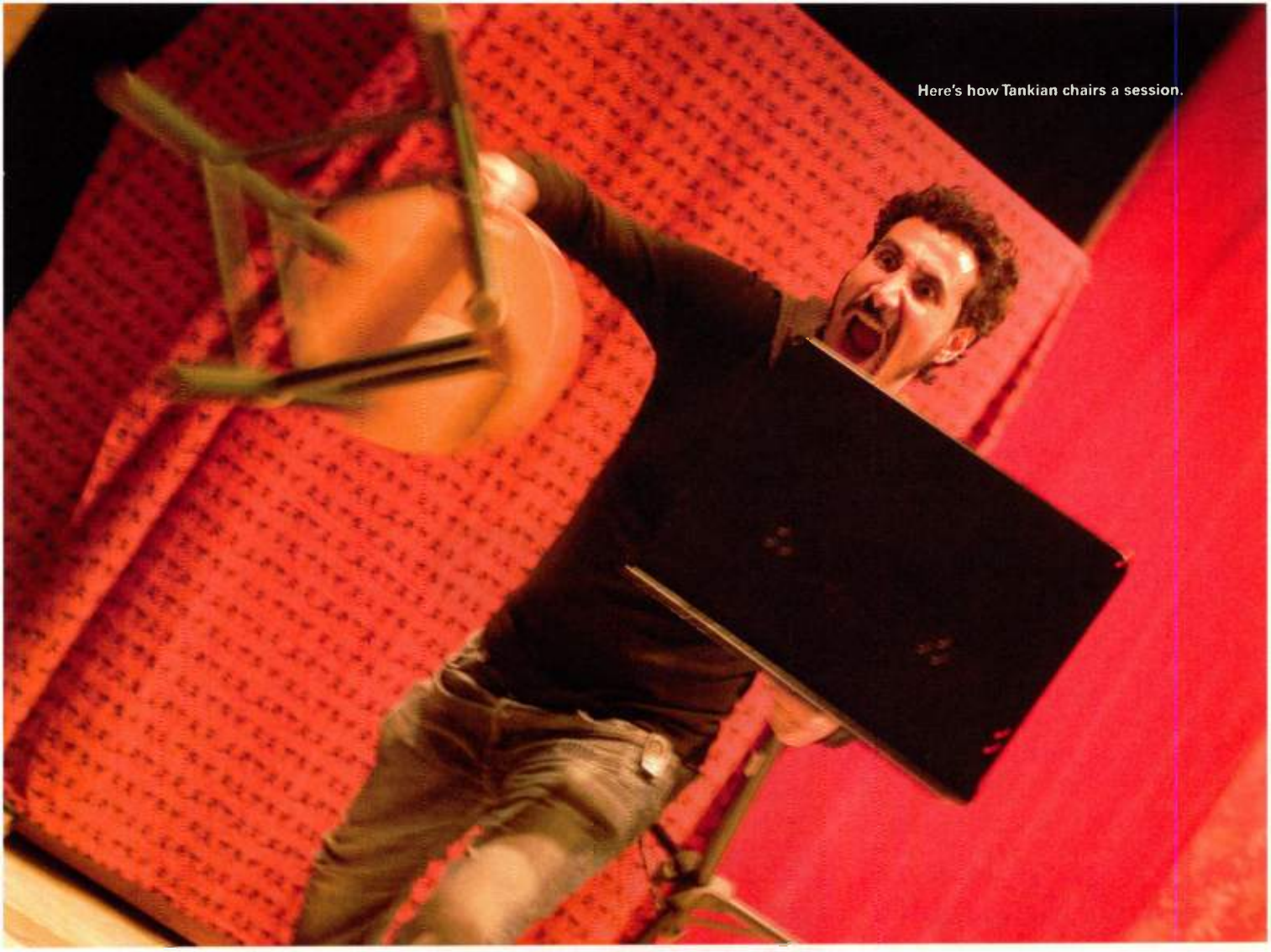
When you need music and audio content, you want to cut through the clutter and locate unique materials that won't blow your budget. The Sony® Sound Series catalog contains over 125 royalty-free content libraries – from single instrument titles and song construction kits to exclusive sound effects and our groundbreaking Artist Integrated™ series of fully synchronized libraries. Our sounds are expertly organized so you can find what you need – fast. Our legendary pitch-shifting and time-stretching metadata has made ACIDized® loops an industry standard that is supported by most music creation applications. Sony Sound Series loops can be imported into any program that supports the .wav file format.

To audition and purchase Sony Sound Series loops or locate a dealer near you, go to www.sony.com/loops.

like.no.other™

Tankian and Dan Monti (right) discuss a few fader moves.

Dead RECKONING



Here's how Tankian chairs a session.

with the music—even if it's not a thematic or conceptual record. You've got to be able to start somewhere, and finish somewhere. That's what's great about putting out a record. If I released a handful of songs that was intended to be downloaded separately—and never be released together as a CD—then I wouldn't care so much. But this *is* a record, and the songs have to connect to each other so that it all makes sense. Of course, the other trick is that not everyone will listen to the entire album. Some people will choose to download only a song or two, and so each song must also stand on its own. What works in the desert should work in the city [*laughs*].

You've had the benefit of working in big studios, and recording in your home studio. Do you operate much the same in the two environments?

Obviously, you're a lot more comfortable in a home studio. You go in when you're inspired, and you work as long as you want. In a major studio you're paying for time, and you're always conscious of that. But I think a home studio works for me, because I'm very hands-on. I'll sit down—sometimes in the middle of the night—write a song, record it into Pro Tools, cut it the way I want it, do a rough vocal melody, and then leave it until I feel like coming back to it and arranging it further. And when it's my own space, I can go and come as I please without having to drive to work [*laughs*].

How is your home studio set up?

My studio is about 1,000 square feet with three rooms—a control room, a live room, and an isolation booth. Everything is recorded on Pro Tools HD. I have a Digidesign Control 24 console with Focusrite mic preamps, and I also have some really cool outboard gear, such as a Neve 1073 preamp and a Urei 1176 compressor.

Can you detail your typical songwriting process for *Elect the Dead*?

I usually wrote the songs on either a piano or an acoustic guitar. Once I had a basic sketch—including some rough vocals—I programmed the drums around it so that I had my rhythms figured out. Then, I recorded a basic electric-guitar track—just for me to have some melody ideas for the guitars, and some rhythmic ideas around the drums. At that point, I developed a bass line—which really told me what my rhythm section is doing. Then, I went back and redid the guitars, or started layering them.

How did you generate the programmed drum tracks?

I used a lot of stuff, but mostly Sony Acid. The only reason I keep a PC around is for Acid—most everything else I do on a Mac. I'd throw the rough vocal and acoustic guitar or piano tracks in there, generate a click track, and then build the drums using samples. Then, I dumped the drums back into Pro Tools, and started recording the rest of the tracks.

And, at some point, real drums were laid on top?

Exactly. We recorded the live drums at The Pass studios in Los Angeles with engineer Krish Sharma and assistant engineer Joe Oreland. But, ultimately, we did whatever worked for the song. The sampled beats were deleted entirely, or left in for certain parts, or used completely if the real drums didn't make sense for a particular song.

What were the main guitars you used during the sessions?

I used a bunch of different stuff—a Gibson SG and a Les Paul Custom, a PRS, a Fender Strat, and a Jerry Jones electric sitar. I also used this really cool custom guitar that First Act made me. It's right between the Gibson- and PRS-type warmth and the more high-pitched Fender sound. It doesn't have a name, I guess.

How did you cast the guitars for each song? What would lead you to pick up the First Act, as opposed to the Les Paul or the Strat?

I used them all in combination with each other, and I did a lot of layering. I also used multiple amps. I'd start with one guitar—let's say a Les Paul—and split the signal through a Little Labs PCP Instrument Distribution Box to a Mesa/Boogie Triple Rectifier, a Marshall Mode Four, and a Vox AC30. I wanted a lot of different tonal options so we could blend and/or construct sounds during the mix. For the rhythm guitars, we'd first find a guitar and amp combination we liked, and then we'd mix and match other amps and guitars until we dialed in sounds that were complementary. Then, on top of those layers, I'd overdub a part using a SansAmp PSA1.1 to get that cool, metal "shhhhhh" tone. I love the tremolo on the AC30, so I did some tracks using that, and I also played some sustaining guitar parts while plugged into a Moogerfooger MuRF pedal. Ultimately, there were at least eight rhythm guitars for each song.

How did you mic the amps?

I wanted some separation between the guitar tones, so I set up the Rectifier and the Mode Four heads in the control room so I could fool with the knobs, and I put the speaker cabinets in the live room with some foam panels between them. The AC30 was in the iso booth. I'd often mix up the speaker cabinets, and put, say, a Marshall 4x12 with the Rectifier, and a Sunn 4x12 with the Marshall. Most of the guitar sounds were recorded with Shure KSM-44s positioned close to the speaker cones—some dead-on, and some angled a bit. The KSM-44s are good for recording guitars. They're pretty transparent mics with a sweet midrange emphasis.

When you layer a ton of guitar textures, out-of-phase signals and other frequency issues can often make the guitars sound *smaller*. How did you manage to avoid this problem, and make your guitars so huge and ferocious?

I had a great engineer, Dan Monti, working with me, so that helped. If he heard anything out of phase, he'd deal with it. And, of course, the mixing phase is critical—you don't want to have the whole kitchen sink playing back on the final track. You let your mixer weed out what he needs, and then go about separating the guitar sounds. That's the beauty of mixing. If I had a *bad* mixer, I imagine all the guitars might sound mushy and small—which is the risk you run when you're layering a lot of different instruments. But Neil Avron came in, and he did a phenomenal mix. He was able to separate the guitars in a really classy way so that they sounded thick, and all the tones supported each other. If something was unnecessary, he'd mute it. As long as it sounded good, I was into it.

Can you offer any specifics as to how Neil mixed the guitars?

I wasn't there the entire time he was mixing over at Paramount Studios in L.A.—although I'd come in when he had something down, and make my comments. Most of the job was using EQ and panning to separate parts, of course. He'd pan some rhythm guitars hard right and hard left, and he'd also position each layer according to its frequency range. For example, a high guitar part might be panned at 4 o'clock, while a low-midrange guitar might be positioned more at 9 o'clock.

Dungeon Toys

HERE'S A SAMPLING OF THE TOOLS TANKIAN HAS AVAILABLE IN HIS HOME STUDIO:

SOFTWARE: PRO TOOLS HD, MCDSP FILTERBANK, MCDSP COMPRESSORBANK, BOMB FACTORY CLASSIC COMPRESSORS BUNDLE, DIGIDESIGN EQ III, ANTARES AUTO-TUNE 3.1

MICROPHONES: KORBY KAT SYSTEM WITH C12, U47, AND U67 HEADS; SHURE 514B, 55SH, BETA 52A, BETA 56A, BETA 57A, BETA 58A, BETA 91, BETA 98D, KSM27, KSM32, KSM44, KSM141, SM7A, SM7B, SM81, VP88

CONSOLE: DIGIDESIGN CONTROL 24

STUDIO MONITORS: GENELEC 1032A

What about your bass tracks? Did you go direct, mic an amp, or use a hybrid approach?

I took the lazy way out! I was doing those tracks by myself a lot of the time, and as I had to engineer them, the easiest thing to do was sit at the board, and plug into a Line 6 Bass Pod*xt*. Later on, I had all these Bass Pod tracks down, and I was too lazy to replay them through a proper amp. So, at the mixdown, I reamped the tracks simultaneously through an Ampeg SVT Classic, an Ashdown bass combo, one of the guitar heads with the Sunn cabinet, and a nice direct box. Just like the guitars, I wanted separate bass tracks available to blend or build tones.

Did you also lay down a modeled tone from the Pod?

No. I bypassed the sounds to get a clean signal. I basically used the Pod as a direct box.

What basses did you use?

I've got a really cool Fender Jazz Bass, as well as a Fender Precision, but I mostly used the Jazz.

How did you approach recording your vocals?

I pretty much went with the same signal path we've used on the System records. I'm comfortable with it, and I have the same setup in my home studio. I just dialed it in the same way, and went at it. The main mic is a tube Korby KAT47. It's a really good mic, and it's reliable. It doesn't change sound throughout the session like some old, vintage tube mics can do. From there, the signal goes to the Neve preamp and the Urei 1176 compressor.

Do you like to hit your voice hard going down, or compress it lightly, and then refine the compression during the mixdown?

I like putting a general level on it—not a lot of compression, just something to warm up the voice and keep the dynamics reasonable. But, sometimes, I'll sing a song section by section, and change the compression for each part, so that I get one type of sound on the intimate passages, and another type of sound for the heavier moments.

Do you like to sing with headphones on, or do you toss them and stand in front of some studio monitors?

I usually wear headphones and keep one ear off. I record vocals in the live studio, and I like hearing the track *and* how my voice sounds in the room.

What about the sweetening parts?

I wrote a lot of parts for cellos and violins using sampled strings. I recorded them myself, and then I brought in some string players—cellist Cameron Stone and violinist Antonio Pontarelli—to listen to the pieces and play the parts for real. The parts on the album are a combination of the live and sampled strings, because I wanted a big sound.

Did you have to notate the parts for the string players?

I wish I knew how to write on paper, but I don't. However, these guys were friends, as well as great musicians. One of them listened to my tracks, and wrote out his part in music notation, and the other just played his parts by ear.

How did you mic the strings?

My engineer did the miking for the drum and violin sessions, but I know we did stereo passes for the strings, and he used a number of different mics to get different vibes. We recorded the violins in the drum room at The Pass to capture some really nice ambience. It's a bigger room than I have. The cello was tracked in the live room of my studio.

As far as how you manipulate arrangements, are you someone who will edit right up to the very last possible moment, or, once you get the main arrangement down, is that how it stays until the final mix?

That's a good question. Generally, once I recorded all the instruments, I didn't change much, except for a bridge here and there. If a certain part of a song isn't strong, then you have to go in and dissect it so you can make it stronger—whether it's through changing the beat, rearranging the instruments, redoing the vocals, or inserting a whole new part. You've got to do that until you're ready to mix.

Did you have any kind of creative epiphanies or other inspiration that told you when a track was done—when there was nothing more you could, or should, do to it?

Well, I wasn't totally finished even when the mixes were done! It's a feel thing—definitely—but I also went back and forth with the mastering about five or six times [*laughs*]. I was frustrated, but I knew what I wanted. I just *knew*. Sometimes, it would be as little as cutting the bass frequencies just a tiny bit so that the drums would come alive in the chorus. You start with the general mastering balance, and you initially like what you hear, but then you start getting nit-picky about every little frequency, pan position, and instrument level [*laughs*].

I had a great mastering guy—Vlado Meller at Sony Music Studios in New York—but if you're going to produce a project yourself, you've got to be very, very critical throughout every step of the process. For mastering, you not only have to hire the best guy, but the guy who is right for the music you're recording. I didn't just pick a mix engineer or a mastering engineer, and go with it. I spent my time doing test mixes and mastering passes with people until I was sure I was working with someone who understood what I wanted, and who shared my vision for what the album should sound like.

Dead RECKONING

Going back to the mixdown process, how did you direct Neil Avron?

I'd listen to the mixes, and say, "Okay, this sounds great, but I think we should have the strings panned this way, and let's mute the bass in the breakdown." We'd make those adjustments, and then I might say, "I think these MuRF guitars in the verses could be louder. Let's build them up. Let's have those guitars ramp up into the chorus, because there's nothing else ramping up except the vocals." I'd make all these notes—the bass could be louder here, the bass should be softer here, the guitars have to be thicker, I need more balls [*laughs*].

Obviously, you've got a little more of a pulpit to stand on because of the success of System of a Down, but it's interesting that, just like a new artist, you still have to market this solo album, and figure out the best way to get it into the public consciousness.

Absolutely! Absolutely! But that pulpit I stand on—which you put so well—is a huge thing for me, because it gives me the comfort of being able to do things the way I want to do them without constantly worrying about how I can live off it. Thanks to the success of System, I don't have that fear of survival. It's harder for a brand new artist who is starting a career without money in his pocket. I realize that, and I'm very grateful. It gives me the strength and confidence to be able to do the right thing, and not compromise.

So, at the end of the line, was all the responsibility you undertook during the recording process an enjoyable experience?

It was awesome. It was a lot of work, but the experience I've gotten putting out records with major labels and for my own indie label has been invigorating and confidence building. I know what I want to present. For example, we made 12 videos—one for every song—and a crazy funny EPK [electronic press kit] that's on YouTube right now. I brought in all these amazing artists to do the artwork and the websites, and everything is done in house—which lets us be really creative for way less than what a major label would typically spend. I told the photographers, "You know those crazy ideas you had when you were first starting out? *That's* what we want." We even pick who we want to be interviewed by, and we ask them to answer a question about civilization that we'll compile and put on the website. So you and I are going to be doing an interview about the record, but then I'm going to be able to get your take on what you think about civilization.

For me, this is not just putting out a record—it's an experience that I want all my creative partners to have fun with. What can we learn from this? How can we put out a rock record in a different way? We're challenging the artistry in everyone around us, because this is an art industry. We've forgotten that, because major record labels are all about selling widgets. But this *is* about art and music and having fun, so let's enjoy it. EQ

SON OF H8000

The Eventide H7600 Ultra-Harmonizer® reviews are in...

"It simply sounds simply stunning, refined, glowing and always useful." ***George Shilling, Resolution***

"Apart from the dual-engine H8000FW, the H7600 is by far the best-sounding and most versatile professional multi-effects unit now available." ***Barry Rudolph, Mix***

"If I had the choice of just one processor unit to have with me in the studio, at this point, the Eventide H7600 would have to be that choice." ***Simon Tillbrook, Audio Media***

"If you haven't been out-of-the-box for a while, do yourself a favor and find your inspiration with a great outboard processor like the H7600...it will long outlast your next several digital workstations."
Adam Kagan, Tape OP

Eventide®

For more information call (201) 641-1200, email audio@eventide.com or visit www.eventide.com
Eventide and Harmonizer are registered trademarks of Eventide Inc. ©2007 Eventide Inc.

Soft Touch

by Craig Anderton

A Crash Course for Exploiting Virtual Instruments

Virtual instruments used to be pretty much gimmicks, with limited high-frequency response, grainy audio, aliasing, and the ability to suck the life out of a CPU. But today's cutting-edge software instruments—aided by improved computers—boast a level of sound quality and innovation that rivals the finest hardware equivalents.

But maybe progress has occurred a little *too* fast, because improvements have happened at a rate faster than some musicians and engineers can assimilate them. So, let's see how to make these babies really work for you.

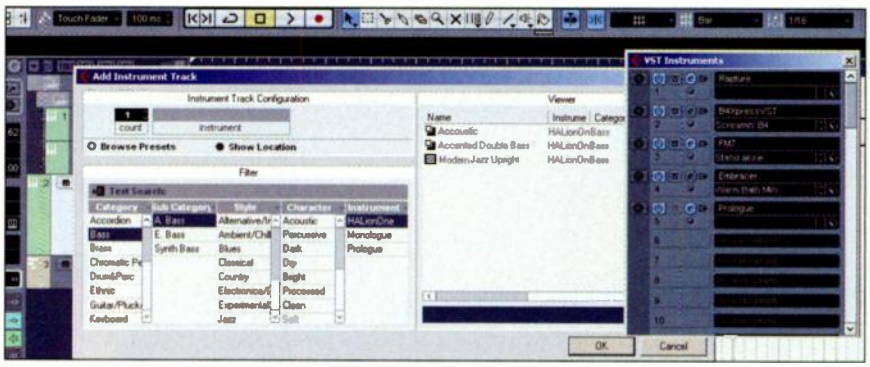


Fig. 2: With Cubase, you can browse presets, and narrow your search by browsing categories and other characteristics as you load an instrument.

INSERTING INSTRUMENTS

Over the years, the way of adding instruments to a project has changed. They were treated initially like signal processor plug-ins—you'd insert the instrument, then create a MIDI track for it. However, a more common approach now is to use one operation to create an Instrument track that deals with both audio and MIDI.

For example, in MOTU's Digital Performer, going *Project > Add Track > Instrument Track* (Figure 1) presents three options: Add an unassigned instrument track with accompanying MIDI track, add an instrument that will need to have a MIDI track added later on, or my preference, Add Instrument. When the Add Instrument Track window appears, you can specify how many Instrument tracks you want, the instrument to be assigned to the track(s), how many MIDI tracks you want to create to drive it (*e.g.*, you might want multiple tracks for a multi-timbral instrument), and whether you want all these tracks to be part of a folder.

Digidesign Pro Tools, Steinberg Cubase, Cakewalk Sonar, and several other programs take the same general approach for inserting a specific instrument track—usually with an accompanying MIDI track. However, all have some unique variations. For example, when you insert an Instrument track in Cubase, there's the option to browse patches, which uses a filtering process so you can seek a patch from a particular instrument category, musical style, plug-in, etc. (Figure 2). Or, you can go *Devices > VST Instruments* and when you choose a device, decide to create an accompanying MIDI track. It's also possible to call up presets from within this window.

Another insertion technique is drag-and-drop. With Ableton Live, dragging an instrument from the Plug-In Device Browser into the Clip/Device Drop area automatically creates an Instrument track that takes the instrument audio as an output, and provides MIDI as an input. Mackie's Tracktion works somewhat similarly. You drag a "filter" (which can be an instrument or audio processor) into a track. During the dragging process, you can specify which instrument to insert. Specify your MIDI controller as the track's input device, and you can record MIDI data into the track that triggers the soft synth.

Multi-timbral instruments may require special handling. In Tracktion, the instrument needs to be wrapped in a "virtual rack," and multiple outputs assigned. Power App Alley in the June 2007 issue of *EQ* describes how to handle multi-timbral instruments in Live. Some programs, like Sonar, let you specify during the insertion process whether you want a stereo mixed out, or tracks for multiple outs.

Fig. 1: Digital Performer already has the Modulo and BassLine instruments (included with the program) inserted. Another Instrument track is about to be created.

VIRTUAL INSTRUMENTS AS SIGNAL PROCESSORS

Some virtual instruments can insert as instruments *or* signal processors that utilize the instrument's filters, envelopes, and other modules. I first saw this with Native Instruments' B4, where you could use its excellent rotating speaker simulator with

Fig. 3: Arturia's Moog Modular V, inserted as a processor in Pro Tools.

any audio signal. Now several synths have opened their doors to external inputs. These include modular synthesizers, like those from SonicCore (formerly Creamware), Arturia (Figure 3), and Korg, as well as individual instruments (like several Minimoog emulations).

With software hosts, the instrument itself may show up on a list of virtual instruments, while the instrument effects show up in the list of audio processing plug-ins. However, there are exceptions. For example, with Sonar, if you insert an external input-friendly synth into a track's effects/instruments slot, the external input becomes available for any signal present on the associated track (see Power App Alley in the October 2006 *EQ* for how to use Sonar's Pentagon I synth as a vocoder with this technique).

Another example: With SonicCore's Minimax (Minimoog emulation), the audio input is a patch point within the Scope system (Figure 4). Thus, you can feed it from any output within Scope (including sequencer host tracks) and its output can be folded back into a host input, sent to the Scope mixer, or routed directly to an output.

Virtual modular synths make the best processors because of their flexibility and various modules. Of these, the MS-20 from Korg's Legacy Collection, Analog Edition 2007 (Figure 5) offers some extremely creative possibilities. Like the old Moog modular synths, it doesn't have a multimode filter. Instead, it has separate resonant highpass and lowpass filters, which when combined, can also give bandpass responses. But also remember that the original MS-20 was designed to handle external signals, and Korg's emulation respects that.

An important element is MS-20's the ability to condition the input signal via controls for level, low cutoff, high cutoff, control voltage adjust, and threshold. For example, you can use the filtering so that only low frequencies—like those from a kick drum—are recognized to provide a control signal.

As to the control process itself, there's a frequency-to-voltage converter, so that pitch can be used as a control element. You'll also find an envelope follower (great for patching into a resonant filter or amp) and trigger output, derived from the input signal, suitable for triggering either or both of the two envelope generators, providing a clock signal for the sample-and-hold module, etc.

Another great synth for effects is Arturia's ARP 2600V—but there are plenty of others.

HARDWARE MEETS SOFTWARE

Virtual Instruments want to be *played*, which doesn't just mean pushing down keys on a keyboard, but assigning parameters to MIDI, then using hardware controls (that generate MIDI controllers) to provide realtime control over those parameters. Being able to vary filter cutoff, envelope times, modulation, and other sonic qualities in real time is what makes synthesized parts come alive.

The typical way to create modulation assignments is with *matrix modulation*—so called because it presents a matrix of available modulation sources and destinations (Figure 6). I can't overemphasize how important this is to creating expressive sounds. Just using velocity to open up the filter a bit to increase brightness with louder sounds adds a significant degree of realism, as does using velocity to speed up the attack time of an amplitude envelope so that louder sounds attack more forcefully. And while mod wheels are typically mapped to vibrato, they are also good for morphing between different sounds, changing the timbre of bass patches, adding octave-above and octave-below layers, and the like.

Keyboard controllers will offer at least velocity, pitchbend, and mod wheel controllers (and, hopefully, aftertouch response). There will also likely be a jack for an expression pedal and sustain switch. However, sometimes you want more physical control options—if not to add expressiveness, then to be able to program sounds by tweaking parameters in real time, as you used to be able to do with analog synthesizers. Currently, there are four main protocols for doing this.

Fig. 4: The SonicCore Minimax is configured in Scope's routing window so that its audio input receives signal from an analog input (circled in red).

Soft Touch

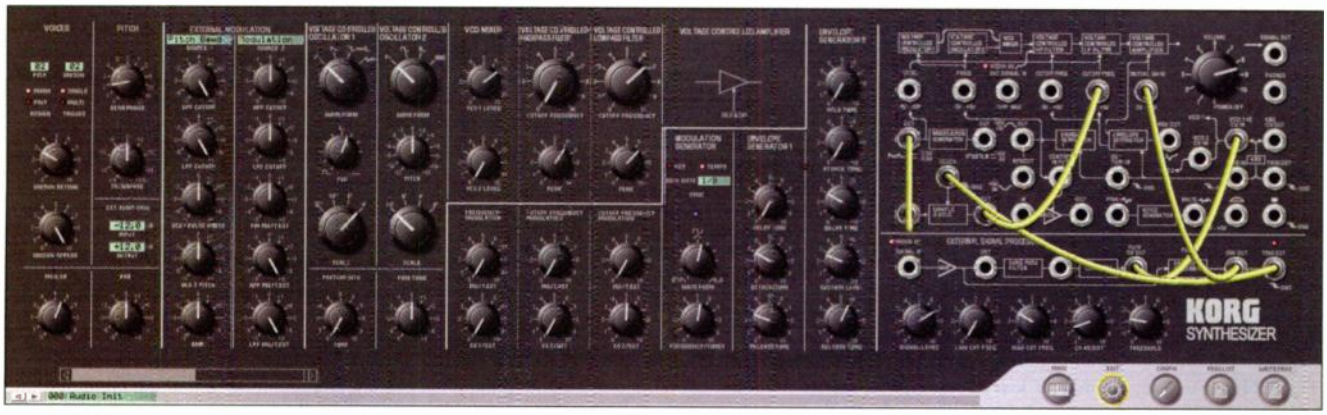


Fig. 5: Korg's MS-20 soft synth has been patched as an external processor for a drum loop. A trigger derived from the loop goes to the sample-and-hold (S/H) clock. The S/H input comes from an LFO, while the module's output controls the lowpass filter frequency. Meanwhile, the envelope follower output controls the VCA to let the signal through, and the Frequency to Voltage converter drives Oscillator 1 to add a sweeping oscillator effect along with the drums.

Native Instruments Kore. Kore (reviewed August 2006) is a hardware/software combination that incorporates an audio/MIDI computer USB interface with a control surface (Figure 7), so you don't have to rely on a mouse and keyboard. Kore's main element is the *KoreSound*—a "container" for VST/AU plug-ins you can save and recall—that can be as simple as a single instrument containing a preset, to a complex combination of instruments, effects, mixing, and routing, treated as a single entity. Kore itself can serve as a plug-in, and all the plug-in parameters within Kore are accessible through the controller.

The KoreSound is not tied to any particular host or program. You could use it with Logic on a Mac, then with Ableton Live on Windows. As long as the same VST plug-ins are installed on both Kore-friendly platforms, the KoreSound is transportable. Also, Kore itself includes many effects of its own. This increases transportability, as you can be sure any Kore system has these effects.

Another aspect of Kore is that it streamlines the process of finding presets because it "tags" presets with attributes, then assembles these in a database. All presets included in NI's Komplete soft synth suite have been tagged, as have some plug-ins from other companies; you can also do your own assignments. Ultimately, Kore's value is in direct proportion to your workflow's complexity. The more complex it is, the more Kore can help you work efficiently.

Active Controller Technology (ACT). This Sonar-specific protocol deals only with hardware control over synth parameters (as opposed to Kore's ability to work with Presets). ACT maps parameters found within Sonar—mixer faders, pan pots, effect settings, virtual instrument controls, and the like—to a hardware controller of your choice. Its biggest plus is context sensitivity, in that it instantly maps physical knobs to software parameters of the application having the focus. As long as a plug-in exposes automatable parameters, those parameters will be controllable via ACT.

For ACT to have the most positive effect, you need to customize it for your own needs. Simply using ACT to assign *every* single parameter to a control can get overwhelming. What works for me is to set up a consistent set of controls for the most-used parameters. For example, with almost all synths, I'm into tweaking filter cutoff and envelope attack, decay, and release times. Therefore, I've used ACT's learn function to assign these to the *same knobs* for all the soft synths I use.

Propellerheads ReMote. ReMote is a Reason-specific protocol for mapping controllers to Reason. Several manufacturers support this so that if you're using, for example, the Novation ReMOTE SL or Korg Kontrol49, you can select them as controllers within Reason, and their controls will be mapped to crucial Reason parameters. Bank switching allows directing the messages to particular instruments or functions within instruments, and best of all, ReMote accommodates multiple controllers. You can use a master keyboard controller for triggering notes and such, while using a fader box for dynamic, realtime control over particular parameters.

Novation Automap Universal. This protocol provides control for Novation controllers over VST Windows/AU Mac plug-ins via their Plug-In Automap, and a variety of host programs via Mixer Automap. These two Automap functions work from different templates within the controller. The templates load automatically, but can also be switched manually.

Plug-in Automap "wraps" plug-ins (as evidenced by a small control panel added to the plug-in with Automap functions), thus mapping their parameters to hardware controls on the controller. There's also a learn function for customization. Control is independent of the host sequencer, as it "talks" directly to the plug-in. (The Mixer Automap function also works from a template, which is different for each sequencer and therefore, not all sequencers are supported.)

Rolling Your Own. Finally, you can create your own mappings with your hardware controller of choice, and save presets corresponding to different instruments. For example, if you have a fader box, assign MIDI controllers to the faders, then assign the controllers to particular parameters within your plug-ins. However, compared to the protocols mentioned above, which do most of the mapping for you, the custom approach is far more labor-intensive.

TROUBLESHOOTING VIRTUAL INSTRUMENT PROBLEMS

Virtual instruments tend to be pretty foolproof, but there are some potential problems.

Not enough CPU power. Instruments place more of a load on CPU than just

"AT THE END OF THE DAY I NEED TOOLS I CAN RELY ON... FOR ME, THAT MEANS KRK."

"Over the past five years I have produced hits for Mary J. Blige, The Pussycat Dolls, The Black Eyed Peas, and Keyshia Cole listening exclusively to KRK's. KRK's are the benchmark for me. In production, in final mixes, in casual listening sessions: it all leads back to KRK's. I have brought them to mix rooms, mastering rooms and living rooms in order to establish that what I'm hearing – I'm hearing. KRK's are refined, pristeen, and analytical, while at the same time kick serious ass, impress the hell out of artists, dazzle the promotion staff and deliver an unforgettable sonic impression".

KRK SYSTEMS

888.361.5528 www.krksys.com

Soft Touch

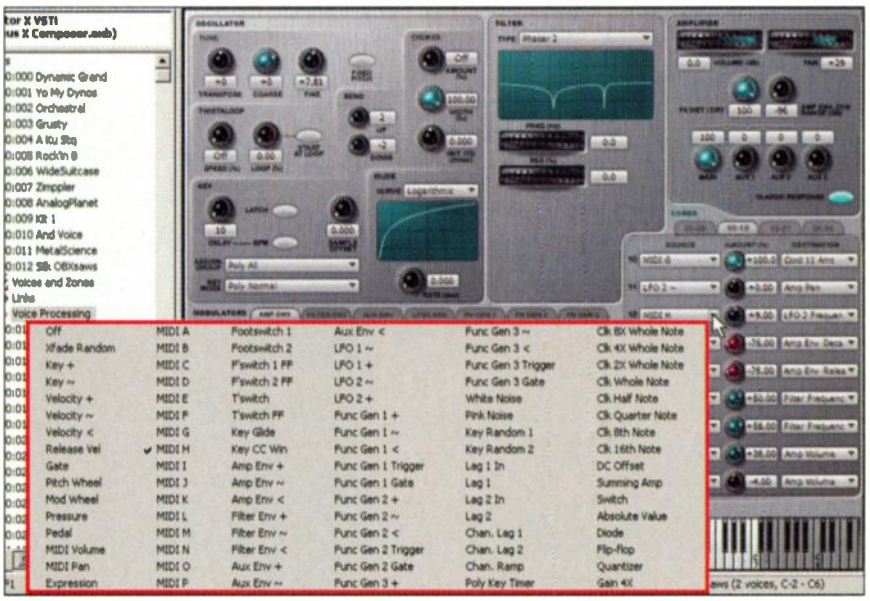


Fig. 6: E-Mu's Emulator X2 presents its modulation matrix as "software patch cords" that patch a modulation source to a modulation destination. This shows the list of modulation sources (outlined in red).

about any other element of a host program. The symptoms of an overloaded CPU include audio crackling or popping, sluggish operation, or the host program crashing upon inserting an instrument. Fortunately, there are several ways to minimize CPU power.

- If the instrument has multiple outputs, enable only the outputs you need.
- The instrument may have a "high quality" or similar mode that gives better audio quality at the expense of greater CPU consumption. Choose lower quality—at least while tracking (you can bump the quality back up when mixing).
- Increase the latency. Lower latencies tax the CPU more, leaving less processing headroom for handling virtual instruments.
- Use the host program's "freeze" function (if available). This renders the instrument's audio output as a hard disk audio track, then disconnects the instrument from the CPU. Note that bypassing or muting the instrument is not the same thing, as the instrument remains connected to the CPU so it can return instantly upon unmuting.
- If your host doesn't have a freeze function, save the virtual instrument patch you're using, bounce the instrument output to a hard disk track, then remove the instrument from your project. If you decide to edit the part later, re-insert the instrument and load the preset you saved. Do your editing, then bounce again.
- Minimize polyphony. Playing more notes requires more CPU power. If you have a long release time where multiple notes sound at the same time, consider using an external processor like reverb instead. Also, most synths let you restrict the polyphony. If you're lucky, this will be a "smart" algorithm that, for example, removes notes with the lowest volume and/or the same pitch if you exceed the available polyphony.
- Bypass unused elements within the instrument itself. For example, there may be several elements or voices, so turn off the ones you aren't using. You also may be able to turn off, say, the filter section independently from the rest of the plug-in.

Host program hangs while loading. Some programs scan plug-ins when they start up, or do a selective start-up where they scan only for plug-ins that have been added. If the program freezes during this process, that may indicate incompatibility with a particular plug-in. Programs often show the names of the plug-ins as they're being scanned, so you may be able to see what was scanned when the program froze. Remove the plug-in temporarily from the folder containing it, and try again. If the host starts without problems, you've isolated the culprit. Check the website for the host and the virtual instrument for updates,

Fig. 7: Kore's hardware controller features eight switches and touch-sensitive controls to provide standardized control over plug-ins. It also allows fast preset searches.

but also realize that some plug-ins just won't work with particular programs.

In addition, the program may scan a plug-in that requires some sort of attention, such as inserting a CD for verification. This warning may appear behind the host's "splash screen," and if you don't see it, you can't take the appropriate action. Move the splash screen if possible—or, with Windows machines, check the Taskbar for a tab that says something like "Error." Use Alt-Tab, or click on the Taskbar tab, to bring the warning window to the front.

General instability upon adding a plug-in. This can happen with shareware and freeware plug-ins that, while well intentioned, may not be coded carefully. If you add these types of plugs, add one at a time, and try it with all the hosts you plan to use before committing it to your system.

Instruments don't show as available. Most hosts have a preference that specifies the folder(s) where your plug-ins reside. Make sure that all folders containing plug-ins you want to use are included. For example, a host may create a folder for its own plug-ins and point to that, without pointing to a folder created by another program that also has plug-ins. It's generally best to consolidate all your plug-ins in a single folder, and point to that in each host's preferences. However, this can be problematic if a plug-in works with some programs, but not with others (either by poor coding or because a plug-in is keyed to a specific program), so you may have to create some custom folders that are pointed to by some, but not all, host programs.

Rewire doesn't work. The order in which you open programs matters. You generally need to open the ReWire client first, and then the host.

The instrument won't accept MIDI input, or deliver audio output. Remember that there are two parts to virtual instruments: the MIDI track providing it with data, and the audio output feeding into the host's mixer. Both aspects have to be functioning properly for the instrument to work.

An instrument loads, but its samples and patches are missing. When you install an instrument, its content goes to a particular folder and the instrument "knows" this file path. If the path changes (*e.g.*, you add a hard drive, or move the content folder), the instrument won't know where to find its content. There should be a preference or option for specifying this path. If all else fails, re-install the instrument with its content.

VIRTUAL INSTRUMENT FAVES AND RAVES

I've played with a lot of virtual instruments—enough to have some definite faves. So, with apologies to the many products I either haven't had time to evaluate, or don't have space to include, here's a look (in alphabetical order) at some instruments

Soft Touch

that have really spun my crank.

A.I.R. Instruments. Xpand, Strike, Velvet, and Structure are all superb. But Velvet stands out for not just emulating, but reproducing with uncanny accuracy, the sound of tine-based electric pianos.

Arturia. While famous for adding tasteful additions to classic instruments (Minimoog, CS-80, ARP 2600, Jupiter 8V, Prophet VS, etc.), to me, the company's *crème de la crème* is the Moog Modular V. The flexibility of a good modular synth is a joy for sound designers.

Cakewalk. Cakewalk has produced a slew of fine instruments since acquiring rgc:audio, but none as innovative as Rapture. It combines DSP, wavetables, and out-of-control modulation options to produce sounds that range from pure analog emulations to hardcore electro. (Give each of its six elements a track in Ableton Live—they're a perfect pair.)

E-mu. The competition in sampler-land is fierce—Steinberg's HALion, NI's Kontakt, MOTU's Mach5, and many others. But with beat slicing, off-the-wall DSP, and the brilliant "synth swipe" feature that transforms your favorite synth sounds into samples, it's worth dealing with the Emulator X2's sometimes dated interface to take advantage of its feature set.

Garritan. For coming as close as possible to a real violin or cello, the go-to products are Stradivari Solo Violin and the Gofriller Cello. They're amazing in the hands of someone who knows how to massage MIDI controllers.

GForce. In a world awash with Minimoog tributes, the Minimonsta not only nails that classic, rich Mini vibe, but incorporates some cool, extra tweaks.

Korg. The Legacy Analog Edition (MS-20, PolySix, and Mono/Poly) is a programmer's delight. But the Digital Edition, with the fabled Wavestation and M1, is something else. The M1 includes *all* the sounds from the original M1 soundcards, placing banks and banks of sounds at your fingertips.

Native Instruments. People have their favorites—Massive's, uh, massive sound; the clever, ambitious Kontakt; Absynth's sophistication; the B4 clonewheel organ; and, of course, Reaktor. But special props go to the FM8, which resurrects FM synthesis in a way that eclipses all hardware units, and makes the technology relevant again.

Propellerheads. Sure, Reason is an all-in-one studio, but thanks to ReWire, it brings a suite of fine instruments (two samplers, a drum machine, subtractive synth, modular synth, and granular synth) to any ReWire-compatible host.

Steinberg. Given that they invented Virtual Studio Technology, it's not surprising that Steinberg has a legacy of soft synths. But two stand out for innovation: Virtual Guitarist and Groove Agent—both of which can create convincing guitar and drum parts, respectively.

Sample libraries with front ends. Once upon a time, sample developers said, "Why are we developing different versions for E-mu, Akai, Roland, Ensoniq, Yamaha, and other samplers? Let's just make plug-ins!" These days, more and more companies are mating their sample libraries with player engines, resulting in a world of plug-ins that's way too big to summarize.

Spectrasonics put this approach on the map with Stylus (drums), Trilogy (bass), and Atmospheres (synth pads and more). It's debatable whether it was the concept or Eric Persing's brilliant sound design that got these products the attention they so richly deserved, but they remain coveted and viable.

Several of these products are software equivalents of workstations, and some stand out: Sonivox's Muse couples the engine from TASCAM's GigaSampler with a comprehensive sound library; IK Multimedia's SampleTank was one of the first software workstations, and its convenience, depth, and roster of effects have kept it relevant; and East West's Colossus is an expansive, rich environment for putting together complete compositions—think of it as General MIDI taken to the max.

String libraries have also taken this approach, from the ultra-high-end Vienna instruments to more budget approaches, such as IK Multimedia's Miroslav Philharmonik and Garritan's Personal Orchestra.

And we haven't even touched on Vir2's Acoustic Legends guitar library or their Drums Overkill, which wraps up just about every vintage drum machine sound into a convenient package. And there's the Yellow Tools series, and Rob Papen's Predator—a synth with a huge, but focused sound for dance productions. It's all good. . . .

DESKTOP AUDIO

THE PROFESSIONAL'S SOURCE

real world solutions from industry professionals!

www.bhproaudio.com

800-947-5518 | 420 Ninth Ave. New York 10001 | We Ship Worldwide

B&H PHOTO - VIDEO - PRO AUDIO

RICHIE KOTZEN ON RECORDING *GO FASTER*

by Michael Molenda

The title of Richie Kotzen's latest solo album, *Go Faster* (Headroom), is apt, because the former Mr. Big and Poison guitarist likes to move fast. Real fast. Whether his fingers are zipping down the fretboard of his signature Fender Telecaster, or whether he's whipping around his personal recording studio capturing guitar sounds, speed is always of the essence.

"I have an extreme lack of patience," admits Kotzen. "When I'm inspired, I don't want to screw around. I want to get my ideas recorded as quickly as possible. One reason why my studio is fully Pro Tools is because I can't even wait for a tape machine to rewind anymore. That would send me right through the roof. I actually sold my analog 24-track deck a long time ago."

Although the guitar tones on *Go Faster* range from absolutely raging to clean and chimey, Kotzen didn't employ a number of cagey microphone strategies, or fire up a battalion of amps. He simply dialed in his signature Cornford RK100 amp (mated with a Cornford 4x12 cabinet loaded with Celestion 30 speakers), and set up one microphone.

"I would say 97 percent of the guitars were recorded with a single Shure SM57 positioned right on the speaker," he says. "The signal chain is the SM57 into a Neve 1073 mic preamp and an Anthony DeMaria stereo compressor. The compression setting—and the EQ—depended on the part I was playing. Sometimes, I even went direct and just compressed the living hell out of the signal. The only other miking option I remember was when I played through my Mesa/Boogie Revolver rotating-speaker cabinet. Then, I just pulled one of the AKG C414s I was using as a room mic for the drums, and stuck it in front of the speaker.

"You see, I'm not sophisticated enough to deal with using multiple microphones and fancy mic positions. If I can just plop the SM57 in front of the speaker, and the guitar sounds awesome when I start playing, I stop right there. I've been on studio sessions where an engineer would place three or four microphones on a 4x12 cabinet, and then screw around for 20 minutes moving the faders around. After all that trouble, the guitar would sound good sometimes, and, other times, it would be the worst sound I ever heard in my life. That kind of hipped me to the fact that you may not need to spend a lot of time messing with microphones in order to get a good—or a bad—guitar sound."

Kotzen's casual approach to crafting guitar sounds obviously hasn't hindered his ability to deliver thrilling performances, or vary his tones according to what he is trying to communicate. (He also manipulates the Volume and Tone controls on his guitar to get different sounds, as well as creates diverse tonal impacts by softly caressing or viciously pummeling his strings—depending upon the mood he wants to convey.) The reason is that Kotzen believes the recording process is secondary to the demands of a performer's artistry.

"It seems to me that some people get so wrapped up in the intricacies of recording applications that they forget the object is to create music," says Kotzen. "You must evoke an emotional response when someone listens to the music you create. The fact of the matter is that you can achieve that response by playing a song on an acoustic guitar, and recording it on a 2-track. On some level, the recording process depends on what you're after, and how complicated the music is, but the end result should still be something affecting. If you get caught up in tweaking a piece of work that doesn't resonate as something true—or if the song is basically crap—then all the recording tools and applications in the world aren't going to mean squat. You need to ask yourself, 'Am I creating something that is even worth recording in the first place?'

"This is why I think technology is a double-edged sword. There was a time when only the artists who were making records were record makers. But now we've gotten to a point where *anybody* can make a record—even if they don't know how to play an instrument. If you don't know how to play a blues progression on the guitar, it doesn't matter. If you're computer savvy, you can make a record. In some ways, this is cool, because more creative ideas and different artistic statements are being generated. But, at the same time, you also get a vast community of people who can claim to be music makers, but who really *aren't* music makers." EQ

BLAST FROM THE PAST
LINK WRAY'S "RUMBLE"

by Michael Molenda

"Rumble," one of the most primitive and exciting guitar instrumentals ever laid to tape, didn't even have a name when it was recorded, and it almost wasn't released at all. The classic track was rescued from obscurity when the 17-year-old daughter of Cadence Records owner Archie Bleyer grabbed the acetate while looking for music to play at a birthday party. The kids flipped, Bleyer smelled a hit (the song reached #16 on the national charts in 1958), and his daughter ended up naming the song because it reminded her of the fight scene in *West Side Story*.

The late Link Wray made up "Rumble" on the spot at one of deejay Milt Grant's record hops after the dancers asked for a stroll.

"I didn't know a stroll, so I just started playing," said Wray, who launched into perhaps the nastiest two-chord intro in rock. "My brother took the vocal mic and put it in front of my amp, and it just distorted the heck out of the small P.A. speakers."

Impressed by the audience hysteria, Grant brought Wray and his band into U.S. Recorders in Washington, D.C.

"That place wasn't even a music studio—politicians went there to record speeches," remembered Wray. "The engineer had never recorded music before, so when he miked my brother's drums, he accidentally put the kick mic behind the drum, by Doug's foot. That's how we got that knocking bass drum sound."

Wray stood in front of the drums and pointed his Premier amp—with two 10" speakers, a 15" speaker, and built-in reverb and tremolo—towards a wall to minimize leakage. Standup bassist Shorty Horton stood to Wray's left, with a mic taped to his internal soundpost through a hole kicked in the instrument during a bar fight. Another Wray brother, Vernon, played acoustic guitar seated just behind the drums, and was tracked with a single boom mic.

"I tried to remember how the song sounded at the gig to make those kids scream," said Wray, who rumbled with a 1953 Les Paul. "Right away, I missed the distortion, so I compensated by punching holes in the amp's 10" speakers with a pencil. Then I put one mic on each of the distorting speakers and another mic on the clean 15".

"It took three takes to get the sound I wanted because everything was mixed down to a one-track Grundig. After the first take, I asked for the kick drum to be louder. The second take was okay, but I wanted to do another one. That third take sounded so good, I said, 'I ain't messing with it any more.'"

Grant paid $57 for the session, cut the deal with Bleyer, and reaped plentiful rewards.

"He stole everything," said Wray. "I was just a nobody." EQ

TIE A RIBBON. . .

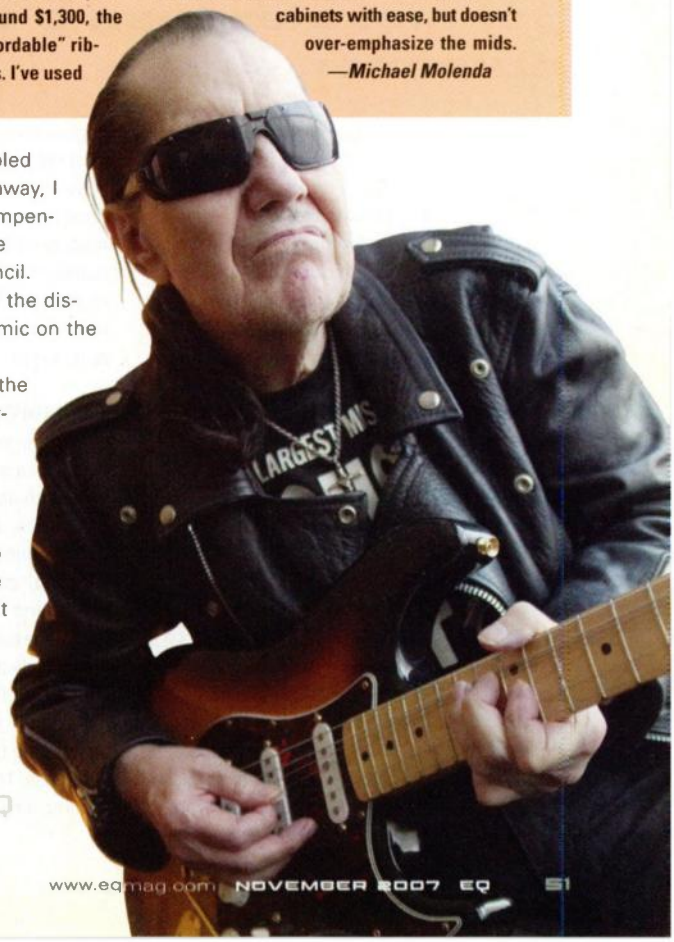
Many professional engineers laud the natural, organic sounds produced by ribbon microphones, and have been using these mics to great effect when recording electric or acoustic guitars. Ribbons have been around since the 1930s, and the thin ribbon that actually captures the signals used to be so delicate that a vocal plosive (such as a "pa" sound)—or even a sudden wind—would shred the sucker like a starved cheetah. These days, ribbons are much stronger, and they can easily stand up to blasting guitars, and even loud drums. The other good thing is that modern technology has reduced the price points of these once-expensive beauties so that the average home-studio owner can add a ribbon to his or her mic cabinet. Here are three models worth checking out.

ROYER LABS R121. While still expensive at a street price of around $1,300, the R121 was one of the first "affordable" ribbons to break into pro studios. I've used this mic on countless guitar sessions—both as a close mic and a room mic—and its natural, almost uncolored focus is fabulous for those times when a more organic, even "vintage" guitar sound is desired.

ELECTRO-HARMONIX EH-R1. The street price on this puppy is just $299, which makes it a pretty fabulous deal. I haven't used this mic personally, but I attended a session where the engineer was using it as the sole mic for an acoustic jumbo, and it captured the shimmer and low-end chunk of the guitar beautifully.

NADY RSM-2. Nady makes an entire line of affordable ribbons, but the $199 (street price) RSM-2 is pretty astounding. It's a versatile mic that is natural, yet very detailed when used to track acoustics, and it totally rocks on electrics. It nails the punch and push of ramaging 4x12 cabinets with ease, but doesn't over-emphasize the mids.

—Michael Molenda



3 PATHS TO BIG BOTTOMS

by Michael Molenda

Guitarists are infamous for overdubbing massive numbers of guitar tracks—typically recording combinations of different guitars, amps, speaker cabinets, and/or digital models—and mixing the stew into a gargantuan armada of 6-string power. Sometimes, the studio zealots outsmart themselves, and actually craft guitar layers that make the instrument sound *smaller*, due to issues with phasing, mismatched tones, and goopy signal processing. But the goal is to document tones that roar with impact and girth, and, for the most part, layering is a pretty successful way to get there.

Interestingly, this technique is not a path often traveled by bassists, even though stacking disparate bass sounds and kick drums have been components of dance and urban music for years. But why *wouldn't* you want to immerse yourself in the thunderous bliss of mixing and matching low-end tones? Put yourself into the headspace of the cat who does the subwoofer mixes for a big action film, and discover some new ways to rumble. Of course, you'll be producing musical parts, rather than cruise-missile explosions or death-ray drones, but putting some cinematic resonance into your bass lines couldn't hurt. Here are some ideas for beefing up your bass.

KEYBOARD KLONES

Doubling bass lines with a synth patch can add a mammoth wallop to the low end. Better yet, you can punch up or lay back on the rhythmic intensity by selecting a buzzy and edgy tone, a clean and transparent patch, or a droning sound. There's also a certain mechanical precision that gets transmitted when you're performing a bass line on keys, rather than busting it out on a fretted (or unfretted) electric bass guitar. The so-called typical method for blending the tonal layers is to feed the synth bass just under the electric bass. The idea is to let the original bass track provide the groove and movement, while the synth timbre provides a little more sizzle or boom. This is a very good and very safe mix option that preserves the original intent of the basic rhythm track.

A more "risky" option is to determine which track best serves the song—a move that might inspire you to blast the synth bass, and tuck the electric bass under it. Either way, make sure that the attacks—pick to bass string, and key-to-synth sound—line up exactly or your groove will get messier than a drunken drummer who keeps smacking his drumsticks together during fills.

Another advantage of sweating the time it will take to precisely match pick attacks with the keyboard performance is that you can do something crazy like panning the original bass hard left, and the synth bass hard right. There are certainly no rules that a bass—or basses—must be positioned in the dead center of the stereo mix, unless the final result will end up on vinyl. In fact, putting the kick drum dead center, and surrounding it with left/right bass lines often delivers a feral roar that could upend an armored truck at 100 yards.

Make sure the attacks line up exactly or your groove will get messier than a drunken drummer.

LAYING LOW

Of course, you don't need to double a bass part with a synth—you can simply layer the lows with another bass. This can be a bit dangerous, as, once again, you need to nail the overdub precisely so that you don't have "dueling" pick or finger attacks. (Guitar doubles sound kind of cool if they're a bit fuzzy—basses, not so much.) If you're timid, you can always just copy your original bass track to another track. Whichever path you choose, the goal is to utilize both tracks to construct a thick, punchy freight train of low-end mass. There are a number of ways to accomplish this feat. For example, you can treat the original bass track as if it were to be the *only* bass track. Make it sound fabulous. Then, depending on the needs of the song, EQ the overdub higher (boost the 3kHz range by 6dB) or lower (boost 80Hz or 100Hz by 6dB). I tend to compress the overdub pretty viciously to produce an unyielding dynamic pulse. Experiment leaving the overdub clean, or adding a bit of overdrive via a guitar plug-in such as Line 6's Amp Farm. Now, fade in the overdub ever so subtly until it provides the desired boom, drive, or snap. The cool thing is that you haven't compromised a well-recorded bass track, you've simply added a new dimension. You should also try cutting all the lows from one bass track and boosting the attack, and then adding some subsonic lows to the other. Then, mix the two tracks fairly even, and perhaps experiment with some slight panning. An expansive, animated bass line might be just the thing to really energize your mix.

OCT-A-VATE

An easy way to layer bass lines without risking phasing problems or performance stutters is to route the original signal through an octave processor such as a Boss OC-3 Super Octave or an Electro-Harmonix Micro Pog. Various multi-effects boxes and plug-ins include octave functions, as well. In this instance, you simply play your bass line, and adjust the unit to generate a higher or lower octave note (depending on the parameters of the processor and whichever direction makes sense for the groove of the song). Obviously, you can also apply this processing to an already recorded bass line, and bounce the effect to another track in order to blend in the octave. However you apply the octave effect, it will add height and/or weight to a conventional bass part.

BADA-BOOM

There's no reason to fear radical (or slightly edgy) bass enhancements. For one thing, you can always default to the ho-hum ordinary. But crafting some supersonic lows or killer punches should illuminate just what you *can* do to rock a groove. And exhausting all options to make a mix more exciting is exactly what savvy home-studio cats should commit to. EQ

THE STANLEY CLARKE METHOD

by Brian Tarquin

In the 1970s, Stanley Clarke redefined and reshaped the way bass players approached their instruments. He took Larry Graham's slap-funk technique, and added ferocious dexterity and a melodic sensibility that pushed the bass to new musical frontiers. He also invented the piccolo bass and the tenor bass.

Clarke has a home studio where he does most of his projects, including his solo albums, and scores for films such as *Boys N the Hood*, *What's Love Got to Do With It*, and *Poetic Justice*. When tracking bass, he typically uses his famous Alembic basses and Alembic F2 preamps, and plugs into two SWR amp heads and two SWR cabinets—one a 2x15, and the other a 4x10. He then mics each cabinet separately, and also takes a direct signal from the amps, which gives him lots of flexibility for blending tones during the mixdown. However, when the project calls for a really huge bass sound, Clarke will book a commercial facility in order to record in a big room. In those situations, he likes to mic the cabinets and capture the room ambience to craft the final bass sound.

The Alembic boasts a pretty wide dynamic range, and, wherever Clarke records, he employs a good limiter to control the bass sound. He's a fan of the hardware Fairchild limiter for these applications, and he feels the Universal Audio Fairchild plug-in sounds "95 percent like the original." EQ

SPLITTING SIGNALS

Part of the recipe for Stanley Clarke's mammoth bass sound is recording direct and miked signals, and then blending those separate tracks to taste during the mixdown. This is a really easy operation, and all you need is an amp and a direct box. For example, when I record bass, I use a Radial J48 Direct Box as a buffer to split the signal. I plug the amp into the Radial's Thru jack, and plug the bass into the input. Then, I take the XLR signal directly to one track, and print the miked signal—which is typically a AKG D112 positioned near the speaker bottom, and run through a tube mic preamp—on another track. This enables me to blend the direct signal a little lower in the mix than the live signal, which adds full body to the bass sound.
—Brian Tarquin

Clarke (far left) with Marcus Miller and Victor Wooten at Bass Player LIVE! in New York City, October 2006.

Photo by PatrickWong

4 WAYS TO BLEND YOUR SYNTH WITH THE BAND

and turn it down for sustained sounds. Here's why: More diffusion adds more reflections, creating a lush type of sound. Less diffusion means fewer reflections, giving more of a discrete echo sound. Discrete echoes added to percussive sounds can give the "steel balls bouncing on a marble plate" type of reverb effect, because you can hear individual echoes from the percussive transients. More diffusion smoothes out these echoes.

Conversely, with sustained sounds such as pads and organs, lots of diffusion piles a rich, full reverb sound on top of a rich, full synth sound. This can take away from the synth's sense of definition. Lowering the amount of diffusion creates a more sparing reverb sound that doesn't step on the main sound.

PLAYING THE SPREAD

One of the characteristics of a real keyboard such as a piano is a definite sense of left/right placement. From the performer's position, for example, bass tones appear predominantly at the left, and treble to the right. While that's not necessarily an inherent part of a synth's sound, it is possible to create a stereo perspective through proper programming.

One way to do this is to use keyboard note position to modulate panning. This puts bass notes toward the left, and higher notes toward the right. In some cases, this may be ideal, as the synth bass frequencies are out of the way of the kick drum and bass part, which are usually centered in the stereo field. With some mixes, though, having bass coming out of the left speaker could unbalance the mix. A simple solution is to avoid spreading the keyboard from left to right, but, instead, spread its image from center to right. You do this by panning the left output to center, and the right output full right (or close to full right). This works particularly well if you have a complementary instrument (such as rhythm guitar with some degree of stereo ambience) that can spread from left to center, thus balancing the keyboard.

I've also had good results by using keyboard split techniques to place the keyboard's bass in the center, with the midrange and upper notes panning from left to right (Figure 2). This takes advantage of the fact that many soft synths let you split the keyboard at arbitrary places. So, create two splits, one containing the lower two octaves, and the other containing the remaining octaves. Pan the split with the lower octaves to the middle, and modulate panning with note position for the other split. This way, the midrange and treble notes will pan from left to right, while the bass comes out the middle.

However, to avoid an overly artificial sound, this works best with keyboard parts that have a fair amount of reverb. The ambience tends to make the stereo placement more diffused so that the notes are perceived as more of a whole instrument, rather than being "points" of sound. EQ

Professional Audio Education Since 1976.
Industry Standard Equipment.
International Recognition.
Individual Studio Time.

Current U.S. Locations:
NEW YORK 212.944.9121
ATLANTA 404.526.9366
LOS ANGELES 323.466.6323
MIAMI 305.944.7494
NASHVILLE 615.244.5848
Plus a further 40 locations around the world!

If you're considering training to become an audio engineer or producer, you know that you need hands-on time in the studio, running your own sessions and working with artists – that's exactly what you get as a student at SAE Institute. Call us today to schedule a private tour of our studio facilities.

SAE INSTITUTE

For more information about jump starting your career call 1-877-27 AUDIO or go to www.sae.edu

SIX STEPS TO A SUPERIOR SNARE SOUND

by Matt Harper

In theory, getting a good snare sound is pretty simple—you just put a hypercardioid dynamic mic up on the top of the drum and you're good to go, right? Well, yes . . . and no. While you do yourself no disservice by first grabbing your trusty SM57, mic placement can be quite tricky when you're looking for maximum drum isolation *and* an awesome sound.

PUT IT WHERE IT BELONGS

First things first: You should *always* rely on room mics and overheads to capture ambience—not individual mics devoted to individual kit components. Given this, I recommend getting approximately 80 percent of your sound dialed in with your overheads and room mics, then bringing your close mics (*i.e.*, kick, snare, tom) in during the mix to "help" your overall drum sound, as opposed to simply trying to build your mix one drum at a time.

After dialing in your overheads and room mics, get to work on your close snare mic. Most engineers will attest that a good first approximation for a snare mic is as follows: Place the mic at the "audience side" of the drum (about 10 o'clock) at a 45 degree angle, pointed as far away from the hi-hat as possible, with the capsule roughly an inch and a half above the head. The angle of the mic and the space between the mic and the head is variable, depending on whether or not you want a more "roomy" sound (in that case, try a 60 degree angle to get more of the area surrounding the drum) or more low end (the closer the mic sits to the drum, the greater the low end). Moving the mic more toward the rim of the drum will give more "ring," which can be beneficial if you play a lot of rim shots in a performance and you really want that snare "overtone" to be present in your tracks. And if you want a snare sound with tons of attack, make sure the mic points to the area of the head that the stick strikes most often (*i.e.*, the sweet spot).

It's not advisable to position the mic at an angle greater than 60 degrees, as a deep angle is key to avoiding pesky bleed from other parts of the kit. Also, don't bring the mic capsule up beyond just a few inches from the head, as you'll run a greater risk of experiencing phase issues.

Fig. 1: A Slingerland snare's "under" mic. This is thrown 180 degrees out of phase compared to the "over" mic, which will be set up next. (Courtesy Brian Hardgroove, Public Enemy/EV2.)

TAKE TWO

If you're not digging the sound you get from one single snare mic, you can always add in another to help get a richer sound. One popular method often used on some of your favorite recordings is what is known as the "over-and-under" technique (Figure 1). The moniker assigned to this technique is self-explanatory: You put one mic on top of the snare, and another mic underneath. Why? Because miking the bottom head of the drum will result in more snare rattle, which can be a cool effect, and will also give your track additional "snap."

When applying the "under" mic, make sure to place it in a similar manner to your top mic (same relative angle and spacing of capsule to head). You must also flip the "under" mic out of phase compared to your "over" mic, lest you end up with a thin sound. Furthermore, as the purpose of the bottom mic is to get a different sound than what the top mic produces, consider using a mic that is very nuanced in the high end (such as an AKG 451) to get a lot of "thwack" and rattle.

A condenser that can handle high SPLs [Sound Pressure Levels] can be really cool as an "under" mic as well, especially if you're going for a real jazzy sound. If you opt for a condenser, use one with a hypercardioid pattern so you're not getting any bleed from your inside kick head (I've found the Audio-Technica 4053a and Neumann KM185 both work really well in this application). And keep your preamp

Fig. 2: Digital Performer's EQ set for a snare drum. Note the steep, low-cut filter below 20Hz, which helps get rid of unwanted subsonics. The high frequency shelf (colored green) adds a little "air."

gain down, cowboy. You don't want your signal to be too hot and distorted.

KEEP 'EM SEPARATED

Always assign each snare mic to its own channel so that you have the maximum amount of flexibility during mixdown; you'll want to be able to manipulate each individual track without compromising the room; it's the combination of the snare with the signals of the overhead mics that give the drum mix believability.

COMPRESSION TIME

If your levels are still consistently unruly, it's compression time. A little will do you most of the time—you don't want to crush the snare sound. Set your ratio at 2:1 or 3:1 with a high threshold. Doing this should quell your "excited hits" without making your snare sound lifeless.

Compression isn't just for taming peaks and keeping drummers that forgot their Ritalin in check.

But compression isn't just for taming peaks and keeping drummers that forgot their Ritalin in check. You can also greatly affect the envelope, or shape, of the snare sound in cool ways by creatively applying compression. Setting the ratio high (4:1 is recommended) with a low threshold and a medium attack will help give your snare sound more punch in a busy mix. And cranking the release will keep your snare loud as the signal decays, which can be flattering to machine gun-esque snare lines (death metal Hessians take note).

BOOKING THEM BANDS

You'll probably need to EQ your snare a bit, as your rough track usually won't sit perfectly in your mix (Figure 2). Boosting a bit in the 100–200Hz range is recommended if there aren't enough lows to your snare sound. Conversely, cutting in that range will help combat any excess lows you may have in your track due to proximity effect (some of the mud lives around 50Hz, so be sure to cut from that area as well if your snare is sounding messy). Pulling out some of the lower midrange frequencies (between 300–700Hz) will usually help combat "boxy-ness." Add just a bit in the upper-mids (between 3–7kHz) to add some extra attack . . . and give a nudge upwards between 7–10kHz if you want to accentuate the highs with a bit of "air," without bringing in the "Rosie O'Donnell effect" of constantly sounding harsh and annoying.

'VERB IS THE WORD

Ah, reverb—the icing on the cake. Reverb and snare go together like me and scotch on a Sunday afternoon—meaning that it can be a wonderful thing, but only if not overdone. Too much . . . and you're looking at jail time.

Well, maybe not jail time per se, but a sloppy time nonetheless. While you can get away with adding more reverb to a snare (the drum doesn't have any serious lows to exaggerate), it's still best to go light and easy on the effects if your goal is to keep your track sounding natural.

Plate reverbs (and emulations) are my personal faves, as they are fast, bright, and don't produce a bunch of early reflections. Fast reverbs tend to lend themselves well to snare tracks, except when used in songs that are really spacious and slow. So if you're not performing a John Cage cover, keep the attack fast and the decay short and sweet as well, following the standard reverb philosophy that the faster the track, the shorter the 'verbs (*i.e.*, around 0.5 seconds).

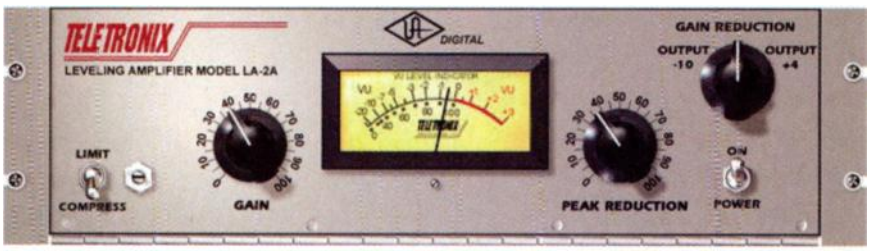
TAMING EXTREME VOCAL DYNAMICS

by Kris Force

When it comes to vocal performances that range from intimate whispering to meter-slamming screaming in the space of just a few measures, we should consider ourselves blessed that we're galaxies beyond yesteryear's technical inadequacies that had us manually riding faders. The tools for solving vocal dynamic problems have truly come a long way. Hardware and software compressors, limiters, expanders, and gates are more versatile (and cheaper) than what we once would have ever dreamt them to be. But it's in honing technique that the experts are separated from the novices, and employing factory presets in these modern tools isn't always enough to get a dynamic vocal mix.

When managing vocal extremes, one can always count on a compressor and/or limiter being an important tool of the trade. Although one of the most common audio processes, compression/limiting is often one of the most misunderstood. Many first-time users don't fully understand how to deploy compression so that it doesn't sound as if it's being used. As a result, they shoot to normalize a vocal part, and end up just crushing the dynamics to dust.

To start from the very beginning, a compressor/limiter is essentially an automatic volume control that consists of four basic parameters: ratio, threshold, attack, and release. Ratio narrows the dynamic range of the performance by reducing the gain once it reaches a predetermined signal threshold. A 3:1 ratio, for example, works like this: When the input signal increases in gain by 3dB, the output from the compressor will increase by only 1dB.



Many products, both hardware and software, pull double duties as compressors/limiters. Pictured are UA's software versions of the famed 1176 and LA-2A—two of the most highly rated compressor/limiters of all time.

This reduction of the output peak volume thereby "compresses" the overall dynamic range of the performance.

Threshold determines at which dB level the compressor engages. Any basic compressor will have what are called hard knee and soft knee threshold settings. Hard-knee compression is when the amplitude of the input signal is a straight line until it reaches the dialed-in threshold setting, at which point the "angle" changes abruptly. The higher the ratio is set, the harder the angle. Conversely, a soft-knee setting allows a wide progressive adjustment as the threshold is approached, therefore creating a curve correction rather than a hard angle. As you can imagine, the soft knee offers a smoother transition, which, depending on the situation, may be preferred. Attack determines the speed at which the compression is applied after the threshold has been reached. Release controls the time the compressor takes to return to pre-threshold settings.

Take note that a short attack, quick release, high ratio, and hard-knee threshold can create an undesirable pumping and breathing effect. Setting your release time to longer than the source signal's natural decay will prevent pumping, and smooth out the tone, making the vocal seem more sustained. The only potential problem with this is if your vocal track has a lot of noise floor, as the noise will be sustained, as well.

There are three common places to apply compression: in the record path, as an insert on a mix channel, and on an output bus. During recording, compression applied to a voice is often positioned in the signal chain directly after the mic preamp. When the compressor is on, the output

There are plenty of options for users who want an all-in-one dynamic processing solution. The Focusrite Compounder is one of many units on the market that works as a combination compressor/limiter/gate/expander.

peaks will be reduced, but the output level, including noise, is increased. Using a compressor first in the chain gives other effects a more consistent signal to process, so it's good practice to place the compressor directly after the mic preamp.

Stereo compression is often used on the output bus as a final process to increase the apparent loudness and presence of an overall mix. Multiple passes of compression in low ratios (such as at the record stage, and later on during the mix) will result in an overall smoothing quality, and will reduce the potential for undesirable byproducts (such as those pumping and breathing anomalies).

Multiband compressors act the same as non-multiband compressors, but instead adjust volumes to both ratio settings *and* predetermined frequency bands. For this reason, multiband compressors are particularly useful when processing the human voice, as it's rare to encounter a human voice that results in a perfectly flat frequency curve. While multiband compressors have historically been regarded as primarily broadcast and mastering tools, their inclusion in many DSP packages have led to frequent use at the mixing stage. A de-esser—which can reduce undesirable sibilance (most notably on "s," "sh," and "zh" sounds) in the 2kHz to 10kHz range—is a commonly recognized example of a multiband compressor. This tool can be especially helpful in keeping harsh high-end sounds at bay (even from cymbals, percussion, and fuzzed-out guitar tracks).

With an incredibly loud vocal performance, compression alone may not be sufficient, and limiting may be needed to help tame the meters. Compressors and limiters behave identically, but limiters typically handle compression ratios of 10:1 and upwards. Limiting is frequently used as a failsafe with a "do not pass" threshold level—with the processor often inserted in the bus output in environments with legal decibel limitations, or to protect live sound gear—as nothing is more damaging to a performance than a huge moment of overloading distortion.

However, the outcome of *over*-limiting can equally be crude and jarring, as a sudden gain reduction disrupts the listener's experience. Extreme limiting also affects the point of perspective, because a sudden reduction results in the source being perceived as being further away spatially. One can certainly argue that the sound of limiting is better than the sound of overload distortion, but, even so, use limiters with extreme caution, and apply them moderately—*unless* you want to squash the life out of your track.

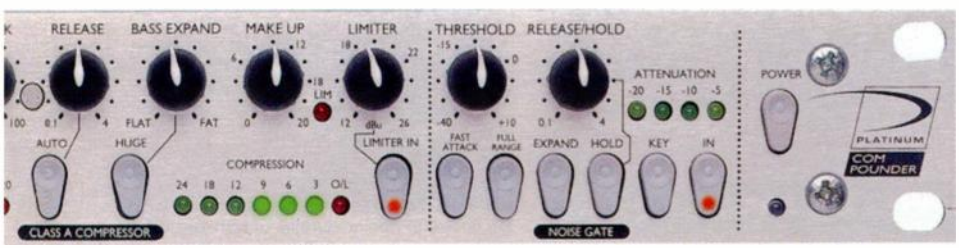
Noise gates and expanders conduct an inverse process of compressing and limiting—they increase dynamic range by attenuating signals that fall below a user-defined threshold. A noise gate, for example, lets through only those signals that are above the gate's threshold. This is great for ensuring that breaths and background noise don't compromise a vocal performance. However, if you set the threshold too high, the gate might also clamp down part of the vocal, causing chopped syllables or weird stutters. Plosives (boomy pops caused by "p" and "b" sounds) can also be battled with a gate. If you're dealing with a plosive, set the attack of the gate moderately fast, so that it attenuates the sound, but doesn't "click" when opening. Try 1ms as a good default setting, as it will preserve intelligibility of the words being sung without making the sound to choppy. Make sure to keep the decay time relatively long (approximately 0.5 seconds), so that you don't snip off the end of any words, and keep the threshold set as low as you possibly can, so as to keep as much of the dynamic performance intact (although unwanted noise levels shouldn't be able to open the gate).

While a noise gate attenuates signals swift as a guillotine blade, an expander offers a *gradual* attenuation. This is a good tool to use when a vocalist takes breaths, say, in the middle of a silent passage (as opposed to just before a lyric line, where using a gate would make more sense), or if noise is apparent when the singer isn't singing (such as during a guitar solo).

Perfecting dynamic range is a matter of understanding your source signal, and working the parameters akin to a sculptor using clay. There are no hard and fast rules. When choosing your tools and planning out your mixing strategy, you can't just look at the vocal performance itself—you must first look at the whole picture. And you can't simply watch your dials and meters when applying any of these tools. Before touching a single button or cranking any knobs, you must ask yourself: Is the voice one part of a dense musical arrangement, or a delicate and exposed soundscape? What is the dynamic range of the vocal performance *vis-à-vis* the dynamic range of the surrounding instrumentation? And then listen, tweak, wash, and repeat. EQ

STEREO MIKING FOR A DYNAMIC VOCAL PERFORMANCE

If you have found your vocal dynamic range has been extreme in the past, and that limiting, while necessary, has altered its natural sense of space in the mix, it may be wise to track your vocals with two matching mics in a stereo pattern. To do this, place both mics equidistant from the source—your mouth—and apply a high-threshold compression setting to mic one, and a noise gate to mic two. One microphone will take off where the other has left off, yet both will have manageable dynamic ranges. Make sure each signal is panned in the same pattern, and match everything in both signal paths to preserve uniformity of your sound. —*Kris Force*



WHAT YOU NEED TO KNOW ABOUT DIGITAL METERING

by Craig Anderton

Meters are your visual cues about what's happening with the audio levels. Are they too low? Too loud? Causing distortion—even distortion you can't hear? The meters will let you know. But they can also tell you a whole lot more than you might realize.

The digital meters in today's DAWs, while arguably not as cool-looking as illuminated analog meters, are a vast improvement in terms of accuracy—although they're still not perfect, as we'll discuss later. And they also provide a lot more options, such as . . .

ADJUSTABLE RESOLUTION

Being able to change meter resolution can be very handy. For example, when tracking, you might want to cover as wide a range as possible (*e.g.*, 90dB so you can see any little glitches or noises). On the other hand, when mixing, you might decide to go for 12dB or thereabouts on the output bus. This lets you monitor the all-important peak values, and gauge the approximate amount of loudness maximization that may be required. For example, if the output meters make it to 0 frequently, but otherwise spend very little time at 12dB, then the song will probably need to be made hotter during the mastering process.

PLACEMENT AND SIZING

Some programs let you change the meter size, as well as choose whether the meter is horizontal or vertical. Generally, the meter will be longer and easier to read in a horizontal view—and offer higher resolution, as well—but it will take up more screen space. With a vertical view, you can try making the meter short enough that it serves more as an activity/clip indicator. This saves a lot of screen space, and, in many cases, tells you all you really need to know (Figure 1).

Note that meters do use up CPU power (albeit not much), so if you really need that last ounce of performance, disable any meters you don't need. This is because they're part of the GUI, and need to be updated frequently. With some programs, you can change the refresh rate so that the meters place less of a strain on the CPU when set to a slow refresh.

WHERE DO THEY GO IN THE SIGNAL PATH?

If you can choose whether playback meters are pre- or post-fader, and whether bus meters are pre-fader, post-fader, or pre-fader and post-FX, so much the better. With a mixer screen, I tend to choose post-fader metering, so it's easy to see at a glance which tracks are contributing the most amount of level. But in a track view, as used when recording, a pre-fader setting lets you monitor track activity so you can check whether a track has signal, regardless of the fader position.

RESPONSE CHARACTERISTICS

A typical choice of responses includes peak, RMS, or peak+RMS response. Ideally, you'd want to be able to set these characteristics independently for the record, playback, and bus meters.

Peak readings are most useful while recording, so that you can determine if any signal levels are loud enough to cause overload problems. Most meters can also hold any peak overload conditions, so even if your eyes are away from the meter when it happens, you can still know that 0 was exceeded.

RMS provides an average reading, and can be helpful while mixing if you want to see the average amount of energy a track contributes to a song. For example, a

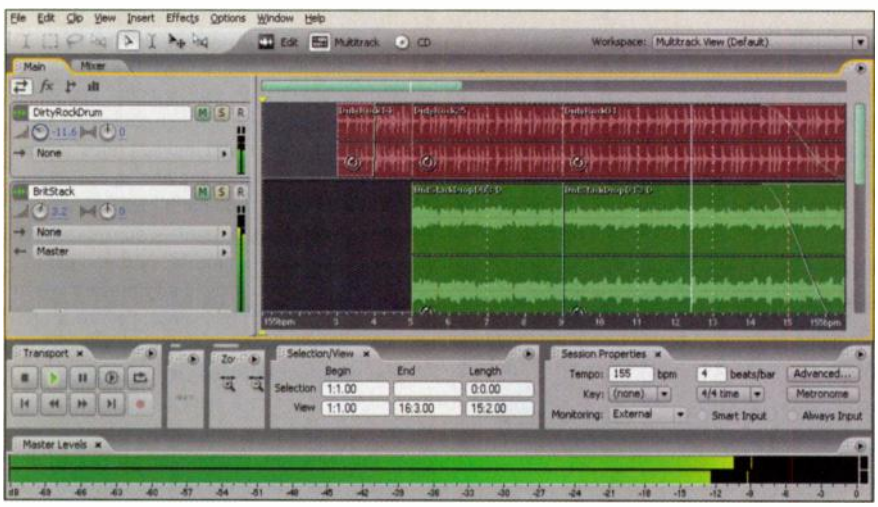
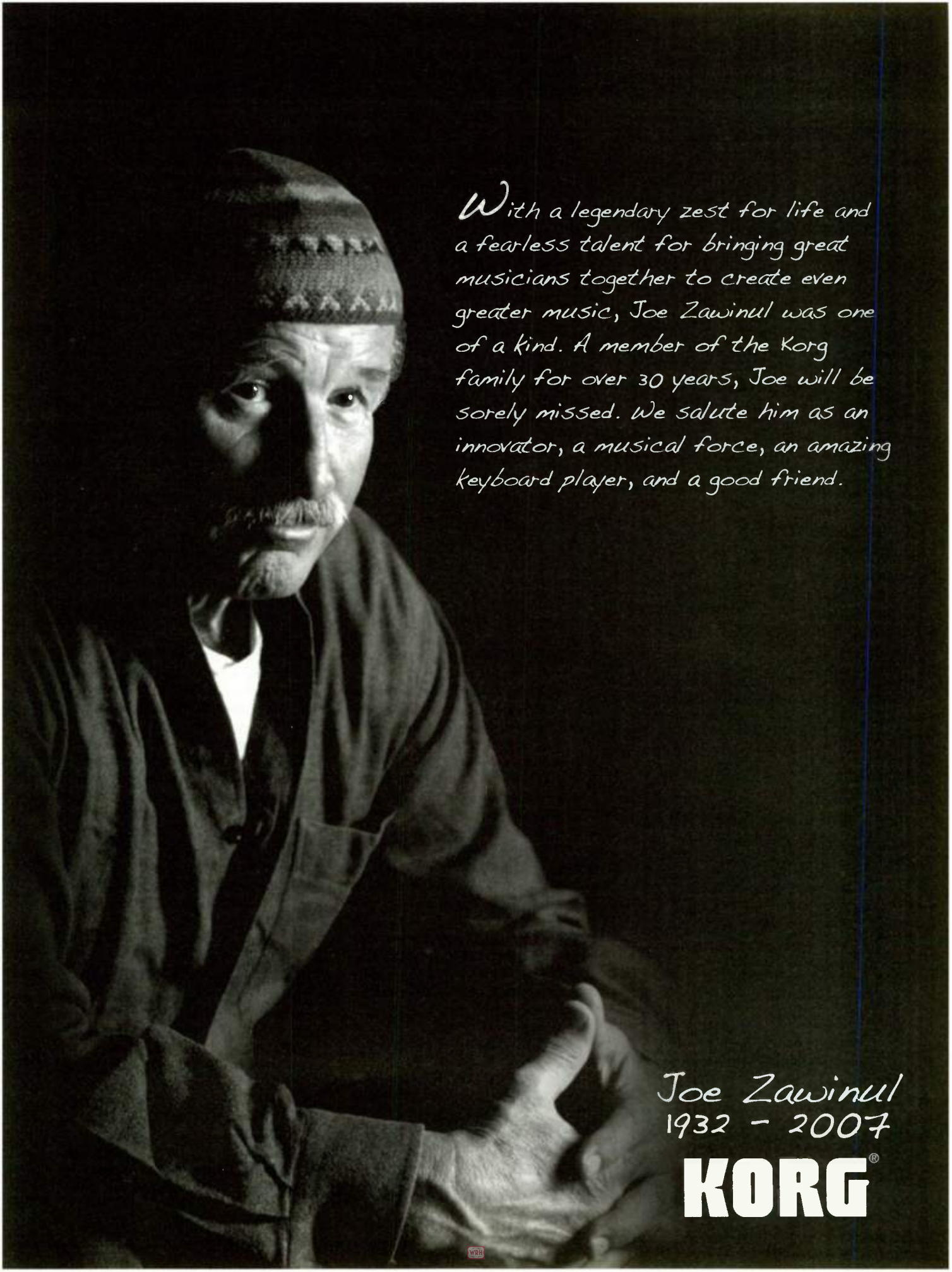
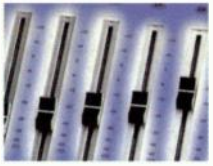


Fig. 1: Audition's Multitrack View has meters for each track, whose height varies depending on the track height. But there's also a very high-resolution bus meter toward the bottom of the screen.

Fig. 2: Sonar offers a choice of horizontal or vertical meters in track view, as well as several options for tailoring the response, placement in the signal chain, and range.


With a legendary zest for life and a fearless talent for bringing great musicians together to create even greater music, Joe Zawinul was one of a kind. A member of the Korg family for over 30 years, Joe will be sorely missed. We salute him as an innovator, a musical force, an amazing keyboard player, and a good friend.
Joe Zawinul
1932 - 2007
KORG®



MIX BUS

WHAT YOU NEED TO KNOW ABOUT DIGITAL METERING

THE EARS HAVE IT

Craig Anderton warns against obsessively locking your eyes to those meter levels, and it seems no less a studio legend than '50s and '60s Columbia Records A&R honcho Mitch Miller (yeah, the Sing Along With Mitch cat) would strongly agree. In an apocryphal tale about a long-forgotten '60s session, Miller is said to have been listening to a mix—eyes closed, arms crossed, lost in his evaluative trance.

"Turn up the horn section," said Miller to the engineer.

"I can't do that," Mr. Miller," answered the console jockey. "The levels are already pegging the meters. See?"

Miller opened his eyes, surveyed the board, and walked out of the control room without saying a word. The next day, every single meter in the studio was hidden under several layers of gaffer's tape. The record man had made his point. His ears were to be obeyed.

—Michael Molenda

percussive sound will typically have a high peak value and a low average value, whereas distorted guitar will have a high average value that almost equals the peaks. In this kind of situation, unless you want to squash lots of dynamics, the peak vs. RMS indications give you a clue about setting the proper levels. You'll want the guitar, with its high average value, lower than the percussive sound to retain a good balance.

My favorite option, though, is peak+RMS response. In the credit-where-credit-is-due department, the first time I saw this was with Peavey mixer LED meter hardware. The average level was lit solid, and the peak was a single LED, tracking peak levels and "floating" above the average reading. What makes this so useful is that you can get a sense of the relationship between the peak and RMS levels, making it easier to determine if a signal needs limiting, or whether it would be better served by compression.

USING METERS TO CHECK FOR LOOP DISTORTION

Meters aren't just for measuring record and playback levels—they can provide valuable insights into the behavior of all audio signals.

For example, many programs these days can handle "Acidized" loops, which include metadata for stretching characteristics so that they can match different arbitrary tempos or keys. However, with these programs, I've noticed a strange problem where even though the Acidized clips were normalized to something less than 0 (*e.g.*, –0.1 dB), the track meters sometimes indicated overload. When I played back the samples normally, they clearly didn't distort. But as I continued to monitor the meters, I would notice occasional flashes of distortion.

After putting in a lot of hours editing a loop library, I finally figured out what's going on. If a file plays back without distortion, is converted into an Acidized loop, and plays back at its "native" tempo, it will not distort. However, speeding up or slowing down the song may cause slight volume increases that push the overall level past the point of overload. This is because the DSP that does the time stretching relies a lot on crossfading, so if phase-coherent audio sections overlap, the signal peaks could add up and clip.

This effect is not at all predictable. For example, a 120 BPM Acidized loop clip might distort at 80 and 140 BPM, but not at 160 BPM. So, when creating acidized files, my final step is now normalizing them to –3dB. During mixdown, watch the levels of any Acidized files to make sure they're not distorting when the tempo changes.

IT'S PERFECT BECAUSE IT'S DIGITAL, RIGHT?

Well . . . not necessarily. As most people know, digital audio takes a series of "snapshots" (the sample rate), processes them, then smoothes these stairsteps at the output to produce a continuous, analog-type waveform. However, the meters almost always monitor the waveform before it has been smoothed, not after. As a result, it's possible that during the process of reconstructing the output signal, the output level may momentarily exceed 0, even if

the meter doesn't show it. This is one reason why many engineers don't use the full headroom available in a mix bus (hey, there's usually at least 24 bits to play with anyway—you're not losing much), and produce final mixes that are down a few dB from 0dB. The mastering engineer can always make sure that the final signal reaches full code.

AND ABOUT THOSE LITTLE NUMBERS . . .

A great feature of many digital meters is that they include a number that indicates the maximum (peak) value obtained during playback. Furthermore, the fader will be calibrated, as well. So for example, if the meter indicates the maximum signal during playback was –3.16dB, then you can move the fader up +3.15dB, and, at least in theory, the signal will never exceed 0.01dB. However, as noted above, you can't always assume this figure is 100 percent accurate, so you'd more likely raise the fader by +2dB or so to leave a margin of error. Conversely, if the number indicates, +4.2dB, odds are you already heard it was distorting, and fixed it! But if not, you can bring the fader down to –5dB and feel a lot safer.

EQ MAGAZINE SAYS YOU SHOULD MIX WITH YOUR EYES!

No we don't! Realistically, you should be able to do a pretty good mix without even seeing the meters, and many people (myself included) would advocate not caring too much about what the meters are showing—it's the sound that matters.

However, meters are like test equipment—they'll point out problems much faster than you might find them on your own. To illustrate, suppose you have a dense mix with 40 tracks, and you swear you hear some distortion in the chorus. The track meters will show you where the track is clipping (or, at the very least, the bus meters will show the part of the song where clipping occurred), and you can find out which track is causing the problem by muting tracks one at a time, observing the meter number that shows the peak value, and then turning down whichever track pushes the level over the edge. The bottom line is the more you know about metering, the more you'll know about what's happening with your audio. EQ

CHEAT SHEET

ABLETON LIVE 6 SESSION VIEW

BY CRAIG ANDERTON

Cheat Sheet delivers concise, explicit information on how to do specific recording/audio-related tasks. This installment describes the most common operations for Ableton Live 6. Note that for the Mac, use Ctrl-click instead of right-click.

TRACK OPERATIONS

CREATE AN AUDIO TRACK: Drag an audio file into the Clip/Device Drop Area, *or* right-click in the Clip/Device Drop Area and choose "Insert Audio Track," *or* type Ctrl-T.

CREATE A MIDI TRACK: Drag a MIDI file or virtual instrument into the Clip/Device Drop Area, *or* right-click in the Clip/Device Drop Area and choose "Insert MIDI Track," *or* type Ctrl-Shift-T.

CREATE A RETURN TRACK: Right-click in the Clip/Device Drop Area and choose "Insert Return Track," *or* type Ctrl-Alt-T.

DELETE A TRACK: Click on the track name and hit the Del key, *or* right-click on the track name and select "Delete," *or* type Ctrl-X.

IMPORT AN AUDIO OR MIDI FILE: Drag into the Clip/Device Drop area if you want to create a new track, *or* drag into an existing track clip slot.

REWIRE

REWIRE A CLIENT INTO LIVE: Open Live. Open the ReWire client. Select the client from the "MIDI To" drop-down menu so it can receive MIDI data. Monitor the client audio from a Live audio track by selecting the client from the "Audio From" drop-down menu.

REWIRE LIVE INTO A REWIRE HOST: Open the ReWire master program first. Open Live. Refer to the host's Help on how to set parameters for rewired client programs.

CONTROLLERS

USE YOUR QWERTY KEYBOARD TO PLAY MIDI NOTES: Go *Options > Computer MIDI Keyboard, or* type Ctrl+Shift+K.

CONTROL A PARAMETER WITH A QWERTY KEYBOARD KEY: Click on the "Key" button in the upper right of the screen to enable assign mode. Click on the parameter you want to control. Type the desired key. Click on "Key" again to return to normal operation. Note: If you're using the QWERTY keyboard as a MIDI controller, it will have priority.

ASSIGN A MIDI CONTROLLER AS A CONTROL SURFACE: Go *Options > Preferences*. If your control surface is supported under the "Control Surface" field, use that. Otherwise, under the "MIDI Ports" section, for the Input parameter associated with the controller, click on the box under "Remote" to turn this functionality On or Off.

CONTROL A PARAMETER WITH A MIDI CONTROLLER: Click on the "MIDI" button in the screen's upper right to enable assign mode. Click on the parameter you want to control. Move the desired MIDI controller. Click on "MIDI" again to return to normal operation.

CLIP OPERATIONS

EDIT LOOP SOUND QUALITY: Click on the Clip you want to edit. Click on the "Clip Overview" tab toward the lower right (the tab showing a waveform). In the Sample section under "Warp," experiment with different algorithms accessed via the drop-down menu. Also experiment with the options under the selected algorithm (transient, grain size, flux). Under Sample, click on "Hi-Q" to invoke a higher-performance transposition algorithm if you're willing to trade off CPU power.

CONVERT A CLIP INTO A RAM CLIP: This prevents stressing out your hard disk if you have lots of clips loaded, or are using a laptop with a slower drive. Click on the clip you want to convert. Click on the Clip Overview tab. Under Sample, click on "RAM." If multiple clips are selected, they all will be converted.

QUANTIZE CLIP LAUNCH (GLOBAL): Clip launch will be quantized to the value selected in the Quantize value in the control bar (to the right of the Transport controls).

QUANTIZE CLIP LAUNCH (PER CLIP): Click on the Clip you want to edit. Click on the "Clip Overview" tab. Under the Clip section, click on the small circled "L" to reveal the Launch Editor. Choose the desired Clip Launch Quantize value from the Quantization drop-down menu.

NAVIGATION

CHANGE ARRANGEMENT POSITION FROM SESSION VIEW: Click on the transport's numerical for number of bars, and drag up or down. For more precise adjustments, click on the beats or sixteenths numerical.

ZOOM IN AND OUT ON A MIDI/AUDIO CLIP: Click on the Clip Overview tab, then click the cursor in the Clip Overview tab, *or* in the timeline above the Clip. Drag up to zoom out, drag down to zoom in.

ENVELOPES

DRAW MIDI OR AUDIO ENVELOPES: Click on the Pencil tool in the control bar. Click on the "Clip Overview" tab. Under the Clip section, click on the small circled "E" to reveal the Envelope Editor. Under "Envelopes," choose the type of envelope you want to draw, and draw it with the Pencil tool.

CHANGE ENVELOPE RESOLUTION

WHEN DRAWING: Envelope resolution is tied to the visible grid. Go *Options > Narrow Grid* or *Options > Widen Grid*.

CREATE AND EDIT BREAKPOINT ENVELOPES: Proceed as if drawing an envelope. Turn off the Pencil tool. Double-click on the envelope line to create a breakpoint, and drag as desired. Double-click on an existing breakpoint to delete it.

COMBINE STEPPED AND BREAKPOINT ENVELOPES: Enable the Pencil tool when you want to draw stepped envelopes, disable to convert a stepped envelope to breakpoints.

GUI CUSTOMIZATION

RENAME A SCENE, CLIP, OR TRACK: Right-click on the name and choose "Rename," *or* type Ctrl-R.

CHANGE CLIP COLOR: Right-click on the Clip and select the desired color from the pop-up menu.

CHANGE SKINS, DEFAULT CLIP COLORS, AND LANGUAGE: Go *Options > Preferences > Look Feel* tab.

CHANGE MIXER CHANNEL WIDTH: Click on the right side of the Track Title Bar and drag right or left.

CHANGE MIXER FADER HEIGHT: Click on the horizontal line above the fader and drag up or down.

SHOW/HIDE VARIOUS MIXER ELEMENTS: Click on the show/hide buttons toward the lower right corner of the mixer window.

GO TO FULL SCREEN VIEW: Type F11. Type F11 again to return to the previous screen size.

SHOW/HIDE INFO WINDOW: The Info Window is extremely helpful, as it describes whatever function the cursor points to. Use the "?" key to show/hide this window.


1-800-VIVALDI
(848 - 2534)
The Largest Insurance Firm
Dedicated to Musicians...Period
30% Off
For EQ Subscribers
*subject to underwriting
not all will qualify
ASSOCIATES INC
CLARION
MUSICAL INSTRUMENT INSURANCE
www.clarionins.com

KRK SYSTEMS VXT8

These Bi-Amplified Studio Monitors Were *Not* What I Was Expecting

by Garrett Haines

With a street price that's a fraction of KRK's Exposé line, I mistakenly assumed the VXT line was a cosmetic redress of old KRK technology. Wrong. The VXT series has benefited from the R&D that went into the Exposé in ways that are not always apparent to the eye, but are sonically obvious.

IN PERSON

The VXT Series (VXT4, VXT6, and VXT8; the number signifies the main driver's diameter) features an almost egg-like shape, with no sharp edges. Each sports KRK's Kevlar driver and a slotted bass port that runs along the front bottom. Rear panel toggle switches (protected by plastic overhangs from accidental changes) let you adjust high and low response, as well as clip and power settings.

The VXT8 is no bookshelf speaker—it weighs 36.5 lbs., and approaches 18" in height. With its non-skid, neoprene-like bottom pad for acoustic decoupling, this speaker won't start bouncing around in the middle of a track.

IN USE

We set up the VXT8s in the main tracking room next to our main 8" active monitors (we'll call them "Brand X"). In a shootout a while back of eight different models, the entire staff agreed on Brand X; we know them well. In comparison, we initially thought the VXTs had a strange midrange and that we wouldn't have much nice to say. Nonetheless, we grabbed our standard test tracks, and got methodical.

In formal listening tests, we immediately noticed the VXT8's wide sweet spot: We could sit high or low, and slide far left or right along the console, and still hear a consistent high end. Chalk up part of this even dispersion to the cabinet's rounded edges.

Listening to mixed tracks, we started to appreciate the VXT8's extended low end. We could hear when tracks were bass-thin, and when we had gone overboard. By coincidence, we were considering a subwoofer to augment our current speakers; we assumed our room was too big for an active 8" monitor. Nope: The VXT8s can throw serious bass at your skull.

I further noticed more detail in the VXT8's upper-mids (for example, reverb decays had greater clarity, and vocal nuances were more evident), and came to the startling conclusion that our beloved Brand X monitors were "flattering" and soft on many key elements of a mix. Instead of letting us hear what was going on in the mids, most things were diffuse or even scooped out. Yes, they were more fun to listen to, but we don't need "fun" from audio monitors.

"Several elements contribute to the VXT's sound quality." For example, the silk (not cloth) tweeter and woven Kevlar (not paper or polymer) woofer cone allow for faster transient response, and more accurate stereo representation. The ABS structural foam cabinet allows more internal volume, which provides significant bass extension. And the lack of internal parallel walls reduces image smearing, common in less expensive designs. In isolation, the VXTs sound solid, but these design improvements are glaringly obvious when compared to other brands.

Worried that I had lost my perspective, I sought the opinions of Sean McDonald, a local guy with national credits, and Claus Trelby, who has worked with projects ranging from the London Philharmonic Orchestra to Iggy Pop. They noticed what we noticed; everyone I asked had something appreciably positive to say about the VXTs.

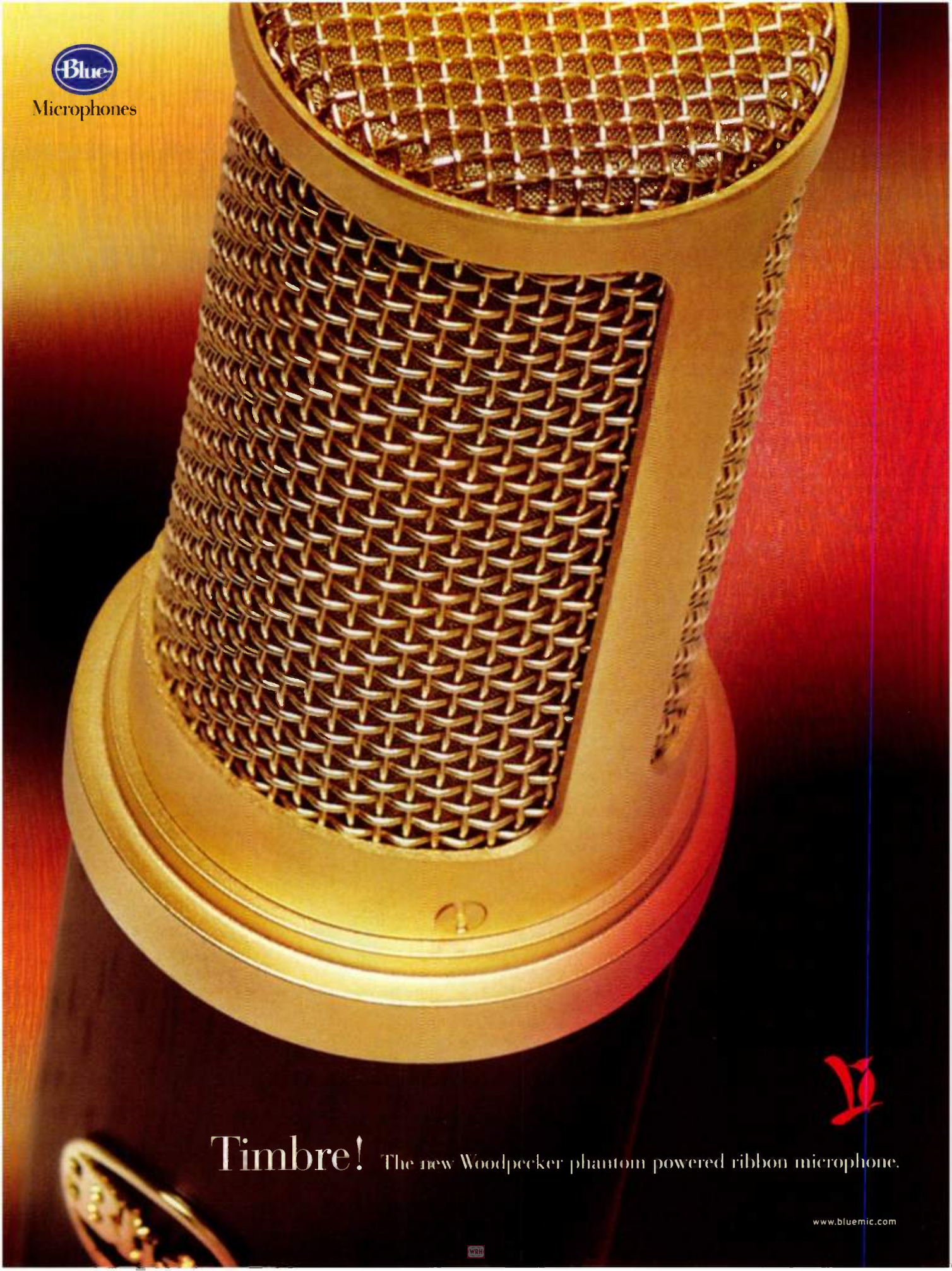
Finally, we used the VXT8s for some mixdowns. When we checked mixes in the car, on iPod earbuds, or a home stereo, the sound translated consistently—no surprises, other than that mixes also sounded better on other, less-accurate speakers.

CONCLUSIONS

Imagine an NS-10 with all of its midrange accuracy, but with immense low-end extension and accurate, but non-fatiguing highs—that's the VXT8. However, carefully choose the right size for your listening environment; the VXT6s were too big for our mixing suite and too small for our tracking room, but the VXT8s were ideal—work with a dealer who allows exchanges. And note that these are accurate, revealing speakers, with perhaps a slightly forward midrange. If you want something that smoothes over errors or makes mixes sound nice, look elsewhere.

After using these speakers for a few months, I believe the VXT line is poised to be the new "stand-up-and-take-notice" speaker on the market. Get the right model for your studio and work with them for a few weeks—I bet you'll agree. EQ

PRODUCT TYPE: Near-field monitor.
TARGET MARKET: Recording studios and production suites.
STRENGTHS: Great bass extension. Tall, wide sweet spot. Can play at LOUD volumes. Configuration toggles protected from accidental changes.
LIMITATIONS: Rear panel power switch. Speaker must be positioned vertically. Accurate midrange might seem overly-forward for those used to hyped monitors.
LIST PRICE: $799 each
CONTACT: www.krksys.com


Blue
Microphones
Timbre! The new Woodpecker phantom powered ribbon microphone.
www.bluemic.com

BLUE WOODPECKER RIBBON MIC

Yes, this Really is an Actual Blue Ribbon

by Jeff Anderson

It's official: Ribbon mics are all the rage these days. The usual suspects (we're talking Royer and AEA) have kept their game up—providing the market with awesome mics and related ribbon-specific products (such as AEA's TRP). And a lot of other manufacturers (specifically CAD and Nady) are taking advantage of decreased manufacturing costs, and putting out some damn fine mics at a fraction of the cost of the historically expensive ribbons we've all coveted over the years.

And now Blue—the company responsible for the prestigious Bottle mic, among many others—has taken a crack at ribbon production. Since it seems that all I do these days is play with ribbons, the editors of this here prestigious magazine decided to send yet another ribbon my way. So I ripped open the box, and . . .

OVERVIEW

Not surprisingly, the Woodpecker is an impressive-looking microphone. From its classy wood finish to the gold highlights, it's obvious that Blue made the effort to engineer a pretty mic. I, for one, appreciate that. It may make no difference in how the unit performs, but you have to appreciate a sexy-looking mic . . . and you know Blue is always good for one of those.

Additionally, the Woodpecker comes complete with a solid brass shockmount and a storage box carved from cherry. Again, you have to give them props for putting out a sharp-looking product. The case by itself is just gorgeous.

But, you say, what about the specs, Jeff? Okay, the Woodpecker (as with all ribbons) is set in a figure-8 polar pattern. What may strike you as strange, though, is that the Woodpecker is an *active* mic, meaning it requires 48V phantom power to wake up in the morning. Its frequency response ranges from 20Hz to 20kHz, and it can handle some decent SPL (up to 114dB). Furthermore the Woodpecker is all hand-made, from the discrete Class A electronics to the aluminum ribbon pressure-gradient transducer.

So it looks nice, is built solidly, and the spec sheet checks out: Time to round up some musicians, warm up some recorders, and see how it performs.

APPLYING THE WOODPECKER

Before we start, I have to say one thing: The shockmount is a pain, plain and simple. The clips in the middle of the mount are buckles, meaning that you have to slide the mic into position and then snap the ring of the mount around the mic and buckle it in. Out of the box, the ring of the mount was way tight and, with the buckle in open position, I had to apply a ton of pressure to get the mic mounted correctly—thereby causing this beautiful, brand new mic to get scratched. And I hadn't even tracked with it yet.

The Blue Woodpecker, in all its glory.

Once you get the Woodpecker set up, however, everything is cool . . . so cool. Doing a quick test on male vocals, I noticed the Woodpecker added a nice amount of gloss to the voice. It also had zero noise floor, which was most impressive. There seemed to be a bit of a natural frequency boost around 400Hz, which may be good or bad depending on the source, but a quick EQ notch of around 4dB in that area made everything nice and smooth, and the vocals were sounding really warm. I was excited.

Up next: acoustic guitar for a country album. The session guitarist for the album was playing a handmade acoustic with a maple body that gave most Martins I've heard a run for their money. The guitar was naturally rich-sounding and had insane sustain, and the album's producer was so taken by its sound that all he requested was that we make it sound "spacious." So I set the mic about 15 inches away from the guitarist, running through a No Toasters Nice Pair preamp with a low gain stage setting to keep from coloring the sound too much. Due to my previous experience with the Woodpecker's low-end frequency boost, I was ready to break out the EQ, but we got exactly the sound we needed without further adjustments. We simply set the level to tape and hit record . . . and it sounded beautiful.

I like to use ribbons as room mics when tracking drums, as I find the ribbon character adds a much-welcomed vintage flavor to the mix. So, I thought it was essential to take the Woodpecker to task in that scenario. Setting up an old '60s Gretsch kit in the live room, I guesstimated my mic placement by putting the Woodpecker about nine feet out from the snare. Hitting playback, I found that the kick was cutting too much and the overall drum sound was rather floppy. Turning the mic backwards to see if the opposite side of the ribbon would give a

different sound, I noticed very little change. Hmmm. . . .

As the Woodpecker has a figure-8 pattern, I decided to try an old trick by turning the ribbon 90 degrees so that the mic's front and back faced the walls of the room and not the instrument itself. Perfect, I thought—the drums sounded even and lively, which was just what I wanted. I simply ran my trusty old Teletronix LA-2A in line with a 4:1 ratio (set to around 10dB of gain reduction, to really smash and smooth out the drum submix), and the result was a drum sound that sat perfectly in the final mix.

CONCLUSIONS

Blue has become a pretty heavy hitter in both the pro and project studio realms in the last few years, and with good reason. With this mic, I can't see the Woodpecker doing anything besides accelerating the company's reputation for making mics that have a lot of character, both sonically and aesthetically. Given its price, this mic is a serious contender and has no problem holding its ground with the higher-end, non-Chinese ribbons on the market. In fact, in certain applications, it beats the competition. To me, the Woodpecker sounds a little different than other ribbon mics available today. Because of the combination of active and ribbon technologies, the Woodpecker offers the warmth and gloss of a ribbon, and the high end and clarity of a condenser. EQ

Ribbon Mic Basics

A ribbon mic is a variation on the dynamic mic, but has high frequency detail more like a condenser mic. It achieves this by placing a thin metal ribbon between the poles of a magnet; when sound waves hit the ribbon, the movement of the ribbon cutting through the field induces a voltage in the ribbon, which is then amplified to bring it up to a useable level. As ribbon mics are bidirectional, they are often used in pairs to form Blumlein Pair arrays.

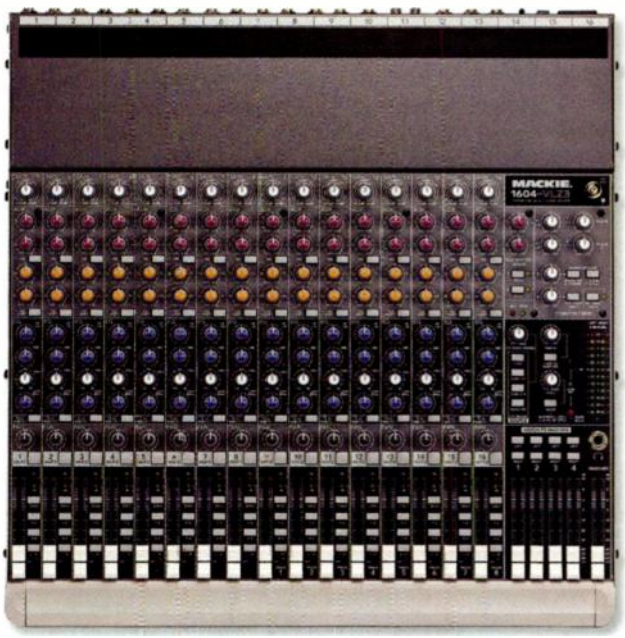
A reputation for fragility endures, because early ribbon mics were indeed very fragile. However, newer materials and construction techniques allow ribbon mics to be used with guitar amps and other loud sources; some are even suitable for live use.
—Craig Anderton

PRODUCT TYPE: Active ribbon microphone.
TARGET MARKET: Mid- to pro-level musicians/engineers who want to round out their mic lockers with a clean-sounding, versatile ribbon mic.
STRENGTHS: Great sounds. Unusually low noise floor. Fair price point. Beautiful packaging.
LIMITATIONS: Shock mount, difficult to use.
LIST PRICE: $1,299
CONTACT: www.bluemic.com

MACKIE 1604-VLZ3

The 1604 Mixer Itself Gets Re-Mixed

by Garrett Haines



In my travels, the Mackie 1604 has become as ubiquitous as the Shure SM57. Visit any studio, production house, or live venue, and the odds are good you'll find one of these workhorse mixers. The latest incarnation, the 1604-VLZ3, adds some creature comforts that studio and live engineers will appreciate.

As the original model has been covered in many reviews over the years, we'll focus on the updates. Rest assured beloved items such as the roto-pod, tank-tough chassis, 60mm faders, and flexible bus routings are still present.

A subtle—but very useful—update is the use of slightly larger fonts for control labels. In low-light situations, the larger print makes it easy to find a particular control. We have several revisions of Mackie mixers in house; in a side-by-side comparison, despite the larger print size, the surface still doesn't appear cluttered. Also note that cosmetically, the darker gray/brown chassis has been lightened to a metallic silver.

SOUNDING OFF

Most of the other updates are not as visible, but they add up to improved sonic performance. Mackie claims the VLZ3's new preamps, dubbed XDR2 (Extended Dynamic Range), have wider gain and dynamic range, as well as extended low frequency response. But no matter what any brochure says, in the real world *sound* is what counts.

First off, this mixer is definitely quieter than previous versions. (And even the first generation Mackies had respectable signal-to-noise characteristics.) We used the review unit for all of the *EQ* Television sessions (www.eqmag.tv/) done at Treelady Studios. Everyone commented that the drums, guitars, and narration sounded clear and solid. Sure, it's easy to be a snob about mic preamps, especially with so many cool pieces out there. But you could record an entire CD with these preamps if needed. They are quiet, clean, and give a lot of gain if required. Of course, they don't have the sense of depth you get with a True Systems, Great River, or John Hardy preamp—but most of those units cost many times (per channel) as much as the Mackie preamps. Comparing apples to apples, you won't find preamps that sound this good on another small format mixer in this price range.

The equalizer has been tweaked to improve its "musicality." This is tough to quantify, because what sounds good to one person might be harsh to another. The low frequency is centered at 80Hz; I would have preferred 90Hz, but 80Hz provides more flexibility to more people. Likewise, the high band is centered at 12kHz—a good spot to grab unruly hi-hats or sibilance. The mid band can sweep from 100Hz to 8kHz, which is just about as good as you can get in an analog unit. We also found that the equalizer sounded more natural than previous versions, especially when applying two bands to neighboring frequencies. Whereas the VLZPro EQs are more surgical, the VLZ3 emphasizes a sweeter interaction of the bands. This is not a bad thing at all, especially when the result is an EQ that sounds less intrusive over a broad range of applications.

Mackie also improved the overall mix bus headroom—a fact we demonstrated in our drum tracking videos. Granted, if you set up your gain stages correctly (via the level setting process common to all Mackie mixers), it's unlikely you'll overload the mixer anyway. But headroom is one spec where more is indeed more, and more headroom is always beneficial. Doing a quick stem mixing comparison against a VLZ model, at the same gain settings I found the 1604-VLZ3 to be a bit more open. It seemed to be less a part of the signal chain . . . more invisible, more out of the way.

CONCLUSIONS

If you already own a Mackie 1604 mixer, you don't necessarily need to run out and upgrade it—although owners of the original version will hear an appreciable difference from one end of the unit to the other. But if you're looking for a keyboard mixer, studio mixer, or just want some solid-sounding preamps, the 1604-VLZ3 delivers a heck of a lot of bang-for-the-buck. You may find less expensive 16-channel mixers, but they won't sound as good or be as durable as the Mackie. Nor will their control labels be as readable! EQ

PRODUCT TYPE: Compact 16-channel mixer.
TARGET MARKET: Small studios, production houses, and live venues.
STRENGTHS: Solid construction. Exceptional mic preamps for the price. More readable front panel. Better mix bus headroom. Sweeter-sounding EQ.
LIMITATIONS: EQ high- and low-band frequencies are fixed.
LIST PRICE: $1,099.99
CONTACT: www.mackie.com/products/1604vlz3/

Virtual. Perfection.

StroboSoft™

StroboSoft Software Strobe Tuner for Mac / PC

Perfect tuning for instruments or samples can be made with razor-sharp precision using StroboSoft's 0.1 cent accuracy borrowed from our legendary hardware tuner line.

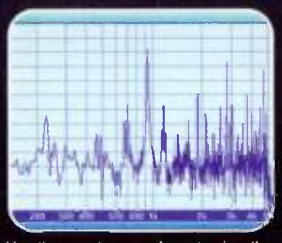
Take advantage of over 30 preset Sweetened Tunings™ that compensate for certain instruments' design flaws and optimize their sound. Use the integrated Buzz Feiten Tuning System® presets for your BF-equipped instruments.

Standard chromatic tuning mode and innovative 'Instrument Mode' provide professional tools for quick tuning or complete instrument set-ups.

With more than 50 preset alternate tunings and unlimited capability to store your own presets, quickly dial in the tuning you need or compile play lists for studio sessions or your live shows.

Note/Octave window offers real-time response and multi-window (below) provides cent offset, Hertz value, and MIDI note number.

Store unlimited presets for all your instruments or create a preset for each song in your set to compile a set list for studio reference or tonight's gig!



Use the spectrum analyzer to view the fundamental note and its harmonics, view noise floors, or to help you isolate tuning issues. Oscilloscope also included.

Legendary, Peterson 0.1 cent accuracy / 52 preset alternate tunings / 26 Instrument presets / Over 30 exclusive Sweetened™ Tunings / 12 selectable temperaments / Buzz Feiten open and intonation presets / Intonate Mode / 96kHz support / 20Hz-5kHz range

VST plug-in available July 2007.

Free 1.22 update to registered customers.

Check it out today at www.strobosoft.com

peterson
STROBE TUNERS
- Industry Standard for 59 Years

ASIO Mac

CAKEWALK SONAR 7

MIDI, Mastering, Plug-ins, and Publishing Get Increased Emphasis

by Craig Anderton

Falling leaves, a nip in the air, an AES show . . . the planets are aligned for another Sonar update, and now we're up to version 7. Unlike Sonar 6, which added some drastic new features like AudioSnap and ACT, Sonar 7 is more about accessorizing a mature program. When you boot it up, there are no major GUI changes; it's a familiar environment. But looks can be deceiving—this time around, the changes lie a little deeper.

MIDI GETS MAGNIFIED

The biggest deal is an overhaul of the MIDI-related tools. Sonar had previously introduced in-track MIDI editing, but using it required frequent track resizing. Good news: Now there's MIDI Magnifier. Pass the cursor over MIDI data in piano roll or inline PRV view and when the notes are below a certain size, they're magnified (you can customize the way this happens, too). You can manipulate notes in the magnified state; this takes some getting used to, but makes the inline view far more useful.

There are lots of other MIDI tweaks: show velocities only for selected notes (lets you edit notes easily in a stacked chord), move controller data that falls during a note with the note, note split and glue, mute notes, show multiple controller lanes in piano roll view, etc. You can also move controller data across lanes; for example, to have one parameter track changes in another parameter, just copy the controller data (it changes automatically to match the controller type in the target lane).

Another addition is Cakewalk's "smart MIDI tools," where the cursor can be any one of a number of configurable rosters of tools, as called up by keyboard modifiers and mouse buttons (left, right, middle). There are three different available toolsets based on the select, draw, and erase functions, and you can implement up to 20 mouse actions per tool. Presets match how these tools work in Cubase, Digital Performer, and Logic. It takes some thought to set this up so that each tool does the most good for your workflow, but as with the other customization options, it's worth the effort.

Drag-quantize is a good example of a new cursor tool: Click on a selected group of notes (or individual note) and drag up to move them closer to the snap grid, or drag down to move them further away. As someone who uses quantize strength a lot, this has joined my default smart MIDI tools.

And if you're into multitrack step sequencers, you'll love what Sonar 7 has to offer—patterns can be up to 64 measures long, with up to 16 steps per beat. You can also add velocity offset to tracks, multiply velocity values by a constant, and edit swing, articulation, and portamento. If you're into those dance mix thangs, the Step Sequencer is straight-out fabulous.

Clockwise from the upper right, the windows show the Boost 11 limiter, Rapture LE soft synth, LP-64 equalizer, and the "Smart MIDI Tool" configuration window. Just to the left of Rapture LE, the MIDI Magnifier displays part of a MIDI track.

NEW INSTRUMENTS

Many sequencer manufacturers add instruments in their updates to provide extra value, and Cakewalk is no exception. This time around, you get the Z3TA+ waveshaping synth, Dimension LE with the Garritan Pocket Orchestra, DropZone sample player, and Rapture LE. If you've already invested in these synths and have a sampler, the inclusion represents dubious value, especially as Dimension and Rapture are "lite" versions. However, if you're synth-shy, there's significant value here. Rapture LE and Dimension LE will both read files created by the full versions (although the editing options are more primitive), and if you've overlooked the Z3TA+, it's a unique, full function synth that can produce metallic, big timbres with a different character than analog synth emulations.

DropZone isn't particularly sophisticated, but will play back two layers or splits of individual samples as well as SFZ-format multisamples, and includes filtering and some other goodies. As you can just drag and drop files into it, DropZone is a fast way to turn sounds into playable instruments. The bundled instruments from V6 are also present (Pentagon I, RXP groove/Rex Player, PSYN II,

TTS-1, Cyclone, Session Drummer 2, etc.), so that's a lot of instrument power.

However, there's also a backstory to all these plug-ins. With Sonar the only sequencer optimized to run under 64-bit Vista, this collection of instruments means that Sonar will cover most, if not all, of your needs out of the box. Sure, you won't be able to ReWire up Reason 4 and use that wonderful Thor modular synth until the rest of the world catches up to 64-bit operating systems, but you'll have the basics—and then some—covered while you're waiting. (And given the current rate of progress, it may take a while before we transition from the 32-bit Vista world to 64 bits.)

NEW PROCESSORS

Sonar 7 also wants to be your inspiration-to-mastering-to-publishing host. While I still don't think Sonar replaces a dedicated two-track stereo editor like Peak, Wavelab, Sound Forge, or Audition (for starters, there's no noise reduction), it comes much closer to the Samplitude/Sequoia model of "this is the only program you'll need."

Start with the LP-64 Linear Phase Mastering EQ, which complements the Sonitus:fx EQ (a basic, low CPU EQ) and VC-64 channel strip (EQ with "character") by providing a transparent, surprisingly versatile mastering-style EQ with 64-bit double-precision processing. Better yet, the LP-64 works with other VST-compatible programs. I also like that you can add 20 control points to the curve, for very detailed responses. Now if it only had a spectrum analysis graph in the background to show the averaged response curve . . . but as an equalizer, the LP-64 can go head-to-head against anything out there.

The LP-64 Multiband Linear Phase Mastering Compressor/Limiter also works with other programs and again, it's a honey. If you're careful with your settings, it's possible to get a good compromise between the "hot" sound that people demand and the clean, dynamic quality you'd rather hear.

But if you want to squash the living daylights out of things (or just tame some peaks), the new Boost 11 Peak Limiter will oblige—it reminds me somewhat of a Waves L1, although you can get severe enough to cross over into special effects territory.

The V-Vocal pitch corrector/processor, now at version 1.5, adds pitch-to-MIDI conversion for monophonic signals. Cakewalk hasn't figured out how to violate the laws of physics, so it does this just about as well as anything else. But I will say that with my Digital Les Paul, which has a separate audio output for each string, you can get reasonable results for guitar-to-MIDI without having to go through the hassles of using a dedicated controller (but you do have to deal with the hassles of editing). I wouldn't give this feature a "must-have" rating, but it's a helpful addition for when you need it.

On a more "under the hood" level, there's now a sidechain input for the Sonitus:fx Compressor, Sonitus:fx Gate, and VC-64 Vintage Channel so you can do keying effects, or have compression controlled by a signal other than the one at the processor's input. In fact, there's now support for all multi-input VST processor plug-ins that support sidechaining.

PUBLISH OR PERISH

These days, musicians are more likely to get their music out into the world on their own (MySpace, "distributors" like TuneCore, and the like) than sit around and wait to sign with a major label. But Cakewalk Publisher does a whole lot more than just let you "save as" your material as MP3 files. The paramount feature is the ability to create custom flash audio players with playlists, which generate XML so you can embed them in many websites. You can even choose an image and URL for each song in a playlist. The concept isn't new: Sony Acid has had its "Acid Planet" component for some time. But this takes distribution to a much higher level, and it's integrated into Sonar.

Of all the Sonar 7 features, this is one where I need to spend a *lot* more time with it; all the other features are easy to test out with a project, but this involves serious Web savvy and testing. Rather than hold up the review, we'll deal with the details at some other time—we've been wanting to do a "how to distribute your music" feature, and this would be a logical addition.

AND THERE'S THE LITTLE STUFF . . .

. . . some of which isn't really so little, such as integrated audio CD ripping and burning. Of these, my favorite is Dim Solo (no, not a Chinese food). This lets you reduce the level of unsoloed tracks by 6, 12, or 18dB, as well as mute—fantastic for hearing a track in isolation and in context at the same time. Another fave is the ability to drag and drop EQ settings across channels in the Console view. I can't tell you how many times I've tweaked the EQ for one channel, wanted duplicate settings in another channel, and then had to either duplicate the settings manually or create a preset and then load it into the other channel. This is much faster.

I also appreciate the inclusion of more import/export options (although there's no AAC export), and you still have to pay for an extra plug-in to do MP3s (although you can use the free LAME encoder instead). Sonar now does FLAC (yeah!), Sony Wave-64 which breaks the 2GB file size limitation, AIF, Apple CAF, and Sound Designer II (it seems Cakewalk is becoming more Mac-aware all the time . . . perhaps not surprising, as I keep hearing reports of Mac enthusiasts running Sonar under Parallels or Boot Camp, which Sonar now officially supports).

Another improvement concerns ACT. When it was released in V6, ACT didn't really have its "act" together. People kind of muddled through, wondering whether the problem was them or the program, until 6.2 came out shortly thereafter and fixed a lot of ACT issues. Now there are more presets for hardware controllers, which makes things more plug-and-play for those with supported devices. If you were initially frustrated by ACT, give it another shot; it's more mature now.

THE VERSION THING

Sonar comes in Producer and Studio Editions; Studio is the "lighter" of the two (it doesn't have the LP-series EQ and compressor, Z3TA+, AudioSnap only works on one track at a time, there's no surround, POW-r dithering, convolution reverb, Pentagon I/RXP.PSYN II instruments, etc.—see the Cakewalk website for a full comparison). However, both versions include 32-bit and 64-bit native applications, and English, French, and German localization. I think the typical *EQ* reader would likely spring for the Producer Edition, if for no other reason than to get the LP-series processors, VC-64 channel strip, AudioSnap, V-Vocal, and the extra instruments, which are indeed useful. If you already have a satisfactory roster of virtual instruments and processors, and don't need AudioSnap, V-Vocal, surround or high-end dithering, the choice becomes more difficult; under these circumstances, the Studio Edition is a very good value.

CONCLUSIONS

I've been using Sonar since version 1, when it was the only program to do hard disk recording, serious MIDI, and Acidized loop support with editing—all of which I needed, so I *had* to use it. Since then, I've come to appreciate the workflow, where it seems to take fewer

CAKEWALK SONAR 7

mouse clicks than some other sequencers to do the same functions. And the customization options in V6—which I originally thought meant that someone at Cakewalk had too much time on his hands—has made a huge difference in the way I work (so much so I wish that V7 migrated colors and custom toolbars properly). Cakewalk also touts the 64-bit double-precision audio engine; while it really doesn't make a huge difference in most situations compared to other programs, there's no doubt that it does reproduce signals and do calculations with a higher degree of accuracy.

Those seeking new, blockbuster features in Sonar 7 may be disappointed that there's nothing with the scope of AudioSnap or ACT. Yet the new features go way beyond "a few fixes here and there." The MIDI makeover is welcome; MIDI Magnification and the Smart MIDI Tools are exceptionally helpful. And normally I wouldn't get excited about another compressor and EQ, but the LP-64 versions really do bring something new to the party, especially for bus insertions. Sonar doesn't take mastering all the way, but if you define mastering as "something that makes my mixes sound better," you're pretty much covered. And the publishing aspect seems promising, even though I haven't had a chance to put it through all its paces.

Some Sonar users skip alternate upgrades, preferring to consolidate learning curve changes every two years instead of every year. For those still using Sonar 5, I would highly recommend upgrading to V7, given the amount of significant changes between the two versions. Users of V6 might feel they still haven't exploited all its possibilities, and may want to wait—although for those using MIDI, the associated overhaul may be all the incentive you need to upgrade.

In the larger picture, over the years Cakewalk has become a major, important player in the sequencer world—and their line of instruments is impressive, as well. Version 7 consolidates and strengthens Cakewalk's position as one of the most actively updated, and technology-driven, sequencers on any platform. EQ

PRODUCT TYPE: Digital audio/MIDI sequencer.
TARGET MARKET: Windows-oriented studios.
STRENGTHS: Much improved MIDI tools. Very cool Step Sequencer. Bundles several new instruments and processors. Publisher module simplifies the process of uploading material to the Web. Comes close to being a mastering program, too. Burns and rips CDs. V-Vocal now does pitch-to-MIDI. Sidechain capability for some plug-ins. Commendably non-intrusive copy protection.
LIMITATIONS: Previous custom color settings and user toolbars don't migrate to Sonar 7. Roster of MIDI effects not as complete as Cubase. TTS-1 soft synth won't work at 88.2kHz (but does at 44.1/48/96kHz). LP mastering processors use major lookahead, requiring longer sample buffers.
LIST PRICE: Sonar Producer Edition, $619; Studio Edition, $369
CONTACT: www.cakewalk.com

UNITED STATES POSTAL SERVICE
STATEMENT OF OWNERSHIP, MANAGEMENT, AND CIRCULATION
(All Periodicals Publications Except Requester Publications)

1. **Publication Title:** EQ
2. **Publication No:** 1050-7868
3. Filing Date: October 1, 2007
4. **Issue Frequency:** Monthly
5. **No. of Issues Published Annually:** 12
7. **Complete Mailing Address of Known Office of Publication (Not Printer):**
   NewBay Media, LLC, 1111 Bayhill Drive, Suite 125, San Bruno, CA 94066-3040
   Contact Person: Cheri McElroy Telephone: 650-238-0308
8. **Complete Mailing Address of Headquarters or General Business Office of Publisher:**
   NewBay Media, LLC, 810 Seventh Avenue, 27th floor, New York, NY 10019
9. **Full Names and Complete Mailing Addresses of Publisher, Editor, and Managing Editor:**
   John Pledger, Publisher; Matt Harper, Editor; Debbie Greenberg, Managing Editor
   NewBay Media, LLC, 1111 Bayhill Drive, Suite 125, San Bruno, CA 94066-3040
10. Owner: The Wicks Group of Companies, LLC
    405 Park Avenue, Suite 702, New York, NY 10022
11. **Known Bondholders, Mortgagees, and Other Security Holders Owning or Holding One Percent or More of Total Amount of Bonds, Mortgages, or other Securities:** None
12. **Tax Status:** The purpose, function, and nonprofit status of this organization and the exempt status for federal income tax purposes has not changed during preceding twelve months
13. **Publication:** EQ
14. **Issue Date for Circulation Data: September 2007**

15: Extent and Nature of Circulation	Avg. No. Copies Each Issue During Preceding 12 Months	No. Copies of Single Issue Published Nearest to Filing Date
a. Total No. Copies (Net Press Run)	43,649	42,918
b. PAID CIRCULATION		
1. Mailed Outside-County Paid Subscriptions Stated on PS Form 3541	3,598	4,014
2. Mailed In-County Paid Subscriptions Stated on PS Form 3541	0	0
3. Paid Distribution Outside the Mails Including Sales Through Dealers and Carriers, Street Vendors, Counter Sales, and Other Paid Distribution Outside USPS	2,172	1,866
4. Paid Distribution by Other Classes of Mail Through the USPS	0	0
c. Total Paid Circulation	5,770	5,880
d. FREE OR NOMINAL RATE DISTRIBUTION		
1. Free or Nominal Rate Outside-Country Copies Included on PS Form 3541	27,035	27,068
2. Free or Nominal Rate In-County Paid Subscriptions Stated on PS Form 3541	0	0
3. Free or Nominal Rate Copies Mailed at Other Classes Through the USPS	0	0
4. Free or Nominal Rate Distribution Outside the Mail	4,811	4,636
e. Total Free or Nominal Rate Distribution	31,846	31,704
f. Total Distribution	37,616	37,584
g. Copies Not Distributed	6,033	5,334
h. Total	43,649	42,918
i. Percent Paid	15.34%	15.65%

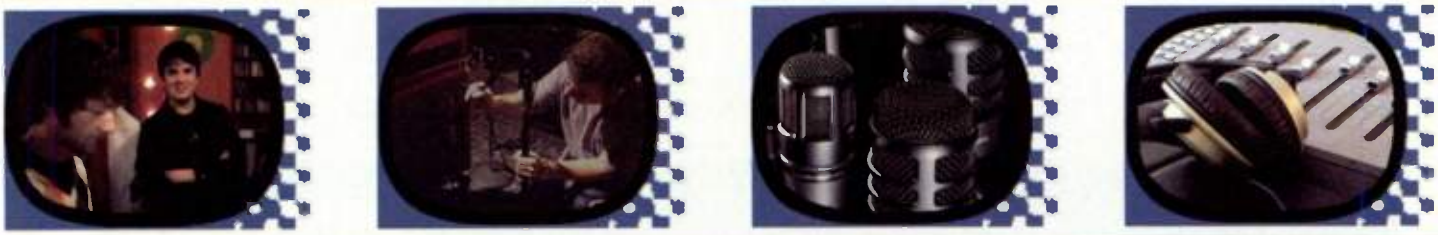
16. Publication of Statement of Ownership: If the publication is a general publication, publication of this statement is required. Will be printed in the November issue of this publication.

I certify that all information furnished on this form is true and complete. I understand that anyone who furnishes false or misleading information on this form or who omits material or information requested on the form may be subject to criminal sanctions (including fines and imprisonment) and/or civil sanctions (including civil penalties)
17. John Pledger, Publisher (signed)

EQTV

Get Educated, Get Entertained

Presenting the best video resource on the web for recording musicians and production pros! Check out EQTV on eqmag.tv



Featuring instructional how-to clips, instrument miking, software tips & tricks, application advice from the pros, home studio techniques and more.

www.eqmag.tv

TrueFire.com
Ignited we stand

EQ
Defining the Future of Recording

MUSIC PLAYER NETWORK

... Now show us YOURS! Upload your own videos at mymusicplayertv.com!

ARTURIA JUPITER-8V

Does Arturia's Take on this '80s Classic Live Up to the Original?

by Mo Volans

In 1981, Roland introduced what many consider to be the greatest polyphonic synth ever made: the Jupiter-8. With its eight-note polyphony, dual filters, arpeggiator, vast array of controls, and eye-catching color scheme, the synth soon found its way onto numerous hit records. Memorable appearances include Duran Duran's "Rio," Harold Faltermeyer's "Axel F," and Michael Jackson's "Thriller;" more recent users include Liam Howlett (of Prodigy), Underworld, Howard Jones, and BT.

This mighty machine was discontinued in 1985, and finding one has become increasingly difficult. Only 2,000 units were manufactured, and few owners are willing to part with this classic. When you do see one for sale, it's often around the same price as a small car!

The original hardware Jupiter-8.

For those of us who've dreamed of owning a Jupiter-8, Arturia now offers the Jupiter-8V, a software emulation of the classic hardware. Not only has the company gone to their usual great lengths to replicate the original's signature sound and controls (including hard sync and cross-modulation), but they have added extra twists like a great preset system, dedicated effects section, and a new

Fig. 1: The Jupiter-8V's graphic look accurately conveys that of the original Jupiter-8.

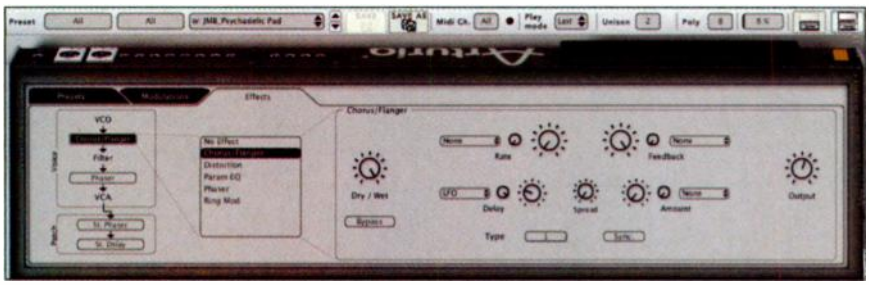


Fig. 2: The effects section goes well beyond the usual chorus and delay.

animated modulation matrix called "Galaxy." Furthermore, they've updated their patented TAE (True Analog Emulation) technology, taking us even closer to the real thing.

SEEING DOUBLE

After a simple installation and authorization process (the Jupiter-8V uses Syncrosoft USB dongle-based copy protection), I was struck by the impeccable graphic representation of the original machine (Figure 1). I once owned one, and it was like being re-introduced to an old friend. A quick flick through some of the fast-loading factory presets made it clear that not only did the software mirror the Jupiter's looks, but also its trademark sound.

The presets meet a very high standard and do an excellent job of showcasing the instrument's strengths. But as with all good synths, its true potential is fulfilled only when you stray from the beaten path and start to twiddle all those knobs and sliders.

Upon editing some of the simpler patches, I felt that each control movement did exactly what I expected—and what I heard was the same as what I remembered from the original Jupiter-8. The filters are smooth and fat, while the envelopes have a lightning-fast response time, making them perfect for percussive sounds and effects. I was able to muster up totally solid basses, searing leads, and truly mind-blowing effects. This soft synth could do all the things my original '80s dream machine could do—and I had yet to explore the extras that had been strapped on.

BIGGER AND BADDER

Taking a closer look at the Jupiter-8V's extra goodies, it's apparent that much thought went into integrating these new elements. While Arturia is known for incorporating extra features in their emulations—and do so with respect for the instrument's original design—they've really outdone themselves here.

One click exposes a new graphical panel with three tabs: Presets, Modulations, and Effects. Although there are some pretty complex routing and effects options, the overall feel is very accessible.

The Preset area follows Arturia's usual philosophy and focuses heavily on sound character, which works very well when in the creative flow. Using this method, you could quite easily find an ambient, abstract pad sound or a hard, edgy bass quickly, without trawling through hundreds of presets. A more traditional way of navigating presets by category is still available if you prefer, and saving your own patches is simple.

Next up is the innovative Galaxy modulation matrix. Unlike most mod matrices, the Galaxy concept is truly graphical, and actually incorporates its own dedicated LFOs. These are represented on an X/Y style grid, and you can achieve some mind-bending results by patching the sources to any destination you choose from the drop down-menus. The whole Galaxy section can then either run free, or sync to the host tempo. This section also includes a fun and easy-to-use analog style sequencer; in conjunction with the mod matrix and onboard arpeggiator, the sky is indeed the limit.

Finally, there's the full-blown multieffects processor (Figure 2). Arturia has included basic chorus and delay units in a lot of their recent products, but the Jupiter-8V's effects section offers four effects slots; two of these are voice-based while two are patch-based, allowing for maximum flexibility. Not only do these processors (including ring modulation and distortion) have lots of editable parameters, but they can be modulated by various sources—further increasing the overall flexibility.

IN THE FIELD

Since having access to this synth, I've included its sounds in three productions. As all were remixes, I sought a synth that supplies instant gratification and inspiration. I'm glad to say the Jupiter-8V delivered.

I usually layer my basses, using several different instruments to get a really "in your face" sound. In one production, I was able to use three instances of the Jupiter-8V to achieve this, without reaching for any other plug-in. The synth's sonic character is flexible enough to create most of the sounds I needed. Also, the internal effects engine's CPU use is modest, allowing you to free up your other native processors.

I also created "out there" effects by using the internal sequencer and Galaxy mod matrix, then inserting them into the mix. I usually reach for sample libraries or dedicated presets for this purpose, but this method achieved a much more creative end result. After using it in these recent mixes I'm seriously considering hooking up a permanent control surface just to control this synth—it's that useful.

HERE TO STAY?

Arturia has done the original Jupiter-8 justice. Not only has the company captured the original instrument's essence and character, but have upped the ante by adding new and creative touches that expand the reach of its sound.

As far as I'm concerned, this is not only the strongest analog emulation on the market, but is quite possibly the best soft synth I have yet encountered. Once hooked up with a good control surface, this could be the only virtual analog you'll ever need—and it will definitely be my go-to instrument of choice in the future. EQ

PRODUCT TYPE: Cross-platform synthesizer for standalone or VST, AU, and RTAS plug-in formats.
TARGET MARKET: Analog synth fans who want an extremely faithful software re-creation.
STRENGTHS: Near perfect re-creation of a true classic. Slick interface. Useable new additions. Includes all the features even a veteran sound designer would expect. Reasonable price.
LIMITATIONS: Large GUI may be inconvenient for laptop users.
LIST PRICE: $249
CONTACT: www.arturia.com

AdIndex

These companies would be very happy to assist you in obtaining more information about their products and services. Please contact them via phone or online. For a better response, tell them "I saw it in EQ!"

COMPANY	PHONE	WEB	PAGE #
Ableton	646-723-4550	www.ableton.com	C4
Argosy Console, Inc.	800-315-0878	www.argosyconsole.com	40
Bass Player LIVE!		www.bassplayerlive.com	31
B&H Pro Audio	800-947-5518	www.bhproaudio.com	29, 49
Blue Microphones	818-879-5200	www.bluemic.com	69
Cakewalk	888-CAKEWALK	www.cakewalk.com/sonar	7
Clarion Musical Instrument Insurance	800-VIVALDI	www.clarionins.com	67
Conservatory Of Recording Arts & Sciences	800-562-6383	www.audiorecordingschool.com/eq.html	28
EQTV		www.eqmag.tv	77
Eventide	201-641-1200	www.eventide.com	41
Full Compass	800-356-5844	www.fullcompass.com	18-19
Furman Sound	877-486-4738	www.furmansound.com	56
John Lennon Songwriting Contest		www.jlsc.com	83
Keyboard TV		www.keyboardmag.tv	85
Klein + Hummel Studio Systems		www.klein-hummel.com	55
KRK Systems	888-361-5528	www.krksys.com	45
Lexicon	801-568-7567	www.lexiconpro.com	1
M-Audio	626-633-9050	www.m-audio.com	23
Mark Of The Unicorn	617-576-2760	www.motu.com	C4
Microtech Gefell	++49 (0)36649/882-0	www.microtechgefell.de	39
Millennia Music & Media Systems	530-647-0750	www.mil-media.com	25
Mojave Audio	818-847-0222	www.mojaveaudio.com	46
Music Player Network		www.mymusicplayertv.com	87
Omnirax	800-332-3393	www.omnirax.com	65
Peterson Electro-Musical Products, Inc.	708-388-3311	www.strobosoft.com	73
Pinta Acoustic	800-662-0032	www.pinta-acoustic.com/eq	76
Rock'n Roll Fantasy Camp	888-762-2263	www.rockcamp.com	79
Royer Ribbons	818-847-0121	www.royerlabs.com	47
SAE Institute	877-27-AUDIO	www.sae.edu	57
sE Sound Engineering	617-623-5581	www.sonicus.net	48
Shure	847-600-2000	www.shure.com	C2
Sonic Circus	888-SC4-GEAR	www.soniccircus.com/1608	30
Sony Creative Software	800-577-6642	www.sony.com/loops	35
Studio Technologies	847-676-9177	www.studio-tech.com	65
Sweetwater Sound	800-222-4700	www.sweetwater.com	11
T.C. Electronic	818-665-4900	www.tcelectronic.com	5
Wright's Reprints	877-652-5295	www.wrightsreprints.com	64

SOUNDS

Check www.eqmag.com for audio examples of sound reviews.

M-AUDIO DISRUPTIVE ELEMENTS

These are long, 24-bit samples that aren't really designed to loop per se, but instead, serve as backgrounds and ambience. (However, you can easily extend the files *ad infinitum* by crossfading a generous amount of the file's end with the beginning of a copy of the file; this is particularly effective with the 17 "Perpetual Pad" files.)

The CD-ROM has 134 files (654MB) and provides textures you'd find in the background of *Alien VII* or *Star Trek XI—An Endless Supply of Hostile Galaxies*, not an Amy Grant comeback single . . . unless she's into a serious goth makeover.

Pinning meaningful titles on this type of material is futile, so you'll need to click around a lot until you find what you want. There are six folders, but frankly, I don't hear all that much difference among them in terms of intent (although "Psycho Sequences" tends to sound more sequenced, while "Re-Valved Rhythms" does has more of a rhythmic component).

While this CD-ROM seems slanted toward twisted post-production, there are also some potentially great transitions/backgrounds for dance music and dark ambient. But the main point is that these are complex, quality sounds, not just someone fooling around with some synths and processors. I'd give it a thumbs-up . . . but I'm afraid this disruptive CD might spin out of its drive, and slice it off. *—Craig Anderton*

CONTACT: M-Audio, www.m-audio.com
FORMAT: CD-ROM with 134 WAV files, 24-bit/44.1kHz
LIST PRICE: $49.95

BIG FISH AUDIO JAZZ SESSIONS

This isn't about construction kits, but a collection of general-purpose loops (drums, percussion, bass, brass, wind, guitar, and keys) accompanied by samples and patches for the double bass, drums, Rhodes, and jazz piano sounds. The loops are Acidized (although not particularly well), but you'll also find REX file versions for most loops, with pretty good rexification.

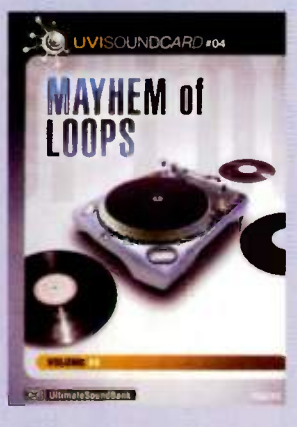
There are different samples for different tempos, ranging from 60 to 270 BPM; some are in 5/4 and 6/8. However, there isn't much consistency among the different folders, which complicates mixing and matching. For example, you might find some great piano loop, but no bass part that matches the chord progression played by the piano.

As a result, these loops seem best for augmenting existing compositions, or serving as inspiration starters. The inclusion of the samples and patches reinforces this, as you can create custom loops or parts. Of course, the advantage of a more "do it yourself" approach is that the end result will be more original.

I expected *Jazz Sessions* to be similar to Big Fish's *Acid Jazz City* and *Nu Jazz Funk* sounds, which I thought were great. I wasn't as taken with *Jazz Sessions* initially, but once I understood how it was meant to be used, I was able to put together some very satisfying compositions with a smoky, organic feel. *—Craig Anderton*

CONTACT: Big Fish Audio, www.bigfishaudio.com
FORMAT: DVD-ROM with WAV loops (many duplicated as REX loops) and sample presets (NN-XT/EXS24/HALion/Kontakt2/SFZ), 16-bit/44.1kHz
LIST PRICE: $149.95

ULTIMATE SOUND BANK MAYHEM OF LOOPS



One of the cool aspects of PlugSound Pro (reviewed 8/07) is that there's a lot of optional-at-extra cost content. However, these expansion sounds can also run as stand-alone virtual instruments (as well as work with MOTU MachFive2); one of my favorites is *Mayhem of Loops*.

This collection is 4.2GB of big beat, breakbeats, dub, drum 'n' bass, electro, jungle, hip-hop, house, industrial, and more—basically a one-stop-shop for modern beats. They also stretch well over a wide range of tempos, and furthermore, include instrument loops (bass, keyboards, etc.) for some of the genres that can help flesh out an arrangement quickly. The browser is good at helping you navigate through the content, too.

Once you've found a loop you like, you don't even need the player: You can drag and drop the audio (at the stretched tempo) into a sequencer's audio track. This is raw audio that doesn't take advantage of any processing. However, if a loop works in slice mode (almost all do), then you can drag over a MIDI file that triggers the audio in the player, and take advantage of the processing options in PlugSound Pro like filtering, envelopes, etc. (the stand-alone player doesn't include these, though).

I realize we've mostly covered the player, because it's very cool . . . but that wouldn't mean anything if the sounds weren't very useful. And they most certainly are. *—Craig Anderton*

CONTACT: Ultimate Sound Bank, www.ultimatesoundbank.com
FORMAT: DVD-ROM with monolithic sound block; requires iLok (not included) for installation, either with PlugSound Pro or as a stand-alone instrument.
LIST PRICE: $99

The John Lennon

SONGWRITING CONTEST

Categories

Rock World
Gospel/Inspirational
Hip-Hop Children's Electronic
Pop Folk Rhythm & Blues
Jazz Country Latin

Awards and Prizes

$20,000 for the Maxell Song of the Year
$60,000 in EMI Music Publishing Contracts
$120,000 in Project Studio Equipment
12,000 Custom CDs Courtesy of Disc Makers
Open to Songwriters and Bands

ENTER your original songs

Download an application or Upload your entries

jlsc.com

Sponsored by:

maxell Roland DISC MAKERS Sibelius Godin guitars

FULL SAIL Real World Education EDIROL by Roland propellerhead audio-technica. Musician's Friend sonicbids ableton

POWER APP ALLEY

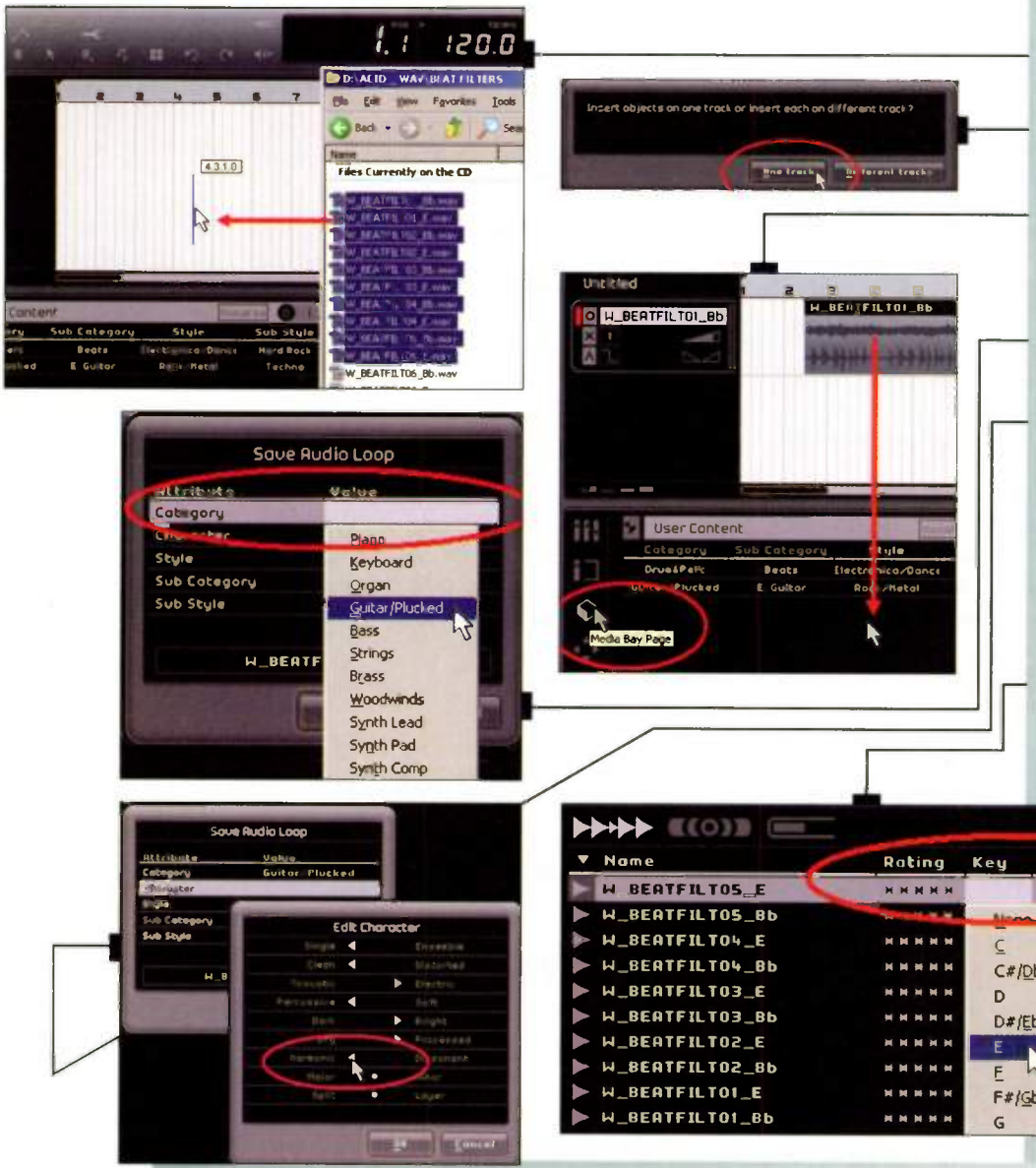
BY CRAIG ANDERTON

STEINBERG SEQUEL

Add custom audio loop content to Sequel's Media Bay

OBJECTIVE: Augment Sequel's content with custom loops that are tagged with metadata for easy searching.

BACKGROUND: Sequel includes a "Media Bay" feature that allows searching for loops (and MIDI instruments) based on particular criteria, such as instrument category, style, and character. Although Sequel comes with almost 5GB of content, it's also possible to add custom content, integrate it with the Media Bay database, and assign tags to make the content searchable.



steps

1. Drag the files you want to add into the Arrange Zone. It doesn't matter where.
2. Select One Track.
3. Click on the Media Bay button, then drag any loop in the track into the Media Bay page. All other loops will be dragged with that one loop.
4. When the Save Audio Loop window appears, click on the Category Attribute, then click in the Value field to choose the Category's characteristic.
5. In the Edit Character window, click to the right or left of the center dot to point to a particular character type. Continue entering attributes; when done, click on OK, then perform the same procedure for each Save Audio Loop window that appears until all loops are saved.
6. To edit more attributes (such as key and tempo), click on the fields in the Media Bay page to the right of the file name.

tips

- To delete a file, select it in the Media Bay page, then click on the Trash Can icon toward the page's upper right.
- In step 6, note that the rating field uses a slider action; to edit the rating, click in the field and drag to the right or left.
- If the category titles don't appear over the displayed user content, hide the Multi Zone by clicking on the small arrow on the Sequel window's lower border, then show it again. This should "refresh" the category titles.

keyboard TV

GET EDUCATED, GET ENTERTAINED

Presenting the best video resource on the web for keyboard players of all ages, styles and abilities. Check out KBTV on keyboardmag.com

CHANNELS

- Top artist techniques
- New gear and tech how-to info
- Lessons
- Live performances
- Keyboard shows and events

CHANNELS

Jordan Rudess, Rachel Z, Chester Thompson and more!

www.keyboardmag.tv

TrueFire.com
ignited we stand

keyboard

MUSIC PLAYER NETWORK

... Now show us YOURS! Upload your own videos at mymusicplayertv.com!

POWER APP ALLEY

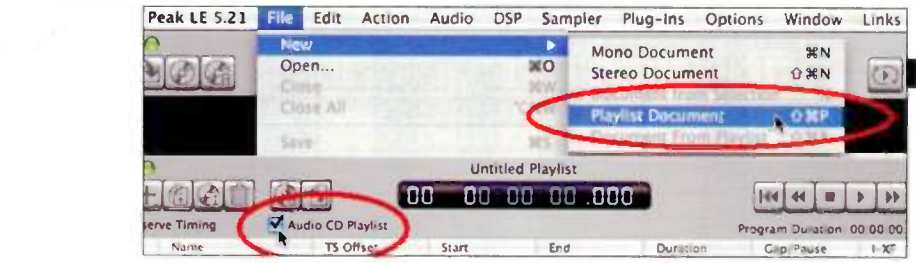
BY CRAIG ANDERTON

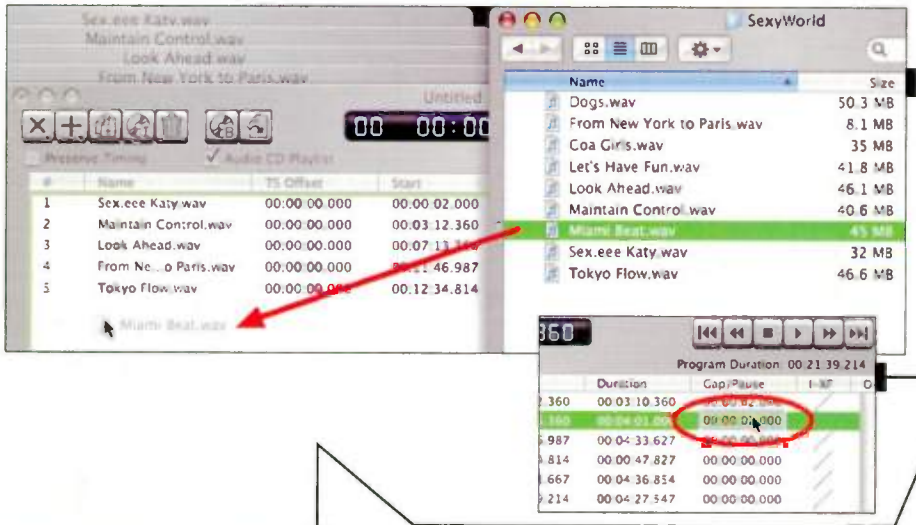
BIAS PEAK LE

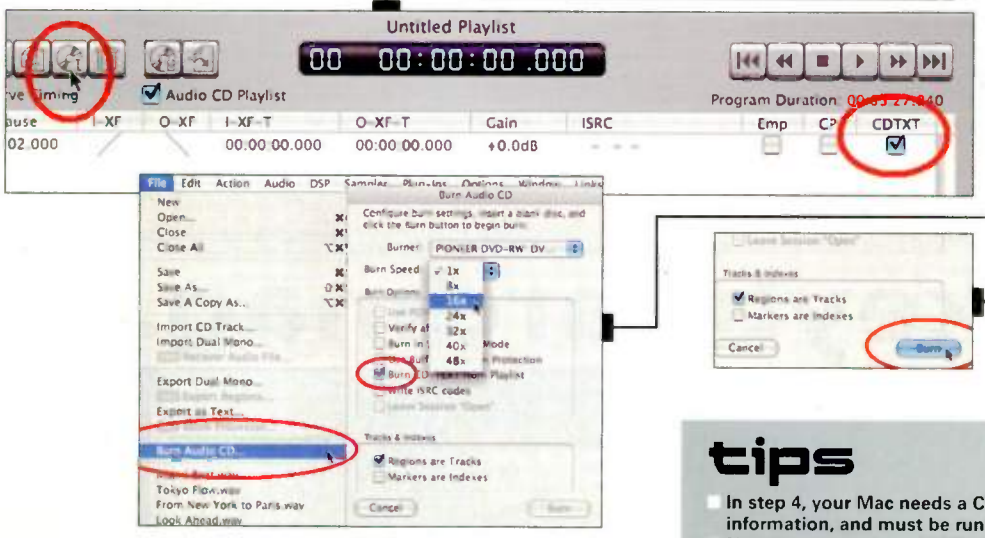
Burn a Red Book CD from within Peak LE 5.2

OBJECTIVE: Create a quick CD of your tracks for proofing and auditioning.

BACKGROUND: Despite its low price, Peak LE can burn Red Book audio CDs, as well as include CD-Text. While there are many sophisticated options, such as the ability to crossfade between cuts with customizable crossfade curves (and define particular regions of larger files as cuts for a CD), it's very easy to create a basic audio CD.







steps

1. Go File > New > Playlist Document. A Playlist window appears; check the Audio CD Playlist box.

2. Drag AIF, WAV, or SDII files into the window. If the order isn't what you want, you can drag an entry up or down to change the order.

3. To add a gap between cuts, enter the desired value in the playlist's Gap/Pause field.

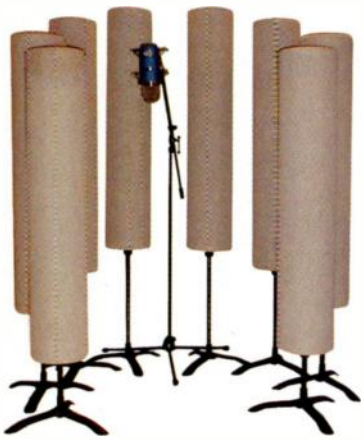
4. To enter CD-Text for the CD, click on the Audio Compact Disc info button toward the left of the Playlist window. Enter the information in the window that appears, then click on OK. To enter CD-Text for individual tracks, click on the CDTXT button (at the extreme right side of the Playlist) for the associated track. Enter the information, then click on OK.

5. Go File > Burn Audio CD, and enter your preferences in the Burn Audio CD window. To include CD Text, check Burn CD-Text from Playlist.

6. Insert a blank CD into your drive, click on Burn, and shortly thereafter you'll have a Red Book-compatible CD.

tips

- In step 4, your Mac needs a CD drive capable of burning CD-Text information, and must be running system 10.4 or higher.
- Note that you can't add MP3 files to the playlist window—only WAV, AIF, or Sound Designer II format.

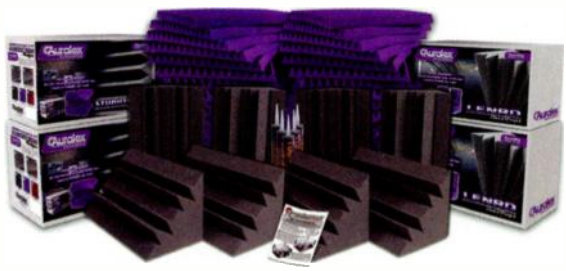


ASC Quick Sound Field

Acoustic Sciences Corp
Now Available
QSF takes the room out of the mic and puts the talent in the mix. It's a free standing live room, where you'll love the mics you used to hate.
SRP: $2998 for set of 8 traps. Free delivery.
www.asc-studio-acoustics.com/qsf.htm
1-800-ASC-TUBE (272-8823)

Brainstorm DCD/8 Word Clock Distripalyzer

plus24
Now Available
Dual Clock Generator/Distributor. Two rates (any) simultaneously output. SD/HD Video, WC, AES, Reference Inputs, VSO, GPS, Pull Up/Down Modes. Units link phase accurate. Optional VSG-4 HD/SD Generator.
SRP: $1,395
www.brainstormtime.com
323-845-1171

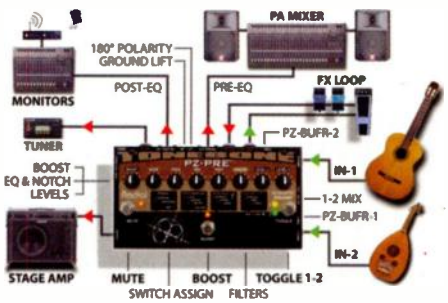


Roominators Project2 SmartPak

Auralex Acoustics, Inc.
The Roominators Project2 SmartPak is a complete acoustical treatment system featuring 24 – 2' x 2' panels of 2" Studiofoam, 8 – LENRD bass traps, 5 tubes of Tubetak Pro adhesive, and a comprehensive installation guide.
SRP: $799
www.auralex.com
317-842-2600

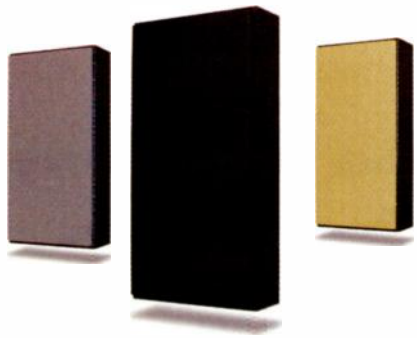
Micro Effects Series

Guyatone
Now Available
5 new Guyatone Micro Effects models offer boutique-grade features like high-quality components, true bypass switching, and updated EQ options in a compact, affordable stompbox format.
SRP: $120-$140 USD
www.guyatone.com
973-777-7477



Radial PZ-Pre™ 2-Channel Acoustic Instrument Preamp

Radial Engineering
Now Available
Compact footpedal preamp featuring great sounding Class-A piezo boosters for virtually any pickup system. Powerful parametric EQ, notch filter, 2 DI-outputs, amp-output, tuner-out, FX-loop, mute and more. Unbelievably clear true sound for your acoustic instruments!
SRP: $375 USD
http://www.tonebone.com
604-942-1001



Primacoustic Fulltrap Bass Traps

Primacoustic
Now Available
Effective and affordable, Primacoustic Fulltrapp bass traps are engineered for maximum absorption and great results. Attractive black wood-composite outer shell, high-density fiberglass panel, inner resonating membrane and easy-mount wall adaptor.
SRP: $349 USD
http://www.primacoustic.com
604-942-1001

ATM450 Cardioid Condenser Instrument Microphone

Audio-Technica
Now Available
Audio-Technica's new ATM450 side-address stick condenser delivers outstanding reproduction/ positioning flexibility. Extended frequency response is ideal for high SPL, live/amplified acoustic instruments. Hi-pass filter, 10 dB pad, isolation clamp, windscreen, pouch.
SRP: $369
www.audio-technica.com
330-686-2600

MV1 Studio Tool

Creation Audio Labs
Now Available
The ultimate tracking and re-amplifying tool for guitar and bass.
SRP: $995 (Introductory)
www.CreationAudioLabs.com
615-884-7520

CL2 Pencil Condenser Mic Pair

Samson
Now Available
A matched pair of true condenser mics ideal for recording or live applications, CL2s include a high pass filter and 10dB pad, omni and cardioid elements, and all the accessories in a durable aluminum flight case.
SRP: $299
www.samsontech.com
631-784-2200

or more information about these products, visit www.bassplayer.com/productspot. To advertise in this section contact;
anne McGowan at 650-238-0296 jmcgewan@musicplayer.com or Mike Montgomery at 650-238-0307 mmontgomery@musicplayer.com

continued on next page →

Mackie HR824mk2 and HR624mk2 Studio Monitors

Mackie
Now Available
The new Zero Edge Baffle™ minimizes diffraction and controls sound waves for wide, even dispersion. Combine this with uniform response and superior bass extension, and your studio's sweet spot becomes a full-or sweet zone.
SRP: HR824mk2 $779.99 each, HR624mk2 $579.99 each
www.mackie.com
1-800-898-3211

Tracktion 3

Mackie
Now Available
Mackie's Tracktion 3 is the fastest, most natural way to produce music on your Mac or PC. Tracktion's single-screen interface and vast array of professional plug-ins will accelerate your project from concept to finished master.
SRP: $129- $319
www.mackie.com
800-898-3211

Mackie Onyx 1200F

Mackie
Now Available
The Onyx 1200F premium 30-Input/34-Output 192kHz Firewire Interface combines all the key recording tools from the professional studio and unites them all in one ultra high quality rack system.
SRP: $2,049.99
www.mackie.com
1-800-898-3211

Mackie Control Pro

Mackie
Now Available
The new Mackie Control Pro series controllers give you ultimate command of your mix and plug-in parameters. Make your music production software whole with complete hands-on control.
SRP: $1,549
www.mackie.com
800-898-3211

VLZ3 Series

Mackie
Now Available
VLZ3 Series Compact Mixers feature Mackie's world-famous functionality, premium XDR2® mic preamps, and a completely reengineered channel EQ. The 1202-, 1402-, 1604-, and 1642-VLZ3s are the next generation of the VLZ Pro line.
From $389.99- $1099.99
www.mackie.com
800-898-3211

Mackie SRM150 Compact Active PA System

Mackie
Now Available
The Mackie SRM150 Compact Active PA System features a built-in mixer and 150-watt Class-D amplifier, making it ideal for multiple applications. No other compact PA is as loud or great sounding as the Mackie SRM150.
SRP: $389.99
www.mackie.com
800-898-3211

Mackie Big Knob

Mackie
Now Available
With its multitude of rear-panel connections, Mackie's Big Knob lets you easily switch between studio monitors, stereo sources, and even send musicians their own custom headphone mixes, all from your desktop.
SRP: $389.99
www.mackie.com
800-898-3211

Mackie HMX-56 Headphone Mixer

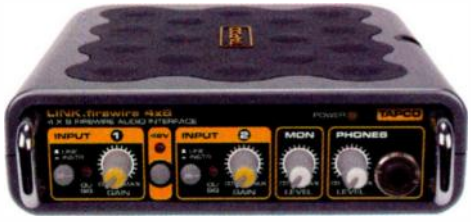
Mackie
Now Available
World-renowned Mackie technology is now available in the HMX-56 Headphone Mixer. The Six-Channel Mixer allows up to six musicians to personalize their headphone and monitor mixes in the studio or on-stage.
SRP: $199.99
www.mackie.com
425-487-4333

TAPCO Mix Series

TAPCO
Now Available
The perfect, cost-effective solution for applications ranging from home recording to live gigs and more, the TAPCO Mix.50, 60, 100 and 120 mixers offer incredible sound quality without sacrificing your desktop real estate- or wallet.
SRP: From $59.99- $119.99
www.tapcoworld.com
877-827-2669

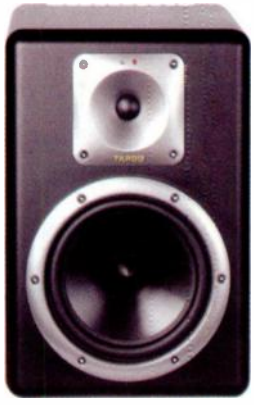
Mackie Onyx 1620

Mackie
Now Available
The FireWire capable Onyx 1620 Premium Analog Mixer features eight of Mackie's acclaimed Onyx mic preamps and their Perkins EQ, a new design that captures the musicality of the classic British EQ.
SRP: $999.99
www.mackie.com
800-898-3211



TAPCO Link FireWire

TAPCO
Now Available
The TAPCO Link.FireWire 4x6 is an ultra-portable recording solution for composers and producers. Thanks to Link's superior sound, low-latency 24-bit/96kHz operation, and Mackie Tracktion Music Production Software, you'll sound like a pro.
SRP: $249
www.tapcoworld.com
877-827-2669



TAPCO S8 Active Studio Monitor

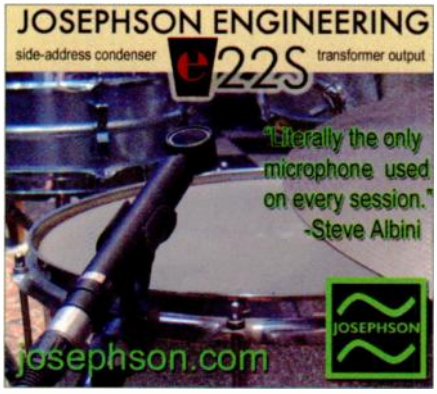
Tapco
Now Available
TAPCO S8 Active Studio Monitors deliver the big, punchy low-end needed for modern music production, thanks to dual 60-Watt amps, an 8" woofer and Active electronics designed by Mackie.
SRP: $619.99 a pair
www.tapcoworld.com
877-827-2669

CLASSIFIEDS

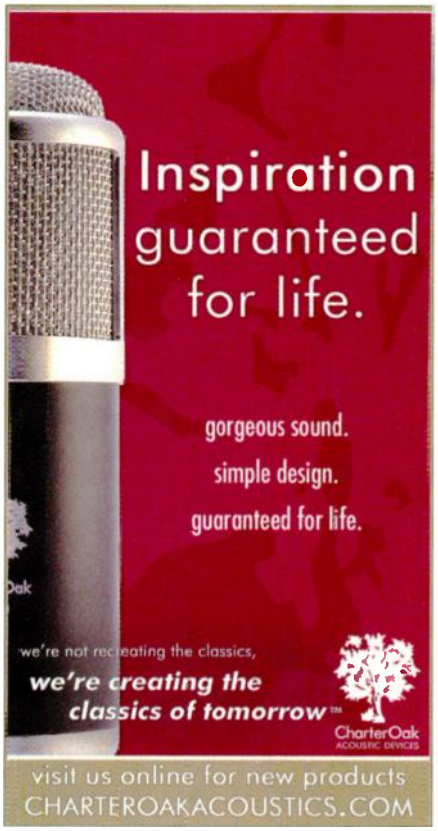
CATEGORIES

Buying or selling instruments through our Classified Ads offers you convenience, a big marketplace, and a wide range of instruments and prices. However, buying mail-order does have its drawbacks, too. EQ Magazine suggests the following guidelines to help the buyer and the seller in these transactions: 1) Get a written description of the instrument, which should include the serial number. 2) Get front and back photos of the instrument. 3) Get a written purchase agreement, with a 24-hour approval clause allowing the buyer to return the instrument for a full refund if it does not meet his/her reasonable expectations.

MARKETPLACE



MARKETPLACE



GEAR FOR SALE

MIXING / MASTERING

TALENT AND EMPLOYMENT

WWW.EQMAG.COM

DUPLICATION

DUPLICATION

STUDIO FURNISHINGS

ACCESSORIES

ACOUSTIC PRODUCTS AND SERVICES

ACOUSTIC PRODUCTS AND SERVICES

EDUCATION/TUTORIALS

WEBSITES

AD ORDER FORM

An ad in EQ's Classifieds reaches more than 35,000* serious musicians for only $2.40 per word plus $7.00 for an address. Minimum charge: $25.00. Please underline words to appear in bold type and add $0.50 per every bold word. Please indicate clearly any words to appear in all caps and add $0.25 per every cap word. Each phone number, e-mail address, or website address counts as one word. Call for display rates. 3 months minimum schedule required. Deadlines are the 8th of the month, 2 months prior to cover date (for example, April 8th for the June issue, on sale in early June. All Classified ads are also posted on our Website (www.eqmag.com) at no additional charge. Businesses must list business name in ad. All ads must be received in writing, paid in full in advance. All ads must be music-related. Retail advertisers may not list discounted prices or percentages on specific models, unless items are used or discontinued. Advertisers must provide us with complete name, street address, and phone number, whether or not included in the ad copy (you may list a PO Box address in your ad, however). Mail ads to: EQ Classifieds, Attn: Darlene Labrecque 1111 Bayhill Dr., Suite 125, San Bruno, CA 94066. FAX (if paying by MasterCard, or Visa): (650) 238-0263. For more information, call Darlene Labrecque at (650) 238-0297; E-mail: dlabrecque@musicplayer.com. (*Audited circulation; does not include pass-along rate.)

Company Name _____ Contact Name _____

Address _____ City _____ State _____ Zip _____

Telephone _____ E-mail _____

Please print your ad clearly. Use a separate sheet of paper if you need more room.

_____

_____

_____

_____

(do not include address when counting words)

Category: ❒ Marketplace ❒ Duplication ❒ Talent and Employment ❒ Sounds/Sequences/Software ❒ Mixing/Mastering ❒ Instruments ❒ Accessories ❒ Gear for Sale ❒ Acoustic Products & Svc's ❒ Studio Furnishings ❒ Other

TO COMPUTE COST OF AD

#_____words x $2.40 = _____

#_____bold words x $ .50 = _____

#_____ ALL CAPS wds x $ .25 = _____

Address $7.00 = _____

Total cost per issue = _____

(minimum $25.00)

x number of issues to run x _____

Total payment = _____

❒ Payment enclosed, or Charge my

❒ Visa ❒ MasterCard

Card # _____

Expiration date: _____

Signed: _____

Maximize Your Music Production With EQ

Musician-focused solutions for players interested in recording and producing music

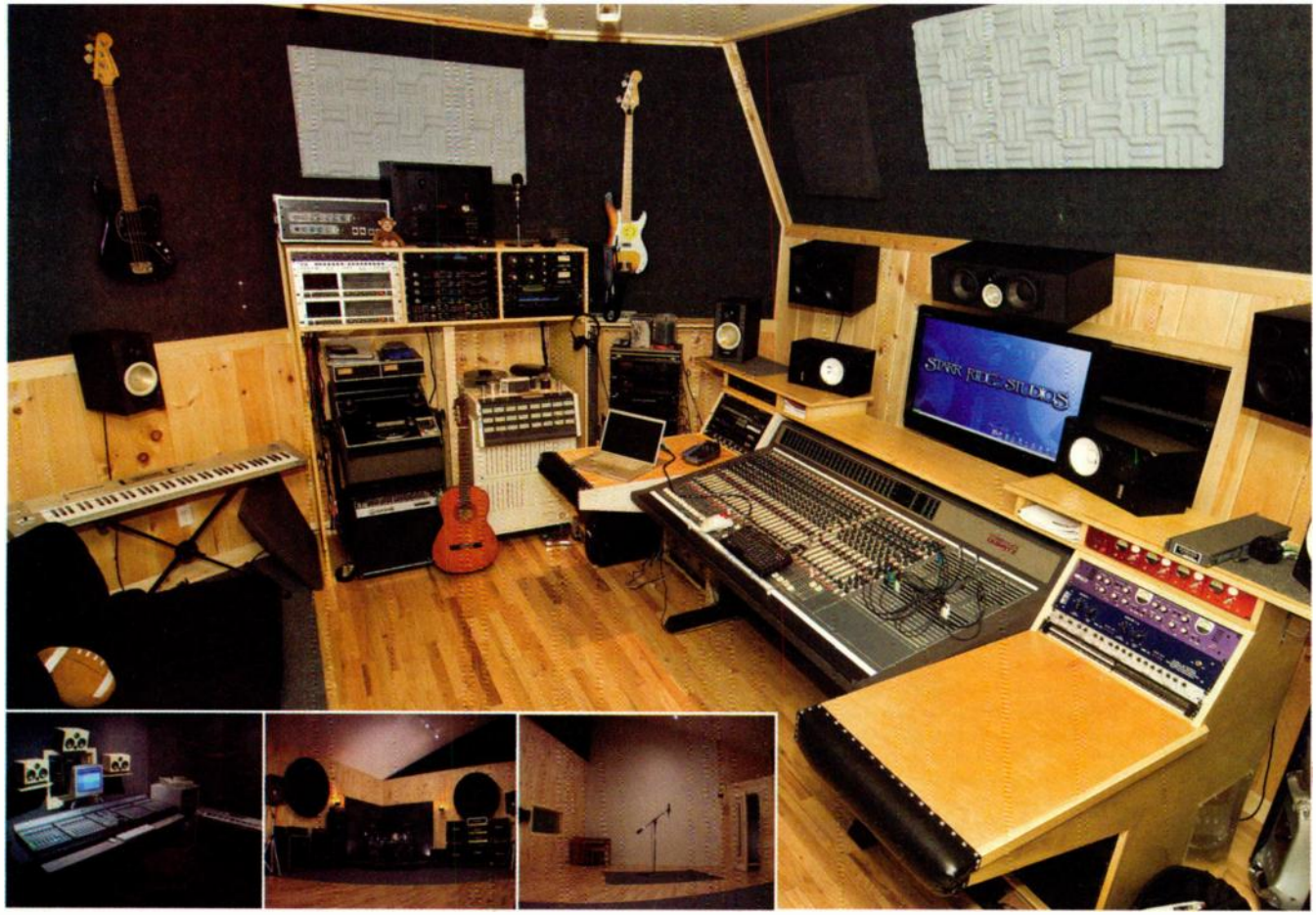
- Learn from the producers behind the sound of today's top artists
- Recording Guitar, Bass, Drums, Keys, and Vocals columns in each issue
- Punch In with new artists, session stories, and reader tips
- Mixing seminars, production concepts, and application ideas you need to make better recordings using today's music production tools
- Music production product reviews

If you are a musician who records, then you need EQ.

Get 12 issues for only $5!

Subscribe today at www.MyEQmag.com/mpn

www.EQmag.com www.EQmag.tv



ROOM w/a VU

by Salvatore Russo

STUDIO NAME: Starr Ridge Studios
LOCATION: Brewster, NY
CONTACT: www.starr-ridge-studios.com
KEY CREW: Anthony Croce, Kristen Koerner, David Powers
CONSOLE: 32-channel Soundtracs Quartz
CONTROL SURFACE: Digidesign ProControl
SOFTWARE: Digidesign Pro Tools HD3; Sony Sound Forge 9, Vegas
RECORDERS: Alesis XT20 (2), Ampex ATR-102, Mackie SDR24/96, Otari MX-80, Sony ES75
MONITORING: Dynaudio (custom mains), Paradigm 5.1 Signature Series, Yamaha NS10
MICS: AKG C 1000 (2), D112; Audio-Technica AT813a (2), AT4050 (2); CAD e300 (2); Electro-Voice ND308 (6), RE20; MXL 2010 (2), 2001 (2); Neumann TLM 93, U87 (2), U89; Sennheiser MD 409 (2); Shure SM57 (2)
MIC PRES: Focusrite Platinum Voicemaster, Red 1; Neve 1064; Vintech 1272
COMPRESSORS: dbx 160x (4), 166 (2); Focusrite Platinum Compounder; HHB Radius 30; Urei 1176 (2)
PROCESSORS: Alesis Quadraverb 2, Lexicon PCM70, Yamana SPX90
INSTRUMENTS: Fender Musicmaster; Gibson Flying V, Les Paul Custom; Hammond B-3 organ; Hardman-Peck Baby Grand piano; Ibanez 760, 7-string Universe; Korg Triton; Michael Kelly electric 6-string bass; Pearl 5-piece Export; Roland JX-3P
AMPLIFICATION: Behringer V Ampire LX112, Budda Superdrive 80, Fender 65 Super Reverb, Gallien-Krueger GK 800 RD, Marshall JCM800, Randall M-80, Soldano Hot Rod 50 Plus
NOTES: In the past 20 years ago, Westchester County's Starr Ridge Studios has grown from a cramped spare bedroom space—not unlike the one you and I probably started tinkering around in—to a massive two-story, 2,200 sq. ft. facility with 20-foot ceilings ripe for picking up great room tones. It's a virtual cornucopia for the recording musician—and this makes sense, as the studio was built to the specifications of chief engineer and certified guitar-geek David Powers, with those specifications largely being "a lot of room to hold a lot of rock."

Powers elucidates: "We specialize mainly in rock and metal—two genres that benefit from the big drum sounds we can get in the live room. But this space also allows us to track all kinds of 'large' acts, from choirs to jazz ensembles. It's especially good for the indie acts. Lower budget bands can accomplish a lot in their living rooms, but they need a space to really get their drum sounds. That's why they come here."

The Starr Ridge roster seems to be growing exponentially by the year, with the most notable of recent clients being London's post-punk heroes The Psychedelic Furs. But what brought them? Surely a band like the Furs could have found a good space to record drums in, I don't know, England or something?

"The allure for most bands is the live room," Powers continues. "We built a 5' x 5' vocal booth in the room, and isolated the drum platform. That way we could really make some nice-sounding live recordings, without compromising the vibe or negatively affecting our ability to separate the elements as needed during the mix."

Big room, hip gear, and a cool atmosphere to boot—how can you really go wrong? Especially given that so many of the nearby NYC studios have shut down throughout the years, it's refreshing to see that it's still possible to build a great studio to indulge your passion as an artist, while also lending a helping hand to the local bands.

HEY, *EQ* READERS. WANT US TO FEATURE YOUR STUDIO? SEND PICS AND INFO TO eq@musicplayer.com.

version 6

live

Also Available
Live 6 LE
Entry Level Version

Creative Explosion

This is Live 6, the latest version of Ableton's award-winning software that composers, producers, DJs and musicians worldwide have taken to heart. Live now includes a versatile, comprehensive collection of sounds ready to play and inspire - from faithfully sampled acoustic and electric instruments to impressive electronic creations. Pre-configured controls let you play expressively without worrying about technical intricacies, or, if you prefer, you can dig deeper and explore endless possibilities for creating your own unique and personal sounds.

Check it out at www.ableton.com.

ableton

MACHFIVE2®

Professional sampling evolved

MachFive 2 merges four worlds of advanced sound design: samples, loops, synthesis and effects. Now you can import, edit, synthesize, process and play multi-sample instruments, loops and phrases in one unified, intuitive window. 32GB (four 8GB DVD's) of instruments and loops are included. Load sounds directly from your favorite libraries, from Giga to EXS24 with no wasted time importing or converting.

Drag, drop and trigger tempo synced loops while playing sampled instruments. Edit samples in MachFive's full-screen editor with unlimited undo/redo. Switch to MachFive's unique LoopLab™ and edit beat-sliced loops down to the smallest beat. For sound design, music production and broadcast, MachFive delivers superb audio quality, powerful programming, fast operation and broad compatibility.

Includes 32GB of sounds

DVD 1 — Universal Loops and Instruments

DVD 2 — MachFive Concert Grand (8GB piano)

DVD 3 — 192K and Surround Instruments

DVD 4 — VSL™ Orchestra MachFive Edition

Direct sample library support

- Reads all major sample library formats directly
- No lengthy importing or converting needed
- Reads multi-layer "dimension" Giga libraries directly
- Reads native sampler discs & disc images directly

Across-the-board compatibility

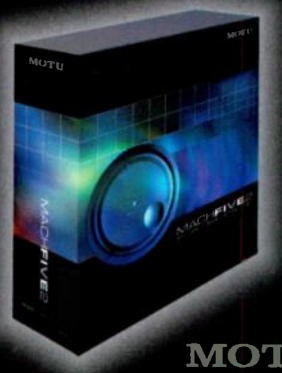
AIFF, SDII, WAV, ACID, Apple Loops, REX, Akai MPC, Akai S Series, E-mu, Ensoniq ASR, Ethno ufs/dat, EXS24 / GarageBand instruments, GigaSampler 1/2/3, Kontakt 1/2, Kurzweil K2xxx, MSi dat, Roland S700 Series, SampleCell, SoundFont, UVI Soundcards, VSampler 2

"An impressive production tool" — EQ

"We were floored" — EM

"The most powerful sampler ever" — Future Music

"Finest in the fleet" — Remix



MOTU
motu.com

Feature highlights

- Unique, unified design provides instruments, loops, phrases and synthesis in one window
- Unlimited parts (unlimited multi-timbral operation)
- Unlimited polyphony and ultra low software latency
- Unlimited sample keygroups and keygroup layers
- 200 audio outputs (with supporting host software)
- 256 MIDI channels (with supporting host software)
- Three-dimensional layered keygroup mapping with unlimited layers and dynamic, programmable rule-based layer switching (supports Giga dimensions)
- Graphic mixer with faders, effects inserts, graphic EQ and effects, VU metering and unlimited channels
- Disk streaming – can be enabled/disabled per part
- New modular synthesis architecture with two synth engines (organ drawbar and vintage analog) — program and play synth sounds side by side with, or as part of, your multi-layer sampled instruments
- Automatic keyboard mapping of imported samples via advanced pitch-detection and velocity algorithm
- One-click MIDI controller mapping for all settings
- Powerful modulation matrix with controller mapping
- Syncable envelopes with unlimited control points
- Drag and drop audio import/export
- Advanced time-stretching and pitch-shifting engine
- LoopLab™ beat-slice editing of multi-channel loops; playback at any tempo with or without stretching
- Full-screen keygroup/sample editor with unlimited undo/redo, batch processing and 47 DSP effects
- Multichannel surround support with multiple outs
- Plug-in or stand-alone operation
- Multi-column browser for fast and easy navigation
- 47 built-in effects with hundreds of presets
- Support for multi-effects with over 100 multi presets
- Flexible effects architecture — apply effects anywhere, from individual samples to your entire mix
- Convolution reverb for stunningly realistic spaces