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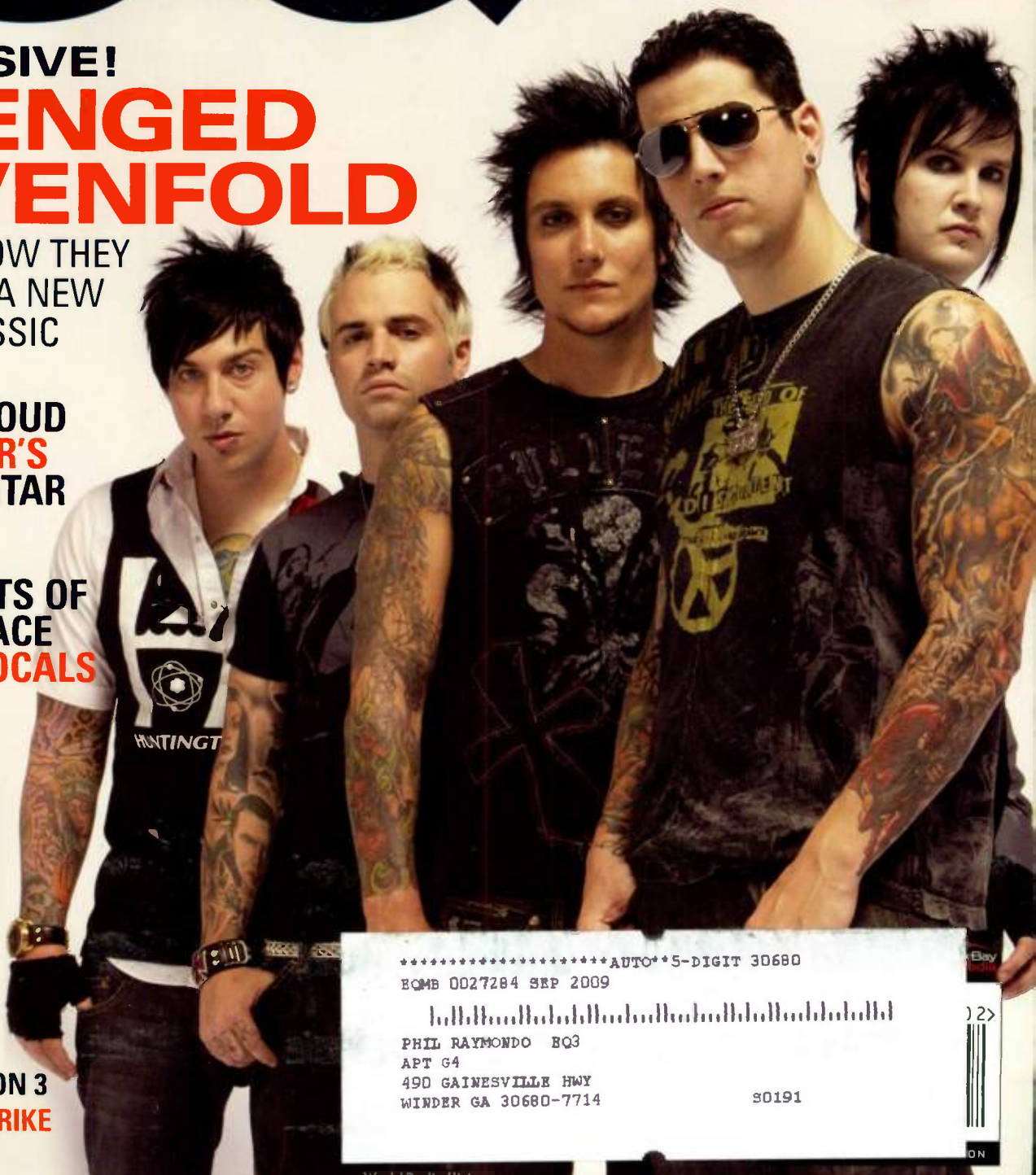

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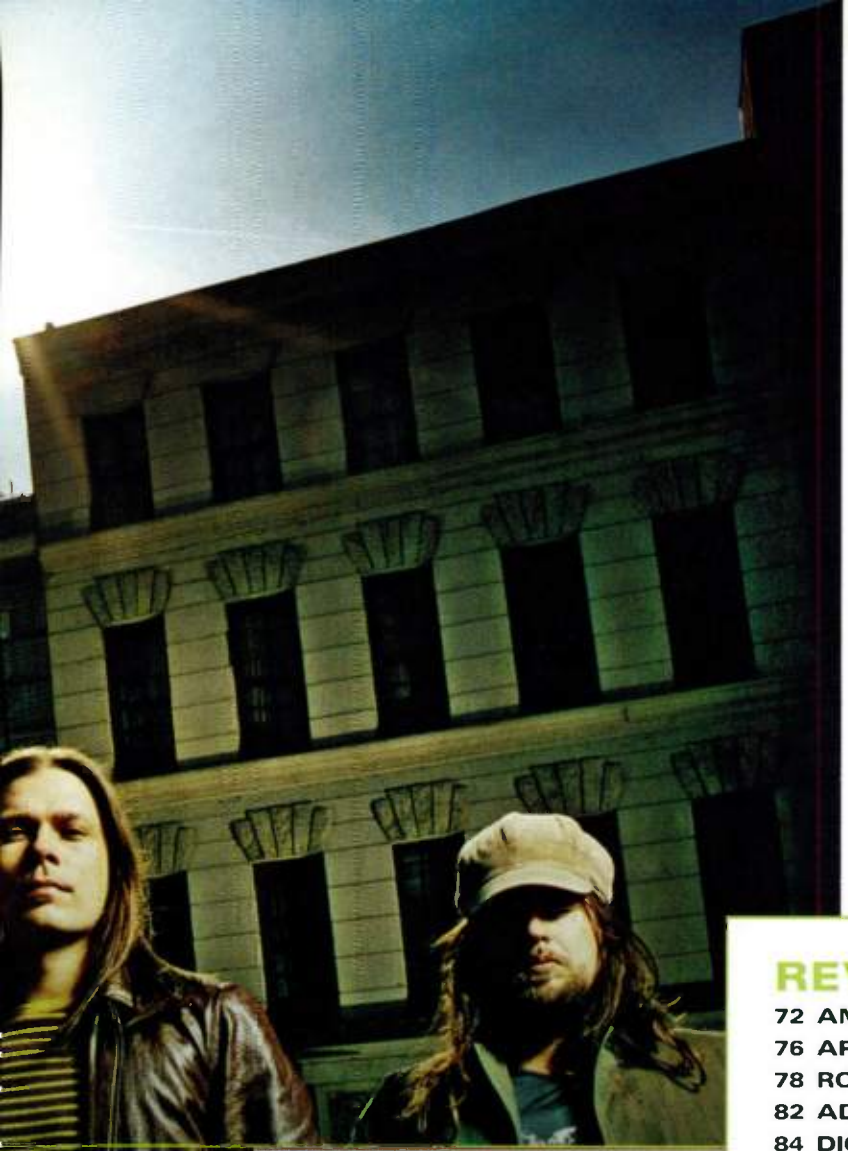

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"IT'S GOOD ENOUGH"—OR IS IT?

For those into computers as a spectator sport, Microsoft's Windows Vista has provided plenty of spectacle: It was hyped for years, delayed, had features cut back, then limped into a marketplace where not only had Windows XP become accepted as a reliable standard, but one where a resurging Apple was regaining traction. To further muddy the waters, Vista had one foot in the 32-bit world, and another in 64 bits. The new OS was greeted mostly with yawns; some computer manufacturers even rejected Vista altogether, and shipped new computers with Windows XP.

No, you're not reading *PC World* . . . there's a larger issue here. I think Vista, while not for everyone, has considerable merit. It's not the paradigm shift Microsoft had promised, but as XP's successor, it delivers quite a few new and compelling features. And regarding 64-bit operation, I've used Sonar 7 and several interfaces with excellent results (both WDM and ASIO).

Yet it seems a growing trend is to feel the status quo is "good enough," and people see no need for Vista because "XP is good enough" (and for most people, it indeed is). Similarly, while some use 96kHz sampling rates, for most people 44.1 or 48kHz is "good enough." And sure, while PCIe soundcards are faster, PCI ones are usually "good enough."

But we need to be careful not to let "good enough" keep us from implementing genuine improvements. Consider the audibly superior SACD format: The standard Red Book CD, and MP3s suitable for loading into portable players, were "good enough" and as a result, we're stuck with lower quality. VHS was "good enough" compared to the better Beta format, and American cars were "good enough"—until the Japanese showed us they really weren't.

There's no need to embrace new and different technology just because it's new and different ("if it ain't broke, don't fix it"). But in the process of being rightfully skeptical of every "latest and greatest" thing that comes along, be careful not to reject something that *does* have value. It's important to look at, say, Vista with an open mind to determine whether it offers features you might find useful, then weigh the pros and cons (and there are plenty of both). Ditto 96kHz recording: Try it, and if you can't hear a difference, no harm done. But if it does sound better, then it's worth the extra hard disk space, computer resources, and time spent vetting which plug-ins work well at higher sample rates.

The bottom line, as always, is "don't believe the hype." Just remember that sometimes, the hype really is true.

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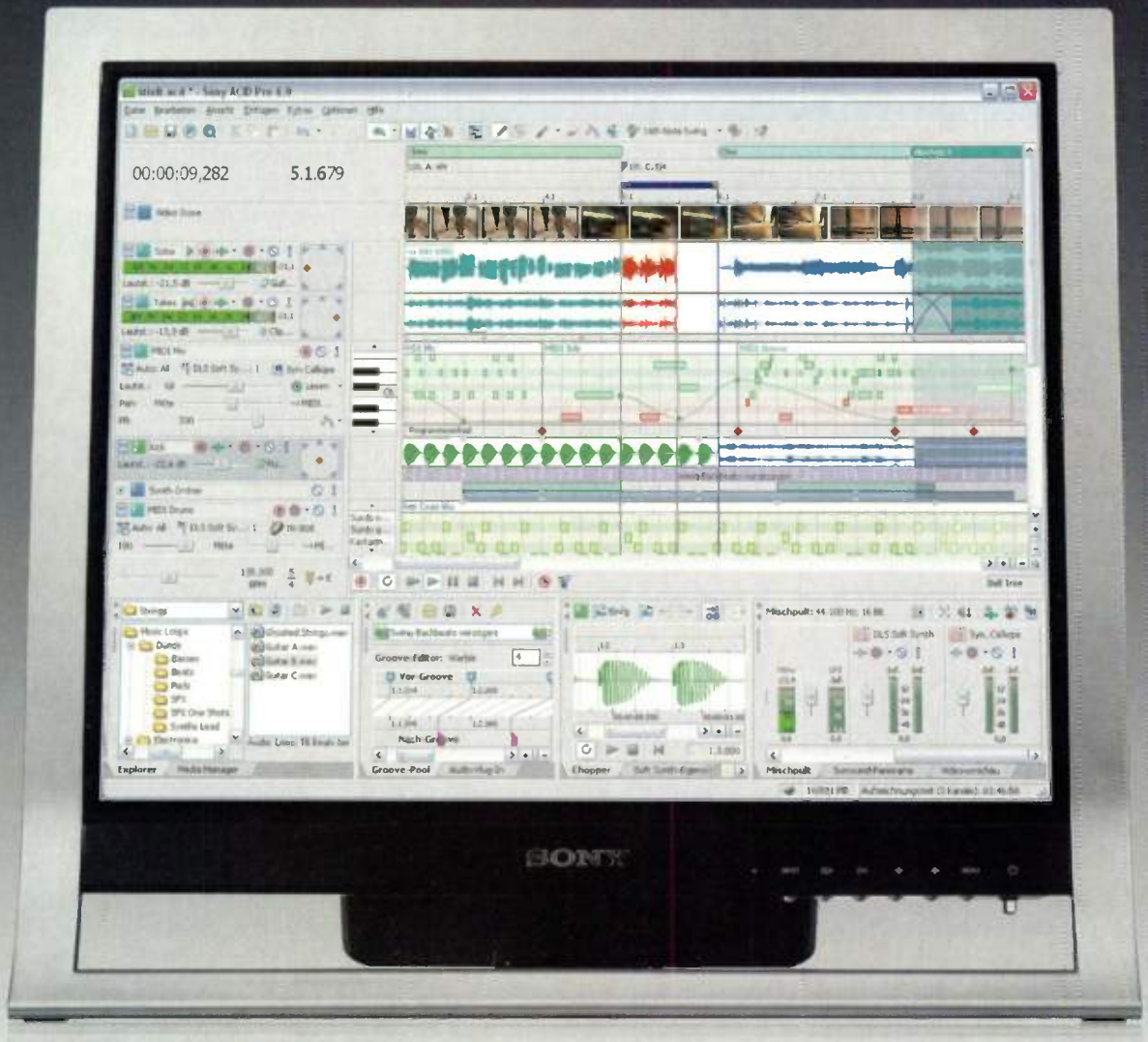

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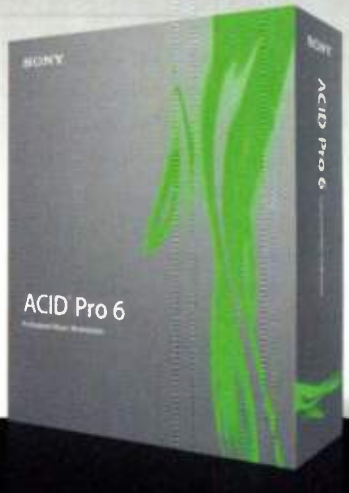

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Awesome piece on Pink Floyd ["Shine on You Crazy Diamond," 12/07]. I couldn't believe that Jeff Touzeau got Norman Smith to talk! Hasn't he been a recluse for the past 20 years? Talk about a great Christmas present! I was stoked when I opened my mailbox that day.

You guys are the best recording magazine around—hands down!

Alex L. (via email)

KISSING THE SKY

I just wanted to add one more tip to Craig's flanging ideas, being a flange junkie then and now. While a plug-in flanger can't achieve that magic zero point of cancellation because the delayed sound can never get ahead of the source sound in real time, instead of using multiple plug-ins and millisecond adjustments as described, a possibly easier method is to duplicate the source audio file and speed up the copied file's pitch (one to three cents is all you need). Just make sure that any pitch changing you do to the copied file does not have time-correction or harmonic correction enabled. (Pro Tools users should ensure the time correction checkbox is off; Logic users, stay in "classic" transposition mode.)

Line up both tracks to start at the same time on your DAW, set them to the same output, and play. You'll instantly hear the zero cancellation effect at the start. By delaying the duplicate track by a few milliseconds you can control exactly where you want the "going in—zero—going out" tunnel effect in your music. Split the region and delay it, then you can do the same thing again and again anywhere in the song. By using crossfades on the dupe track you can shape and control the entrance and exit of the flange. And whether you use the dupe track in phase or 180 degrees out of phase, each method will give you a different but equally wild result. Finally if you're working with stereo material, then pan the dupe pair to mono or even swap left and right. You'll get some dizzying Doppler effects!

Hope this helps. That was a great article—long live *EQ*!

Amin Bhatia (via email)

HE GETS IT

I've been getting *EQ* for about a year now and I've noticed the magazine improves every month. The Pink Floyd issue [12/07] had a number of excellent articles such as Bass Management ["The Evil Bliss of Butchered Bass Tones"], Key Issues ["10 Ways to Exploit Your Oscillators"], and Drum Heads ["Drum Fun with Vintages Boxes"]. I particularly appreciate these types of articles, as they provide real insight into how to get the most out of my equipment.

I also get two other recording magazines [*Note: Mark's list has been edited out because we are gentlemen here—Editor*]. Those magazines seem more concerned about trying to sell me new equipment, rather than trying to help me use what I already have.

It's because of your helpful articles that I now look forward to receiving *EQ* far more than those other magazines.

Keep up the good work!

Mark (via email)

SOMETIMES EVEN FACT-CHECKS FAIL

I just read the K + H review in this month's issue [01/08]. It's a great piece and I thank you for the coverage. However, there are a couple of points I wanted to clarify to your readers.

EQ wrote: "The horizontal form makes it easy to get your ears at tweeter level, though K+H tells us that doing so narrows the sweet spot significantly."

Sadly, this is inaccurate. The O 300 is specifically designed for horizontal use, and the elliptical shape of the waveguide loading the high frequency driver is designed to promote maximum "sweet spot" width on the horizontal plane and minimize vertical dispersion to avoid early reflections off the console surface. Mounting the O 300 vertically, rotating the box by 90 degrees (which would not be suggested under normal circumstances), would narrow the sweet spot significantly.

The cabinet of the O 300 is also unported, contrary to the review. This design point plays a prominent role in the accuracy and detail delivered by the O 300.

Finally, to address the pricing, the largest contributor to the cost of these speakers is in the quality of components, materials, and engineering. These are high-end professional tools and accordingly are designed and manufactured to the uncompromising standards of the situations in which they are used. No corners are cut. While the exchange rate certainly plays into the final cost of the O 300, by no means do we feel that it has priced it out of range with competitive models of professional grade three-way near field studio monitors.

Our sincere thanks again for the coverage and thank you for your time.

Dan Radin, Product Manager,
Klein + Hummel Studio Systems

BACK TO THE BASICS

I am a player and subscriber for many years and applaud the publication, specifically the new upgrades and additions. I would really like to see more "how to" 101 articles such as a "MIDI 101," or an ongoing refresher course for those of us who have either forgotten, lost a brain cell or two, or just need keep keeping up with new technologies so we can better interface the new with the old.

Ricci (via email)

Got something to say? Questions, comments, concerns? Head on over to www.eqmag.com and drop us a line in 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 possible inclusion in the Sounding Board.

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WORK FLOW

AESOP ROCK'S HOME RECORDING METHODS

by Merrick Angle

Coming out of the same scene as Cannibal Ox and Company Flow, Ian Bavitz, *a.k.a.* Aesop Rock, writes and produces his own dense slices of hip-hop, and is generating quite a buzz doing so. With his gravely "Noo Yawk" baritone voice and his loping beats, Bavitz is clearly approaching hip-hop from a different angle. And the kids love it. But Bavitz, for all his critical acclaim, sighs when the subject of recording is brought up.

"Recording music, to me, is a deep process," he says from his home in San Francisco. "I get caught up in the details. I comb over songs for a long time—probably too long. I think I got that from art school, where I was taught to constantly re-look at my work in an attempt to see it in a way that I had never seen it before. Because of that, I'm never afraid to scrap a beat or a lyric, or to cut a song down and then rebuild it. That's a blessing, but it's also a curse."

Bavitz's latest release, *None Shall Pass* [Definitive Jux], is the first recorded in his new studio digs, but it doesn't lack the fidelity of his past albums—all recorded in decidedly more posh surroundings.

"I just moved all of my old gear into a room in my new house, and started laying down tracks like nothing had ever changed," he says. "I wanted to try out a new mic, so I splurged on a Neumann M 147. I fell in love with it—it's so warm sounding. But, besides that, it's just my old gear, and I'm recording my old way."

Rising at the crack of noon, coffee in hand, Bavitz says he starts his daily ses-

CHRISSY PIPER

Aesop Rock, sans sleep, after a late-night session.

CHRISSY PIPER

Breaker-breaker 1/9—Aesop Rock rushing home for a session.

sions by digging through his crates of records, sampling source material into his Ensoniq ASR-10.

"I'll always start with a drum loop run off a Technics 1200 turntable into my ASR-10, which I then use to trigger and manipulate the beat," he explains. "Once I get something I can work with, I put the loop into Pro Tools, because you're working with more than 2MB worth of memory there. I could never cut the ASR-10 out of the loop, though. It's much more intuitive for me to construct the beat on the ASR-10 than it is to cut and paste in Pro Tools."

The next step for Bavitz is recording his bass lines, which he does with "an old Fender P" into a Radial JDV Super DI through a Mackie Onyx 1640 16-channel mixer. After laying down his rhythm section, Bavitz spends the rest of the time writing lyrics to match his loops.

"I can be pretty scattered, so I make a point to sit and write to a loop for the rest of the day, and into the night. Sitting down and just doing it is the only way it will ever get done."

Thankfully, the majority of Bavitz's gear manages to take up just a few corners of his bedroom.

"A Mac G5, a Roland Fantom, an ASR-10, a Technics 1200, the Mackie board, and my Event 20/20 monitors are pretty much my entire set-up," he says. "But I put a WhisperRoom [isolation booth] in the middle of the room, and ran an extra computer monitor and a mouse into it. That way, I can engineer and perform at the same time, and I'm not rapping in an office chair."

Although Aesop Rock is certainly part of the "personal studio overground" that produces hip-hop and dance albums at home, he is scrupulous about not following the crowd.

"There has always been a norm in the hip-hop mainstream," he says. "Nowadays, it's all Roland TR-808s and fake-ass soft synth strings. I'm not into all that. Thankfully, there are a lot of people now who are trying more interesting things. You have to dig deep to find them, but they are out there. They are not just taking a canned beat and rapping on top of it. They are using Theremins, collecting found sounds, and playing the instruments themselves. Besides the drums, I make sure everything else is live. I like blending the samples with the live elements to get a hybrid effect. I like to be able to actually compose a song—even if I'm sampling. Don't get me wrong—I appreciate a great loop as much as the next guy, but I'll never leave a loop untouched. It's too much fun chopping it up, and adding in your own instrumentation. That's what makes the difference, I think, in songs that get played once, and songs that get a few rewinds." EQ

PUNCH IN

OUTFOXING YOUR LIMITATIONS

Ian Eagleson's Tips for Tracking Big Sounds with Budget Gear

NOEL KUPERSMITH

Extra Golden (left to right)—Alex Minoff, Opiyo Bilongo, Ian Eagleson, and Onyango Wuod Omari.

by Craig Reynolds

There's more than one way to bake a cake, but you don't have to be a world-class chef in a state-of-the-art facility to cook up some tasty treats. Ian Eagleson—co-founder of the band known as Extra Golden—knows all about maximizing what you've got, and recording great-sounding albums on a budget. Tracking the entirety of Extra Golden's newest foray into the world of American rock and Kenyan Benga dance music fusion, *Hera Ma Nono* [Thrill Jockey], at his portable "Nyathi Otenga Flying Studio," Eagleson is an ardent believer in the power of low-budget gear.

"My entire set-up is a PC running Sony Vegas, an RME Multiface II, a Mackie 1604-VLZ3, and just a handful of mics," he says.

Here are his tips for making the most of what you have.

EXPERIMENT WITH WHAT YOU HAVE

"When I was recording on location in Kenya—and working on my doctorate in Ethnomusicology—every session was totally improvised. I'd be working in very remote places—at times having to bring a generator to even track. And I could never bring much gear, so I had to make what I did have work. I learned a lot from those times, so, on this album, when I was in a similar position, I used an old Electro-Voice PL11—which is designed as a broadcast mic—to record the drums. I only had a PL11 and a Shure SM57 to work with on the source sounds, so I put the PL11 inside the kick, as it could translate the low end, and the SM57 on the beater side of the head to get the attack. Then, I just blended the sounds together. It came out great. You'll probably never see anyone else using those two mics to record a kick drum, but they work if you just apply them creatively."

WHEN IN DOUBT, RECORD DIRECT

"Whilke traveling around Kenya, it's not like I could bring a ton of guitar amps to get the perfect sound. So I recorded everything I could direct. Afterwards, I could treat the signals however I wanted with plug-ins. In my opinion, it's more important just to get the performance, than it is to worry about the sound. And it's better to get a nice, clean, and malleable direct signal, than it is to get stuck with a bad sound due to poor miking techniques."

GO SMALL

"If you're recording in your bedroom, you don't need to spend thousands on a 100-watt Marshall stack to get a good guitar sound. For this last album, we recorded the majority of our guitars in a small room using a Z.Vex Nano Amp, which is a really cool half-watt amp that lets you get a very thick, overdriven sound without turning it up loud. It was great for recording in a small space. So use little combos. They'll save you cash, and, if all else fails, you can always reamp later. EQ

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JOYFUL MISERY

TOCA and Producer Colin Fairbairn On How to Make an Album the Hard Way

by Shane Mehling

No one ever said that making a record is easy. TOCA—an L.A.-based band that seamlessly combines indie rock, hip-hop, and Latin music into a refreshing musical gumbo—can attest to that fact; as can their producer, Colin Fairbairn, who was tasked with helping the band assemble their self-titled album from over 50 songs, culled from five years' worth of various lo-fi home recordings. "It was hard," Fairbairn modestly states. "Really, really hard."

"These songs are mostly from 16-track demos recorded at our rehearsal space with just a handful of [Shure] SM57s and 58s," admits Ceschi Ramos, TOCA's guitarist and primary songwriter. "After five years, we decided to take them to Sounden Studios and get Colin to help us put them all together. I don't think he knew what he was getting into."

"About 60 percent of the album is rehearsal space recordings," Fairbairn adds, "but those recordings were already huge on the production front. We're talking upwards of 30 vocal performances, bounced down to four tracks, that we had to then fit into the re-recorded tracks from recent sessions."

Ceschi Ramos recording guitar at Sounden.

These re-recorded tracks, Fairbairn says, were originally planned to be just the drums, bass, and guitar, though the band ended up bringing in everything from a Fender Rhodes to "a bunch of weird, cheap drum machines," running the signals through a Korg KAOSS pad and manually sweeping the panning parameters to further twist their already significantly twisted sound. "It was a lot of fun to track, but we ran out of time at Sounden and had to retreat to this small, Guitar Center-built studio that only had a Pro Tools HD 2 system," Fairbairn laments. "It would constantly crash because our track counts were so high. We were murdering that system. And, to top it off, the place didn't even have a console, so we had to mix completely in the box. I've never done that before, and I hated it—going from a Neve 8026 to a laptop. It was a learning experience, but I never want to go through that again."

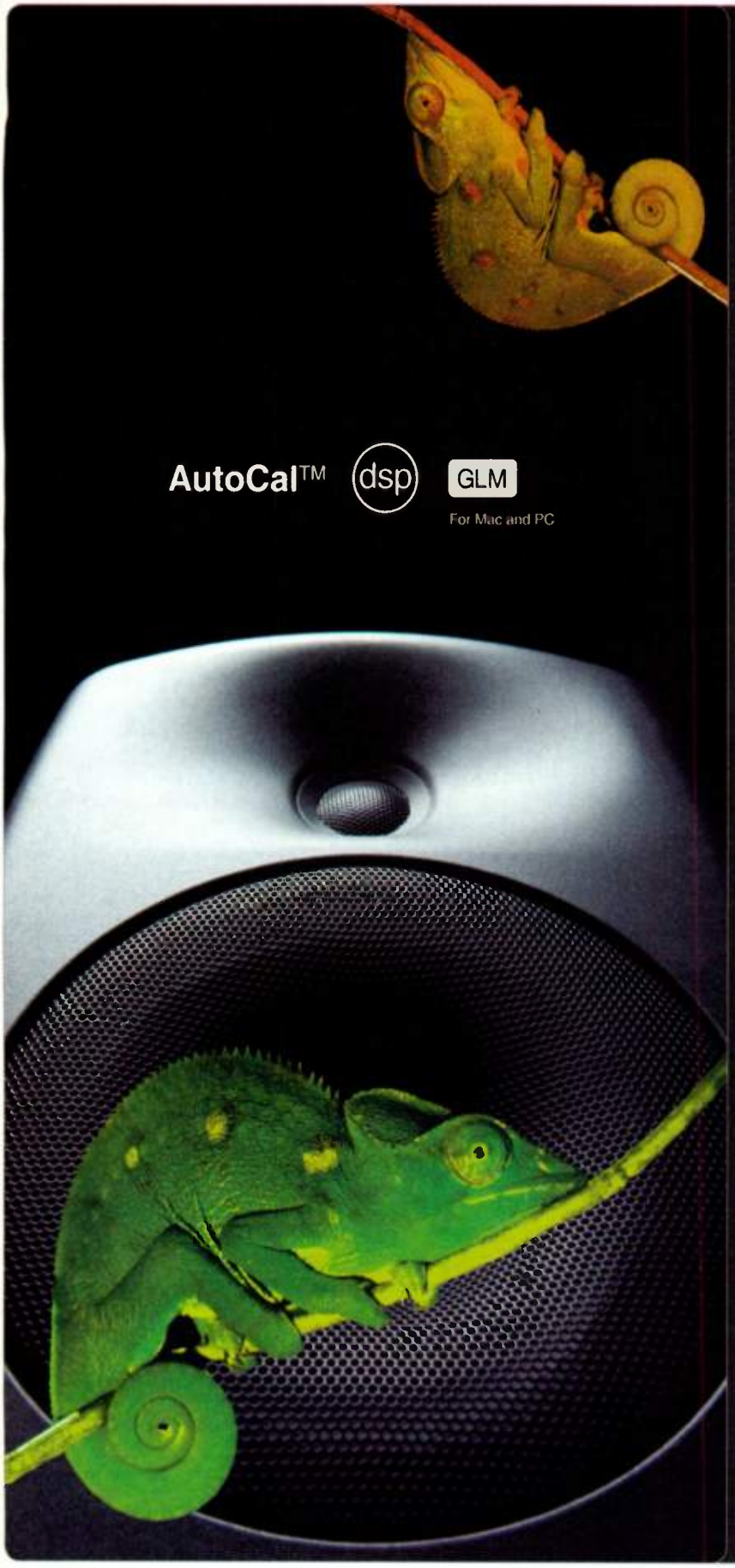
The basic tracks that the band recorded at Sounden, however, were a joy to engineer, Fairbairn admits. "I had a lot of fun with the drums. I got to bring in my old '60s Rogers Holiday kit for David [Ramos, drummer/percussionist/vocalist], which is always good for a real garage-y sound."

To capture the kit, Fairbairn applied one of his favorite miking strategies. "I used two AEA R88s as overheads," he says. "Since they are ribbons, they naturally give a darker, warmer sound than, for instance, a pair of [AKG] C12s. We then put a [Sennheiser] MD 441 on the snares. We used anywhere between five and eight snares for each song, layering the tracks to get a really fat, thick snare sound. The rest of the kit's sounds were all from a pair of [beyerdynamic] M 160s that I set up about 12 feet back from the kit, running the signals through a pair of Neve 2254 compressor/limiters that were crushing the tracks. Compressing the hell out of room mics is a great way to get a real nasty drum sound. If you take that sound and use it as your main tracks, just adding the snare and overheads in a bit in the mix, you can get an awesome lo-fi drum sound out of your kit."

David Ramos behind the drums for the TOCA sessions.

When it came time to re-record Ramos' guitar tracks, Fairbairn describes the process as "a mad dash to get the perfect sound." With the clock ticking, Fairbairn quickly set up a Royer R-121 in front of a Silverface Fender Vibro Champ, laid on its back, and a Peavey Classic 50/410. "I'd split them out. I would use a pedal, flip the signal to the two amps and do stereo modulation," Fairbairn informs. "I have this Electro-Harmonix CloneTheory pedal with a dry out and chorus out, and the dry out went to the Classic, the chorus out to the Fender. Beyond that, Ceschi just grabbed the knobs, gave each amp a quick tweak, and we just laid down all the guitar tracks.

"We were still arranging the songs when it came time to mix," Fairbairn adds. "We knew we had the drum parts, so I'd put together a submix of the drums, drop the faders on every other track, and then we'd just build the songs up around the basic rhythm tracks, editing as we went. It was a difficult process for me, since I'm used to finishing a mix of a song in less than six hours. This album, however, is just a series of revisions and overdubs, and we spent days on each song just finalizing the compositions. Psychologically, it's a hard way to work, but I feel like I'm better at recording now for it. When I listen to that album now, knowing what went into it, I know we're all very proud of what we did." EQ



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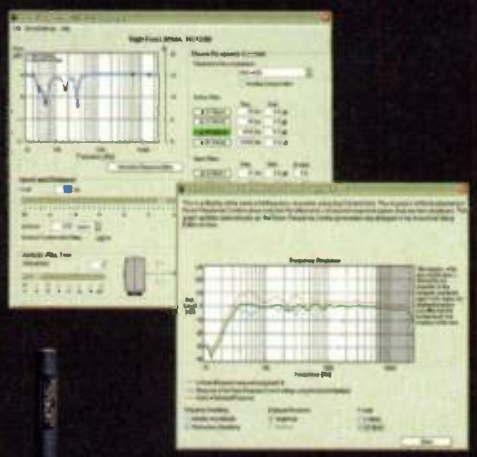

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THE GLORIOUS RIGHTEOUSNESS OF OPTIMISM & REASON

The Music Industry is Growing. Record Labels Will Live. Really.

by Mosers Avalon

Now that we've rang in the New Year, I'd like to share a story with all you recording musicians. The short of it is this: At the beginning of 2007, I made my lawyer a bet. He's a tough litigator with one of the best track records for winning cases against bastard major labels. But, just because I love him like a big brother who is always right, doesn't mean that he's actually right all the time. So I bet him lunch that CD sales would not drop more than 20 percent this year (he thought 50 percent), and that this drop would be more than offset by other revenue in the music space. He laughed, informed me that he eats very expensive lunches, and said that he was already making the reservation at Chez Second Mortgage on my dime.

See, he subscribes to the ultra-technocratic viewpoint that CD sales being down and widespread usage of P2P services is the death knell for the major labels, and that technology (mostly net-based) will provide answers for artists that will "obsoletize" the need for large labels and their ridiculous deals.

I have been saying something a bit different for some time: It's very possible that labels could get completely absorbed into technology companies. If it happens it would be another decade or so. But the most likely scenario is that labels will eventually, albeit slowly, embrace digital distribution, and provide the same services (and ridiculous contracts) they always have to artists: marketing, promotion, and security in distribution. They will grow bigger and more powerful via the Long Tail philosophy, and the business will be making more money than ever. And so will the artists.

But it seems that we have some very pathologically pessimistic people in our space, people who want to believe that everything is going to hell in a Hummer. They jump on this "the business sucks" bandwagon as if it's taking the scenic route to Graceland. I've covered the reasons why we have these misanthropes in other articles, so I won't bore you here with a reprise.

I have always maintained that a drop in CD sales is not really an indicator of anything much except a changing business model. For this, I have been ridiculed to no end. But that's okay. Being right tastes even sweeter when it seems the world was against you from the beginning. Of course, no one ever remembers that you were the first to say it—or admits they were wrong. Not unless you put it in a magazine, and shove it in people's faces.

No matter. Just a month back, I won my bet.

Chris Anderson, author of the most talked about new theories of commerce and economics, *The Long Tail,* posted a bit of good news for those out there still mourning the death of the music business and whining over the dip in CD sales.

In a nutshell he says this: "Believe it or not, the music business is growing."

Actually that's not even a nutshell. It's exactly what opens his thesis, which can be viewed online at: http://archive.mediaor.com/post/16622264.

Now, I know my lawyer, he's a very good arguer. He'll come up with some contorted logic about how this is irrelevant. He'll say: "But labels are firing tons of people." He'll ignore the fact that they are hiring far more than are being fired—in the marketing and licensing departments.

Yes, it's true. The firings at labels are mostly in the A&R department. How can this be a bad thing? Wasn't it just a few years ago that most independent musicians (and those scorned by the majors) were saying they should fire most of these good-for-nothings? Now it's happening, and tech-loving spinmasters have turned *this* into bad news.

Many big-firm lawyers argue that they know the business is falling because their billable hours are down. In other words, *they* are making less money, so therefore things must be bad. They often forget that a $600 an hour lawyer is something of a luxury to musicians and record guys alike. One of the first things that any industry will look to do when it's in transition (notice that I didn't say "decline") is cut back on overhead, and that often means legal bills. Cutting back to $300 an hour attorneys is not the end of the world, fellas.

We're coming out of the fourth quarter of 2007 as this article goes to print. The worst is behind us, and the holidays are staring us in the face. An 18-percent dip is as bad as it will get for now. This brings our total "downturn" adjustment to about a 38-percent drop in CD sales over the last four years. This is almost the same as 1991's sales levels for CDs—a very good year for the music business, even without all the new digital revenue. Think this is bad? Talk to the cougars on Wall Street. Not too many weeks ago, the Dow Jones dropped about 300 points—almost a three-percent loss in a single day. You don't see them saying it's doomsday. They know that things are cyclical and will rebound. We music types sure could learn a thing or two from them in terms of optimism.

So, who owes whom lunch? Me? I'm making my reservation at Chez Kiss-My-Ass right now. And I'm ordering two desserts. EQ

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The KS3 is a large-diaphragm, omni-directional mic with a low self-noise of 14dB-A, dynamic range of 132dB, and max SPL of 173dB. An external control unit has both mic-level and line-level outs so it can be used with a preamp or connected directly to A/D converters. www.md-mics.com

■ SIMS AUDIO PCI RECORDING INTERFACE

The Infrasonic Quartet ($199) is a 4x4 PCI audio interface that supports sampling rates up to 192kHz, with 24-bit recording. In addition to line level, digital, MIDI, and word clock I/O, the Quartet features a mic preamp with +48V phantom power and low pass filter. Also new: The PRIMUS a25 ($270) 25-key MIDI/USB controller/audio interface with both line and mic level inputs; and the BLOW5D ($550) studio monitors with on-board D/A converters that support up to 24-bit/192kHz audio. Room compensation controls and a bass response switch allow tweaking the monitors' frequency response. www.simsaudio.com

■ WWAYM NWRCFIL2 FILTER VST PLUG-IN

The NWRCFil2 ($88.11) VST plug-in is based on classic hardware filters using selectable 2-pole and 4-pole filter algorithms. It features crossfadeable lowpass, bandpass and highpass modes, an envelope follower with adjustable attack and release parameters, and LFO with MIDI trigger support. www.wwaym.com

■ TASCAM DP-02 PORTASTUDIO RECORDER

The DP-02 all-in-one studio contains a multieffects processor, two-band EQ per track, built-in chromatic tuner, and internal metronome. The unit records to an internal 40GB hard drive, has a built-in CD-RW for CD masters and song backup, and utilizes USB 2.0 for transferring tracks, mixes, and backups to a computer. www.tascam.com

■ M-AUDIO USB 2.0 AUDIO INTERFACE

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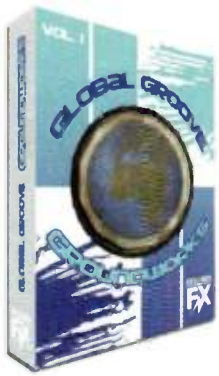

■ POWERFX GLOBAL GROOVE LOOP LIBRARY

The Global Groove Groundworks Vol. 1 percussion loop library ($69) contains 396 RMX compatible REX2 files for Reason, Stylus, and all other software that supports REX2 files. These indigenous percussion loops from Middle Eastern and African countries feature instruments like the darbuka, djembe, tabla, bendir, udu, ganjira, mridangram, and more. www.powerfx.com

■ IK MULTIMEDIA ARC SYSTEM

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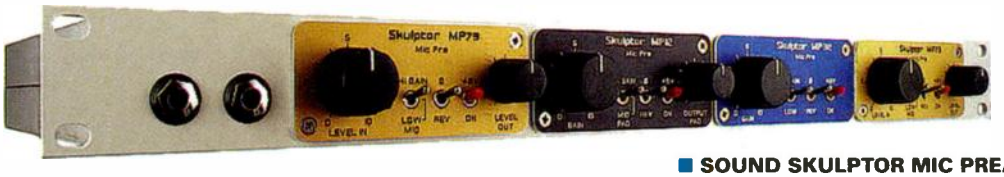

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The M-16DX digital production console can handle 18 inputs and features 24-bit/96kHz processing, a 3-band sweepable EQ, and three effects. With v2 software, the M-16DX mixer can also serve as a software control surface. www.rolandus.com/edirol

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Acoustic Feedback ($99) is a VST/AU/RTAS plug-in that emulates the feedback effect that occurs between the guitar and amplifier. The effect responds to vibratos, bends, slides, and tremolos, and is compatible with foot controls via MIDI. www.softube.com

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A MOTLEY CREW

BY MERRICK ANGLE

Above: The Book of Knots (from left to right)—Matthias Bossi, Tony Maimone, Carla Kihlstedt, and Joel Hamilton.

Below: A tasting of some of Studio G's outboard arsenal.

"With this band, we try to do everything wrong," proclaims guitarist/producer/engineer Joel Hamilton from the confines of his control room at Studio G Brooklyn. An established sonic wizard in his own right (his résumé includes everyone from Tom Waits and Elvis Costello to Sparklehorse and Unsane), Hamilton is speaking to the subject of practicing chaos theory in the studio with The Book of Knots—a collective that's highly regarded in underground circles for its highly experimental and hard-edged style.

While based around Hamilton and core members Tony Maimone (Pere Ubu), Carla Kihlstedt (Tin Hat Trio, Sleepytime Gorilla Museum), and Matthias Bossi (Skeleton Key, Sleepytime Gorilla Museum), the "band" is a seemingly endless revolving door of guest musicians—all of whom are subject to Hamilton's admittedly reckless tracking techniques.

"I love sabotaging myself by limiting my choices to just one or two microphone makes for the majority of sources on an album," he says. "I think it gives each project a fingerprint. And I'm not afraid of using mics that are less than ideal on a session, either. I'll toss in some crappy mic that I don't mind dropping. I'll throw it behind the piano, or put it behind the floor tom. Mic placement is not premeditated. I won't sit there listening and scratching my chin while the drummer is playing in the room. I will literally walk in and drop a mic on the floor, just to see what I get in the mix."

Harvesting spontaneous sounds is the name of the game on TBOK's newest album, *Traineater* [Anti-]. But, according to Hamilton—who produced the album entirely at Studio G through his newly acquired Neve desk ("Thirty-two Neve 33114s in a 5316 frame, eight buses, and two channels of 33314 compressors built right in")—gear-geekery is also a vital component in achieving The Book of Knots sound.

"The more obscure the gear, the better," says Hamilton. "I love using pieces that distort the incoming signal until it's almost unrecognizable. Take the McMartin limiter, for instance. It's this incredibly strange piece of equipment that was manufactured in the Midwest in the '70s. I don't know whether the person who made it actually *liked* audio, because it seems to destroy everything it touches!"

A good example of this unit in use is on the intro of "View from the Water Tower" where the music sounds "broken."

"The speakers just shake with this insane amount of intermodulation distortion—like somebody hit a gigantic vibraslap in the room," says Hamilton. "That's the McMartin limiter coming up to speed on a parallel bus across the drums and bass. It sounds as if the whole recording system is just reeling from the impact of that downbeat. I wanted to give the impression that we were so huge that recording technology

Joel Hamilton, producer/engineer and guitarist for The Book of Knots.

JOEL HAMILTON AND COMPANY CALL IN THE TROOPS FOR THE BOOKS OF KNOTS' *TRAINEATER*

couldn't even handle our giant gestures. I'll also use the SPL Transient Designer just to have one mono fader that sucks up all the air around the drums. The image almost sounds reversed when you solo the drums, as the SPL totally reshapes the envelope. Couple that track with a reverb, and it gives the source sound the illusion of being more focused and punchy than it was when it was tracked. Another thing that works well is the Thermionic Culture Culture Vulture—which is basically an incredibly expensive distortion box. It can create everything from an old Ampex tape machine-esque sound with all that rich, harmonic lushness to an overdriven guitar amp that's powering down. I typically use it across a drum bus, and it sounds just elegant."

Essential to the end product is the previously mentioned slew of high-profile guest contributors. The roster on *Traineater* includes bassist-extraordinaire Mike Watt, Mr. Bungle/Secret Chiefs 3 wünderkind Trey Spruance, and Norman Westberg of SWANS fame. Perhaps the most noteworthy contribution on the album comes from Tom Waits, who adds his papal seal of approval in the form of a demented preacher vocal on "Pray." Collaborating with Waits was a Waits-ian process—meaning no trips to a proper studio to track, no live link ups, and no gear created post 1970.

"Tom requested a 4-track cassette," Hamilton reveals. "There is definitely no FTP business with Tom—not unless someone is making a wooden laptop now [*laughs*]. I made him a mono submix on track one, and he sang on track two. Then, he and Kathleen [Brennan, Waits' wife and co-writer] sang the back-up vocals on track three, and he played guitar on track four."

Hamilton decided to record first, and worry about context later when it came to the album's guest musicians, which left the band with the Herculean task of editing down tons of source material.

"Sticking with my 'do everything wrong' work ethic, we wound up with 98 tracks for me to edit," says Hamilton. "The Mute button became my best friend for the first 15 minutes of the mix, as I looked for emotional cues from which to assemble the story, or to frame a lyric or a riff we recorded. We don't think very much before we hit Record. I like it that way."

Not surprisingly, TBOK songs go through a series of evolutions before the band settles on a final structure.

"We'll hack out entire parts of a song, or rework them," Hamilton says. "I have mixes of the record where a couple of the songs are unrecognizable from what was ultimately released—apart from maybe the drums and vocals. As far as I'm concerned, a mix isn't done until something reaches out and grabs me—until I forget I am the one who is playing on it. When a song makes me shut my eyes and not make any mix moves because it's making me think about something greater than where the kick drum is sitting—when it speaks to me as a piece of art instead of a series of technical issues—*that's* when a track is done. If you spend 17 hours tweaking some parameter in Pro Tools, then you have already lost perspective. Making an album is more about utilizing basic skills than having a Universal Audio 1176, or knowing where a Fairchild 660's sweet spot is. It's about having the right approach—not necessarily the right technique." EQ

so I can easily switch between the drier signals and more ambient room sounds within the arrangement of a song.

One thing I did that was interesting was to sample all the toms individually before we tracked the drums. Then, as I had so much time in my hotel room to edit, I carefully cleaned up the signal leakage between the tom fills we recorded. I kept the original performance, but I pasted the natural decay back in from the individual samples. I did this because I think it sounds great to have natural decay ringing out over the end of the drum fills.

How did you handle the bass guitar?

Palmer: We first ran Migé Amour's '76 Fender P-bass through a DI box, and then we sent it out to a combination of amps. The main bass sound is a combination of a Mesa/Boogie and a Prince Piggy Bass 115 combo, which has killer growl. We also tracked a distorted bass track for each song that we later introduced into the mix as needed. For that track, he ran his Fender into a Budda Superdrive II Series 18, using an EBS MultiDrive Bass overdrive. His Hamer CH-12 12-string bass was put through the same chain. I'd record three or four takes of each instrument, and comp together the best parts on my laptop. I had a lot of fun tracking Migé—especially when he'd stop and scream into his pickups in the middle of a song [*laughs*].

What was the approach for tracking guitars?

Palmer: We set up Mikko Lindström with a splitter box so we could route his guitar to three separate amps. The main source was his Laney VH100R with matching 4x12 cabinet. We miked the speakers with SM57s placed right against the grille, and positioned off-axis to the cone. We submixed all the signals down to one track. Mostly, Mikko used a Gibson SG for the rhythm parts, but we also layered the tracks with an old Levin acoustic from the '40s that Ville brought in, as well as a Danelectro baritone for the really low sections. The solos were all tracked with a Telecaster or an ESP baritone guitar through the same amp setup. We also recorded a direct signal for everything in case we needed to re-amp later on—which we didn't. But it was still good to have that direct track available, because we didn't track the guitars live with the drums and bass, and getting a good guitar sound in isolation doesn't necessarily translate to having a good sound in the context of a mix.

What did you use to capture Ville's voice?

Palmer: A Neumann U67. It has a really warm, full sound, and, if you crank the preamp, the mic distorts beautifully when the vocalist screams.

Valo: We went for a more baritone vocal sound for this album. We also wanted to stack the vocals so that there was some dissonance. We'd record takes where I was sharp or flat, and then mix them together with the main vocal. A lot of the sound is dependant on my body position, so I did overdubs sitting down, or contorting my body into weird positions. The body is a fragile instrument, and posture and placement really affects the sounds you get.

How did you arrange the keyboard textures so that you could realize the stripped-down sound you wanted for the album?

Valo: Instead of just playing synth pads all the way through a song, we used the

instrument to play countermelodies, or to harmonize specific parts—such as playing a seventh or ninth on top of a guitar playing a root and a fifth. Things like that add dimension, not clutter.

Palmer: Janne Puurtinen used a Roland V-Synth, a Fantom X6, and a Clavia Nord Modular for most of the album. But we also recorded MIDI tracks for all his parts, so that we could apply plug-ins to the data, and then print a combination of the keyboard tracks.

Many rock bands are going back to analog tape to record their basics, so why did you choose to track the entire album on Pro Tools?

Palmer: I'm not one of these people who romanticize the days of tape. I put in my fair share of years biasing 24-track machines! Of course, I love the sound of tape, and I still see its sonic appeal. But when you look at what you gain with the control and creativity of Pro Tools, then Pro Tools wins hands down. You can make extreme changes to sound and arrangements faster, and every change is non-destructive. With tape, crafting a completely new arrangement at the mix stage was a nightmare, and it was often avoided—often to the detriment of the song. With Pro Tools, I can try radical ideas that may really improve the song—right up to the last minute before printing.

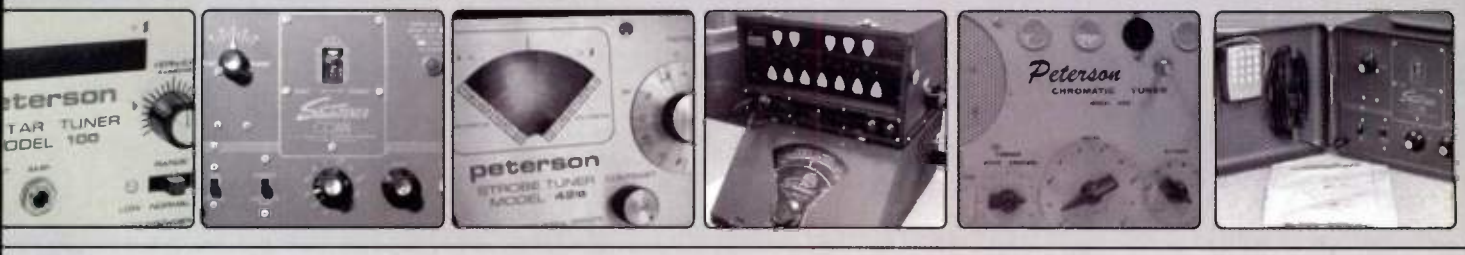
What was your bit rate?

Palmer: I'm not one for 96K recordings—24-bit/44.1K is fine for me. It's funny that as we increase sampling and bit rates on the production side of music, the public is moving the other way, and downgrading to mp3s. They are showing us they really care about the songs, artists, and performances. That's not an excuse for poor production and mixing, but it's a reminder about what makes someone want to own a piece of music. The song is—and always will be—king. If the song is great, the recording is automatically in a good place. As they say, "The best cure for a bad mix is a great song."

So how did you approach the mix for *Venus Doom*.

Palmer: I mixed as I went. I feel that I can't put off a decision until the mixing stage. I need to get it right immediately. I'll put all my samples on the drums, ride vocals, put delays and effects on parts, and try my best to simulate the finished product. Doing this really helps prevent over-producing an album, and it keeps the music in perspective. I also think it really helps prevent "over recording."

The HIM sound is not a meat-and-potatoes rock sound—there's a lot of depth. The challenge is in keeping the raw, rock elements without taking anything away from the band's soundscapes. As the arrangements are complex, telling the story successfully with the mix is vital. I try to keep the sound moving in sync with the musical changes so, when mixing, I'll change my monitoring depending on what I'm listening to. To get a tight sound with the bass and the bass drum knitting together, I'll play the music on the big speakers where I'll get the best bass response. When I'm balancing vocal levels, I'll listen more quietly on Yamaha NS10s. For checking the overall fidelity, I go more for Genelec 1031As or the big speakers. And, of course, the car is a good point of reference. That's where everyone is going to hear the album anyhow. EQ



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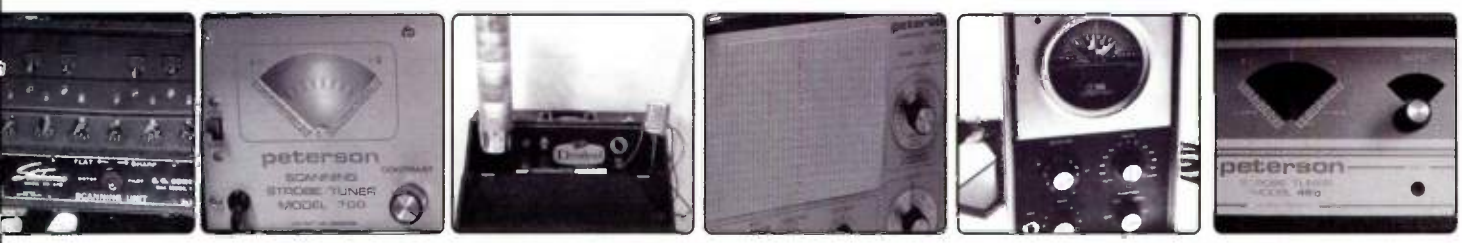

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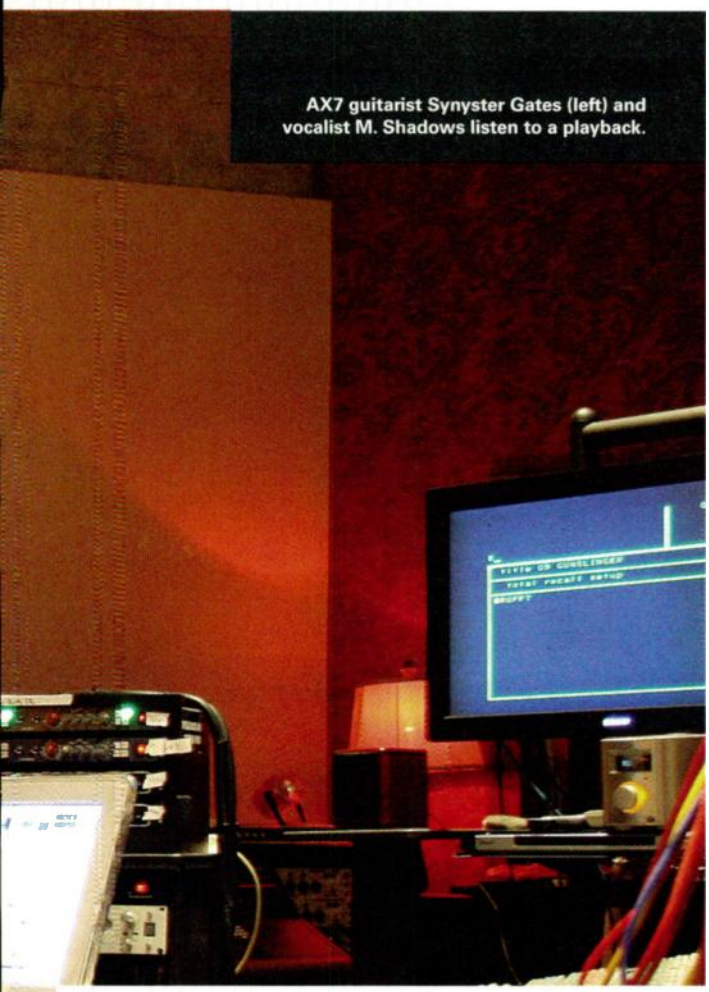

AX7 guitarist Synyster Gates (left) and vocalist M. Shadows listen to a playback.

By MICHAEL MOLENDA

MIXED METAL

SEIZING THE PRODUCTION REINS, AVENGED SEVENFOLD UNCORKS A STYLISTICALLY DIVERSE MODERN ROCK ALBUM

Photography By RAFA ALCANTARA

If first impressions are lasting, then we're all pretty much screwed. While it has been proven time and time again that the brute who looks like a Hells Angel can have the heart and demeanor of a bunny rabbit, society tends to read the glossy billboard and avoid the small print of one's deeper persona.

As a "for example," consider the seemingly cartoon-isn visage of Avenged Sevenfold: five dudes bearing arms rife with tattoos, wearing clothes that could have been snatched from the wardrobe of a reality show stylist, and exhibiting perfect L.A. hair. Cynics could mouth off that the Huntington Beach, California, group is some major-label Smurf's wet dream of a corporate-metal money-making machine, rather than an example of a

Drummer The Rev works out a piano riff.

genuine and impassioned rock band.

But the cynics would be full of whale poop.

First, consider that only extremely dedicated wackos would dare play loud guitars if they were seeking fame and fortune in the current music culture. Check out the charts—unless Alicia Keys and Carrie Underwood and Fergie are shredding (they're not), then guitar-based music is not exactly conquering the download generation.

Then, there's the fact that Avenged guitarists Synyster Gates and Zacky Vengeance made the cover of the October 2006 issue of *Guitar Player* because they mastered the incredibly intricate and technically brutal art of sweep picking, stole licks from Steely Dan, and crushed nu-metal convention by taking long, blistering guitar solos. They even played—*gasp*—harmony lines that would have made Thin Lizzy and Lynyrd Skynyrd cry tears of joy.

And let's not underestimate the band's old-school work ethic and lofty creative aspirations. These guys seem to work as hard to make their music absolutely right as Bruce Springsteen does. And, arguably, they might even be a bit more, um, progressive than The Boss.

Finally—and here's the f**k you moment for the cynical militias of the world—Avenged Sevenfold's sweat has actually paid off. They have a rabid fan base, their songs consistently hit the Top 40 of U.S. singles charts, they copped Best New Artist honors at the 2006 MTV *Video Music Awards*, and they clocked tons of acts to reach the final four of FUSE's "Best of 2007."

For its recent self-titled album [Warner Bros.], the group upped its game some more, deciding to produce the record itself—a move that also inspired the members (which, in addition to Gates and Vengeance, includes vocalist M. Shadows, bassist Johnny Christ, and drummer The Rev) to crank up the excitement level by incorporating their varied stylistic influences into the band's signature sound. The result is a soundscape that slams rock hard, while simultaneously molding prog-rock, hip-hop, electronica, orchestral, and music-theater elements into its core. Here, engineer Fred Archambault and Gates detail how Avenged Sevenfold crafted a metal epic.

What made the band want to take on the production duties for *Avenged Sevenfold*?

Gates: The easy answer is that we met with a bunch of producers, and no one really worked out. But there's also the fact that we're really into audio production. When we listen to CDs, we listen to *all* the elements. As much as we check out other artists, we follow producers, as well.

When you met with those producers, what were some of the things that prompted the band to think, "We don't want these guys. Let's do it ourselves"?

Gates: There were a couple of things. One producer was like, "I'm there to challenge you. One of the coolest things a band said to me was that I'm the best arguer around—I could talk them into anything." And we thought, "Whoa, that's a little too much for us." We've grown up together, we're all really great friends, and we didn't *want* to argue about anything. We're all pretty much on the same wavelength regarding tones, lyrics, and performances—there's never a struggle. We also had a clear vision of the record, and we didn't want anything to mess with our chemistry as a band.

What types of records were you referencing during pre-production?

Gates: We listened to lots of hip-hop records, as well as to Metallica to check out the bigger sonic stuff. A lot of metal records don't really sound that huge, and we really wanted to produce a big and diverse sound—just like our favorite hip-hop records. If you play this record on a really nice stereo, it's going to give you everything from the lowest lows to the highest highs.

Fred, what was your first impression of the material when you were confronted with it?

Archambault: I've been involved with the band since *Waking the Fallen* in 2003, so I've gotten to know the guys not only as musicians, but as friends. I know how musically diverse they are, so the album's mix of influences and styles was not really a surprise to me. If you hang out in their tour bus when the band is on the road, you hear everything from Big & Rich to Danny Elfman. We actually talked about this record quite a bit—starting six months before the band decided to produce it. There was a lot of preparation and forethought going on. I'd listen to the demos, meet with the band, and talk with David Schiffman—who was the other engineer on the project—to start developing a game plan for the studio. And, ultimately, our job became trying to get the sounds the band was talking about out of the speakers.

Did you and David develop any overriding strategies to manifest the band's sonic ideas?

Archambault: We quickly determined that we couldn't rely on defaults. The normal things we often do—such as

put a Sennheiser MD 421 on the kick drum and a Shure SM57 on the snare—might not produce the nuances of sound the band wanted. We also decided to craft sounds from the source. If Syn was unhappy with a rhythm-guitar sound, for example, we'd go to a different amp head, or a different speaker cabinet, or add a Tube Screamer to the signal chain. We wouldn't just reach for the EQ, or manipulate a compression plug-in. There was a lot of purity in the crafting of sounds.

Which recording medium did you choose for the album?

Archambault: We recorded to Pro Tools, but the band wanted to use it more as a tape machine, rather than solving problems with mouse clicks. There was very limited plug-in action during the sessions—I think we used a total of three. The band really wanted an organic sound.

Gates: We're a spur-of-the-moment group. We'll certainly have ideas and melodies in place when we're in the studio, but when you hit the Record button, we're often flying by the seat of our pants, and we wanted that spontaneous energy reflected in the final tracks.

How did you record The Rev's drums?

Archambault: The Rev has this amazing custom DW kit—two kick drums, seven toms, and a really wide variety of cymbals. So we had to customize our engineering decisions to make sure we were capturing all the nuances of his performance. For example, it's a little bit frustrating trying to get two kick drums to sound exactly the same, because there are so many variables that can make each kick sound different, from the exact spacing of the microphones to the sonic characteristics of the mic cables. So we ended up using three different microphones. One was a Sennheiser MD 421—which is kind of your stock, go-to kick-drum mic. We also used a Shure SM91 cardioid condenser to capture some attack—as The Rev plays those kicks pretty fast—and an AKG D112 to supplement the SM91s with some low-end oomph. Each kick drum got the same mic arrangement, and all the mics were submixed to a single track so that we had one track for each kick drum. During the submix, the fader for the SM91 was set at 0dB, the 421 was at –5dB, and the D112 was at –15dB.

What about the overheads?

Archambault: We found that using Neumann U67s—our usual default mics for overheads—didn't capture every single cymbal. For example, there would be a chime over his right side that we weren't hearing as prominently as a crash up front over a tom. So we put up two additional overheads, positioning two Coles 4038 ribbon mics about ten feet over the kit. These kind of became room/overhead mics, and they pretty much got what we needed, but we added some AKG C 451s as supplemental "spot" mics to ensure we captured his intricate cymbal work.

That sounds like a phase nightmare.

Archambault: The biggest issue with the kit was getting the phase correct. It definitely came down to following old-school mic placement, such as adhering to the three-to-one rule [*Editor's note: This is a strategy for positioning multiple mics that typically prevents one microphone from interfering with the pickup of another.*

Simply put, the rule states that two mics must be placed at least three times the distance that either mic is positioned from the source sound it is capturing]. But, to be quite honest, it went pretty quick. We had a sound up within a day, and we pretty much had our phase correct.

Was it also a hassle dealing with seven toms?

Archambault: Once again, we had to customize our approach, because we had all these mic stands around the kit, and it was hard to get some mics in there. The first three toms were an 8"x8", a 10"x8", and a 12"x10", and we couldn't squeeze MD 421s in there, because space was pretty tight over the hi-hat, and there were other cymbals around, as well. We ended up using Shure SM98s, which are miniature condensers often used to mic drums live. I haven't really used them in the studio before, but I thought, "Well, that's the only mic I can fit in there!" But they sounded great, and the signal rejection between toms was really good, because the 98s are hypercardioid. For the rest of the toms, we used MD 421s. The Rev's two floor toms are 18" and 16"1 in diameter, and we weren't getting enough

Gates assessing an off-camera critique.

body, so we added a bottom 421 along with the 421 on top. That helped us get the girth, and we didn't have any phase issues.

Did you do stereo submixes to manage the toms a bit easier?

Archambault: We left the toms on separate tracks. The only things we bused

were the kick-drum mics, and the top and bottom snare mics—which were a Shure SM57 [top] and an AKG C 451. I feel that when you bus your toms together, you're too restricted at the mixing stage. We ended up with 33 drum tracks between the source and room mics, and I didn't feel we were herding tracks that bad during the mix. The only concession we made was not going to analog tape for the drums—which the band wanted to do. Locking up two tape machines would have been a huge hassle, so it was way easier just staying with Pro Tools. Actually, I felt our track assignments were pretty conservative considering the drum setup we had—although it definitely helped to submix the kick drums and the snare mics.

What types of guitar gear made the scene during the sessions?

Gates: Our guitar tech, Walter Rice, brought in all these crazy amps, so we took the best elements of Bogners, Marshalls, and other models. We'd blend four different amps together to get the right tone. And, of course, I had my Schecter Synyster Gates signature model guitar, and Zacky used his Schecter Zacky Vengeance Custom.

Archambault: We had a lot of guitar stuff. Marshall sent over a bunch of amps, and I think we had every Bogner amp that has ever been made. Walter was absolutely an amazing resource for cool, one-of-a-kind amplifiers, and that gave us the palette to go straight for the tone from the source. We didn't need to use multiple mics and mic positions. In fact, the whole record was done with one Shure SM57. We put that mic dead center to one of the cones in the speaker cabinet—an inch off the grille cloth—and ran the signal through a Neve 1073 or a Neve 1084 preamp with a touch of compression from either a Fairchild 670 or a dbx 160 VU. That was the signal chain.

What speaker cabinets were you using?

Gates: We mostly used a Marshall 4x12 cabinet loaded with Celestions G12L-35s, a Bogner cab, and a Mesa/Boogie 2x12.

Did you use a splitter to route the guitar into several amps at once, and then decide which tone worked best for the song at hand?

Gates: We didn't split signals. We constructed the tones by choosing an amp for each track, and we layered the sounds one by one. When you play through several different amps at once, I've found that you lose the magic of what each individual amp can do. I'd rather play each overdub through one amp at a time. We also wanted each part to have its own unique feel, so we didn't paste, say, a guitar part for a chorus onto all the other choruses.

Archambault: We did split the guitar signals for *City of Evil* and *Waking the Fallen*. We'd use a Little Labs PCP Instrument Distro, or a Radial JD7—which I'm a huge fan of—to send the signal to two or three amps simultaneously, and then blend the various amp sounds to taste. But, for this record, we wanted the best thing we could get out of, say, the Bogner Uberschall with just one microphone, and pan the signal to the left. Then, we'd get what we liked out of a Marshall DSL100, and pan it to the right. Like Syn said, we wanted the vibe and tone of the individual amps to be maintained throughout the rhythm layers.

In what type of room were the guitar amps recorded?

Archambault: We did the guitars at Eldorado Recording Studios in Burbank, California. It's a one-room facility, but it has

More Goodies!—Shadows looking for the reverb send.

30-foot ceilings, cinder-block walls, and a concrete floor. You can just imagine how reflective that room is! We had to cut down the room a bit using packing blankets in order to ensure we weren't hearing the slap of the room in the microphone.

You were still getting significant room bleed into the SM57, even though it was positioned just an inch from the speaker cone?

Archambault: The mistake a lot of musicians make is forgetting that close miking doesn't completely cancel the interaction of the room with the source sound. Avenged Sevenfold has these incredibly tight rhythm-guitar parts, and any amount of slap, echo, or natural reverb can compromise the sound. There aren't many guitar players who can stack three or four rhythm overdubs, and still sound tight, but Syn is great at that. Obviously, we didn't want the room to smear his performances.

Synyster mentioned that he and Zacky would record one song at a time, rather than doing an assembly line of rhythm or lead tracks for every song on the album.

Archambault: Yeah. Unfortunately, we had to take the assembly line approach for the drums, and run through all of The Rev's drum tracks for the entire album, but we had the luxury of focusing on one song at a time for the guitars. Not many rock records get made that way. We'd typically finish all the guitars for a song in three or four days. This obviously allowed us to get the right tones for the right song, but another cool advantage was that it prevented Syn and Zacky from getting burnt out doing nothing but rhythm tracks for two weeks. This is especially important for Syn, who always leaves a lot of room in the studio for spontaneity. He'll write a solo while he's tracking, and, a couple of hours later, the other guys will run into the studio to check out what he did. That kind of stuff keeps the energy level way up throughout the whole process of making a record. I think the album required 72 hours of tracking time, but it never felt like a drag. Everything always sounded fresh, and that energy is really evident in the final sound of the album.

What was the typical approach for the bass?

Archambault: We would always track rhythm guitars before doing the bass tracks. I did this for two main reasons. First, if you record the bass before the guitars, you can get this great big bass sound that leaves no room for the guitars—and Avenged is definitely a guitar-driven band. This is not to say that the bass isn't important to the band—it very much is—but I felt we had to find the sound of the rhythm guitars first, and then determine where the bass tone would live. The second reason has to do with intonation. The worst situation is when your bass tracks are slightly out of tune. Then, your rhythm guitars become nearly impossible to track. And, actually, a third reason is that Johnny sometimes follows Syn's rhythmic accents, and doing the bass after the guitars allows him to better define what the bass line is doing.

How many sources do you juggle for the bass tone?

Archambault: I always record three sources for the bass. The first is a direct signal, and the second is an amp sound that's pretty clean, but with a little grit. The third source is either a super-distorted signal through a guitar amp, or a patch on the Line 6 Bass POD. The bass amp was the main sound, but I liked having three sources to blend together. Being able to add a little bit of distortion, for example, can help glue the bass sound together.

What was the bass amp?

Archambault: There was a SVT classic—which Johnny has had for a long time—and a Gallien-Krueger 2001RB. That's a great two-channel head that puts out an amazing distortion sound on channel B. We ran it through a Gallien-Krueger 410—which was the main bass sound—and a Gallien-Kreuger 115 for the distorted-amp sound.

How did you mic the cabs?

Archambault: We put MD 421s on both cabinets. The mics were maybe two or three inches from the dead center of the speaker cone, and the preamps were Neve 1073s. The bass cabs were set up in a vocal booth, so I didn't have the big-room challenge that I had when tracking the guitars.

I'm assuming that, unlike the guitar tracks, you ran the bass into all three sources simultaneously?

Archambault: Yes. We used the Little Labs PCP to split Johnny's bass signal to the bass amp, the guitar amp or Bass POD, and the direct line. All three signals were also routed to Empirical Labs Distressors set to about 6dB of gain reduction, a 4:1 ratio, and a really fast release with a very slow attack. This is usually a safe, yet aggressive compression that retains much of the high end of the bass tone. I also did something pretty cool with the DI signal. I ran it through a Little Labs IBP—which stands for "In Between Phase"—that allows you to dial in precise phase adjustments. I'd start by checking the phase relationship between the bass amp and direct signals before moving on to check the phase on all three sources.

Basically, I was adjusting phase to get the most low-end clarity. When everything sounded good, I'd pop on the subwoofer for Johnny so he could *feel* the bass as he played.

Did you submix the three sources into a single, monaural bass track?

Archambault: The bass tracks were not submixed at all, because I wanted [album mixer] Andy Wallace to have three distinct bass sounds available. I didn't want to deny him the opportunity to, say, add some distortion to a chorus in order to pump up a song's dynamics.

Considering the range of sounds and styles on the record, did Johnny use a number of different basses for his tracks?

Archambault: Johnny's Ernie Ball 30th Anniversary Stingray with ebony fretboard is on 80 percent of the record. There were a couple of other basses around, and we'd definitely record a bit, and then sit back and decide which bass would be best for the track. On "Dear God" he used an ESP 5-string bass, and he had him overdub a wall of bass for "Scream," which he really dug.

Did you use any effects?

Archambault: I'm a huge fan of the Visual Sound H2O Liquid Chorus & Echo pedal, and it sounds great on bass. You don't want to overdo it, but if you use it lightly, it's really cool. So we used it on the cleaner parts of the slower songs to add that Guns 'N Roses chorus sound that the guys love.

How did you audition mics for M. Shadows, and which model was selected as the main vocal mic?

Archambault: Actually, when we cut *City of Evil* at Ocean Way Recording in 2005, we discovered the perfect vocal mic for Shadows. That studio is like a microphone museum, and we literally put up every mic they had, from an AKG C 414 to a Neumann U47. The mic that brought up every nuance of Shadows' voice was an AKG C12. Luckily, when we did the vocals at Eldorado for the new album, the studio had not one, but two C12s. This allowed us to A/B the two mics, and you'd be surprised at how different they sounded. We ended up using C12 number two, because it enhanced Shadows' gritty vocals with a little more bite than C12 number one exhibited.

Can you detail the vocal signal chain?

Archambault: As with the guitar and bass sessions, we kept our single chain pretty much the same from song to song. The mic preamp was an Aurora GTQ2 Mark 3—which is a Neve 1073 copy made by Jeff Tanner—and the compressor was a silver-face Universal Audio 1176. The compression ratio was usually 8:1, with what we call the "Dr. Pepper" settings: a slow attack, a fast release, input level at 10 o'clock, and output level at 2 o'clock. I also set up the other C12 in another part of the studio to accommodate low vocals or background vocals. This mic was routed through a Vintech X73 preamp and a dbx 160 compressor, and this is what most of the guest singers used. However, I put up a Neumann U47 to round out the girls in the choir parts, because the C12 really has a bite to its sound. Also, as the brighter C12 was Shadows' mic, I wanted to make sure he had his own sonic space in the mix.

Does Shadows like to record his vocals by listening through headphones, or is he more comfortable with the tunes blasting through studio monitors?

Archambault: For the last record, he switched between headphones and studio monitors, but, this time, he used only headphones. He doesn't seem to have a preference, and he has learned to help himself sing on pitch by removing one of the earpieces if he's having intonation issues. Another thing that helped is that we used guide keyboard lines.

You mean you'd actually play the vocal melody on a keyboard, and pump that into his headphones?

Archambault: Oh, yeah. Distorted guitars can be so much white noise that it's sometimes hard to find a solid pitch reference. So, a few hours before Shadows was set to sing, we'd sit down and literally play out the entire vocal melody on a keyboard. I wish more people knew about this, because when there's definitely no issue as to what the note should be, it often frees up the singer to concentrate on phrasing and delivery.

How did you approach the engineering of "A Little Piece of Heaven"—which is almost a bizarre theater piece. Were you challenged to expand the sonic landscape in order to address the song's cinematic layers?

Archambault: I was around when that song was written, so I was aware of its strong visual focus. There are very few songs out there that actually transport you somewhere when you close your eyes. It's an amazing work, and it's based on a piano part that The Rev had written. When he played it for the band, all you heard was piano, drums, and a vocal melody for like eight minutes. Everyone looked at each other, and said, "This is like a Danny Elfman score." At that point, we pictured using strings, brass, and woodwinds, and that's why we brought in former Oingo Boingo member Steve Bartek as the arranger. In a sense, that song was the hardest to pull off technically, but it really wasn't any trouble—if *that* makes any sense. I mean, there were a lot of elements, but we really didn't do anything different from an engineering standpoint. We put up Neumann U67s and Earthworks M50s for the strings—a combination of close mics and room mics. But the core of the piece is actually pretty simple: drums, guitar, bass, a main vocal, and 12 tracks of strings, woodwind, and brass. When the bed was laid down, we did some sound effects—like breaking bottles outside—and crazy vocals with all five guys in the vocal booth.

Gates: We didn't establish any rules for Steve's arrangement, or for ourselves. And doing the vocals was awesome. We had one big, fun, drunk day, and laid down a million and one vocals.

There was a time—not too long ago—when a diverse album such as *Avenged Sevenfold* would have been considered commercial suicide by major-label A&R squads.

Archambault: Absolutely. So I say, "Hats off to management, hats off to the label for allowing them to do this, and hats off to the band for pulling it off." In a sense—because the band conceptualized and wrote this whole epic—I think Avenged Sevenfold may have the biggest pair of balls ever. EQ

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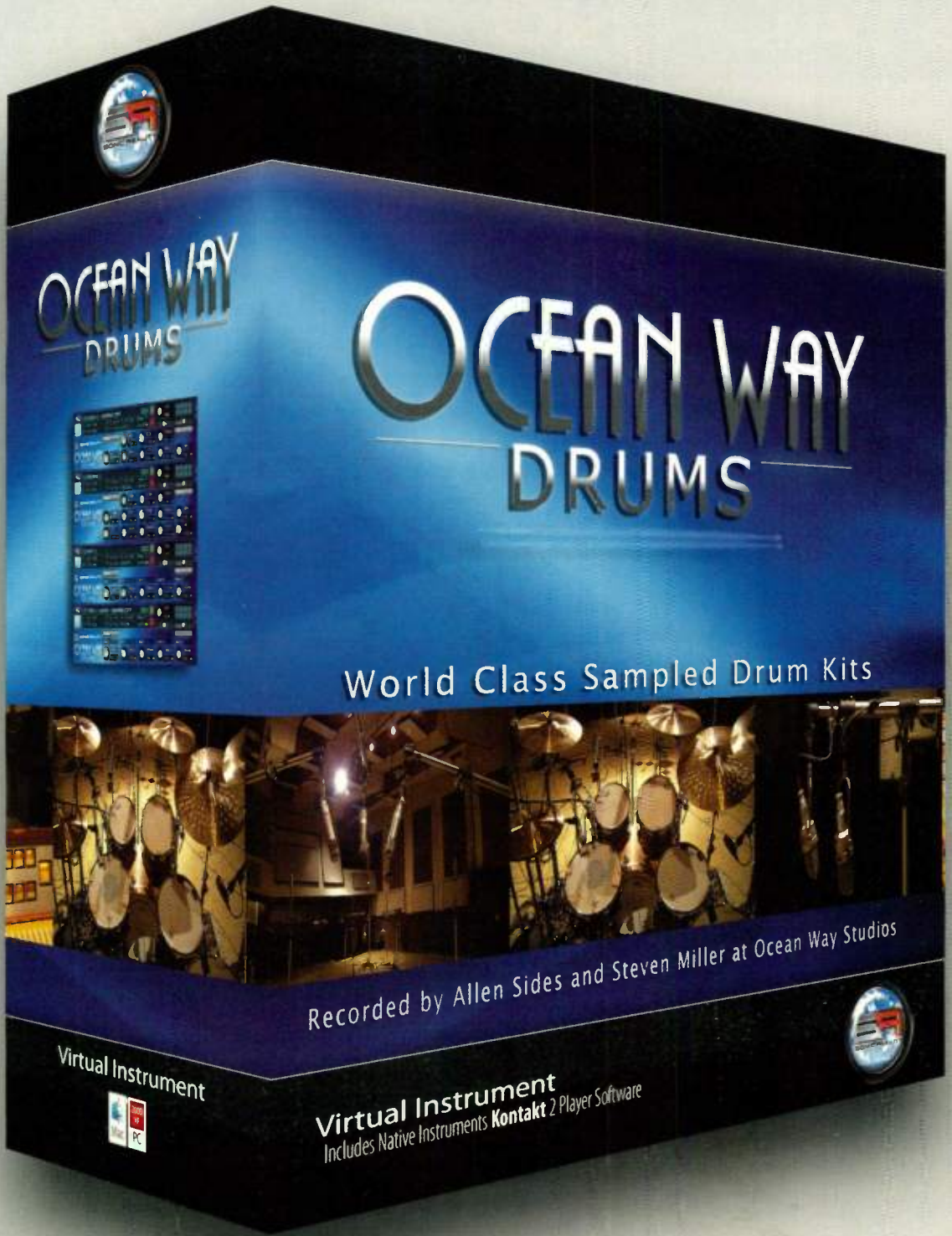

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STALKING THE WILD

by Bruce Bartlett

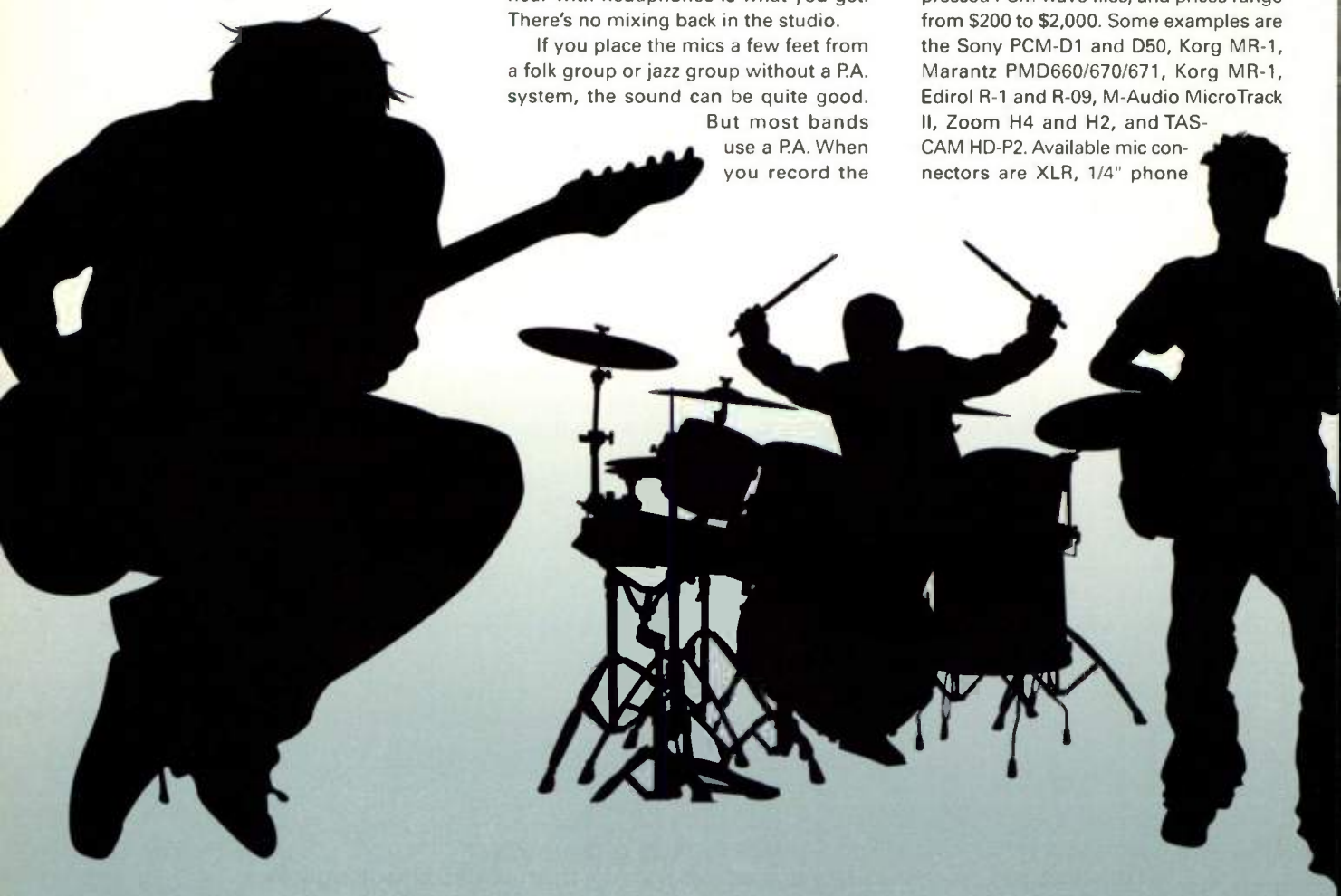
How To Record Your Band with a Portable 2-Track

Need a simple way to record a live gig? Try a portable stereo recorder. The process is easy, and the required gear can cost as little as $200. You can use a recorder with built-in microphones, or you can plug two mics (or a stereo mic) into a recorder. You'll pick up the group as a whole from several feet away. The mics will capture not only the musicians, but also the room reverb and background noise. You could call it a documentary or "audio snapshot" recording. As you record, what you hear with headphones is what you get. There's no mixing back in the studio.

If you place the mics a few feet from a folk group or jazz group without a P.A. system, the sound can be quite good. But most bands use a P.A. When you record the band, you're also recording the sound of the P.A. speakers, so the mix you get often depends on the house engineer's skill. Also, the sound will be more distant and ambient than what you'd capture using several close-mic positions.

The Gear

You'll need a portable digital recorder. These typically record onto a Compact Flash or a Secure Digital card or hard drive. You can record mp3 or uncompressed PCM wave files, and prices range from $200 to $2,000. Some examples are the Sony PCM-D1 and D50, Korg MR-1, Marantz PMD660/670/671, Korg MR-1, Edirol R-1 and R-09, M-Audio MicroTrack II, Zoom H4 and H2, and TASCAM HD-P2. Available mic connectors are XLR, 1/4" phone



STEREO IMAGE

Three portable two-track recorders: M-Audio Microtrack II, Zoom H2, and Edirol R-09W.

or 1/8" phone, with or without 48-volt phantom power for the mics.

Headphones let you know whether the mics are working correctly, and let you hear what the mics are picking up. If the band and P.A. are loud, it's hard to hear what's being recorded unless you use *isolating* headphones (Remote Audio HN-7506) or earphones (Etymotic ER-4S, ER-4P and ER-6i; Shure E3 and E4).

Preparation

Before going on the road, install fresh batteries (and bring some spares), recharge any rechargeable batteries, bring along spare flash memory and an AC adapter, and clean the connectors with isopropyl alcohol or Caig Labs DeOxit. Do a trial recording to make sure everything works. If possible, record the gig in a room where the audience is attentive, and the background noise is low. You might visit some potential venues to check out the noise and acoustics. Avoid very live rooms because they can make the recording muddy. Be sure you have enough free space on your flash-memory card before going on location. Listed below are approximate recording times for a 1GB card (double these times for a 2GB card):

- 24-bit/44.1kHz stereo WAV: 1 hour
- 16-bit/44.1kHz stereo WAV: 1.5 hours
- 256kbps stereo mp3: 9 hours
- 128kbps stereo mp3: 16.5 hours

The recorder might have a record-level switch labeled Manual and Auto. Set it to Manual in order to retain the dynamics of the performance. If the switch is labeled AGC (Automatic Gain Control), set it to off.

The Gig

Plug in some headphones and turn on Record Monitor mode. You'll hear the room acoustics and any background noise (audience, air conditioning, traffic). Room noises that you wouldn't otherwise notice become obvious when you listen on headphones. Also listen for buzzes, distortion, and crackles from bad cables or connections.

Where do you place the mics? The closer the mics are to the group, the clearer and cleaner the sound will typically be. In other words, close placement captures more of the music, and less of the room sound and background noise. Try to place the mics as close as possible to the stage where you still pick up the house speakers—about a stage-width away from the stage (Figure 1). Keep the mics away from the bar and other obvious noise sources. If there are dancers near the stage, and the ceiling is low, you might try boundary mics (such as two PZMs) gaffer-taped to the ceiling, or mini mics hung from the ceiling.

To eliminate muddy room sound, you can record acoustic groups off the P.A. mixer into a line input. Or try placing the recorder/mics on stage (on a stool

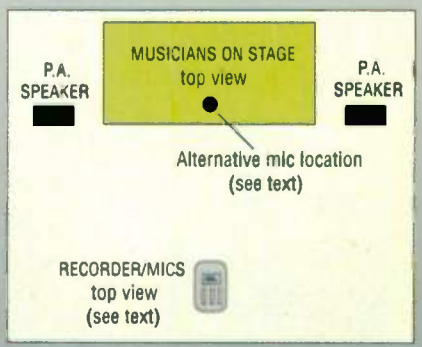


Fig.1. A typical mic placement.

or mic stand), and record the sound of the monitor speakers plus the band. Some bluegrass or old-time bands sing into one microphone, so mount the recorder and mics next to that mic.

To record a small folk group or an acoustic jazz group without a P.A., place the mics about three to six feet from the ensemble (Figure 2). If the group plays in a circle (as in an outdoor jam or rehearsal), try placing omnidirectional mics in the center. Or just walk around with the mics while monitoring the mics over headphones. Find a spot where you hear a good balance, and put the mics there. If you're not working with your own band, ask the musicians' permission to record them during a break in the music, and here's a word to the wise: Get that permission in writing.

While the band is rehearsing a loud song, make a trial recording. Set the recording level so the meter reads about –6dB maximum. This allows some headroom for surprises. Peak levels above 0dB result in distortion. Some portable recorders include a limiter that prevents recording levels above 0dB, and others have a clip or peak LED that flashes when the recording level is too high. Pay attention to these levels, because while you can always bring up levels if they're too low, fixing a clipped signal is extremely difficult.

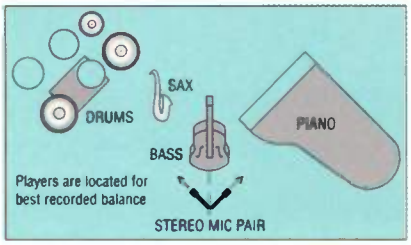


Fig. 2. One method of stereo miking a jazz group.

Think this is a picture of a camcorder? Actually, it's a DAT in disguise, and you can use it to record your band.

Tips For Savvy Live-Recording Engineers

Bring a Limiter. If you're going to be recording from the mixer outputs, bring a limiter. Yes, most portable recorders have internal limiters, and if that's all you have, it may be better than nothing—*if* you use it only to trap significant peaks, rather than compress the overall signal. However, a quality limiter will give quality results.

Consider a FireWire Mixer. Got laptop? Then a FireWire mixer can do double-duty for mixing your band, and providing a recording feed to your laptop. For live use, this saves a lot of gear lugging.

Avoid distortion. If you have enough tracks, it's worth splitting crucial tracks (such as lead vocals) into two tracks—one trimmed down about –6dB to –10dB compared to the other. If the main track hits an "over" by exceeding 0, you can use the –10dB track to cover for you, and bring up the gain to match that of the main track.

Move Up. Several companies make portable *multitrack* hard-disk recorders with varying capabilities (mic preamps, expansion ports for external hard drives, etc.). These will let you add some audience mics, have a separate track for the lead parts, and the like. However, as you check out various units, make sure they're fast on their feet. Some units make you wait for a while when saving to disk—which is not very helpful for live gigs. Also make sure there's a sufficiently big hard drive to record entire sets, or, if not, the ability to expand via FireWire or USB 2.0 drives.

Multitask Your Camcorder. A typical miniDV handheld camera's audio is basically a DAT, and most have line inputs so you can bypass the mics with either a feed from a P.A., or the output of a couple of preamps fed with quality mics (although the mics in camcorders can sometimes give surprisingly good results). However, before you do any recording, check the menus for audio, as there may be an option for 12- or 16-bit recording. Of course, you want the 16-bit version. Also, check whether "wind cut" is enabled. This is a lowpass filter that you don't need if you don't have to worry about wind in your recording environment. If you do use the internal mic, remember that they are mostly stereo, with two mics in the same capsule. As a result, keep side-to-side motion to a minimum, or you'll hear phasing/flanging effects owing to the right and left channels being so close together.

Where's the Audience? In a typical 2-track recording scenario, you won't have audience mics, so you won't capture applause very well. No problem: Invest in a sound effects record and throw some audience noises in the background. I won't tell. Promise.

Read *EQ*. There was a really great article in the April 2007 issue about how to set up your mixer to do double-duty for P.A. work and live recording. Check it out for some great tips. —*Craig Anderton*

STALKING THE WILD STEREO IMAGE

Recorders often have a mic-gain switch or a pad to prevent mic-preamp distortion. If you have to set the recording-level control less than 1/3 up to achieve a 0dB recording level, use the low-gain setting, or switch in the pad. If you are using large-diaphragm condenser mics plugged into a phantom power supply, you might need to plug the output of the supply into the recorder line input—rather than the mic input—to prevent distortion.

You might be able to record an acoustic group on location without a P.A. system or an audience. This situation gives you some freedom to improve the sound:

- If the room is too live (reverberant), put up some packing blankets, comforters, rugs, acoustic foam, or cushions.
- Often a good-sounding spot for the musicians is near the center of a large room. Place the musicians around the stereo mic pair where you want them to appear in the recording. For example, you might place two singing guitarists on the left and right, with the bass in the center.
- Experiment with microphone height to vary the vocal and guitar balance. Try different mic distances to vary the amount of ambience or room sound. A distance of three to six feet is typical.
- As the musicians are playing (and during playbacks), monitor the mic signals with headphones. If some instruments or vocalists are too quiet, move them closer to the mics—and vice versa—until the balance sounds right.
- If someone makes a mistake, either record another take of the entire tune, or record starting from a few bars before the mistake, and edit the takes together later.

After the Gig

Back in the studio, connect the USB port in the recorder to the USB port in your computer. The recorder shows up as a storage device on your computer screen. Drag-and-drop the recorded sound files to the computer's hard drive for editing and CD burning. The files transfer in a few minutes. Then, the flash-memory card is empty, and free to make *more* recordings.

Now you can edit the recording, and adjust its tonal balance with DAW software. You might cut a few dB around 300Hz to reduce boomy reverb, as well as to get a clearer recording. If your mics are weak in the bass, compensate by boosting a few dB around the 50Hz to 100Hz range. Try out these tips, and enjoy your recordings! eq

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ROMPLER ROOMS

Make Workstation Software Work Harder for You

by Craig Anderton

When hardware still ruled the earth, the Korg M1 workstation created quite a stir with its combination of sounds, multitimbrality opps, a MIDI sequencer, and effects. And when MIDI sequencing took off, workstations became essential elements of many a MIDI studio because they could play back multiple sounds from a single, economical piece of hardware.

Today's software workstations can not only do pretty much anything their hardware ancestors could do, but a lot more. Let's discuss how to maximize the potential of these virtual instruments, from your initial songwriting inspiration to the final mix.

Ready To Rock (or Jazz, or Techno, or . . .)

Software workstations are exceptionally useful for songwriting, because with one instrument, you can create eight, 16, or even more tracks. As a result, you can simply keep loading instruments into MIDI channels, create new MIDI tracks, and lay down overdub after overdub.

However, as you add more instruments, CPU consumption increases—sometimes dramatically. Many workstations let you adjust polyphony for particular channels (Figure 1), so take advantage of this feature to minimize the number of voices that need to sound at once. With many bass lines, for example, you probably won't need more than two voices. Sounds with long decays—such as pads—tend to eat polyphony, so restrict these, as well. Often, the voices that are cut off are at a low enough volume—or are masked by other notes—so that you won't notice they're missing.

Another way to reduce CPU consumption is to use bus effects within the instrument (if present), rather than insert effects, whenever multiple sounds use the same effect. For example, insert reverb on a bus, and send signal to it from the instruments you want processed, rather than inserting reverb on all channels requiring reverb.

The MIDI Advantage

A big advantage to using a MIDI-based software workstation is that MIDI data is so malleable. During the songwriting process, if you decide to change the key or tempo, it's much easier to do with a bunch of MIDI tracks than with digital audio. (Having said that, many workstations can also stretch digital audio loops with respect to timing, and, possibly, pitch.)

A corollary MIDI advantage is that when it's time to mix, you can replace the sounds of individual tracks with individual instruments that may offer a better sound quality. For example, you can use the workstation to lay down a piano part, but then switch over to something such as Ivory or another dedicated piano program to get the best piano sound possible. The same thing goes for drums, as you can use a program like FXpansion's BFD to replace the simpler drum sounds found in a workstation.

Also note that hosts with MIDI plug-ins get along very well with workstations. When you're laying down tracks in quick succession, rather than deal with quantizing or tweaking as you record, you can often use MIDI plug-ins to temporarily do quantization, scale velocities, and the like. After you're finished laying down tracks, then you can get into the editing process and tweak the MIDI data.

Content Management

There's a growing trend with workstations and multitimbral samplers to include ever-greater amounts of content. Generally, this defaults to being installed in the root drive where the program lives. If you use enough of these products, your main drive can run out of space pretty fast.

So, dedicate a big hard drive (250GB–500GB) just to content and samples. If your computer's motherboard has some unused connections for hard drives, you can mount the drive internally, but, if not, an external FireWire or USB drive will do the job. However, you will likely have to instruct the program where to look for the content. Check any available preferences dialogs, as that's usually where you specify a path to the content.

With some workstations, you can create an alias/shortcut for the target samples in the original folder on your root drive. For example, if the workstation has a folder named "sounds," you may be able to create an alias for the drive containing your samples and put it in the "sounds" folder. Other workstations recommend against moving factory content to a different

Fig. 1: Several techniques mentioned in the text are applied here to Native Instruments' Kontakt 2. Polyphony has been limited to save on CPU consumption (outlined in yellow; Electrik Guitar is limited to 16 voices, Drawbar Organ to 32, and Fretless bass to 4). The blue outline shows that reverb has been added as a bus effect to avoid putting a reverb in each channel. The red outlines show that each instrument has been sent to its own output channel.

To Do:

- ☑ Record entire band
- ☑ Program drum beats
- ☑ Comp together guitar solo
- ☑ Fix vocal pitch problems
- ☑ Tighten timing in bridge
- ☑ Add string section
- ☑ Add synth bass
- ☑ Warm tracks with VC-64
- ☑ Add convolution reverb
- ☑ Mix with 64-bit engine
- ☑ Create dance remixes
- ☑ Master mixes with LP-64 Effects
- ☑ Burn CD + upload to web

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Fig. 2: SampleTank 2.5 comes with 33 effects, including a convolution reverb.

location, as any updates might be written to the current location, thus making it difficult to keep file structures in sync—although you can create a path to your own custom content. The instrument's documentation should mention any considerations involved in moving content.

Streaming vs. Loading into RAM

Some workstations can stream long samples from the hard disk, while others are restricted to what you can load into RAM. While using RAM is generally faster and smoother, you're not going to load a 30GB piano into a computer with 2GB of RAM. Streaming often needs to be enabled—sometimes for

Software-Based Workstations

Here are thumbnail descriptions of some common softwarebased workstations, listed alphabetically by manufacturer.

Apple EXS24. This is available only as part of Logic. While showing its age a bit, the EXS24 broke open the virtual sampler market.

Big Fish Vir2 instruments. Based on the Native Instruments' Kontakt Player engine, these offer effects, mixing, streaming from hard disk, and other features derived from NI's flagship sampler.

Cakewalk TTS-1. Available only within Sonar, this basic (and CPU-friendly) workstation is useful for blocking out parts while songwriting.

East West Colossus. While originally offered as a sort of overachieving General MIDI set, this Kontakt player-based instrument contains an excellent assortment of high-quality sounds.

E-mu Emulator X2. This brilliant sampler boasts excellent sound library support and innovative features, such as SynthSwipe for sampling older synths. It's a good choice for those wanting to go beyond basic workstation applications.

IK Multimedia SampleTank, Miroslav Philharmonik, Sonik Synth, SampleMoog. These are all characterized by large sound libraries, and offer a comprehensive set of effects.

Korg Digital Legacy Collection. This set of virtual instruments includes a software M1, but it's not your father's M1. The sound is much cleaner, and it comes with all the M1's expansion card sounds.

MOTU MachFive 2. The latest version includes advanced features such as beat-slicing, REX file importation, convolution reverb, the ability to import samples in just about any format, and a 32GB sound library.

Native Instruments Kontakt 3. This ambitious sampler includes features not found elsewhere, such as MIDI scripting (think of it as MIDI plug-ins you can write yourself). It also has highly developed slice-oriented beat-machine functions.

Propellerhead Software Reason. While not a workstation per se, the ability to ReWire it into any major host is compelling, and you can insert as many instances of the included synths and samplers as your computer can handle.

Sonivox Muse. Based on GigaStudio technology, this isn't as editable as some of the "pure" samplers, but it has a ton of sounds, and it fulfills the concept of a hardware ROMpler brought to software.

Steinberg HALion and Hypersonic. HALion is a traditional sampler that can stream samples from hard disk and offers multitimbral operation. Hypersonic resembles a keyboard ROMpler—it's less editable than HALion, but is faster to set up and easier to use.

TASCAM GigaStudio. This darling of the movie-scoring crowd was the first sampling system to stream from hard disk, and that process has been refined to the point where the program can play back really long samples and do so very efficiently.

Ultimate Sound Bank PlugSound Pro. This workstation also handles loops and beats well, with excellent time-stretching options. Optional-at-extra-cost "virtual soundcards" are available for expansion.

individual instruments within a multitimbral instrument, or sometimes for the instrument as a whole. Just remember that neither solution is without issues. Streaming lots of samples from a hard drive can limit the number of audio tracks you can stream from the same hard drive simultaneously (which is why a dedicated drive for content is helpful), and pushing RAM to the limit can cause instability because that same RAM is shared with the operating system and your host program.

Effects

I formerly advised bypassing the included effects with virtual instruments, as you could likely do better by adding other plug-ins into the signal path hosting the instrument. But times have changed. With better computers, instruments can include far more CPU-intensive, and better-sounding, plug-ins (Figure 2). Some—such as Kontakt, MachFive 2, and GigaStudio—include features like convolution reverb.

However, the issue here isn't just about sound quality. Using effects included within the instrument makes projects more transportable and archivable. As long as you can load the instrument, you're loading the effects as well.

Separate or Stereo Outputs?

Workstations usually offer multiple outputs so you can take advantage of your host mixer's features to process individual sounds. Keeping in mind the above comments about effects, though, if you can do all your mixing and processing within the workstation, you again have a more ergonomic and transportable project. You can even save the workstation setup as a preset, and import it into a different host, knowing that the sounds and mix will be as you intended.

Freeze!

If you really load up the channels of a multitimbral instrument, you may need to *freeze* tracks to free up CPU power. Freezing essentially disconnects the instrument from the CPU, replacing it temporarily with an audio track that makes much fewer demands on the CPU.

However, freezing works a little differently with multitimbral instruments compared to single-channel instruments, and it varies from program to program. For example, you may be able to freeze one particular instrument of a multitimbral instrument, or you may only be able to freeze the entire instrument. Check your host program's documentation for details. EQ

BLUE CHEER AND THE TEMPLE OF LOUD

by Michael Molenda

Long ago, in the foggy galaxy of the '60s, strode the loudest, rudest, and heaviest blues-rock trio to roar out of the San Francisco ballroom scene. While other Frisco hippie bands sought to wrap their audiences in a fuzzy web of psychedelic bliss, Blue Cheer was pummeling listeners' synapses into submission with gargantuan doses of volume. Bassist Dickie Peterson, guitarist Leigh Stephens, and drummer Paul Whaley had just one bona fide hit—a belligerent cover of "Summertime Blues" off 1968's *Vincebus Eruptum*—but the band definitely struck one hell of a deafening chord, and is credited with being an originator of heavy metal and grunge.

Despite various personnel changes throughout the years, Blue Cheer is still active today. Original members Peterson and Whaley are back in the fold, along with guitarist Duck McDonald, who produced the band's most recent release, 2007's *What Doesn't Kill You*... (Rainman/Evangeline). Of course, many acts have battled for the title of "World's Loudest Rock Band" since Blue Cheer first assaulted eardrums during the twilight of the Summer of Love. But the band's iconic stature as blaring blues devils remains, and McDonald is keeping the guitars loud and proud.

Is there any secret gear you use to unleash the guitar volume?

No [*laughs*]. I just play a Fender Texas Special Fat Strat that's loaded with a humbucker, and a bunch of good old amps that include a plexi Marshall, a Sunn Model T, and an Orange.

How do you typically set the mics on the amps?

Chris Kozlowski, the engineer for *What Doesn't Kill You*, used a few different setups, but I like Shure SM57s positioned very close to the speaker to get more low end. The farther away you move the mic, the more high end you get.

Given the high volume levels you put out, is it possible to track basics with everyone in the same room without risking massive signal bleed?

We definitely want that older sound and feel, so we do track as a live trio. But we want separation, as well, so the live guitars were considered scratch tracks and replaced with overdubs. We also stuck Dickie's SVT amps in the basement of a barn to ensure his bass didn't bleed into the drum mics. No surprise—but we do play really really loud. It's a balls-out sound.

JON KOZLOWSKI

Dickie Peterson.

THE ART OF CONTROL

"Playing loud isn't about turning up your amps louder than anyone else," says Dickie Peterson, the original bassist and vocalist for Blue Cheer. "You have to know how to find musical uses for the feedback and overtones, and you have to control the beast. Here's a story from the old days that illustrates what I'm talking about. Back in the '60s or '70s, we'd be on a bill with some band, and they'd say something like, "We're gonna blow Blue Cheer away. We're gonna play even louder than them." And they'd come out and crank up their amps, and they'd sound horrible. This was because they didn't adapt their performance to the higher volume. They played the normal way they played, and the volume just turned all their parts into mush. You see, volume is part of the sound, and you have to incorporate it into your style as much as you'd work out your vocal approach, guitar tones, and everything else. It took a lot of experimenting until we learned how to play loud, and the main thing we discovered was that you have to play almost nothing. You have to be deliberate, play as few notes as possible, and give the overtones room to ring out. It's definitely a less-is-more world when you're cranked up, and you really have to make the things you *do* play count."

How loud is it when you do your guitar overdubs?

I don't usually play through a full stack, but I'll use one 4x12 cabinet and crank up the amp as far as it can go. I want the speakers to sound like they're going to pop right out of the cabinet. Of course, there's a point where you can go *too* far with the volume, so you have to find the spot where the speakers are raging, but just before they collapse into mush.

I'm assuming you're not a guy who uses Master Volume knobs?

Absolutely not. You lose low end with those.

I sense a recurring theme, here. . . .

Yes. Low end is the recurring theme with us. We're living for the bass. The power of Blue Cheer isn't just in the volume, it's that we really chunk out the lows for our rhythm tracks. The low end also gives us what we consider a more old-school sound. I don't have anything against high end—I love screaming guitars—but fat rhythm tracks are a big part of the band's signature style. Ultimately, you want to hear what everyone is playing, but we don't compromise the power you hear—and feel—in the low end just to make the mix crystal clear.

Do you have to make any sonic adjustments to ensure the lows are properly translated?

I just try to keep as much low end in my rhythm-guitar sound as possible. I'll also tweak Dickie's bass sound so that our frequencies cross over a bit. Where the bass starts to leave off in the low mids, the guitar will start there, and then take on the higher frequencies. I also scoop out some of the high-midrange frequencies to diminish any harshness. The hard part is usually the mix, because high end can be pumped up louder than low end. You can only make bass so loud before the meters go into the red. It's a constant battle putting in as much low end as we can get without making the tracks too muddy or distorted.

What about compression?

I don't like it on the guitars. At the volumes I play, the speakers have kind of an automatic compression effect, and everything gets squashed down at the mastering stage, anyway. So I don't compress the guitars very much as they're being recorded.

Does the recording medium affect the low end in any way?

A little. We tracked to Alesis ADATs, but we mastered off a 1/4" analog mix that was running at 15ips.

What is your view on getting an old-school guitar sound these days?

First, let me say that I'm absolutely into modern technology. But you have to use the technology in a way that's true to your artistic vision. Like I said, super-clean, high-fidelity sound isn't necessarily the thing for Blue Cheer. Analog tape was obviously the sound of classic rock and roll, but tape-based studios are difficult to find these days, and they can be expensive. So, given that fact, I believe that using as much tube gear as you can is the way to find that vintage sound in a digital age. I've used PODs and other modeling processors, and I like

BLUE CHEER AND THE TEMPLE OF LOUD

JON KOZLOWSKI

Blue Cheer's Andrew "Duck" MacDonald (left) and album engineer Chris Kozlowski tracking guitars for *What Doesn't Kill You...* in the studio control room.

them, but don't be fooled by those sounds. Tube amps have the sound.

Is there an overriding production concept you carry through as you try to make a classic Blue Cheer record?

The members of this band have accumulated decades of exposure to other artists, different producers, old and new recording techniques, and musical evolution. All that stuff is put into a basket, shaken up, and the end result is us. No matter what we do, we are representing ourselves. So we try not to over-think what we're doing. We just go for it. The Blue Cheer approach is very spontaneous. If you spend too much time with a song, you can beat it to death—just like you can beat a performance to death in the studio. My little secret is to take the bad stuff, and crank it up. You're documenting a space in time, and it's the imperfections that make you unique. It's like having sex. Sometimes, you're good at it, and, other times, you're not, but even bad sex is good. EQ

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WAVES

9 ESSENTIAL ELEMENTS OF A VIBRANT BASS SOUND


Cragin in the studio.

by Shane Mehling

Hal Cragin has played with everyone from Iggy Pop to They Might Be Giants, and from Frank Black to Sarah McLachlan. He's also an accomplished producer, and his most recent project is A Fine Frenzy's One Cell In the Sea *[Virgin] with vocalist Alison Sudol.*

MY FAVORITE GEAR

I really love the API 312 preamp—it's the perfect bass DI. Another great choice is the preamp section of the Ampeg SVT head. The XLR output lets you send a line to the recording engineer, but still have control of your sound. I can't recommend the Crane Song Phoenix TDM plug-in suite enough. It emulates the second- and third-order harmonic distortion and coloration of analog tape machines. I'll often start with the Dark Essence plug-in, choose the bright Sapphire coloration, and set the input trim to -4.0. If you put that on the bass track, it'll give the bass more power without increasing its volume. It won't be louder, but it will *feel* louder.

EQ AND COMPRESSION

I love boosting around 100Hz, and then boosting at 7kHz to get some string attack in there. It's nice to know it's a bass guitar when you hear it. I also suck out a bit of 500Hz to clear things up. Compression is more variable. I've found a good starting point is a ratio of 4:1 with a medium attack and release.

GO FOR IT

If you're running two tracks of the bass signal, try something really extreme with one track—with EQ or compression—and tuck it under the other track. This type of thing is more common with drum subgroups to get some snap and glue into the instrument, but it's also fun with bass.

DISTORTION

It's nice to have some kind of distortion. An Ibanez Tube Screamer—or any distortion pedal with a Tone or Blend control—is great. Just be sure to track a direct [unaffected] signal, as well, so that you don't lose your low end.

ENVELOPE FILTERS

The DOD FX-25 Envelope Filter is great if you set it up to where the envelope *doesn't* open. This goes against what the pedal is for, but it produces a beautiful, keyboard-like dub sound. This is an easy trick that I've used on many records with success.

BOUTIQUE EFFECTS

Nobody is going to *ask* for an effect on your bass. If you can force it on them, great, but if you show up to the studio without the latest $500 boutique pedal, you won't get fired, because they won't ask for it.

RIGHT-HAND MAN

Be aware of where you're playing with your right hand, as moving it around will significantly change the tone of the bass. The only way, for example, to get that Rolling Stones' "Street Fighting Man" sound is to play closer to the neck. And if you move your right hand throughout a song, the changing shades of tone can really perk up a track.

LISTEN TO THE GUITAR

Even if the bass is playing something basic, learn the melody line of the guitar. If you add a note or two that matches up with the guitar, it'll help punch up the melody line.

FOLLOW YOUR BLISS

My band A Fine Frenzy is very bass-guitar driven. The emphasis is less on the all-mighty kick drum, and more into letting the vocals and the bass intertwine. So much modern rock is about the kick, and having the bass follow the guitarist's barre chords, which, to me, is not really "bass guitar" in the full glory of Leo Fender's brainchild. EQ

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VIRTUAL PIANO RECORDING TIPS

by Cliff Goldmacher

While nothing beats a well-played, well-recorded grand piano for feel, tone, and authenticity, most of us don't have the income or studio space required for the real thing. Fortunately, today's sampled software pianos are getting so good that in some (though certainly not all) instances they are virtually indistinguishable from the real thing. But let's not even try to compare real and sampled pianos; instead, let's take full advantage of the options that software pianos provide.

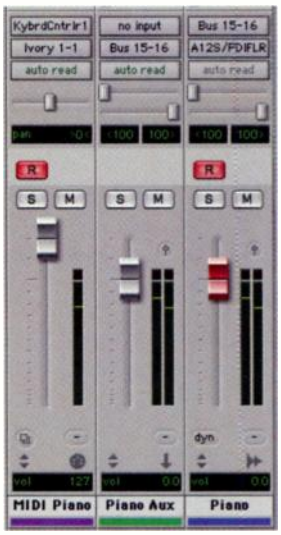

Fig. 1: Set up your DAW to record both audio and MIDI tracks.

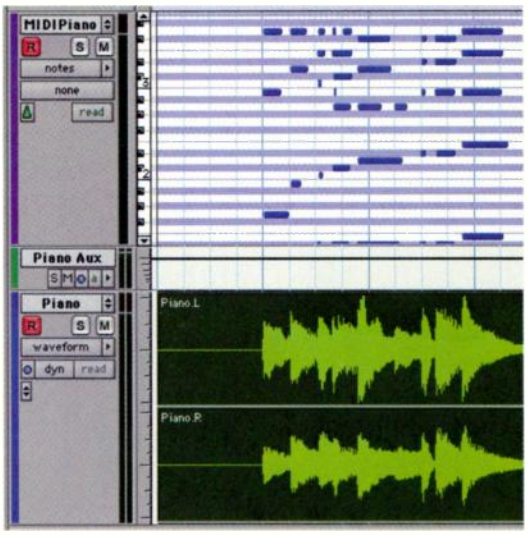

Fig. 2: Bouncing the MIDI data to audio provides a degree of "future-proofing."

MIDI

The primary advantage of recording using a software piano is that you're using MIDI to trigger it, and MIDI gives you *options*. From fixing a wrong note to quantizing an entire piano part, recording a MIDI track of your sampled piano allows you to go in after the fact and tweak elements of your performance. This helps avoid tedious punching or multiple takes.

For example, if you've recorded a piano part and (as is often the case these days) you're playing to a drum loop, even subtle tempo fluctuations become glaringly obvious. By quantizing the piano part (I prefer an 1/8-note quantize in most instances), you can lock it to the loop's groove of the loop in a way that would be almost impossible otherwise.

However, before I quantize, I'll copy the original MIDI track, label it, and bury it in a playlist so that if something goes horribly wrong in the quantize, I'll still have the original. Also, there are often places in a piano part that shouldn't be quantized, like grace notes or little flams. In those cases, I'll copy the needed section from the original track, and paste it into the quantized track.

MIDI also gives you the luxury of auditioning alternate sounds. Let's say that as you listen back to a mix you decide that a honky-tonk piano would sound better: Just go back to your software piano, and find the right sound. Or, even more dramatically, you can experiment with entirely different instrument sounds triggered by the MIDI information from the piano part.

Finally, although there are programs that can change the pitch and time of digital audio, key and tempo changes in MIDI are more precise and don't produce distracting audio artifacts—no matter how dramatic the difference between the new key or tempo and the original.

PIANO SESSION SETUP

Let's start with the basics of recording your piano, as you need to capture not only audio but also MIDI information (Figure 1). Let's begin with the MIDI track. In some recording software, the MIDI track also acts as a playback track so you can actually assign the software piano plug-in directly to the MIDI track itself. In others, you'll need a stand-alone MIDI track and an auxiliary track to which you'll assign your software piano plug-in. You'll also need an audio (usually stereo) track to record the actual audio—we'll see why this is important in a bit.

As setting up a session to record a software piano involves several steps, consider creating a software piano recording template that's ready to copy into a new session. You can then avoid having to create these tracks every time you want to make a piano recording.

PRINTING TO AUDIO

I highly recommend bouncing your MIDI-triggered sampled piano to a digital audio track when you're satisfied with the part (Figure 2). One reason is that software pianos can be a real drain on your system. By printing the audio and de-activating the piano plug-in, you'll end up with more available processing power for other plug-ins in your mix.

Another reason is that if your software piano company goes under, then the next time there are updates to the version of your recording software or your computer's OS, you'll be in trouble if your sampled piano software is no longer compatible. If you go back to an old session and find that the plug-in no longer works, you'll be *very* glad you printed the audio. Recording your MIDI piano's audio is your insurance policy against software-related headaches. EQ

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NATIVE INSTRUMENTS

THE FUTURE OF SOUND

4 EXAMPLES OF THE PREPARED DRUM SET

by Jake Wood

Neurotic engineers with deep pockets pay enormous sums of money to eliminate all the unwanted buzzing, ringing, hissing, rattling, and clicking within their recording setups. Bad channels spitting out dirty electrical signals are replaced, rooms are acoustically treated to silence odd reverberations, and coloring books are strategically placed in the lobby to keep noodling guitar players busy and out of the control room. Going to these lengths provides an environment for capturing great tones, but now a portion of the drumming community is moving in the opposite artistic direction, and engineers need to play catch-up.

Lead by avant-garde percussionists—as well as a desire to reproduce electronic music—this phenomenon has quickly spread into the worlds of rock, pop, and jazz. The drummers of this growing field are plowing through a new territory that embraces all the buzzing and rattling of the drums that would normally make engineers cringe. I call this style "Prepared Drum Set." Identical in approach to the prepared piano styles of the maverick composer John Cage, the basic concept is to alter the timbre of a drum or cymbal using foreign objects—car keys, wallets, springs, and so on. The practice has grown significantly in popularity with the rise of live electronica bands, with drummers simulating break-beats, hip-hop grooves, jungle rhythms, and unique sonic colors. It's a cutting-edge field that only some drummers know about, and of which even fewer engineers are aware. From an engineering standpoint, adding these tricks to the arsenal of sound manipulation can come in handy when a drum track isn't sitting comfortably in the mix. The results of these methods vary, but they typically alter a tone in the same manner that an effects processor would. Because it's the engineer's job to capture and enhance the sound, it makes sense to have a working knowledge of how to alter the drum tones at the source, rather than solely through miking and mixing.

If sampling, triggering, and sequencing are also part of the job, prepared drums can add a whole new section to the sampling library. Think about it. First came the drum, then there was the electronic sample of the drum, and now drums are prepared to emulate the sample that imitated real drums—thereby allowing you to sample the emulated tone of an electronic reproduction of an acoustic origin. Still with me? It's four layers deep, rich with history, and the subtleties make all the difference. Do not attempt to discuss this concept while inebriated.

TRASHY SPLASHY SNARE DRUM DELIGHT

If the drums could speak a language, a swearing snare drum would be a high priority among many drummers. This first preparation is about as close to reaching a percussive expletive as you can get. Brought to the spotlight by drummer extraordinaire Johnny Rabb, the most common drum preparation these days is to lay a splash cymbal on the top snare head, and strike the remaining available head surface. Different sized splashes give different results, but the essential effect is a harsh and trashy snare sound. I've found that 8" splashes work well on 14" snares because they leave some space for the stick to come down on the snare head. But a 10" splash also sounds great—smacking the top of the bell produces an awesome attack. To achieve the best results, tune the top snare head fairly tight, as a loosely tuned, heavy rock snare won't bring out the bite of the cymbal. If there are no splash cymbals handy, a hefty set of car keys resting on the head will produce a fairly similar, but slightly tamer tone.

DEAD RINGER

Heavy dampening of the top head has also grown in popularity. Placing a wallet on a taught snare head creates a staccato punch that is reminiscent of many lo-fi, old-school drum machines. For the extreme dampener, putting a folder or thin notebook over the snare head really changes the tone. The downside is that the volume of the snare is lowered drastically, so it helps if the drummer can accommodate, and play the other drums a bit quieter. Pumping that snare through a compressor will fatten the tone and make some heads bob.

FRANKENSTEIN'S HI-HATS

If you are stuck in a session with a Stewart Copeland 32nd-note hi-hat wannabe who lacks taste, and the parts aren't fitting the song, consider changing out the hi-hat cymbals with something a little less responsive. For a real heavy sound, two crash cymbals from 16" to 18" on the hi-hat stand sound enormous. Put two ride cymbals on there, and it will feel like Phil Rudd of AC/DC is behind the kit. Alternately, to match the previously mentioned lo-fi snare preparations, two splash cymbals mounted on the stand bring out both a 1930s jazz vibe *and* a fashionably indie children's-toy-instrument element.

EXTREME TAMBOURINE

Pried from the asthmatically stuttering hands of prima donna backup singers, tambourines have made their way to the drum kit as rack-mounted instruments. Not only do they provide a new voice on their own, but lay one on top of a ride cymbal, and you create a whole new dimension of steroid-enhanced sizzle. It takes up

a lot of space in the higher frequencies, but, if you have an open mind and the right situation, the sounds it produces can be inspired. Conventional hand-held tambourines are a little too big for this application, but, luckily, RhythmTech makes the Hat Trick—a miniature tambourine that's perfect for such situations. It's small enough not to get in the way of your sticks, but it still rattles like crazy. Just hang it from the wing nut of a cymbal stand, draped over the bell of the ride, and *voilà*—you're cooking bacon. **eq**

LEARN MORE!

To further study the concept of prepared drum set, here are some drummers who are inserting these techniques (and others) into various styles of music: Stanton Moore, Johnny Rabb, Matt Chamberlain, Yuval Gabay, Ahmir Khalib Thompson (also known as Questlove), and Glenn Kotche. Google 'em, because they are forging ahead with new tones, and they're reinventing what a drum mix is all about. *—JW*

EMPEROR'S TRYM TORSON ON RECORDING MAMMOTH TOMS

by Roberto Martinelli

Formed in 1991, Norway's Emperor became one of black metal's most influential bands, incorporating classical motifs, and even elements of Norwegian folk music in its bombastic, pagan assault. The band's violent history—three members were separately charged with arson, burglary, and murder—prompted it to implode in 2001, but it reformed in 2005. Here drummer Trym Torson details how he records his signature tom sound.

"I use Premier Maple Classic drums, and when I first signed with Premier, I asked for the biggest toms they had because I wanted them to sound huge," says Torson. "But when I tried the 12" in both the deep model and the shorter, power tom, the

DRUM HEADS

TRYM TORSON

shorter tom sounded way better. If the shell is too deep, the sound reflects back upon itself, and sound against sound reduces the volume. So I'm really glad I ordered the power toms—I didn't even change the heads that came with them. I've got 8"x7", 10"x8", 12"x9", and 14"x11" rack toms, and 16"x16" and 18"x16" floor toms.

"Here's what I do to get a big tom sound. First, I make sure the tone of the toms is optimum for producing a thundering recorded sound. I have this tuning instrument that you just put on the rim, and it tells you the tension of the head. The manual gives you rough guidelines about how a tom should be tuned if you want a certain sound. So I did all the top-head tunings to make the toms fairly high, so you can hear the attack. Then, I tuned the bottom heads a pitch down to get the low frequencies. It can be a nightmare to get the lower heads right—it's so easy for them to get out of tune.

"We used one microphone to record each tom, but we split the signal to two separate tracks. One track was used to bring out the attack, so you could hear all the hits I played. There was no low end at all on this track. The other track was worked to make the tom sound as good as possible—round, rumbling, and intense. This method allowed us to produce all the power *and* all the attack of the toms.

TAKEN WITH PERMISSION FROM EMPERORHORDE.COM

Torson unleashing his mammoth tom-tom assault in the studio.

"Even with this method, though, the recordings never come out the way I want them. It's the best we can do, but I guess you can't have everything you want. I can't make my drums sound like machine guns and like an old Led Zeppelin record at the same time. In addition, the drums have to sound good with the rest of the music. For example, when I recorded *IX Equilibrium*, I had an amazing tom sound in the beginning, but there was so much low end that it drowned out the guitar. So we had to cut a lot of the tone in the toms. It seems you always have to compromise!" eq

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CRAFTING HARD-HITTING HIP-HOP VOCALS

by Devine Evans

In today's world of major label hip-hop and R&B music, the need for an in-your-face vocal is near the top of the priority list for almost all of the artists I've worked with [*Note: Evans has worked with Mary J. Blige, Britney Spears, Outkast, and Janai Malee, among others*]. For years, people have employed digital reverbs to add the illusion of depth, but this can make your lead vocal sound distant, and not sit well in the mix. So I've devised a way of using other effects—delay, tremolo, and stereo expanders—to get that face-slapping hip-hop vocal sound.

Here are the plug-ins I use on the lead vocal channel to establish gain control: compression (2:1 ratio), limiting (3:1 ratio), de-essing (cuts at 6kHz and 12kHz), and EQ (high- and low-pass filters engaged with a slight boost on the high shelf, a boost in the low and high mids, and a small dip in the lows).

All three EchoBoy plugs in-line.

This screen shot displays the last channel of my effects chain with the Tremolator at the top—adding sort of a weird vibrato to my delays—followed by the Doppler plug-in to further enhance the stereo image.

DYNAMICS PROCESSING

My first insert in an effects chain is always a compressor—such as the Waves R-Comp—in order to dial in a gentle compression that tames unruly dynamic fluctuations in the vocal track. Next in line is always a more aggressive limiter that will catch the more dramatic dynamic shifts that can make a performance sound horribly uneven. I like the Waves L1. It's heavy-handed, but its sound is pretty invisible. The third and fourth inserts in my effects chain are two separate Waves DeEsser plug-ins—each with different frequency settings so that I can control the sibilance. I set the first plug to focus on the 6kHz range, and the second to address the 12kHz range. The last insert is Digidesign's 7-Band EQ.

SENDS & RETURNS

In Pro Tools, I then create five stereo aux returns. This is where I will set up an array of digital delay, tremolo, and imaging plug-ins. On the vocal channel, I activate Send A, and bus the output to the corresponding input of my first aux return. Next, I route the output of effects Return A into B, B into C, C into D, and D into E. Personally, I prefer dark delays, so the first thing I do on Channel A is insert Digidesign's 1-Band EQ with the low-pass filter engaged. Doing this ensures that the delays I set up afterwards all react to the frequencies I've specified. For organizational purposes, it's a good idea to name all of the inputs and outputs after the appropriate plug-in. It's also smart to color code, group, name, and add comments to every track in the Effects Return section for quick and easy recognition. This is especially critical for sessions with a high track count.

ECHO BOYS

Next, I dive into a ton of delays. I will often take one SoundToys EchoBoy plug-in, and feed it into another EchoBoy, which feeds into yet another EchoBoy. I set all three plugs to different EchoTime, Mix, and Feedback settings—the first plug to an eighth-note delay, the second to a 16th-note delay, and the third to a 32nd-note delay. Doing this results in so many delays going in so many places that it creates a sort of false reverb that doesn't seem to add any "distance" to the sound of the lead vocal in the mix.

TREM-U-LATION

Once the settings on all the EchoBoy plugs are dialed in, I route all of my delays into the SoundToys Tremolator—a vintage tremolo simulator. The setting of this plug will vary greatly from song to song, but, generally, I start by setting the Depth to around two o'clock, Groove at nine o'clock, Accent all the way to the left Sync setting, and Rhythm to an eighth-note pulse (or whatever time the first delay was set to).

STEREO IMAGING

Though things should sound pretty good at this point, it might be cool to alter the panning a bit using a few imaging plugs. I really like the Waves S1 Stereo Imager, Waves Mondo Mod, and GRM Tools Doppler plug-ins. There are no perfect settings for any of these—everything depends on what kind of effect you want to achieve. Play around with them, trying settings that result in a sound that's not so traditional (such as panning the vocal straight down the middle). Try some sweeps from left-to-right, or widen your stereo image to get a big, all-encompassing vocal mix. Remember, mixing is an art form, and the key to success lies in every decision you make—just as every small stroke from a painter's brush determines the final masterpiece. EQ

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HOW TO PREVENT DIGITITIS

by Bruce Bartlett

You're recording and mixing in a DAW set to 24-bit resolution, and you think you're doing everything right, but the final product sounds vaguely distorted. The tracks were recorded at just under 0dB, and your meters don't show any clipping, but the sound of your mixes is harsh and fatiguing. What's going on? Chances are the signal *is* clipping, but the meters don't indicate it. Here are three ways to fix the problem.

But first, let's go over some digital-audio basics. The A/D converter in your audio interface measures or samples the incoming analog signal many thousand times a second, and it converts those samples to the binary numbers that you record on your hard drive. Figure 1 shows an analog sine wave that is sampled periodically. The digital meters in your DAW show *those* sample levels—not necessarily the peak signal levels that appear *between* samples. In your D/A converter, a lowpass filter (or reconstruction filter) creates occasional peaks between the samples that can be up to 3dB higher than the measured level at certain high frequencies. If your recorded levels are near 0dB, this can cause clipping that the meters don't display. Similarly, some plug-ins can create larger peaks than those that went in—without any increase in volume, or audible change in the program. Even an EQ cut or roll-off can increase signal levels by producing intersample peaks due to ringing at the cut frequency.

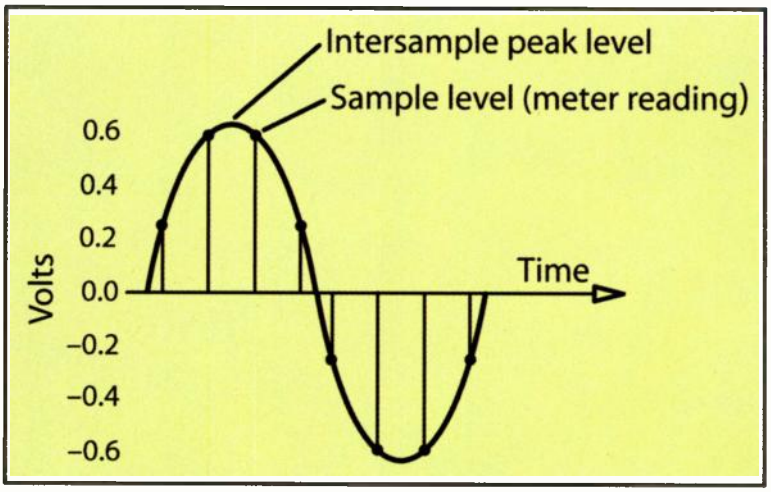

Fig. 1. Analog signal levels can exceed digital sample levels.

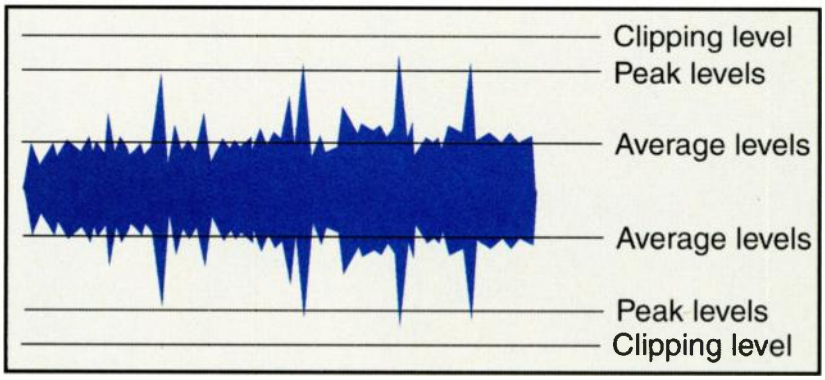

Fig. 2. Average levels and peak levels in a musical signal.

LOVE THE PEAKS

As shown in Figure 2, a musical signal changes in level continuously as it plays. Imagine a musical passage with a low-level synth pad, but with high-level drum hits. The average level or volume of the passage is low, but the transient peak levels are high. Peak levels may be up to 24dB above average levels, depending on the type of signal. Percussive sounds have much higher peaks than continuous sounds (synth pads, organ, flute)—even if the two signals have similar average levels.

The meters in your DAW can show signal levels in two modes: RMS and peak. RMS readings correspond to the average levels, and peak readings show the level of peaks, or short transients. The average or RMS level indicates approximately how loud the sound is, and the peak level shows how close the signal is to clipping. So, as you don't want to clip or distort the signal while recording or mixing, use peak metering—not RMS.

REDUCE RECORDING LEVELS

In general, reduce your recording levels by turning down the gain in your mic preamps. Record at about –6dB maximum in peak-meter mode. One benefit of lower recording levels is that you won't overdrive your mic preamp. The distortion of most analog gear increases as you approach maximum gain. Another benefit of reduced levels is that it creates some headroom for your plug-ins. Going for 6dB of headroom should eliminate any invisible overs. Then, you can set up your mix balances without having to adjust levels every time you insert a plug-in.

Remember, the recorded level on each track drives the plug-ins—not the track fader, which comes after the plugs. If a track was recorded at 0dB, use your DAW's trim control (or insert a –6dB trim plug-in)

CONTINUED ON PAGE 65

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HOW TO PREVENT DIGITITIS

9 TIPS FOR SMOOTHER SOUNDS

- Dirty contacts can cause signal rectification and distortion. Clean connectors with agents such as Caig Labs DeoxIT (www.caig.com).
- Use audio interfaces designed for low jitter.
- During mastering, add dither when truncating from 24-bit to 16-bit. This eliminates quantization distortion at low-signal levels.
- Minimize the calculations your DAW has to do. Each calculation creates some error or distortion by increasing the word length beyond 24 bits. Rather than boosting the gain of a musical section and dropping it later, undo the changes, and apply only the correct amount of gain change.
- Consider using fader moves rather than compression. Compression adds distortion because it changes the waveform. Too-fast release times cause distortion in low-frequency notes, and also cause pumping. A fader-setting change on a series of notes is less audible than a compressor working on each note.
- When you peak-limit and normalize a stereo mix to make a hot CD, try not to exceed 6dB to 7dB of peak reduction, as limiting adds distortion. Use even less limiting on mixes that don't have loud transients. Rather than normalizing the limited signal to 0dB, normalize to –3dB to –1dB to allow for intersample peaks.
- Experiment with an analog tape plug-in to warm up a track, or record to analog multitrack tape and dump the tracks to your DAW for editing/mixing.
- To reduce graininess in a reverb plug-in, use a high-density setting, or use a convolution reverb.
- When recording source sounds, consider using less high-frequency boosts. Cutting around 3kHz to 7kHz tends to reduce harshness. Try a 4kHz lowpass filter on distorted guitar amps to take the edge off, and try multiband compression on voices that get raspy. Set the compressor so it kicks in above 3kHz or so during loud passages. Also, many condenser mics have high-frequency peaks that can sound brittle or sibilant. Apply a high-frequency roll-off, or better yet, use ribbon mics or flat-response condensers. Try moving the mic to a spot that sounds mellower, as well. Mic a vocal at nose height if it sounds too edgy when miked at mouth height. Mic a trumpet off-axis, and position the mic near the edge of the speaker cone on a guitar amp. Finally, use a tube mic preamp that can produce euphonic even-order distortion that's toasty warm. —*Bruce Bartlett*

CONTINUED FROM PAGE 62

to reduce the signal level going to your plugs. Do this before setting up a compressor or gate, because trim affects their gain reduction.

But doesn't a low recording level increase noise? Yes—but in practice, it's not a problem. A 24-bit recording has a theoretical signal-to-noise ratio of 144dB. So even if you record at -6dB, you're still far above the noise floor, and you won't hear any increase in hiss.

KEEP REDUCING

Here are some other spots in the signal path you can set your level to a maximum of –6dB:

Stems. At the output of each plug-in, check for clipping—especially in an equalizer set to boost. Find the plug's output gain control, and turn it down until clipping stops. You might have several plugs in series, and you don't want the output of one plug to overload the input of another.

Upsampling plug-in outputs. These DSP processes upsample the signal (increase its sample rate), do the effect, and then downsample the signal at the output. The downsampler acts like a partial reconstruction filter. So even with the effect turned off, the up and down sampling can create overs.

Mixdown master. Set the maximum peak level of the master output bus to –3dB to –6dB to allow for signal peaks that do not display on sample-reading meters. You can make up the gain later during mastering, where you boost the overall level or peak-limit/normalize to make a hot CD. Try to keep the master faders at or near 0dB. If the master faders are set low, you will turn up the channel faders to get a good mix level, which results in high-level signals that can overload the mix bus. [*Note: Mix bus overload is not a serious problem if the DAW uses 48-bit or 64-bit floating-point math in the mix bus, because float processing can pass signals above 0dB without clipping. Still, it's good practice to control the levels hitting the mix bus. Summing calculations become less accurate when signals exceed full scale.*]

Compressor makeup gain. Don't do it! Makeup gain can cause the initial uncompressed transient to clip by boosting its level. Raise the track fader instead.

A TIP FOR MASTERING

If your mix is going to a mastering engineer, omit any stereo-bus processing. Record the file at a maximum of –6dB to –3dB so that the mix doesn't overload the mastering engineer's D/A converter. If you are mastering your own mixes, use an oversampling meter that displays the actual reconstructed signal, or set the limiter ceiling at –3dB to –0.3dB—not 0dB or higher. This avoids making files that sound distorted on listeners' playback systems, as many CD players distort with samples above the clip level. EQ

CHEAT SHEET

KONTAKT 2 PLAYER

BY CRAIG ANDERTON

Cheat Sheet delivers concise, explicit information on how to do specific recording/audio-related tasks. This installment describes features in the Kontakt 2 player (KP2), which is used as an audio engine for several sample libraries. It's not particularly well-documented, but we'll take care of that.

REVEAL OUTPUT MIXER AND AUX RETURNS

Click on the Outputs button in the top control strip.

ADD AN EFFECT TO A MIXER CHANNEL OR AUX CHANNEL

In the mixer section, click on Show Inserts. Click on one of the four downward-pointing arrows in a channel, and choose the desired effect.

CONFIGURE MIXER CHANNEL OUTPUT

Click on the Conf. Button at the bottom of the channel strip. You can name it, set the number of audio channels, and assign a corresponding physical output.

EDIT OR BYPASS A MIXER EFFECT

Click on an insert effect name in the mixer, then click on Edit Effect to reveal the effect's GUI above the mixer, ready for tweaking. The GUI also contains the effect's bypass switch.

SAVE AN EFFECTS PRESET

When the insert effect GUI is visible, click on the Pre button to the right of the Byp button. Name the preset; saving creates a Kontakt Module Preset (.nkp suffix).

MORE EDITING OPTIONS

The data set of sounds loaded for KP2 is also compatible with the full version of Kontakt (now up to version 3), which provides more editing features

CONVOLUTION REVERB

Kontakt Player 2 instruments that offer convolution reverb will accept impulses from Kontakt 2 and other sources, but only in stand-alone mode.

SHOW INSTRUMENT AUX SENDS

Click on the Aux button in the upper right of an instrument, to the immediate right of the MIDI port selection. Each instrument has four aux send controls, which appear in a strip immediately below the instrument. The four aux returns (which feed the mixer output along with the various individual outputs) can have up to four insert effects, using the same roster of effects as the individual outputs.

LOCATE THE DOCUMENTATION

The documentation is the instrument folder that was installed onto your hard drive when you installed the program.

DELAY TEMPO SYNC

For delay Time parameters, right-click on "ms" and choose the rhythmic value. This works only for Delay processors used as mixer inserts (output channel or aux return), not the Delay effect in the instrument itself.

CREATE A MUTE GROUP

A mute group creates a group of sounds where triggering a sound cuts off any existing sounds that are playing, such as having closed hi-hat cut off an open hi-hat. Although you cannot do mute groups per se, you can create an instrument that consists only of the sounds you want to have as part of a mute group, then set polyphony to one voice using the Max parameter in the instrument GUI.

MERGE MULTIS

Merging Multis places the instruments in the merged multi below the instruments in the existing Multi. To merge, click on "No" in the Replace Multi dialog box when you call up the new Multi.

CHANGE LIBRARY LOCATION

You can place the library anywhere, not just on the root drive. If you change its location, choose Options > Handling tab, then select the library location in the Installation Base Path field along the bottom of the window.

AUTOMATE A PARAMETER VIA HOST AUTOMATION OR MIDI CONTINUOUS CONTROLLER

Although the manual recommends using MIDI automation in standalone mode and host automation when used as a plug-in, you can control plug-ins via MIDI. Under the Kontakt Player 2 logo in the browser, click on Auto and select Host or MIDI automation. Drag the MIDI CC# or Host Parameter number to the knob or fader you want to automate. If it's a valid assignment, the cursor will turn into a hand. Note that automatable parameters may be restricted to volume, tune, and pan.

CANCEL AUTOMATION ASSIGNMENT

To cancel an automation assignment, click on it in the list of Host Parameters or MIDI CC in the browser under the appropriate Auto tab, then click on the Remove button toward the browser's lower right.

VELOCITY SPLITS AND NOTE SPLITS

Click on the Instrument Options button (the button to the left of the instrument title that looks like a couple of gears). Under the Instrument tab, you can restrict the instrument to a particular Key Range or Velocity Range.

TRACK HOST TEMPO CHANGES

Click on the MasterKontrol button, then click on Extern Sync. The tempo value will not change until you start playback in the host sequencer, but will thereupon track any host tempo changes.

ACCESS THE BUILT-IN TUNING REFERENCE

Click on the MasterKontrol button. The Reference Tone module is toward the right of the MasterKontrol strip, with the option to change volume, turn the tone on or off, and select the desired pitch. This is great for fixing out-of-tune samples: Play the sample and adjust its tuning until it matches the reference.

CHECK THE VIRTUAL KEYBOARD

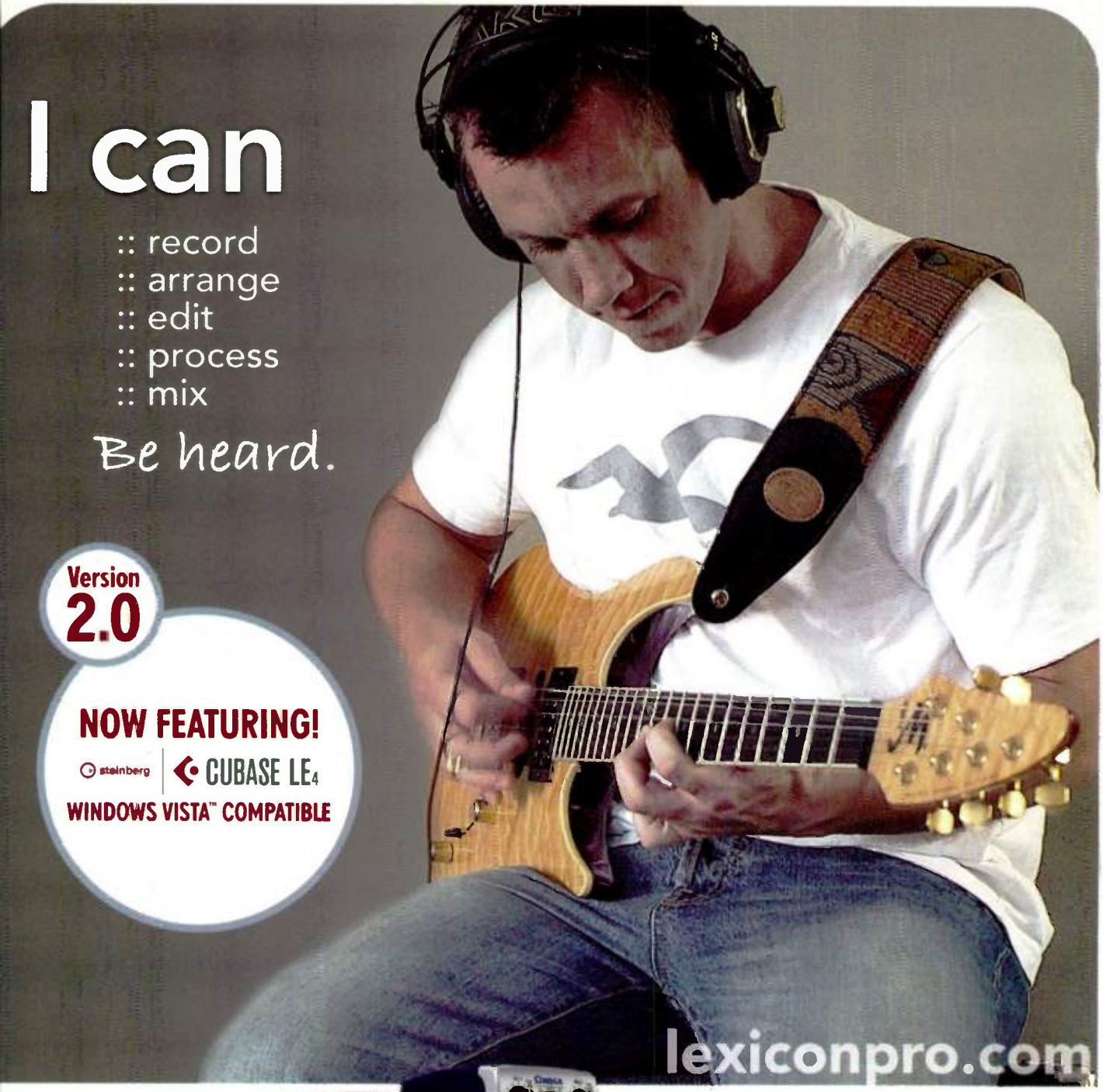
Click on the Keyboard button in the top control strip to display the virtual keyboard. You can use it to trigger notes, but the color coding of the keys has meaning: Blue indicates keys with samples assigned, white means no samples are assigned, and red keys are Keyswitches where holding them down affects other aspects of the sound.

SAVE RAM

After finishing a part or parts, you can remove any unused samples on a global or instrument level. For a global purge, click on Purge in the top control strip and select Update Sample Pool. To purge samples associated with a particular instrument, click on Purge within the instrument interface and select Update Sample Pool. Note the indicator below the purge button: Green = all samples are loaded, Orange = samples have been purged, Red showing up when you hit a note means the sample is no longer available for that note. Click on Purge and select Reload All Samples if you want all samples to be available.

TRADE OFF RAM FOR VOICES

Under Options, DFD tab, reserve less RAM if your system needs RAM for other applications, or allocate more RAM if you need to minimize hard drive activity, which may allow for more voices. EQ



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POWER APP ALLEY

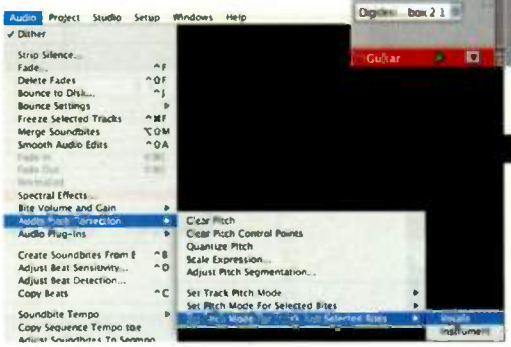
BY CRAIG ANDERTON

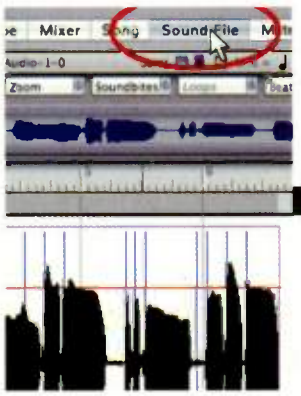
MOTU DIGITAL PERFORMER

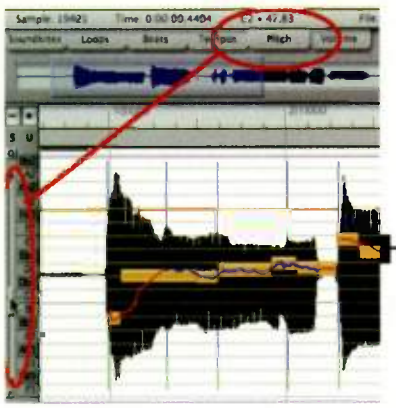
Repair vocals quickly with pitch correction

OBJECTIVE: Use Digital Performer's audio pitch correction features to fix pitch problems with vocals, or add special effects.

BACKGROUND: Because DP has very sophisticated pitch correction options for monophonic vocals and instruments, it's easy to overlook the fact that you can make simple, effective vocal edits very quickly by drawing in correction curves with the pencil tool. Although there are different ways to do this, here's one possible workflow.







STEPS

1. Select Memory Cycle and loop the area you want to edit. Although you can work on the entire vocal, I find it more efficient to isolate specific area that need fixes.

2. Select the soundbite to be processed.

3. Set the processing mode to "Vocals." It's generally best to choose "Set Pitch Mode for Track and Selected Bites" unless you need different pitch modes for the bites in a track.

4. Click on the Sound File tab. Although you can edit pitch in the Sequence view by selecting Pitch (instead of Soundbites, Volume, Pan, etc.), I prefer the more expansive Sound File vie

5. Click on the Pitch tab, then use the scroll bar on the left to bring the pitch curves into vie

6. Go to the Studio menu and select "Tools."

7. Use the Pencil tool to modify the pitch correction curve as much or as little as you want.

TIPS

- Make sure you do step 3 before making any adjustments, as selecting the processing mode influences DP's DSP analysis function
- DP has additional ways to tailor the pitch analysis process, as well as multiple methods (including pitch quantization) you can use to modify pitch. It's well worth spending some quality time with the manua to exploit these features to the fullest.

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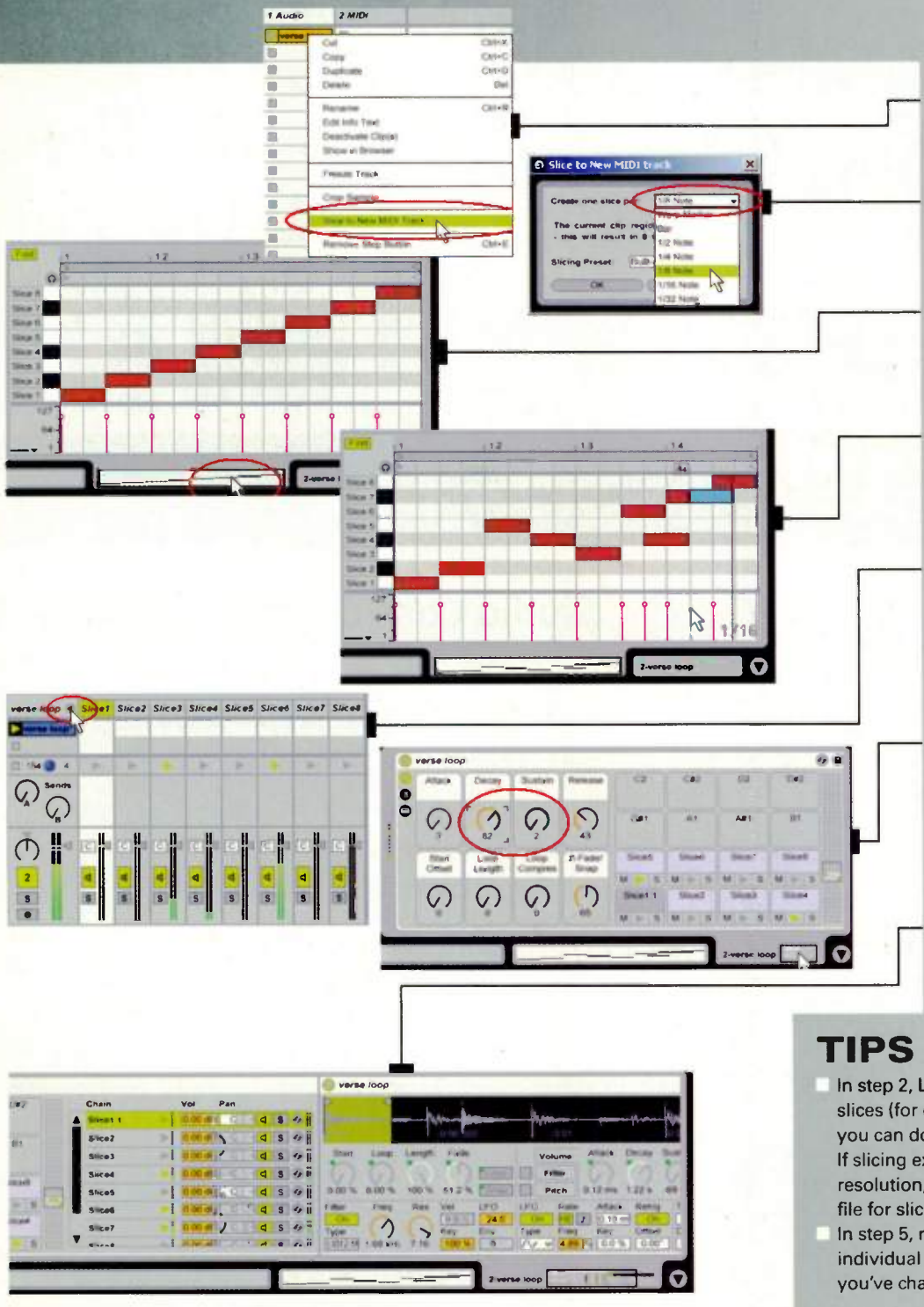
BY CRAIG ANDERTON

ABLETON LIVE 7

Fun with slicing and dicing

OBJECTIVE: Deconstruct an audio file into individual slices, and then apply editing options to alter the file in various ways.

BACKGROUND: Live 7 allows dividing an audio file into slices. You can then edit each slice with respect to placement in a loop, filtering, envelope characteristics, and much more.



STEPS

1. Right-click on a clip (Mac, Ctrl-click) and select "Slice to New MIDI Track."

2. Select the slicing division, typically 1/8th or 1/16th note, then click "OK."

3. Live creates a MIDI track. Call up the MIDI Note Editor to see a chromatic series of notes representing each slice.

4. To edit the loop, change the pitch and/or location of slices, as well as alter slice velocities, in the MIDI Note Editor.

5. Click the newly created MIDI track's "unfold" button to reveal a mixer channel for each slice. You can now mute or solo individual slices, change their levels, etc.

6. Live also creates a device chain for the loop, which has macro controls for the Simplers used to play back the slices. As one example of editing fun, turn down Sustain, and vary Decay to create more percussive slices.

7. You can also show the Chain List to select individual slices, then use Show/Hide Devices to change characteristics of the selected slice (*e.g.*, add filtering, LFO, envelope response, etc.). Cool!

TIPS

- In step 2, Live won't allow more than 128 slices (for example, with a 32 beat long file, you can do a maximum of 1/16th note slices). If slicing exceeds this limit, set a lower slice resolution, or select a smaller region of the file for slicing.
- In step 5, remember that the controls affect individual slices, regardless of whether you've changed their positions or not.

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AMS NEVE 8801 CHANNEL STRIP

That Fabled Neve Sound in a Brand New Box?

Behold! The Neve 8801 Channel Strip.

by Jay Matheson

Unl've tried my fair share of mic pres over the years, amassing a veritable arsenal of vintage Neve strips (and more than a few clones) in the process. In my opinion, the clones have always lacked in delivering that unique Neve sonic signature, but what they failed to deliver in terms of sound they've made up for in reliability.

I don't really want to choose sound quality over reliability, or vice versa, so I picked up one of these new 8801 channel strips, hoping that it could solve some of my problems . . . yet I remained skeptical about AMS Neve's claims that a new product could offer the same magical sounds as the originals. Here's what I found out.

OUT OF THE BOX

The 8801 has a whopping 26 switches and 22 knobs on the front panel—a design feat achieved by having most of the controls double as switches (*i.e.*, when the gain pot is pushed in it acts as an input selector; all eight knobs in the dynamics section double as switches for functions such as attack time and auto release; and most of the pots in the EQ section switch functions between bandwidths, including shelving and bell curve presets).

Scared? Don't be. The 8801's basic features are easily understandable (these include a mic pre with variable gain pad, phase invert, and 48V phantom power; a line mic/DI switch, front/rear panel connectors, high and low pass filters, digital output option, an output control with meter, and the ability to recall all settings stored by the user). The compressor and expander sections are full-featured with control over threshold, ratio, attack, release, makeup gain, separate gain reduction meters, and the ability to side-chain the filters or the EQ section. Note that this is a very useful feature, as side-chaining allows using EQ or filters to affect the signal that triggers the compression. For example, side-chaining a high pass filter can keep lows from pumping too much, and by boosting selected highs when an EQ is side-chained, the compressor can act more as a de-esser.

The EQ has four bands, with the middle two being fully parametric. Conversely, the top and bottom bands are sweepable, with switches to select bandwidth and shelving. This is a serious upgrade from earlier units—such as the 1073—where all the values were fixed, and thus offered less flexibility in terms of creative and/or corrective use of the EQ section.

APPLYING THE 8801

For testing, I patched the mics into the 8801's rear panel input and ran the preamp's analog output into the analog input of my Digidesign 192 I/O. From there, I tackled a variety of sources, starting with a 22" '70s Ludwig kick that was miked with an Electro-Voice N/D 868, just inside the opening of the outside head. The pre on the 8801 sounded clean and accurate—not as colorful as some of my Daking or Neve clone pres that I tend to prefer for kicks. The EQ, however, added some incredible top end snap that some of my vintage pieces (and retro clones) couldn't touch. Not too shabby a performance, I must say.

Up next was a distorted electric guitar sent through an old Marshall cabinet, miked with both a Shure SM57 and an SM7 running through the 8801. The flexibility of the EQ section was simply a godsend. I used the high pass filters around 100Hz, boosted some of the low mids, and added about 3dB of compression on the SM57 for some added thickening. The result: a track that needed no fixing in the mix. The 8801 could quickly become my go-to pre for recording guitar cabs, as it's one of the few pres that still comes equipped with high and low pass filters—features I find especially useful in ridding guitar tracks of unruly subsonics and overly-airy highs.

Next in line was a loud male rock vocalist who had settled on a Blue Bottle mic to scream into. Generally I would use a more colorful pre for rock vocals, but, again, the 8801s exceptional EQ gave me all I needed to ensure the tracks sounded energetic, with plenty of body and presence to boot. The compressor section sounded good on this source, though I must say that when we got into 4–5dB of reduction the 8801 seemed to take the edge off the top end. I would recommend inserting a Universal Audio 1176 or maybe an optical compressor into the signal chain as well when using the 8801 on vocals of this type, as I'm into the habit of using a pair of compressors on loud vocals; I feel that the 8801 works best when paired with another compressor in an application that needs heavy

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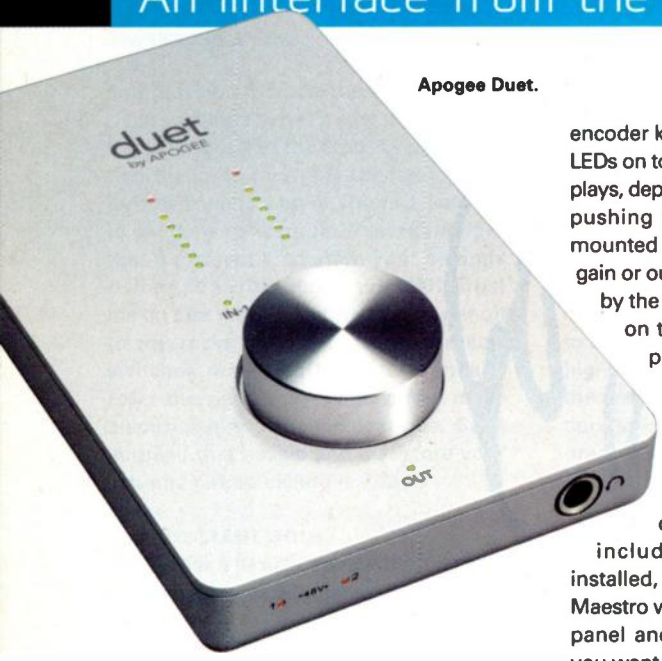
APOGEE DUET

An iInterface from the Masters of AD/DA



Apogee Duet.

by Michael Ross

Shortly after releasing my first record, recorded through a ProTools Digi 001, I purchased an Apogee Mini-Me that made me want to go back in time and re-record everything—it furnished incontrovertible proof that analog-to-digital conversion makes a huge difference in sound quality. Apogee now offers the Duet, for half the price of what that Mini-Me cost me, bringing high-quality conversion to the low-income recordist.

OVERVIEW

The Duet is a portable audio interface with control functions built directly into Apple's Logic Pro and GarageBand software. It works only with Apple computers, mimicking the Mac's elegant design with its brushed metal housing and "iPod white" breakout cables. The breakout system's Medusa vibe is somewhat at odds with the box's elegant looks; still, the unit is compact and FireWire-powered for portability.

The breakout cables include two XLR mic ins (each with phantom power and up to 75dB of gain), two 1/4" high impedance ins, and twin 1/4" outs for powered speakers. The box itself sports inputs for the I/O connector, FireWire cable, and headphones, as well as a large, multifunction encoder knob and tiny LEDs. Two rows of LEDs on top act as input or output level displays, depending on the function chosen by pushing the encoder knob. Other top-mounted lights indicate which level, input gain or output level, is currently controlled by the top panel encoder, and a couple on the side let you know when the phantom power is engaged for each channel.

If you don't use Logic and your GarageBand is lower than the recommended version 4.0, you can still have an on-screen control panel using Apogee's included Maestro software. Once installed, anytime you plug in the Duet, Maestro will automatically open its control panel and mixer screens, asking you if you want to choose the Duet for your Mac sound output or retain your current audio settings (you can always choose Duet within your DAW).

On its Level page, Maestro's control panel provides three virtual encoder knobs that let you choose XLR Line +4dBu, XLR –10dBv, XLR mic, or instrument level for each input, and line out or instrument amp out for the output. Here you can also reverse the phase of the inputs, add phantom power, and mute outputs. The Advanced page lets you specify whether headphones, speakers, or both get muted when you click the mute button, provides four virtual MIDI controller knobs (controlled with the hardware knob), and can send MIDI control data or Song Position data to software applications. A separate Mixer page contains sliders for balancing direct monitoring with the DAW signal, and overall output volume controls. You can also bypass the Maestro Mixer and control the outs directly from your DAW.

IN USE

The Maestro software and drivers installed easily, and once installed, I just plugged in the Duet, a set of headphones, and was ready to go—simple. First I did an A/B test, listening to iTunes alternately through my M-Audio 1814 interface (used for all comparisons) and the Duet. The 1814 sounds good, but through the Duet the separation and definition of individual parts was distinctly better.

Booting up Ableton Live I recorded a shaker and an acoustic guitar using a Soundelux mic and a Shure SM57, then captured a Danelectro electric guitar overdriving an Orange Tiny Terror head powering a 12" Eminence speaker. I recorded everything first through the 1814 and then through the Duet; the shaker's transients were less distorted through the Duet, the rhythm was more defined, and more right-hand finger attack on the acoustic guitar was revealed, as well as a tighter low end. Only the distorted electric guitar seemed slightly preferable through the 1814, as it sounded warmer and smoother. Actually, combining the crystalline sound of the Duet with the creamier but murkier M-Audio device produced the ideal electric tone.

CONCLUSIONS

The Duet makes the dream of an affordable Apogee come true. Compared to many other interfaces, the Duet is on another sonic level, with its amazing definition and minimal distortion. I would prefer a slightly larger box that accommodated the I/Os, and unlike the 1814's multiple outputs, the Duet's single set of stereo outputs won't work for those requiring separate monitoring ability for live performance. Still, these are quibbles about an interface that offers legendary sound at a real-world price. EQ

PRODUCT TYPE: Portable audio interface for the Macintosh enthusiast.
TARGET MARKET: Mobile and home recordists who have lusted after heretofore-expensive Apogee sound.
STRENGTHS: A must for those seeking big studio sound in a small package at a great price.
LIMITATIONS: Not optimal for live performance. A few debatable form over function issues.
LIST PRICE: $495
CONTACT: www.apogeedigital.com

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ROLAND VG-99

The Future of Guitar? That's What the Box Says . . .

by Brian Murphy

Roland VG-99.

Quick question: Have you ever been in the middle of a session and thought "nothing would flesh out this song better than a 12-string," but you didn't have one on hand? Ever think to yourself, mid-album, that the next song could really use a Rickenbacker 360 to help you get that classic vibe you've been looking for? Ever needed the sound of an old tube amp, but you had only a solid-state combo on hand . . . or wanted a tripped-out synth accompaniment, but couldn't find a keyboardist—or even a keyboard—on short notice? The VG-99 claims to make this all possible and more, and seems clearly designed for the studio as well as the stage. But does this little box really live up to its marketing hype? Let's explore.

OVERVIEW

The VG-99 is a guitar processor/USB 2.0 recording interface that uses Roland's much-hyped COSM (Composite Object Sound Modeling) technology. If you want to read more about the COSM, we suggest you head to Roland's website (www.rolandus.com), because we simply don't have the real estate here to do a deep treatise on the subject of COSM.

To start using the VG-99, you need either a guitar outfitted with a 13-pin jack (like a Godin) or you need to purchase Roland's GK-3 pickup and attach that to your axe of choice (a Brian Moore MC1p.13, in my case). The VG-99 is plug-and-play, but if you want to use the included cross-platform editor/librarian software, you'll need to install some software and drivers. Also, depending on the application, in order to use the MIDI converter you may need to take advantage of the advanced driver setting, which allows using Roland's drivers instead of your computer's standard driver. However, for going straight into a MIDI keyboard or sound module, this isn't necessary.

After getting these basics squared away, you should find that the VG-99 makes life much, much easier for you. For those learned in the arts of guitar modeling processors, you could look at the VG-99 as a single box containing two updated VG-88s, a GI-20 MIDI converter, and two equivalent GT Pro effect processors. Working from the front end of the two channels there are numerous choices in guitars, poly FX, pedal effects, and amps . . . and any of these can be switched, combined, or layered. This is where the unit really demonstrates its power: The programmability makes the VG-99 a significant step up from previous models.

APPLYING THE VG-99

The guitar selection is one area where the VG-99 really smokes. You can choose from nine electric guitars (including two Strats, Les Paul, Rickenbacker, Tele, Dano, 335, and an L4). Acoustic models include the Martin D-28, Gibson J-45, Guild D-40, banjo, sitar, dobro, and a pretty good-sounding nylon. You also have two bass models (a Fender Jazz and Precision bass). Do they sound good? They sound good. Great, even. Of the 200 presets, I found that the steel string presets sound suitably dark and warm, some of the distortion settings are downright screaming, and the 12-string and open/alternate tuning options are particularly nice for recording on the fly, or in a small studio that isn't equipped with a million in-house guitars. Add into this the 11 synth models (including the infamous GR-300, considered by some to be the most "natural" guitar synth ever made), and you really are looking at a new world for guitarists.

But as far as I'm concerned, the variable model aspect of the COSM guitars is where the real fun is. After all, the market is flooded with guitar and amp sims. What makes the variable model cool? For example, on the variable model you can place the pickups at an extreme angle, anywhere between the bridge and the 15th fret. This allows for some truly unique sounds. In short, the variable model is nearly 100 percent fully customizable—you can even choose virtual roundwound or flatwound strings.

Another unique feature is the Poly Fx. As the 13-pin cable is sending audio from the six strings separately, the VG-99 can apply a compressor, distortion, octave shift, or a "slowgear" (attack delay) effect individually to each string. There are almost 100 effects pedals to choose from (the wah is particularly special, as you can change the sweep of the pedal and the tone within the custom setting), and some unusual features like Defret (to simulate a fretless guitar). Pretty rad.

As a guitarist who prefers to perform live in the studio, I found using the VG-99's D-Beam feature and the Ribbon Controller especially fun to play with while tracking. For those not familiar with the D-Beam, it's a controller with two little

lights over which you can pass your hand or the guitar neck (vertically or horizontally) to produce effects such as a filter, or simulate the bending a vibrato tailpiece arm. You can even determine what type of vibrato: Fender, Bigsby, Floyd Rose, or Steinberger's Transtrem. Sick! Furthermore, the VG-99's dynamic settings allow switching between the A/B chains by how hard or soft you pick the guitar, effectively achieving guitar parts that are otherwise very difficult to produce. Speaking of cool guitar parts, the Freeze function is unique: It provides infinite sustain by "holding" notes, controllable by the D-Beam.

The ability to control/record software instruments and audio production software right from the VG-99 via the MIDI converter and the USB 2.0 and S/PDIF outs is of specific interest to recording guitarists, making the VG-99 a powerful studio tool. For example, using the A/B chain and the MIDI converter allows you to record three tracks at once (two audio tracks and a MIDI track), making it easy to capture exact, multiple, fast lead lines or rhythm parts without the need to overdub.

CONCLUSIONS

The bottom line is impressive: The VG-99 is a fully portable, feature-rich toolbox that is equally at home in the studio or on the stage, giving the recording guitarist an arsenal of audio and MIDI capabilities to help create and record his/her music. There are a few things that could be improved upon (the Defret option, for example, doesn't sound quite right), I would have appreciated some extra instrument and amp models (MusicMan bass, please!), and the ability to further tweak some of the presets would be welcome—although the 4,000 tweakable parameters in each preset will take you pretty far.

Still, all in all I have to say that the VG-99 is both versatile and useful. Is it the "future of guitar?" Possibly, but that will depend more on guitar players opening up and utilizing this kind of technology than it will on this technology simply existing! EQ

PRODUCT TYPE: Guitar processor and audio/MIDI interface.
TARGET MARKET: Recording guitarists looking for a slew of tonal possibilities without having to procure additional gear.
STRENGTHS: COSM guitars, amps, and effects sound good. MIDI conversion that can send separate MIDI data over separate MIDI channels on each string. USB, S/PDIF, and (analog) XLR and 1/4" connectivity. Increased programming options. Presets organized by category makes it easy to plug-and-play. Includes cross-platform graphical editing software.
LIMITATIONS: Certain desirable guitar and amp models are absent.
LIST PRICE: $1,399 ($1,599 with GK-3 pickup)
CONTACT: www.rolandus.com

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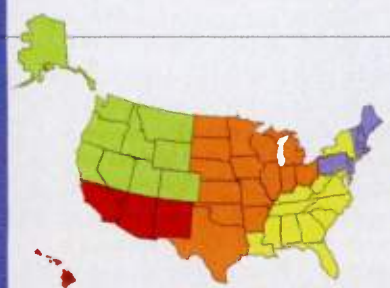

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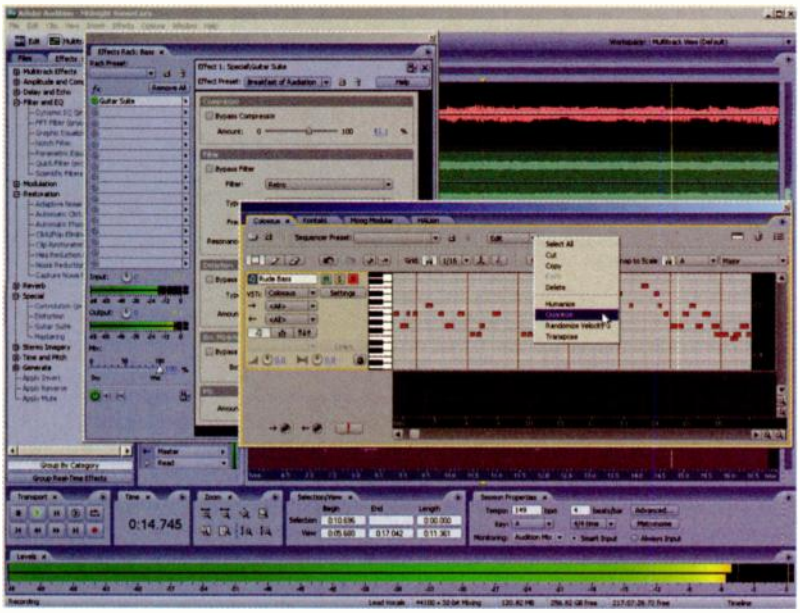
ADOBE AUDITION 3

From Digital Audio Editor to Digital Audio Workstation



Yes, that's a MIDI piano roll you're seeing, tabbed with additional ones for other MIDI tracks. Also note the Guitar Suite (along with several new effects listed in the effects section), and the customized blue tint for the interface.

by Craig Anderton

When a big company buys a small one, anything's possible. Fans of Syntrillium's Cool Edit Pro feared the worst when it was bought by Adobe, but the company has more than done right by the program—not just with updates, but by keeping the price reasonable and remaining true to Cool Edit's spirit.

As a fully-functional trial download is available (Windows only), there's little point in delving into pages of details; the question is whether you want to take the time to check out the software. While Audition 3 may just seem like version 2 with MIDI piano roll editing and VSTi support, look deeper. . . .

MIDI: ON A (PIANO) ROLL

Although Audition could import SMFs and handle ReWire, it couldn't record or edit MIDI. V3's piano roll editor changes that, but it's different from the norm: Each MIDI track has its own "sequencer" (piano roll with other options), which pops up into a window with tabs for each sequencer. Another difference: If you want to undo a recording, there's individual undo for *each* sequencer. MIDI guitarists, or those recording a MIDI band, rejoice—you can record into multiple tracks, yet undo recording for each track individually.

The piano roll has editing views for notes, velocity, and controllers, with the usual marquee/pencil/eraser tools. Again, though, things are a little different, as these tools are designed for precision editing, not freehand drawing or erasing. Clicking on a velocity "tail" colors the note being affected, which is handy if you're working on chords or clusters of notes; you can also ctrl-click multiple lines to adjust them, with values changing linearly (not ratiometrically—it would be nice to have the option).

Of course, you can snap, quantize, and the like, as well as randomize velocity and "humanize" timing. There's also a MIDI controller Learn function, and more interestingly, you can constrain to a scale, with a choice of key and mode (major, minor, and harmonic minor). This is cool, but a bit of a missed opportunity as there are certainly plenty of other scale options. Also, note that this happens on playback; you won't hear the scale snap as you play.

While there's no support for MIDI effect plug-ins, or sophisticated filtering/editing options like Cubase's Logical Editor, the VSTi/MIDI support adds an important creative element. However, I found that Cakewalk's instruments (Dimension, Rapture, z3ta+, etc.) opened into a window that didn't show the entire GUI and couldn't be resized; on the plus side for instruments, the program bundles in a basic poly synth, bass synth, and sound font player.

MIXING AND RECORDING TWEAKS

Several changes beef up Audition's multitrack functionality: Clip handles make it easy to adjust fade curves (including grouped clips), auto-crossfades happen simply by overlapping clips, you can clone tracks, and do "ripple" edits that close the gap when you cut part of a clip. And while this feature isn't new with Audition 3, it bears repeating: With two clicks, you can open multitrack audio in the edit view and apply sophisticated digital audio editing techniques. While other programs let you open up audio in other editors, Audition builds extremely sophisticated editing into the DAW itself.

Another interesting twist is that Audition can save multitrack sessions in XML format. I have a feeling this is more of a "future-proof" feature than something that will change your life today, but with OMF fading to some degree, XML support is welcome.

EFFECTIVE EFFECTS

Audition has always had a superb roster of processors, and version 3 builds on that with a tube-modeled compressor, iZotope's Radius pitch shift algorithms, a couple new restoration processors (adaptive noise reduction and automatic phase correction), mastering section that looks like a lite version of Ozone 3, "guitar suite" (with compressor, filter, distortion, and cabinet modeling—fun stuff!), and convolution reverb.

Many people who habitually use other DAWs or editors purchased Audition for the restoration effects alone, and that won't change. The noise reduction has always been outstanding; the frequency space editing introduced in version 2 deserves special mention. I've used it to do everything from excise the breathing noises of a classical

guitarist to removing the kick drum from a loop so I could add my own kick—powerful stuff. The sample rate conversion is also high quality, and the price is less than that of many restoration suites alone.

Don't let the "plain Jane" look of the effects interfaces fool you: Audition is a processing powerhouse.

FUN WITH SYNESTHESIA

While Audition didn't introduce the concept of turning images into sound or exporting sounds as images, version 3 makes it mainstream. You can bring in a picture of, say, a flower or mountain and after Audition treats it as a spectral view, listen to what it "sounds" like. But there's much more. I exported a drum loop in spectral view, and edited it in Paint by taking the part of the spectrum that had the kick, copying it, and pasting it offbeat at a higher frequency. I then painted in some squares at what seemed to be appropriate rhythmic intervals, and whaddya know—cool percussion hit overdubs. This is a sound designer's dream.

The only damper was trying to use the frequency space editing features, like the "healing brush," marquee, effects brush, or even the lasso, to cut or copy specific sections (or change gain): If I tried editing with anything other than the usual I-beam cursor, Audition would crash. It turns out that Adobe implemented support for up to eight CPU cores for multithreading, however a bug exists that only allocates memory for four. The company plans to address this soon, but using Task Manager to set processor affinity to four cores (I tested this on an eight-core PC Audio Labs computer) provides a workaround . . . and once again, I was a happy spectral camper.

CONCLUSIONS

I use Audition a lot for mastering work, particularly for restoration. While I haven't found the multitrack implementation compelling enough to switch from other programs, version 3 gives DAW features that put its multitrack implementation in the big leagues. And when it comes to value—there's even about 5GB of content included (loops, sessions, beds)—Audition 3 is *very* tough to beat.

As is typical these days for major updates, there are a couple "version X.0" issues that need to be addressed. But overall, Audition 3 will keep the faithful happy, gain new fans (including those who install Audition as a "Swiss Army knife" complement to their existing DAW), and most importantly, continue the tradition of being deep yet easy to use. eq

PRODUCT TYPE: Digital audio/MIDI sequencer with advanced digital audio editing.
TARGET MARKET: Windows-oriented recording studios, budget mastering suites.
STRENGTHS: Adds MIDI recording and editing, with VSTi support. Outstanding mastering tools. Clean, and somewhat more customizable, interface. Benign copy protection (no dongle). Cost-effective. Bundles lots of content. Deep editing available for multitrack audio.
LIMITATIONS: MIDI not as sophisticated as many other DAWs. Some "version X.0" issues with soft synths and multicore support need attention.
LIST PRICE: $349; upgrade from any version of Audition (including versions bought as part of a suite) $99
CONTACT: www.adobe.com

DIGIDESIGN STRIKE

Striking a Blow for Expressiveness

Digidesign Strike.

by Phil O'Keefe

Strike, an RTAS virtual instrument for Pro Tools LE/HD 7.0 or higher, combines drum patterns, samples, and the ability to sequence various patterns. It also offers user-editable patterns, pre-configured into stylistically-related arrangements. Packaged on two DVDs, Strike's 6GB sample library is based around multi-miked recordings of five basic drum kits, with up to 12 individual instruments (snares, hi hats, toms, etc.) per kit. Copy protection is via iLok key.

GETTING AROUND STRIKE

The Main page is where you can modify drum sounds and the rhythmic pattern playback—swing, timbre, timing tightness, basic EQ/compression settings, and the mix of the various mic positions (close mics, overheads, room mics, and even a "talkback" mic).

Two very interesting controls, Complexity and Intensity, map to MIDI pitch-bend and modulation controllers by default; they let you adjust, in real time, the drum hits' intensity and complexity of the parts being played in the current rhythmic pattern. For example, increase complexity and the hi-hat gets more complex, the snare drum plays more grace notes, and the whole kit gets busier—but still within the same general rhythmic pattern structure. You can even specify which individual instruments will be affected by these controls, and to what degree. These controls make Strike a far more expressively playable instrument than most drum machines and samplers. (Of course, you can record this data as automation and edit it.)

When played from a keyboard, the lowest octave provides individual mute switches for each of the 12 instruments in a Strike drum kit. The highest two octaves trigger individual samples, while the middle three octaves trigger various patterns (the bottom octave white keys trigger verse patterns, while the black keys trigger intro fills). The next octave up triggers bridge patterns on the white keys and fills on the black keys, while the highest octave handles choruses and outros. (There's a virtual keyboard for auditioning patterns, but it doesn't generate MIDI note data—so you still need a hardware controller.)

A QUESTION OF STYLE

Each Style has 35 (editable) patterns containing various alternative rhythms for different song sections—intros, verses, choruses, etc.—mapped in a consistent way from Style to Style. So, once you've charted out your arrangement, you can "switch drummers" by calling up a different Style. How would that country tune sound as a bossa nova? It's easy to find out. And, you can adjust the relative response to complexity and intensity for each instrument, as well as the playing dynamics, hit variations, timing, and offset of each kit element. Want the snare to vary tonally with each hit? Just raise the hit variation control.

The Kits are editable for timbre shift, attack and decay times, and sample sizes; note that you can't load individual samples within a kit (*e.g.*, load just the kick drum samples from a different kit). Also, no expansion packs are available yet. But the quality of the included sounds is generally very good, and the amount of sonic variations exceed what you'd expect from the relatively modest number of drum kits.

The Mix screen is extremely comprehensive, and in addition to having lots of choices for level, panning, multi-mic mixes, etc., each instrument can have up to three insert effects using Strike's onboard processors. There are also multiple outputs if you want to add other effects.

STRIKE, OR FOUL BALL?

Part sample library, part sound module/drum machine, and part expressive instrument, Strike is a knob twiddler's delight—without being so complicated that it intimidates new users. Drawbacks are few; overall, Strike offers flexibility, control, cool grooves, good drum sounds, impressive realism, stability, and the superb "playability" characteristic of AIR's virtual instruments. If you need a great virtual drummer for your ProTools sessions, audition Strike. I'd say it's a hit. EQ

PRODUCT TYPE: RTAS plug-in virtual drum instrument (Mac/Windows XP).
TARGET MARKET: ProTools users needing powerful, expressive drum sounds.
STRENGTHS: Exceptional sonic control and stylistic nuance. Complexity control edits a style's "busy-ness." Intuitive user interface. Musically useful and stylistically-related drum patterns for each Style.
LIMITATIONS: Can't load your own sample libraries, or individual samples from the included sample library into a Kit. Can't change panning perspective (audience vs. player) quickly for the entire kit without adjusting numerous panning controls. Lacks many percussion sounds (*e.g.*, tambourines, shakers).
LIST PRICE: $299
CONTACT: www.digidesign.com

TIME TO DEAL WITH YOUR MONITORING SETUP

by Craig Anderton

No matter what you do in the studio, you need to hear it properly—and this month's selection of gadgets is intended to give you a better monitoring environment.

IK Multimedia ARC

Don't believe that equalization can fix room acoustics? Neither do I. Acoustical treatment is the solution. But I'm going to have to revise my thinking, because the ARC (Advanced Room Correction) system really can help compensate effectively for room problems.

The package ($699 retail) includes a calibrated measurement mic, measurement software (ASIO Windows, Core Audio Mac), and a VST/RTAS/AU correction plug-in. You sample your room at multiple places, the measurement software figures out what's going on, and generates a compensation curve to load into the correction plug-in—which inserts in your DAW's stereo master bus (no surround yet).

It sounds like a gimmick, yet after testing it extensively, I have to say it works very well. You can move around within the area you measured, and the sound doesn't change. If you're concerned about the plug-in degrading the final sound, don't be: It's for the mixing/recording/monitoring process, so you bypass it before exporting to a final audio file anyway.

I can't quite believe I'm saying an electronic room tuning system works this well, but I listen to high-quality headphones a fair amount while mastering in order to catch any little glitches, and they of course aren't influenced by room acoustics. Simply stated, ARC makes my speakers sound like my headphones. I started out skeptical—I'm not skeptical any more. www.ikmultimedia.com.

Primacoustic Recoil Stabilizers

And speaking of skeptical . . . Primacoustic sent me an email claiming their Recoil Stabilizer pads ($99 each retail) for near-field monitors made a dramatic improvement in the sound. Sure. Did it come with a green felt tip pen you could run around the rim of a CD to improve the sound, too?

Apparently resigned to people not believing them, Primacoustic sent out prototypes to a bunch of high-profile engineers and reviewers—including yours truly—and asked for comments. None of us compared notes, but we all noticed the same thing: more consistent bass, superior imaging, and a tighter sound. How can a stupid foam pad do this?

It turns out the "secret" is the thick metal plate on top of the pad. When sitting on a standard foam pad and pushing air, a speaker sways back and forth ever so slightly. The plate stabilizes the speaker, which is supposedly the reason for the improved sound and imaging. Whatever; it works.

Primacoustic set up an A/B demo at AES of the same speakers on the Recoil Stabilizers and regular foam pads, and I heard the same difference I heard in my studio—as did other showgoers. We're not talking a subtle difference, but one you can pick out with your eyes closed. Bottom line: Before you replace your monitors, give the Recoil Stabilizers a shot—they could be the most cost-effective upgrade you can make to your monitoring setup. www.primacoustic.com.

ModTrap Acoustical Absorber

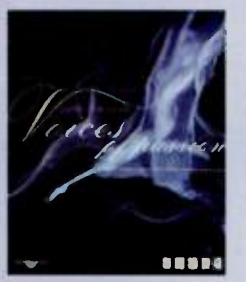
The ModTrap panel is a 2" thick broadband absorber wrapped in fire-rated acoustical fabric, and available in two sizes: 16"x24" (reviewed here; $99 retail) and 24"x30" ($129). The "special sauce" is that they attach to a bracket that can swivel about the center of the panel, and fit on a mic stand. Thus, it's easy to put the panel where you want: between a mic and wall, to isolate a guitar amp, serve as a divider between drums, reduce cymbal splash, and so on.

I tried placing it between a mic positioned about 5' away from a fairly reflective wall, and could hear there were less reflections coming back. So I tried putting two in a corner, to check if they could reduce the bass bump detected when I was testing out the ARC. Sure enough, there was a slight reduction in two "humps" at about 120Hz and 45Hz, but no effect on the mids or highs. Although not sold as conventional bass traps, it was interesting to note the ModTraps made a difference anyway.

But the main value is for isolating instruments and mics somewhat from the effects of a room, or from other instruments or sounds playing simultaneously. In that respect, the mic stand mounting thing is smart—you can tweak placement almost as easily as you'd tweak a knob, and the panels are light enough you don't need a super-heavy mic stand. Overall, the ModTrap is useful, functional, and affordable. www.modtrap.com.

SOUNDS

QUANTUM LEAP VOICES OF PASSION



Wow.

Well, that's the short-form review. To elaborate, these are superbly-recorded "ethnic" samples from Bulgaria, Syria, India, Wales (think Enya), and America (which tend toward more multisampled notes rather than idiosyncratic phrases). They play back through East West's Play engine (32/64-bit, Windows/Mac, ASIO/Core Audio, VST/AU/RTAS) which itself is pretty cool, with convolution reverb, delay, artificial double tracking (chorus), amplitude envelope, and lowpass filter.

There are also interesting tricks, like assigning mod wheel to sample start point—start anywhere within a phrase. Another is you can use the left, right, or both channels of a vocal (different channels use different mics) but if you use only one, you can synthesize a stereo spread. The Welsh vocal "phrase generator" is cool, too.

Although initially it seems there are only a handful of presets, the "master" ones use keyswitching to access multiple articulations. There are also presets with individual articulations, and some with true legato. Print out the PDF manual—lots of options are hidden in those presets.

While obviously geared for video and movies, the vocals are great for replacing that *clichéd* diva sound used in a lot of trance music, as well as for any kind of chill. Three words: evocative, fascinating, and unique. Oh, and add "wow," too. —*Craig Anderton*

CONTACT: Quantum Leap, www.soundsonline.com
FORMAT: Virtual instrument with about 7.3GB of 24-bit/44.1kHz samples; requires iLok (not included) for installation
LIST PRICE: $495

BIG FISH AUDIO ROCK STAR

Actually I'd go more for "pop star," as this is more like a melodic strain of alternative rock than, say, punk or heavy metal. I could easily picture these loops behind a Hilary Duff single, which is *not* damning with faint praise, but telling it like it is: This is commercial, top 40, mainstream music that could easily slide into the soundtrack of a zillion TV shows. Instrumentation is drums, bass, guitar, acoustic guitar, and some piano; the recording quality is good (bass and guitar more so, drums less so), but not particularly exceptional.

End of story? Not quite. Each construction kit typically has 30–50 files—much more than average, and they're relatively short. These loops are like little micro-hooks, and you could cherry-pick a few files from several construction kits to provide a foundation for your own original material. Once you overdub some leads, percussion, voice, and some pads, you'll end up with something no one would ever recognize as being inspired by a sample CD.

While not incredibly original or unusual, I use sample CDs such as this quite a bit as they present a neutral canvas on which you can imprint your own personality, and work well in many contexts. And like other Big Fish construction kit CDs, they're great when you have to put together something stylistically coherent under a tight deadline. —*Craig Anderton*

CONTACT: Big Fish Audio, www.bigfishaudio.com
FORMAT: DVD-ROM with 25 construction kits duplicated as WAV, REX, RMX, and Apple Loops files; 24-bit/44.1kHz
LIST PRICE: $99.95

SONY SEVEN MINUTES TO MIDNIGHT

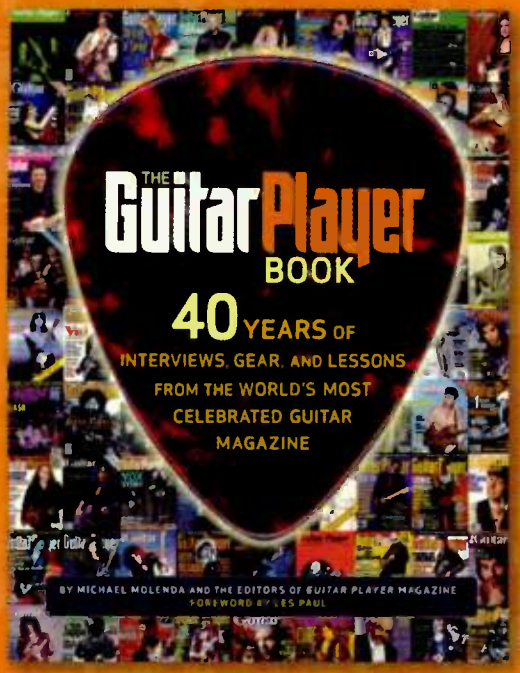
This is a construction kit with a twist: Each "kit" contains the loops used in a song from the Rondo Brothers' album, *Seven Minutes to Midnight*. (Interestingly, I proposed this concept to Sony almost a decade ago, because I thought Acid would make a good multitrack music delivery/remix system.) We're talking trip-hop, with a bit of funk thrown in too.

Given the source, as you'd expect there's a very wide variety of loops, including guitars, drum figures, bass, synths, piano, organ, various effects, and percussion. Furthermore, the one-shots contribute drum hits, stabs, scratches, and the like—even a few vocal elements.

While it might seem like these loops would pigeonhole you into the Rondo Brothers' sound (more specifically, that of their CD), actually the reverse is true. The acidization is up to Sony's usual standards, so I would see this more as a source for mining loops to add to existing compositions. Even though it's organized like a construction kit, I'd recommend treating this like a general-purpose sample library with a wide variety of samples rather than a "kit" sample CD. You might even want to group all the drums, basses, guitars, etc. into their own folders to encourage this type of approach.

In any event, for trip-hop fans there are lots of solid loops—and that's the bottom line for any sample library. —*Craig Anderton*

CONTACT: Sony, www.sonycreativesoftware.com
FORMAT: CD-ROM with 455 Acidized WAV files and 136 one-shots; 16-bit/44.1kHz
LIST PRICE: $59.95



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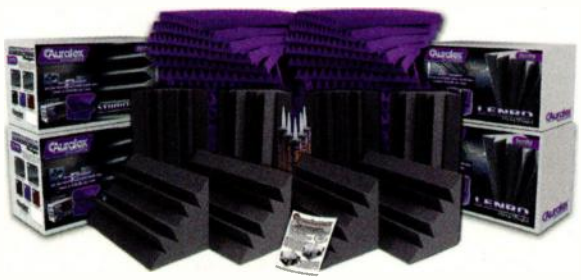

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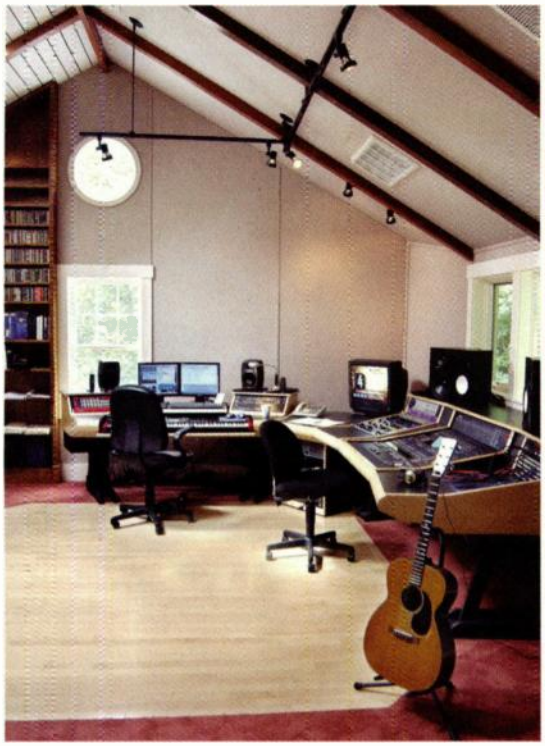

ROOM w/a VU

by Greg Reynolds

STUDIO NAME: Verité Music
LOCATION: Sherborn, MA
CONTACT: info@veritemusic.com
KEY CREW: Sheldon Mirowitz, Ardys Flavelle, Andreas Bjorck
CONSOLE: Yamaha DM2000 96-channel digital mixer
CONVERTERS: Apogee PSX-100 AD/DA Converter
SOFTWARE: Digidesign ProTools|HD1 System w/ Digidesign 192 Digital I/O; 4 PC VI Farm (Windows XP), running TASCAM GigaStudio 3; Native Instruments Kontakt 2; numerous soft synths, plug-ins, commercial and custom sound libraries
RECORDERS: Otari MX-5050 BII analog 2-track; Sony PCM-2700 DAT; TASCAM DA-78HR 24-bit 8-track
MONITORING: Ashley M500 Power Amp; Auratone 5C (2); Genelec 1031A (2), 8030A (2); Yamaha NS-10M (2)
VIDEO PLAYBACK: Apple Powermac 8600/300 w/ Aurora FUSE card; Mitsubishi VHS Recorder; Sony DVD Player
MICROPHONES: AKG C461 (2); Electro-Voice RE20; Neumann U47, U67, U87; Shure SM57, SM58, SM81 (3)
OUTBOARD: Focusrite ISA 115 Preamp/EQ; Lexicon PCM91 Digital Reverberator; SansAmp PSA-1; Tube-Tech LCA 2B stereo compressor/limiter; Urei 1176 LN compressor/limiter (2)
INSTRUMENTS: Black Mountain Dulcimer; Epiphone 1965 Texan; Fender 1965 Stratocaster, 1968 Telecaster; Gibson 1956 Les Paul Special, 1972 Les Paul Custom; Guild 1970s D35; Hohner D6 Clavinet; Jerry Jones "Danelectro" Electric Baritone; Kala ukulele; Martin 1957 OOO-18; Moog Minimoog D; Nord Modular; Oberheim Xpander; Russian zither; Stradolin mandolin
AMPLIFICATION: Fender 1962 Bassman w/ 2x12 cab, 1956 Princeton, 1965 Princeton Reverb, 1959 Tremolux 1x12, 1965 Tremolux Head; Mesa Boogie Mark III 1x12 combo, 1964 Vox AC30 2x12 w/ top boost
NOTES: Picture a panoramic landscape of farms, orchards, and tree-lined roads. On the edge of a forest, a two-story bungalow absorbs the scenery in silence. Zoom. Focus. Cue music. Inside, New York and Hollywood flash across the screen as Verité Music provides the soundtrack. Averaging 80 hours a week composing and recording music for television and film can be intensely confining and consuming; Sheldon Mirowitz calls his home-based studio "an antidote to the insanity of the work I do."

For nearly 20 years Verité Music has been keeping the pace to meet the industry's demands. When the lease ran out on his downtown Boston studio in 2004, Mirowitz not only envisioned his current facility as a sonic workhorse, but a means to some serenity: "I wanted it to be a place that invites ideas—that meant comfort, light, high ceilings, room to move. The control room needed to sound great, but never at the expense of the feeling of being in the woods, or near the sky, or the great cozy feeling of being in a home."

With the help of studio designer Rob Rosati, Verité Music's new home is situated on top of a detached two-car garage overlooking the lush surroundings of suburban Massachusetts. Both visually and sonically appeasing, Mirowitz couldn't be more pleased with the finished product: "Rob designed the room brilliantly, with a lot of wood, a lot of diffusion, broadband absorption—he even used the area over the stairs (next to the booth) as a giant bass-trap/helmholtz resonator—the room sounds great."

Add an isolation booth, additional workstation, and office downstairs, and Verité Music was off and running again at its usual pace. This time, Mirowitz surrounds himself in sound and sight with what he calls a "technically first-rate, acoustically quiet—and truly inspiring—environment totally optimized for writing scores." EQ

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

Axel, BEHRINGER Germany Systems Engineer, was the proud father of the ground-breaking XENYX mic preamp.

Thomas, BEHRINGER Germany Technical Director drove the technology of the 2442FX to the limits of physics and then half a kilometer beyond.

Thomas, BEHRINGER Germany Software Engineer, steered the USB interface and ASIO drivers for the 2442FX.

Shou Long helps assemble the XENYX 2442FX at BEHRINGER City, our highly advanced manufacturing complex. He may very well have built a 2442FX 4U!

Bing, one of our R&D Assistant Test Engineers, helped make sure that the prototype 2442FX complied with all internationally-recognized safety and RF emissions standards.

GEAR GRINDER **HARD DRIVER** **BIT MECHANIC** **PIT CREW MEMBER** **SAFETY FREAK**

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XENYX 2442FX. Not every marvel of German engineering has four wheels.

Some have twelve suhweet-sounding XENYX mic preamps, four buses and an outboard 2-in/2-out USB interface. But the Thomas's, Alex and the rest of the BEHRINGER Germany development team didn't stop there. They included four bands of mellow "British" EQ on each channel. And, so you don't have to spend more bucks on plug-ins or outboard processors, they added 100 24-bit effects including reverbs, delays, phasing and flanging.

Uli Behringer (center) isn't an easy person to please.

When everyone, including the hyper-picky Uli Behringer was satisfied, the XENYX 2442FX emigrated to BEHRINGER City where it underwent grueling stress, life cycle and safety testing...and then final meticulous production.

Learn more about the 2442FX and other XENYX mixers at your BEHRINGER dealer. And then get behind the knobs of a truly fine piece of high-performance German design.

There are 13 XENYX mixer models from which to choose. Two and four mix buses. Five to twenty-four inputs.

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Why is the 8PRE more than just an interface?

Because it's two products in one.

16x12 FireWire audio interface with 8 mic inputs

As a 16-input, 12-output FireWire audio interface, the MOTU 8PRE delivers 8 mic inputs in one rack space, complete with five-segment level meters, phantom power switch, 20 dB pad switch and trim knob for each input. Now add eight more channels of ADAT optical digital I/O, even at 88.2 or 96 kHz. Top it off with separate main outs, MIDI I/O and on-board monitor mixing, and you can turn your Mac or PC into a complete desktop studio that records your entire band.

Mic pre to optical converter with 8 mic inputs

The 8PRE is really two products in one: it's a 16 x 12 FireWire audio interface, and it's also an analog-to-optical expander that converts 8 mic inputs to 8-channel optical. If you already own an interface or mixer with ADAT optical connectors, you can connect the 8PRE optically to add 8 mic inputs to your existing rig and seamlessly integrate them into your current mixing environment, with no extra strain on the FireWire bus. Two for the price of one: now that's value.

- FireWire audio interface for Mac & PC
- 8-channel mic input-to-optical converter
- 16 inputs and 12 outputs
- One rack space
- 96 kHz recording
- On-board CueMix DSP mixing
- 8 mic/line/instrument inputs
- Individual 48V phantom & 20 dB pad
- Individual front panel trim knobs
- 8 channels of optical I/O up to 96 kHz
- Main outs w/front panel volume knob
- Five-segment metering for mic inputs
- On-board SMPTE sync
- Sample-accurate MIDI
- Expandable — connect up to four I/Os
- 2 FireWire ports for daisy-chaining
- Mix & match with other interfaces
- Includes AudioDesk™ software for Mac
- Across-the-board compatibility

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