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FEATURES

WORKFLOW WIZARDRY:
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COLUMNS

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On the cover: CTR Studios' 48-input Sound Workshop console. Special thanks to Cookie Marenco. photo by Marty Sconduto

Talk Box

WHAT WOULD MAKE COPY PROTECTION PALATABLE?

The easy answer: a protocol that makes sure people pay for the software they use, but doesn't inconvenience legitimate users.

Unfortunately, no current copy protection system has attained either goal. Just about anything can be cracked eventually, and sometimes, the people who pay for their software have to deal with more problems than the people who don't.

There are some success stories. I've been told that Apple was able to lower the price on the "Logic Pro with everything bundle" because Emagic's dongle really did cut piracy to practically nothing (a telling sign that stopping piracy might very well lower prices for everyone). And companies like Propellerheads, Cakewalk, and Sony's Madison Media Software division have been able to not just survive, but thrive, using minimal copy protection.

But then there are the horror stories, all of which have happened to me: accidentally using an older dongle driver that crashed my computer so badly it had to be repaired with the help of a tech, a pre-USB dongle that fried my motherboard ("Well, it can happen, but it's rare," said the company), going to a session thousands of miles from home and encountering an "insert CD for verification" message, having software that could never be re-installed because the company went out of business, and doing a hard drive reformat because the documentation for a certain type of copy protection was inaccurate — then taking hours to re-install legitimate, protected software after the re-format.

Stealing software is wrong, and a major problem. Intellectual property deserves the full protection of the law. But all companies could make copy protection more palatable by publishing accurate, crystal-clear documentation on their copy protection protocol, and providing "insurance" so that if a dongle breaks, the user doesn't have to break out in a cold sweat. And, there should be a repository for non-copy-protected versions of software (maintained by IMSTA, perhaps) so that if a company goes out of business and the assets of the company aren't sold after a year, registered owners can download the software for free. And a uniform method of copy protection instead of the plethora of protocols would really help, too.

Manufacturers deserve a break, no question about it . . . but so do honest customers.

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Punch In

TUNE IN, TURN ON, PUNCH OUT BY
MATT HARPER AND THE *EQ* STAFF

THREE ON A TREE

Steve Fisk

Paul Manousos sits down with legendary Seattle, WA, super-producer Steve Fisk (Nirvana, Soundgarden, Low) for a quick question blast.

EQ: How has the change in technology changed your view on how we make music or how you record it? Are you more inclined to work "in the box" these days?

Steve Fisk: I really miss having a booth, or any really acoustically isolated spot. Lots of birds, dogs, and cars get into recordings these days. Anytime I need to record more than one person at a time I'm better off in a real studio. But my own work lives happily "in the box." I use a Folcrom RMS216 and a Summit TPA200 to get in and out of Pro Tools, but that doesn't affect what I mix on Robert Lang's SSL. There used to be an API in there and we used the pres for the electric guitar work, in order to sound more like "Stairway to Heaven." Most of my projects go to tape first, as drums and guitars benefit from this treatment. I tend to use 456 tape so I can actually overdrive it; the "modern" tape stocks don't crunch that much.

EQ: When using plug-ins, do you find yourself not having to think ahead as much as you do when going the "old fashioned" route and just running everything outboard?

SF: I never think ahead and I try to use plug-ins the same way I use hardware. Digital de-essing seems to work a little better than the boxes I've used, but other than that I prefer to steer clear of plug-ins and just utilize my hardware.

EQ: In the case of *For Better or Worse* [Editor's note: author Paul Manousos' solo debut], would you have made the same record if you stayed 100% in the box instead of using Robert Lang's SSL and mixing to tape? What made you choose the hybrid approach?

SF: I wanted to use a real studio to get live takes from the three-piece rhythm section, I hate starting a record relying on click tracks. You need a good room, the proper isolation, as well as the good microphones, and so on.

Most people I work with tell me they get better vocal takes in a home studio — for a myriad reasons. They know they're saving money. There isn't an owner or intern, or often other band members hanging out while they're trying to "deliver." I wouldn't be the first person to complain about the sterile studio environment but, on the other hand, Lang's piano wouldn't fit in my living room. So as we muddle thru this hellish period of rethinking music production, it seems like everyone has a reason, or an excuse. I'm sorry to see studios close down, and I don't like being part of the problem. —*Paul Manousos*

Practical Workflow Tip #1
Signal Flow Cheat Sheets

Make a studio session signal flow cheat sheet in Excel, and fill in the blanks as you set up. Take this example: From far left to right — instrument, mic, tie line #, preamp, fx1, fx2, fx3, DAW input #, and so on. You get the idea. This makes it easy if you have a technical problem, or need to change your setup quickly mid-session. —*Kent Vederico*

Practical Workflow Tip #2
Preventative Patch Bay-ing Part 1

Make each patch from tie line to DAW input using the same colored cables. I like to signify the right sides of stereo mic configurations using red cables; same goes for insert sends — red cables are your insert returns. Regarding stereo channels, odd numbers are always lefts, even numbers are always rights. Sure, it may be borderline OCD, but it will save you a lot of hassle. —*Kent Vederico*

Practical Workflow Tip #3
Archive Organization

Keep it organized. Create dedicated folders, or even partition a drive as a holding tank for your files, organized in subfolders by session or even cue number. And have a back-up system. Whatever archive method you use — DVD, AIT, CDR — employ a system that can easily identify if a project has been archived. It could be as simple as copying the enclosing folder to a dedicated back-up hard drive, or maybe an Excel spreadsheet with dates and a job number. For Mac users, you can highlight folders in OSX. My system is this: "Red" means it needs a safety copy made; "green" means it's already backed-up; "orange" means it's been backed up already but revisions have been made to the original, thus a back-up update is necessary; "gray" means the project is completed, fully archived, and ready for deletion. —*Kent Vederico*

ZERO 7 by Lily Moayeri

Henry Binns, one half of Zero 7, packed up and moved his family to the hippie capital of Glastonbury in the Somerset countryside of England because, "London was doing my nut in." But when it came time for Binns and his musical partner, Sam Hardaker, to start creating the third Zero 7 album, *The Garden*, Binns essentially kicked his family out in order to be able to work.

"I ended up building a makeshift studio," says Binns, shuffling around in said facility, which looks more like an outhouse than anything else. "It wasn't intended to be for always. It is a bit 'shambolic' in here."

And "here" is where all Zero 7's musical decisions are made, in the studio, albeit a bit of a messy one. "Sam and I are always trying to get some dirt in there," Binns admits. "Otherwise it feels too clean and a bit uncomfortable."

If by dirt Binns means natural, organic, real, live, then *The Garden* is an absolutely filthy yet obviously comfortable place for the duo. A soothing combination of lush grooves and seductive croons, digital soul is the blueprint Zero 7 works off of, characterized by the at times clear but hollow tones of tracks like "Futures" and "Left Behind," and the sultry atmosphere that permeates songs such as "Throw It All Away" and "Your Place."

When it comes to capturing/creating Zero 7's patented mellow down-tempo/alt-electronica amalgamations, the band relies on old favorite mics such as the U47, outboard compressors such as the Urei 1176, and a handful of analog synths — as well as a refurbished Audiotronix console to mix into Pro Tools before mastering to vinyl, a process credited as contributing to the album's retro, analog character. The central piece of the album, however, is the GEM F30 organ, an ancient machine that Binns found in a junk shop and later utilized for its preset drum machine canned beats. Throwing the U47 up to the GEM F30, which has no line out, resulted in a ton of hum and buzz, something attributable to the GEM's construction (a wooden body which resonates from the two cabinets built into the back of the unit) — sounds that the group credits for adding a certain quality, a certain soul to *The Garden*.

The occasional sample signal is oftentimes manipulated by being put through the Tech21 SansAmp, the Electro Harmonix Memoryman, or Hardaker's malfunctioning, glitchy EH Small Clone Chorus. Guitars were recorded primarily through a small Pignose practice amp for its "buzzy" personality, a sound akin to the Isley Brothers' distorted guitar moments. Furthermore, an API Lunchbox loaded with 512C Pres and 550B EQs, the band confesses, is the "secret weapon" part of the chain nearly every time, except in the case of the drum tracks where only the kick and snare are treated with the box — mostly for the benefit of the 512C sound with only minimal EQing.

"More than not, with the drums, it's as simple as rolling off the top end," says Binns. "I find most things are too bright and too annoying to listen to. We all rely these days on Pro Tools, and it's really hard to get away from that 'style' of recording. The good thing is that you can edit easily, but you can edit things so much that you refine things too much, where it actually loses character. You just hit 'tab-apple-A' for half an hour, have a cup of tea, and your whole drum track is spliced into bits. We were trying not to do that as much as possible."

To further aid in moving away from too refined of a sound, Binns demanded that *The Garden* be mastering onto vinyl. "I find a lot of digitally mastered records, when you turn them up really loud they fall apart," he says. "If you ever listened to Steely Dan, that band were the first people to spend two weeks in the studio just working on the perfect hi-hat sound. They made all those records on analog equipment and they sounded so much better than the records made now relying on Pro Tools. We're trying to recreate that, to end up with something not too squeaky clean."

Punch In

Debate: To Track or not to Track with EQ

by Jeff Anderson

Though the bottom line is always turning out the best possible product, the means to which to achieve this end have long been debated, particularly in regards to treating your signal to tape. Many engineers are fierce proponents of the idea that it's only proper to send the cleanest, most malleable tracks to the mix, and thus are hesitant to record using external effects. In fact, you would be hard-pressed to find any engineer who would suggest applying time-based effects such as reverb or delay prior to the mix and, likewise, the general consensus is that compression is best left to the second stage as well. Yet in the case of EQing while tracking, the issue becomes a bit more controversial.

Some engineers feel that it's imperative to achieve the best tones as soon as possible, and therefore are much more likely to apply EQ right off the bat. When asked, they'll oftentimes explain that it's important to EQ to tape in order to gain a greater sonic perspective, to help show how each individual instrument is ultimately going to fit in the grand scheme of things, and to communicate the original intent of the engineer (and the band, for that matter) to whomever may be mixing the album.

But on the flip side there are those individuals who argue against EQing signals before the mix, claiming that the shortest possible signal path to tape is the best. Any time you apply a piece of external equipment to the path you run the risk of adding unwanted noise. And do you really want to chance never being able to pull that same perfect vocal take from the artist you are recording because you got froggy with that vintage Urei and ended up with a pop in the middle of the track?

Though EQ is a much more "reversible" treatment than others, certain pieces of equipment are definitely going to leave a thumbprint on your track, albeit to varying degrees. A good example is in EQing a vocal track using a Pultec, which is known for adding a very distinct sonic character, a frequency boost which will add a "growl" to your vocal track and will be very difficult, if not impossible, to reverse in the mixing stage.

But nowadays it's much less of an either/or situation than it was in the past. With literally hundreds of tracks at our disposal, we have the option of splitting our source signals by using a multi on a patch bay, applying the proper cabling tricks, or using a re-amping device. With two identical signals to work with, we can affect one yet leave the other dry, which is a great way to experiment with EQ for the purpose of communicating your ideas to the mixing engineer without being married to the consequences.

And even if you don't have a million great EQs at your disposal, you can always record your source signal sans EQ and then, at a later time, route the signal from the output of the recorder to your EQ and then back in on a new track; and as an added bonus, as the effected material will be tracked into your recorder, you can easily recall your settings. This very much improves your overall workflow.

It's more of a session-specific issue these days — something that's very much dependant on the mixing engineer with whom you work, and to what level you trust that individual to apply the proper EQ settings to the songs you've tracked. If you're tracking for a mixing engineer whose work you admire, and in whose skills you have confidence, it may not be a bad idea to either process to tape minimally or at least have dry sources available for them to allow maximum flex room. But if you are engineering and mixing a project, I suppose it's safe to say that you can trust both the tracking engineer and the mixing engineer (at least one would hope) and if there are time/budget constraints, or even if you are simply trying to expedite the process, perhaps you should start treating those tracks and prepare for the fastest mix possible.

The choice is yours. . . .

SESSION FILE: KAADA

John Erik Kaada looks for signs of new life in the circuits

by John Dylan Keith

Norwegian composer John Erik Kaada's new album, *Music for Moviebikers* — a gentle collection of lush acoustic arrangements set as soundtracks to imaginary films — could well double as a shared dream between Tom Waits and Ennio Morricone had they been residing in a Bergen hostel during choice career moments. It's a kind of chamber orchestra suite, quite different from Kaada's earlier work, though approached with the same sleight-of-hand production techniques that have made Kaada a favorite among DIY musician-producers in recent years.

Kaada's artistic approach is non-traditional, in that oftentimes what you hear has been achieved through rather bizarre means, and not only in the pure performance

Why be normal? Kaada embraces happy accidents.

aspect (as he tells *EQ*, "People think that on my albums it's all sampling — there are some samples, but also we tried to play like samples"). His unconventional ethos is also apparent in the mixing process, where a deep understanding of the science of sound comes into play, outlining the boundaries which Kaada calculatedly, and continuously, pushes. Never has this been more evident than on *Moviebikers* — an album where Kaada has further stepped outside of the box, both figuratively and literally speaking. "When I sum up the last 10 years, I've spent most of it in front of my monitors," Kaada tell us via phone from his home country. "I wanted to do this away from the computer as much as possible, achieve something very organic sounding."

A GRAVE SCENARIO

After the initial tracking of drums, bass, and placeholder keyboards (recorded to a click track at Kaada's own home studio, Wrongroom, for the purpose of more efficient supplementary tracking in different "facilities"), Kaada was privileged enough to stage a session at Oslo's Vigeland Mausoleum — a vaulted, windowless tomb built by Norwegian painter and stained-glass artist Emanuel Vigeland to store and display his artwork (as well as his remains). "The whole idea was to get that fantastic reverb on tape," says Kaada, who is known for his elaborate signal treatments via various "natural" locales for their characteristic resonance(s). "I recorded some claps and snaps to import into my convolution reverb, like impulse/response, for future reference," information that came in handy for overdubs and mixing, when three Eastern European street musicians caught his attention and were corralled into a taxi and taken to contribute overdubs at Wrongroom. This snapshot of the Vigeland acoustics enabled Kaada to place the trio in essentially the same acoustic space as the initial Vigeland session.

Access to the Mausoleum was fairly restricted — the Norwegian government disallowed the presence of drums, most electric/amplified instruments, and even additional lighting (to preserve the paintings stored within). Originally Kaada expected to combat these harsh impositions by recording to a Mac G5 through three rented Daniel Weiss ADC1 MK2, 24/96 AD converters in a bus outside, but soon found that a vestry at the mausoleum's entrance could serve as a control room, crammed with not only the recording gear but a separate mixing desk to accommodate monitoring of the pre-recorded tracks.

Running a 75-foot snake out to the cavernous main room of the mausoleum, Kaada found the expected low mid-tone pile-up, a problem solved by activating the low-cut switches on all the mics, plus a little more roll-off bottom and mid, and boosting a little high end on his Millenia HV3D preamps (units he tells us he employed due to their transparency) before going into the desk. However, there still existed many barriers, not so much in achieving a good sound but more so in the performance aspect — a lack of adequate lighting coupled with the fact that *Moviebikers*, composed by Kaada and relayed via notation, left many of the session musicians simply squinting in the dim light, while Kaada himself tried, in the dark, to properly manipulate his outboard gear.

AT WHAT TIME YE HEAR THE SOUND OF . . .

The aforementioned issues, however, did not stop Kaada from employing a wide array of "government sanctioned" instruments, from dulcimers to Harding-fiddles to autoharps, in the Vigeland sessions — traditional

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instruments that Kaada hardly treated with reverence. He rarely adhered to traditional strategies of producing or capturing their sounds, whether by putting rubber over, under, or between strings, using bows for plucked instruments, unorthodox mic placements, and/or a healthy dose of effects processing and re-amping after the fact. A fine example of this is the piano track on the album-opening "Smiger," a ghostly sound achieved by placing an E-bow over the piano's strings — captured with an AKG-414, placed near the pedals to give a looming quality to the piano's gentle, quiet sound.

Kaada also favors the 414 as a room mic — telling us that he put up four, arrayed around the mausoleum. ("These were recorded straight into the desk," Kaada notes, "I've often found myself dropping the preamp to let the high timbres shine through.") Though none of the ambient mics made it into Kaada's end mix, he does save them in case the album ever gets treated to a surround mix.

Other mics in Kaada's arsenal ranged from the standby SM57s and U87s to the esoteric: On most of the Eastern-European string instruments he used a rare mic called the Symfon BC-10. This is a hyper-cardioid

Kaada advises, "Get things *out* of the computer and through some fresh air — do anything to add the complexity of the sounds."

condenser built with military radar technology — one he likens to an improved AKG C535 — developed and made by the father of the director of one of Kaada's videos. "He came up to me and said, 'My Dad made this great mic.' I assumed there would be a reason it never made it into production, but it was fantastic — it supposedly picks up over 50kHz, sounds we will never hear, but it's nice to know they are there."

DI(Y) BOXES
One of Kaada's favorite tools for tonal coloring is the use, or misuse, of his many amplifiers — most of which are custom jobs he's modified personally. He uses a DI box to lower the noise that arises from impedance mismatches, though at times he's been known to push the signal backwards through the DI, or skip even applying the box in his chain.

At Kaada's blogspot (http://wrongroom.blogspot.com), he discusses how he used an old 1930s Philips speaker due to it's "really thin, far off sound," telling us "it brings the instrument over, on top of everything; an extra very special timbre, that EQing doesn't always allow." Pointing to the track "From Here on It Got Rough," Kaada attributes the use of the speaker to the "exhalation sound" of the autoharp, adding that a good portion of the dominant-melodic instruments were run through the Phillips to achieve a similar effect.

ORDERING CHAOS
Kaada's prime recommendation to those looking to augment their sounds, especially those relying on soft synths, is to "get things *out* of the computer and through some fresh air — re-amp tracks, and move the mics *away* from the speaker. Do anything to add the complexity of the sounds." An admitted spring reverb fetishist, he prefers to use the reverb of naturally resonating physical objects. For example, he sometimes amends his re-amping tactic by playing the re-amped signal into a piano with the damper pedal, or certain keys, held down, then miking the combined sound with that of the resonating piano strings. Basically, the piano becomes a bulky spring reverb.

Kaada has also added springs to some of his rack-mounted effects units, and even built one specialized four-channel reverb — a unit showcased on the guitar tracks of "The Mosquito and the Abandoned Old Woman." Kaada tells us, "I installed four different stereo pairs of springs into a metal frame, and split the signal into four, with volume pots on each input. It has an amazingly rich and flexible sound, maybe because the springs have different lengths, and the frame itself is stretchable."

Another method Kaada employs to add complexity to his sounds is that of running a signal through one of many old tape delays loaded with heavily worn tape, a technique used to impart a unique distortion. "Everything sounds better with a small touch of distortion"; distortion he oftentimes exploits by pitch-shifting extreme intervals with an Eventide H8000FW, rather than simply changing key. He speaks of regularly pitch-shifting a track one octave down, then removing the artificial top and bottom via EQ — a technique Kaada promises will result in bigger, darker sounds for percussive instruments.

IN THE NAME OF SYMMETRY
The mausoleum takes on *Moviebikers* clock out to 59 minutes, 58 seconds of audio, to which he added two seconds during mixing, a running time Kaada found rather significant: "The fact that we ended up with almost exactly an hour of music was spooky, as the last composition is called 'In Hora Mortis,' which means something like 'In the Last Hour.' Being in a tomb and everything, it kind of freaked us out." While this smacks of Led Zeppelin studio lore (*i.e.* the appearance of the mysterious black dog), it loosely speaks to what Kaada sees as the lessons of *Moviebikers*, and his general working philosophy. Kaada encourages others not only to run with happy accident, but also to experiment freely with the tools they have. "It's really time-consuming, but it's worth it. Because people react to it, and it's more fun." It's an easy concept to lose sight of while the studio clock is ticking, perhaps, but well worth the effort to remember. EQ

John Dylan Keith is a freelance writer and musician vibing mostly within Brooklyn, NY. He's a regular EQ contributor and has appeared everywhere from the New York Press to TapeOp, and many publications in-between.

SUCCESS STORY: RICHARD DEVINE

Variations in Serialist Sound

by Matt Harper

You would be hard pressed to find anybody busier than Richard Devine. Between time spent on the requisite composing and engineering/recording of his own projects, he's been moonlighting too — recording and scoring music in various commercial fields (from video games to movies), as well as undertaking sound design and various software projects for Native Instruments, Universal Audio, Korg, and many others. Having collaborated with a vast panoply of electronic-based musicians/producers (BT, Aphex Twin, Venetian Snares, Keith Whitman, and so on), Devine has also guest instructed at Berklee and lent his skills as a tutor to end users of many of the aforementioned manufacturers — all while being barely old enough to rent a car.

A classically trained musician since his early childhood, Devine drew inspiration from composers such as Chopin and Kabalevsky, marrying the complexity of their

The science is tight: Richard Devine hard at work engineering laptop generated IDM.

A quick peak into Devine's home studio, barnyard animals not included.

arrangements with the DIY ethos of acts like My Bloody Valentine and the harsh, electronically generated textures of Coil, SPK, and Brian Lustmord. Before Devine had even received his high school diploma, he was collaborating with the late Oberheim sequential tech Tim Adams. "He taught me about circuit bending: how to modify, to understand the circuit flow," Devine recalls. "I had done a lot of research into Don Buchla's designs, and the records made from his pieces, particularly those by Morton Subotnik. The sounds were so alien, I was immediately like 'Whoa, I'm going down this path.'"

SCULPTING AUDIO

Fast forward not many years and Devine seems, almost miraculously, to have a slew of solo releases under his belt on labels as diverse as Warp, Schematic, and Asphodel; as well as an impressive remix résumé including Aphex Twin's "Come To Daddy" and Matthew Herbert's "Leave Me Now." Having gained worldwide notoriety for his composition, production, and remixing capabilities, Devine partially credits the application of architectural theory to the achievement of his highly regarded sonic creations — listing famed deconstructivist architect Frank Gehry as a major influence in his approach to production. "I was telling the students at Berklee how I work in a very visual manner when I work musically," states Devine, who holds a degree in Graphic Design. "I see things more in shapes and textures, and apply like the principles of design found in the world of visual arts into sound. The world of art and audio are very closely related — color and texture, tone and rhythm — all in various forms of repetition. The elements of design can be applied to how you work with frequency, modulation, and amplitude. I always think, 'what if I make a sound shape, control the contour and the envelopes and the filtering to sound like what an artist created?' I call it 'drawing audio sculptures.'"

SURROUNDING SOUNDS

3-D audio, if you will, is perhaps the most succinct description for Devine's sonic creations; the parallels between the multi-dimensionality of the post-modernists art that he takes cue from and his own deeply layered, dark and (at times) disturbing work are legion. Taking into consideration Devine's intent to approach the listener from multiple angles, it's no surprise that he's delved quite deeply into the realm of mixing in 5.1 surround — a tactic he's employed lately when collaborating with BT on the soundtrack for director Adam Rifkin's new film *Surveillance*.

"I turned BT on to mixing in Nuendo, as he was previously doing everything in Logic," Devine informs. "I converted him right in the middle of the project. I actually program everything, record all the sessions in Logic 7.2, but then I'll export everything out and import back into Nuendo, mixing everything in 5.1. We were doing the sessions in Logic, programming all the cues, and then

dumping everything into Nuendo at the end of the day. Nuendo is absolutely amazing for it's panning capabilities; the way you can automate and get a clear visual of each individual channel to help — it's a logical layout for mixing."

But Devine's insistence on manipulating paths of trajectory in the mix is but a mere nod towards his non-traditional approach to gathering sound sources, many of which are "found" as opposed to played, for compositional purpose, a tactic oftentimes used by one of Devine's biggest influences: Aleatoric composer John Cage. Approaching the *Surveillance* sessions, Devine confesses, "we thought 'why don't we produce this score in the most non-conventional sense?' Undertaking a field-recording mission of epic proportions, M-Audio MicroTrack in hand, Devine and BT found themselves culling tracks from a local Best Buy — tracking refrigerators, fax machines, and even police scanners. "Those are what we ended up using to generate our rhythm tracks," he adds, "and then we recorded electric dulcimer, didgeridoos, circuit-bent Speak & Spells; we used every kind of strange tool and sounds found in the real world, manipulated, time corrected, then processed — programmed into complicated polyrhythms and 'glitched out' so as to match the film's disturbing vibe. We really tried to challenge ourselves, produce something that people would be like 'wow, this is really strange.'"

GOING MODULAR, THINKING ANGULAR AND GETTING BENT

Though his artistic mindset is certainly "out of the box," Devine spends a good portion of time recording "in the box," first creating sounds within certain applications such as Cycling '74s Max/MSP, GRM Tools, and the multitude of Native Instruments software packages that he has held hand in forging — oftentimes using Vokator for spectral vocoding purposes, Absynth for wave fractalization, Reaktor for interpolation delays, and so on in regards to sound design/treatment before dumping the sounds into either Logic 7.2 or Pro Tools 7, only to mix in Nuendo.

However, a sizeable amount of his sound can still be attributed to his Tim Adams custom-built analog synthesizers, including a home built frequency shifter and a full three-filter/three-oscillator modular system — tools that are indicative of the passion that has followed him from his formative years. But though he reveres those tools, he's also quite the mad scientist himself, amassing a huge collection of modified instruments, from synths to circuit bent toys. "I used to walk around with an analog synth handbook just buying everything. 'Alright, this keyboard made the sounds from *Close Encounters of the*

Third Kind — I've got to get an ARP 2600!' A friend of mine, Richard Goodsell, told me to hook up with Tim Adams, one of the best techs in the Atlanta area. He taught me all about circuit flow, what certain chips, op amps, and diodes do; how to surgically analyze your gear, from the transformer to the power supply and modify it. Anything to make a bizarre noise. . . ."

And bizarre noises are sure to continually abound, whether generated within the box or, perhaps, especially out of the box now that Devine has acquired what he calls a "Cage-prepared piano sculpture" — a baby grand piano which he plans to "hang from the ceiling by chains," put up a stereo mic pair, and record the sounds of "throwing stuff at it," a true example of tracking chance music and a far cry from his rather stringent classical upbringing.

TAKING IT UP A NOTCH

The future appears to only hold bigger and greater things for Devine, with his remix and general production duties expanding to include everybody from Sound Tribe Section 9 to Peeping Tom, the latter of which Devine chalks up as

Schooling the masses. Devine guest lecturing at Berklee on sound generation and manipulation.

being one of his more interesting projects. "They got ahold of me and said, 'we have to do some crazy, backwards circus/DSP/heavy metal/noise/everything stuff,' flipping through 20 different genres in a matter of three minutes. We did some intense programming and DSP processing, manipulating voices and then chopping them up. Anything is fair game. Tire breaking off a car? We'll use that as the hi-hat. Stepping on a monkey's tail? I even recorded the sounds of a pig breathing as a transition for one of my songs, to break from one section to another. That's what I'm looking at working on, collaborating with artists and producing music that's really on the next level." EQ

TECH BENCH
Make A Function Generator . . . in Software

by Craig Anderton

Anyone here remember the Intersil 8038 integrated circuit? Okay, I see one hand raised in the back. . . .

Well, one of the important elements of any well-equipped test bench was an audio function generator capable of generating tones and sweeps, and the 8038 — a breakthough multi-waveform audio generator on a chip — made it far easier for hobbyists to build a decent function generator, for a fraction of the cost of commercial models.

But nowadays, when you need test tones, just reach for your software: Wavelab, Sound Forge, Audition, and many others can create test tones and even do sweeps. These are very helpful not just to play back through your system, but to save as a file and load into samplers and other devices to evaluate how they perform. And of course, you can burn these to an audio CD if you want to have a test CD playable through any CD playback system.

A SWEEPING STATEMENT

I find a 20Hz–20kHz sweep very helpful, so here's how to create one using Adobe Audition.

1. Select Edit view (not Multitrack view).
2. Go "Generate > Tones (Process)" and choose the sample rate, channels, and bit resolution. Click on OK.
3. Under the "Initial Settings" tab, select a "Base Frequency" of 20Hz, "dB Volume" of 0 (assuming you want a "full code" signal), Sine for "Flavor," and under "Duration," enter the desired sweep length (15 seconds is a good choice). Under "Frequency Components," the "1" slider should be at max, and under "Flavor Characteristics," select "1." All other fields should be set to 0 and all checkboxes should be unchecked (Figure 1).
4. Click on the Final Settings tab, then click on "Copy From Initial Settings."

5. For "Base Frequency," enter 20,000Hz. You don't need to change any other settings. The Final Settings screen should look like Figure 2.
6. Click on "Preview," and you'll hear the sweep. Cool!
7. After you're satisfied that it works, click on "Stop," click on "OK" to actually generate the file, then save the file.

STANDARD TONES

Of course, you can also generate fixed frequency tones using a similar procedure; here's how to do it in Steinberg's Wavelab (Figure 3). Go "Tools > Audio Signal Generator" and on the "Source" page, choose "Sine," phase angle of 0, and the format (stereo or mono, bit resolution, etc.). Under the "Frequency" tab, set all Time values to 0, and the desired frequency with the "Median Freq. 2" parameter (set here to 440Hz). The other frequency parameters don't matter.

On the "Level" tab, set the tone's duration with the "Sustain Time" parameter. All other values can be left at the defaults.

Click on "Generate," and a window will open with the waveform. Save it, and you're done.

MORE FREQUENCIES!

In the days of analog tape, common alignment tones were 100Hz, 1kHZ, and 10kHz. But of course, using software lets you generate tones of any frequency. For your reference, standard 2/3 octave frequencies are 25Hz, 40Hz, 63Hz, 100Hz, 160Hz, 250Hz, 400Hz, 630Hz, 1kHz, 1.6kHz, 2.5kHz, 4kHz, 6.3kHz, 10kHz, and 16kHz. Standard 1/3 octave frequencies are 25Hz, 31.5Hz, 40Hz, 50Hz, 63Hz, 80Hz, 100Hz, 125Hz, 160Hz, 200Hz, 250Hz, 315Hz, 400Hz, 500Hz, 630Hz, 800Hz, 1kHz, 1.25kHz, 1.6kHz, 2kHz, 2.5kHz, 3.15kHz, 4kHz, 5kHz, 6.3kHz, 8kHz, 10kHz, 12.5kHz, 16kHz, and 20kHz. EQ
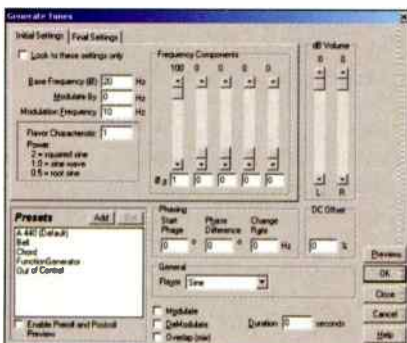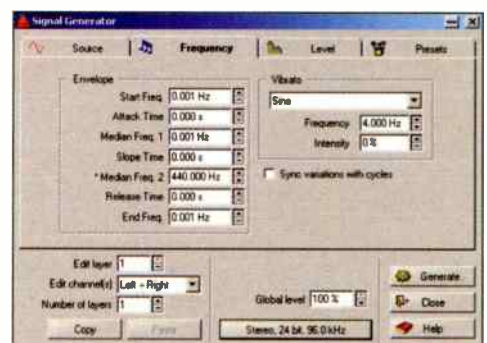
Fig. 1.

Fig. 2.

Fig. 3.

World Radio History

Tool Box

Blue Cat Audio Flanger and Stereo Flanger v2.0
New versions of Blue Cat Audio's freeware Flanger and Stereo Flanger plug-ins (for Windows DirectX and VST) sport advanced MIDI control with MIDI learn, full automation support, and skinnable user interface.
www.bluecataudio.com

Voice Vol. 2 Liquid Instrument
Consisting of 300 choir-based vocal phrases (about 2,400 loops), The Voice Vol. 2 Liquid Instrument ($199.95) is a cross-platform plug-in for VST, Audio Units, and RTAS. Voices are individually editable within the choir.
www.ueberschall.com

Visualizer 1.0 Beta 3
Introducing more functionality and several fixes for this audio analysis tool, NuGen Audio has just announced the release of Visualizer 1.0 beta 3 — available as a free trial download from their website.
www.nugenaudio.com

VSTi Host Now Available in Universal Binary Version
Defective Records' VSTi Host Version 3 ($35, update from Version 2 $10) allows a VST instrument and up to four effect plug-ins to be used outside of larger host applications. It is available for Intel Mac architectures as well as PowerPC, and also supports ReWire and DirectConnect.
www.defectiverecords.com

Feldspar Special Edition Multi-FX VST Plug-In
This Special Edition of the Feldspar Multi FX VST is a slightly altered version of the original with a few notable changes: It has one envelope and settings instead of two, and four modulation routings in the matrix instead of six. A reverb effect has been added, the CPU usage has gone down by about 1/3, and the number of patches has doubled.
www.perimetersound.com

PM8 Passive Summing Mixer
The PM8 from SM Pro Audio is an 8x4 passive mixer/summing unit designed for DAW users to avoid unnecessary A/D and D/A conversions. Merging up to eight analog channels into clean stereo pairs, with all the sonic depth and dynamics intact, the PM8's level control, panning, and muting controls give the unit a distinct analog sonic quality. www.smproaudio.com

Mbox2 Pro
The Mbox2 Pro audio/MIDI interface ($799) provides high-speed FireWire-powered connectivity, up to 24-bit/96kHz audio resolution, and a wide range of analog and digital I/O — as well as MIDI Time Stamping support, Word Clock I/O, and dedicated studio monitor outputs.
www.digidesign.com

RND Universal Binary Versions of DAW Plug-Ins
Roger Nichols Digital's DAW plug-ins (Dynam-izer, Uniquel-izer, Frequal-izer, Finis, and Inspector XL; $249–$299) are now available in Universal Binary versions for Intel-based Apple Macintosh Core Duo systems, in VST, RTAS, and AU formats. All are available for free 14-day evaluation from the company's website.
www.rogernicholsdigital.com

RED Type A Tube Mic
Utilizing a single hand-selected ECC88 vacuum tube, the Type A ($1,345) mic is based on a system of interchangeable capsules ($249–$849 each). The package includes a custom-built flight case, high-definition cable, shock mount, and the PowerStream power supply, which offers a soft-start feature that maximizes tube life and minimizes startup time.
www.redmic.com

RiffWorks Line 6 Edition Version 2.0
Available as a stand-alone software guitar recording application for Line 6 USB and GuitarPort Rifftracker bundles, RiffWorks Ver 2.0 is compatible with OS X and XP alike. It includes a 60-day free trial for RiffLink online collaboration, and RiffCaster instant podcasting technology. Pricing: $99 as an add-on for Line 6 GuitarPort, TonePort, and PODxt; $265.99 in boxed version with the Line 6 GuitarPort.
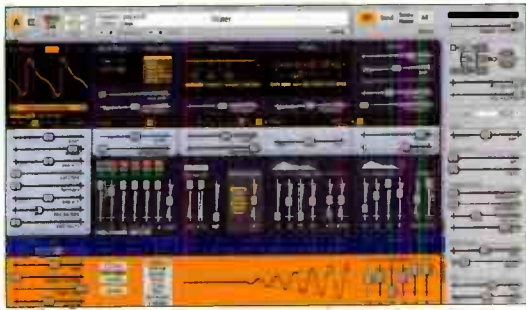www.line6.com

Compadre Beatpuncher VST 1.1
OtiumFX has announced an update for the Compadre Beatpuncher VST plug-in compressor, which fixes initial glitches in regards to registration issues and panning behavior. The upgrade is free for all 1.0 registered users.
www.otiumfx.com

Surge VST Plug-In Ver 1.2

Surge (approx. $169) combines analog-style and wavetable oscillators, configurable dual filters with a feedback loop, and massive modulation capabilities. It's compatible with Windows XP, XP x64, Server 2003, and OS X.
www.vemberaudio.com

R-122V Vacuum Tube Ribbon Mic

Royer Labs is now shipping its newest addition to their R-series line, the R-122V ($2,995). This ribbon-based tube mic comes complete with a camera-style case with a dedicated power supply and shock mount; the included cable set is equipped with military-grade locking type XLR connectors.
www.royerlabs.com

Audiowarrior Master Songwriter Ver. 1.0

Just released for Reason, the 4GB Master Songwriter ($199.99) sample workstation Reason Refill comes complete with acoustic/electric drum kits, guitars, basses, analog synths, pianos, organs, sound effects, and more.
www.audiowarrior.com

Audio Kontrol 1 USB 2.0 Audio Interface

Audio Kontrol 1 is equipped with studio-grade 24-bit/192kHz Cirrus Logic converters and offers a 2-in/4-out configuration complete with mic pre, "Hi-Z" instrument input, and headphone amp. Four freely assignable control elements on the top side of the unit make it possible to adjust any software parameter directly from the interface, allowing it to double as a basic controller device.
www.native-instruments.com

Aviom AN-16/o Output Module

Now complete with DB25 connectors, Aviom's new AN-16/o ($944.95) converts 16 channels of uncompressed, 24-bit digital data transmitted via A-Net into 16 discrete mic or line-level analog audio output channels.
www.aviom.com

Xboard 61 USB/MIDI Controller

E-mu has shipped its new Xboard 61 USB/MIDI Controller for PC and Mac, featuring 61 premium full-size keys with aftertouch, 16 programmable real-time control knobs, 16 new patch select/program change buttons, Xboard Control editing software, and a full version of E-mu's Proteus X Version 1.5 Desktop Sound Module with over 3GB of sounds.
www.emu.com

Nine Volt Audio Stylin Basslines

Stylin Basslines ($69.99) is the first bass collection designed for Stylus RMX users. With over 550 loops, this collection covers a wide range of styles and grooves using vintage, modern, fretless, and electric upright basses, along with vintage analog, modern analog, and computer-based synths.
www.ninevoltaudio.com

Fireface 400 36-Channel FireWire Interface

The Fireface 400 ($1,149), a portable, DSP-based 36-channel FireWire interface, boasts active jitter suppression, additional stand-alone functionality, direct device operation, professional quality I/Os, and a 648-channel matrix router — all of which is operational up to 192kHz.
www.rme-audio.com

Furman MAX In-Wall PowerKit-TL

Used in conjunction with a Furman power conditioner, this in-wall electronic extension system discreetly supplies safe, pure power from the main equipment rack to any remote display device, amplifier, or powered subwoofer, while at the same time improving sound quality.
www.furmansound.com

Sterling Audio Studio Condenser Mic Line

Sterling Audio has introduced a new line of condenser mics — the ST66 ($699.99), ST55 ($499.99), ST44 ($699.99), ST33 ($499.99), and ST31 ($149.99). With the exception of the ST31, all models feature an attenuation switch, a 75Hz-12dB/octave low frequency roll-off switch, cardioid polar pattern, and transformer-isolated output
www.sterlingaudio.com

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

Mastering Workflow

Make your studio environment work with you, not against you.

by Craig Anderton

J ust so we're clear: By "work-flow," we mean the flow of work. As in, whether a project flows effortlessly from start to finish, or gets snagged on riptides and eddy currents.

Workflow isn't just about a more efficient experience. It's also about a more *artistic* experience, because ideas flow when projects flow. And it's also about a more *profitable* experience, because projects that flow well get done faster.

One of the main issues with defining good workflow is the left brain/right brain dichotomy, which was covered at length in the December 2004 issue.

Rather than go over that ground again, here's the somewhat oversimplified short form: You don't want to have to think analytically when you're trying to be creative, because your brain processes those activities differently. A smoother workflow means you can stay in a creative space.

Mastering Workflow

DISRUPTIONS TO WORKFLOW

First, we need to identify what disrupts workflow, because only then can we figure out how to avoid those disruptions. Here are the prime offenders.

■ Not being able to locate what you need, when you need it. Whether it's a piece of gear, a particular sound, files, session notes, whatever — if you have to spend time looking for something, it breaks the flow.

■ Doing unnecessary work. A good example is creating the same thing over and over again when you could be using a template; or not reading a manual to discover helpful shortcuts.

■ Equipment failures. Also known as "instant workflow death."

■ Using the wrong tool for the job. Forcing something to be something it isn't means you'll waste time trying to cajole the tool into doing your bidding.

Now let's zoom out. What all of these have in common is that they create interruptions. So basically, the pursuit of workflow is about minimizing interruptions — whether in the equipment you use, the procedures you adopt, or the mechanics of running a session.

PART I
LOST AND FOUND

There are two main things you'll look for during a session:

■ Hardware (mics, accessories, spare parts, etc.)
■ Media (files, tapes, older sessions, sample CDs, manuals, and the like)

Each requires a different approach if you are to find what you want, when you want it.

HARDWARE CATEGORIZATION

The key to finding anything is a "tree" structure, not unlike how you browse for files on a computer.

For example, I have several large, three-drawer cabinets, each dedicated to a particular function. One is for "computers." The bottom drawer (because it's accessed the least) is for crucial computer items needed to recover a drive or make system changes: operating system disks, printouts of authorization codes, driver disks, motherboard manuals, and the like.

The middle drawer has parts, such as ribbon cables, hard drive jumpers, screws, spare power supply, etc. Anything

mechanical needed to fix a computer goes here.

The top drawer is for oft-accessed accessories, such as USB memory sticks, USB and FireWire cables, adapters, and so on.

Another cabinet is for "studio accessories." The top drawer is for recording and mixing gear, like mics, pop screens, a stopwatch, and stand adapters. The middle drawer is instrument-specific: strings, picks, tuners, cleaning cloths, string winder, straps, etc. The bottom drawer is for miscellanea such as analog recorder essentials (tape de-magnetizer, head cleaner), batteries, tape splicer, recorder lubricant, and audio cable adapters.

There are also smaller "project" drawers that provide temporary storage for particular projects. For example, with an audio-for-video project, a drawer will hold the videotapes, audio media, and take sheets.

Audio/MIDI cables go in large, pull-out drawers. Drawer categories are cables with similar ends (phone-to-phone, XLR-to-XLR, MIDI-to-MIDI, and phono-to-phono), with another drawer for cables with dissimilar ends. These don't get used a lot, but if I need a phone-to-XLR or whatever, this is the place to go.

That's enough examples. The point is when you want to find something, you should know where it lives. And here's one more tip: If you can't find something and start looking for it, remember the first place you looked. That's where your instincts told you it was, so when you find whatever you're looking for, put it in that place.

MEDIA CATEGORIZATION

The problem of categorizing media increases exponentially as you acquire more media to categorize. Here are some suggestions for media organization methods.

■ Sample CDs/DVDs: The "Oskar" shelving system from Mid-Atlantic industries (www.atlantic-inc.com) holds 1,080 CDs (you can mix and match CDs/DVDs), and costs about $130 — well worth the bucks (Figure 1).
■ Archival CD-ROMs and DVD-ROMs: Use quality CD binders, available at office supply stores. However, label each disc (e.g., 950418 for something created on April 18, 1995) and maintain a spreadsheet with keywords for what's on each disc. This simplifies finding the "needle in the haystack" at a later date.
■ Distribution discs for programs: These also work well with binders (Figure 2).

The disks don't require frequent access, so this stores them safely in a minimal amount of space.

■ Mass storage backup: If I need to back up or archive dozens of gigabytes, one option is double-layer DVD-ROMs. But for interim storage, I use USB or FireWire external drives (see last month's Tech Bench on how to build your own). Removable drive bays also provide for quickly interchangeable storage, but the advantage of external drives is that you can easily spin them up periodically to make sure the bearings stay in shape.
■ Funny format storage: Ah, yes, old ADAT tapes, DATs, Hi8 videos . . . while they wait to get transferred to more modern formats, they sit in individual boxes on shelves.
■ Manuals: In today's world, it's easy to forget how to do some crucial function. Dedicate some bookshelves to manuals, arranged alphabetically by manufacturer. But also maintain a CD with PDF versions (sometimes it's easier to do a "find" with an electronic manual than to search through a paper manual), and scans of those little, weird, quarter-size sheets of paper that some companies call a "manual" (they have an amazing talent for getting lost).

Remember, the point of all this is to make it easy to find whatever you want. If you can't lay your hands on the media you want within minutes, expect a workflow interruption.

PART II
AVOIDING UNNECESSARY WORK

Now let's shift our focus over to the gear we use. I hate having to do the same thing twice if I don't have to, so. . . .

TEMPLATES ARE YOUR FRIEND

Almost all modern programs let you determine your program's initial workspace. But there are ways to customize this to your own needs. For example, my "songwriting" template (Figure 3) starts off with a workstation soft synth loaded into the DAW, with eight minimized MIDI tracks and three already pre-assigned to sounds. Typical workstation choices are IK Multimedia's SampleTank, NI's Bandstand, the Emulator X General MIDI set (and GM banks designed for other samplers), East West's Colossus, Steinberg's Hypersonic, and for Pro Tools, the free Xpand! Plug-in (see this month's

Fig. 1: Mid-Atlantic's Oskar cabinet, available at places like Circuit City, can fit a lot of sample CDs in a small space.

Fig. 2: CD binders provide space-efficient storage, but don't cheap out — get a durable one with a strong cover.

Power App Alley). The template also has eight audio tracks for loading in loops and recording scratch parts.

The aux buses have a "minimal CPU" reverb (it can always be replaced later), tempo-synced delay, and distortion.

For my "live-in-the-studio" recording setup I have an Ableton Live template with 16 tracks pre-assigned for MIDI fader control from my Peavey PC1600x controller, and solo buttons assigned to the controller's 16 buttons. As soon as the loops

> There's a downside to reliance on templates and presets: You can get stuck in a rut.

start rolling, I'm ready to mix/automate.

Your audio interface may have an applet that sets up your system for a specific application. It's worth taking the time to create an optimized one (Figure 4).

Furthermore, some host programs let you create "track templates" with particular setups of processors, sends, and so on. For example, if you're mixing a singer who always sounds better with certain EQ settings, you can set up a track template with the EQ already in place, using the settings you want.

DIGGING DEEPER

Most hosts also have provisions for other customizations that can help your personal workflow. For example, Sonar has a dialog box that shows up when you do a drag-and-drop that presents a zillion drag and drop options. But 95% of the time, I use the same option — so I unchecked the "Ask this every time" box, and didn't have to deal with it (it's accessible via a menu, anyway).

Layouts of windows can be saved, so that hitting a key on your QWERTY keyboard calls up an optimized environment for a particular task. Just don't go too nuts; use three or four so they're easy to remember. (If you have to spend time trying to remember a keyboard shortcut, you've just interrupted your workflow.) It's a good idea to print a list of custom keyboard shortcuts so you can refer to it, and memorize the shortcuts over time.

Color can also save time, as parsing an image can take less time than parsing words. Loading a picture of a track icon into a track may take a few seconds, but you'll save time overall by being able to

differentiate that track without having to read a text label.

Signal processor presets are another time-saver. If you come up with a good preset, save it — you'll likely want to use it again someday.

However, there is a downside to reliance on templates and presets: You can get stuck in a rut. If you always load something with the same drum sounds, the same bass, the same piano, and so on, you're doing yourself a disservice. When you create a working environment, create one that encourages spontaneity.

OTHER WORKFLOW ENHANCERS

Native Instruments' Kore (reviewed in the August 2006 issue) underlines several aspects of workflow enhancement. However, note that it takes time to learn, so its benefits are not immediate. However, once you learn how to zip through parameters using the controller, and think in terms of calling up "sounds" rather than specific presets from specific instruments, you'll find you really can improve efficiency. Just be aware that the greatest benefits occur with complex setups involving lots of instruments and presets; if all you have are a couple favorite soft synths, Kore would be overkill.

The Frontier Design TranzPort is another great piece of workflow gear. This wireless controller lets you arm tracks, record,

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Fig. 3: A songwriting template, set up in Cakewalk Sonar. Note the inclusion of East West's Colossus GM workstation synth so that a lot of sounds are available instantly.

playback, and the like from wherever you are in the studio. Playing keyboards? Place the TranzPort on the top panel, and you don't have to go back and forth between your computer and your keyboard as you do different takes. And it's revolutionary to be able to work in a vocal booth while controlling your rig from inside it.

Re-patching is a workflow stopper, but there are a few ways to simplify patching. One is to use an audio interface with multiple inputs, such as those from MOTU and many others, even if you're just working solo in the studio by yourself. Although you might think that as a solo performer you need to do only a couple tracks at a time, the advantage of a multi-input interface is that you can have everything patched in and read to go — for example, if there are four mic inputs, you can have a couple condensers, a dynamic, and a ribbon all ready to go. (Just remember to turn down the gain on whatever you're not using.)

For more flexibility, you can use an interface with multiple ADAT light pipe interfaces (Figure 5). Then you can choose outboard gear optimized for your situation, such as converters that change analog signals into eight channels of light pipe outs, or an octal mic pre-to-ADAT converter.

In a similar vein, check out the latest generation of FireWire mixers, such as the Mackie Onyx series, or similar offerings from Alesis and Phonic. These have two talents: They're mixers, but they also interface with your computer. You can

patch all your signal sources right into the mixer, whereupon they show up in your host program.

DOCUMENTATION

It's a pain in the butt to document sessions, but if you ever need to come back to something, you'll save a boatload of time. Even just making sure that your host's tracks say "background vocals" instead of "audio-1" really helps.

As you document, the most important goal is to note things that might not be obvious later. For example, if there's a track that you left in for some reason but it's not supposed to be in the mix, note that.

PART III
EQUIPMENT FAILURES

The issue really isn't about equipment failing as much as it is about getting back to work as fast as possible.

Attempting to determine the root cause of a problem is a major interruption. A lesser interruption is replacement or substitution. Having redundant backup systems is great (although the downside is maintaining two or more systems). For example, consider computers. Having a second one primed and ready to go can be extremely useful. With today's laptops being so capable, pressing a good laptop into service while your desktop ails can often save a tracking session — particularly if you use external drives, and can

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Fig. 4: E-mu's PatchMix DSP section includes effects and routing that you can define in a template. Open your system, and everything is ready to go; you can also create different templates for different situations.

just unplug the data from the desktop and replug into the laptop.

But having dual computers is tremendously useful even beyond providing a redundant system. For example, I have two Windows machines and when doing video work, I use one for rendering (which pretty much ties up the computer) so I can continue to edit on the other one — a major workflow improvement. I also have a Mac, which handles my office tasks, online activities, and of course, doubles as another "music" computer. Note that protocols like Steinberg's System Link also allow getting more juice out of multiple computer systems.

It's also worth investing a few hundred dollars in spare parts. If a power supply goes, everything grinds to a halt — so a spare power supply is essential. CPU fans are also deal-breakers if they go south, so have a couple of those as well (might as well get lower noise ones too, so that when the fan does go, you'll have a quieter setup). CD and DVD-ROM drives can fail; an external one cannot only step in for a dead unit in a desktop, but can be used with other computers as well. AC "wall warts" don't fail too often, but it's easy enough to keep a spare "universal" adapter on hand just in case.

It's also important to have a good tool set handy with Philips-head and flat-head screwdrivers in multiple sizes, a soldering iron with both fine-point and heavy duty tips, hex wrenches, and Torx wrenches.

Nothing is more frustrating than needing to open up a unit, and not being able to.

A final caution involves proprietary cables. If I buy something with a proprietary cable (i.e., a cable you can't find at Radio Shack or a computer store), I get on the phone with the parts department and order a spare.

PART IV
THE RIGHT TOOL
FOR THE RIGHT JOB

Sequencer manufacturers talk a lot about "workflow." And well, they should, because they try to make their programs as seamless to use as possible. However, it's important to remember that some hosts are better for some tasks than others, and that's why I've made the effort to learn several programs. The time saved by using the right program for the right job has saved lots of time in the long run.

For example, for songwriting, just about any host works fine for me. But for doing quick film scores, I always found Acid to be the fastest way to build up a lot of tracks, using Acidized sample CDs (that is, until Cinescore appeared, which now handles a lot of my "standard" soundtrack work). For creating original samples and patterns, Reason gets the nod. For live performance, given the kind of music I do, I've yet to find anything better than Ableton Live — but as a DAW, I think there are more efficient solutions, like

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Fig. 5: Creamware's SCOPE system virtualizes a studio — mixers, effects, and soft synths — within its interface/DSP card system. This version has dual ADAT light pipe inputs, so mics, line ins, and other signal sources can patch into converters that feed the signals down the light pipe and into your computer.

Sonar, Cubase, or Samplitude. And when I receive a Pro Tools session, it's far easier just to fire up Pro Tools (and hope we have a compatible set of plug-ins) than it is to worry about file conversions.

Even for two-track editing, different tools do different jobs. Peak on the Mac is superb for sample rate conversion, and I like the way it handles playlists. If a file needs lots of noise reduction, Adobe Audition gets the call. For general-purpose editing and mastering of multiple files, I can really fly around Wavelab.

Sure, the more programs you have, the more of an investment they require in both time and money. But I've found that some programs are so good at a particular task, that using the right program saves enough time to more than justify the effort to learn it.

FINAL TIPS

Before closing, here are a few more general principles that help improve workflow.
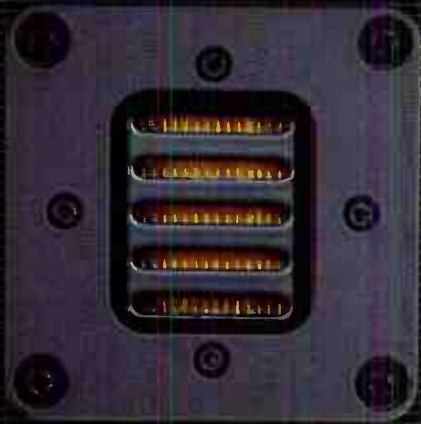
■ Multitask as much as possible, providing the tasks don't require concentration. If you have to think about tasks in parallel, that's tough. But it's easy to set up something to download a large file, back up a hard disk, or render a video while you're doing something else.

■ Pay attention to when you're most efficient. If you're at peak performance, you can accomplish twice as much as when you're out of sorts. Schedule your tasks

so you do the mindless ones — billing clients, backing up data, cleaning your studio, changing guitar strings — when you're not at your peak. Then, when you're in top form, all the mundane stuff will be taken care of and you'll have fewer interruptions to your workflow.

■ Find out your "duty cycle." If you go too long without a break, you'll become less efficient. I find that the musician's union concept of a three-hour session works pretty well for me. Three hours gives me the chance to build up some momentum and get quite a bit done before I start to fade. After a little break, I'm back up to speed for another three hours. Your mileage may vary, but you will likely find that certain schedules work better for you than others.

■ "Batch" emails and phone calls. This probably upsets some of the people trying to get in touch with me, but answering the phone or an email is an interruption, and by definition, it messes with your workflow. Get someone else to take your phone calls, or turn off the phone and check messages during breaks.

And that's all for now. Hopefully, the individual tips and comments have been helpful, but if not, just thinking about the concept of workflow can be beneficial. Many people don't "step outside" and analyze themselves, as if doing a time and motion study. But it's well worth doing if you want to be more productive, more inspired, and improve the quality of your work.

If you want to hear what Mike Shipley's been doing lately, just turn on the closest radio or change the channel to MTV. A 30-year-plus music industry veteran, Shipley has been behind the board applying his mixing wizardry to some of the most widely regarded pop, rock, and country albums of our time. From his early years working at the famed Wessex studios in London, to his career-boosting stint with Joni Mitchell in the mid-'80s, this Melbourne native has spent recent years working with the likes of Aerosmith, AC/DC, Faith Hill, and Maroon 5. As the go-to man for everyone from Michael Beinhorn to Mutt Lange, Shipley has become a truly invaluable mainstay in the recording industry.

We decided to catch up with Shipley and wax in-depth about subjects from outboard compressor applications to those pesky modern mastering engineers that like to choke the life out of your mixes, and everything in-between. So sit back, relax, throw a bag of popcorn in the microwave, put the children to bed, and join us for . . .

AN EVENING WITH MIKE SHIPLEY
He knows what he's talking about . . . so listen up!

by Jeff Anderson

EQ: I understand that you're doing a good portion of your work these days in your home studio. What's the facility like?

Mike Shipley: It's fantastic. We turned the whole house into a studio and, since it's my studio, if I want to spend a bit of extra time on a mix, I don't have to go to the studio manager and beg for hours. To me, it's much more relaxed than being in a big studio. I'm working on an ICON — bypassing the whole SSL deal. About a year and a half ago I got sick of how big the projects were getting; it was maddening jumping from one side of a 100-channel console to another. We were doing lots of A-B mixing on a SSL 9000 J and an ICON, comparing the two. If you read some of the forums on the Net there's so much discussion about the pros and cons of recording either digital or analog, and a lot of that discussion is focused on the ICON, but I've found my own way to work. I really enjoy it and it seems to be okay with my clients. For example, we recently did a mix here, the Nickelback song "Photograph," which was a big hit for them. Everyone was happy with the results. People call the ICON a big mouse, but to me it's an amazing piece of equipment. I'm 110% convinced that I can do a better mix on this system than I can going into a big studio and wasting a lot of time and money.

EQ: As your studio is based around the ICON, what's your Pro Tools system like?

MS: I'm a beta tester for a lot of stuff, so fortunately we get a lot of plug-ins early. We basically just have an HD system with six or seven XL cards, and the ICON with 32 faders.

EQ: Did it take a long time to get used to the ICON, switching banks for example?

MS: Not at all. It's so easy to do things on the ICON that you can't actually do in Pro Tools, like make custom faders so that you can rearrange the board however you want internally. What they've done for the automation is quite phenomenal; they've really made it feel like it should.
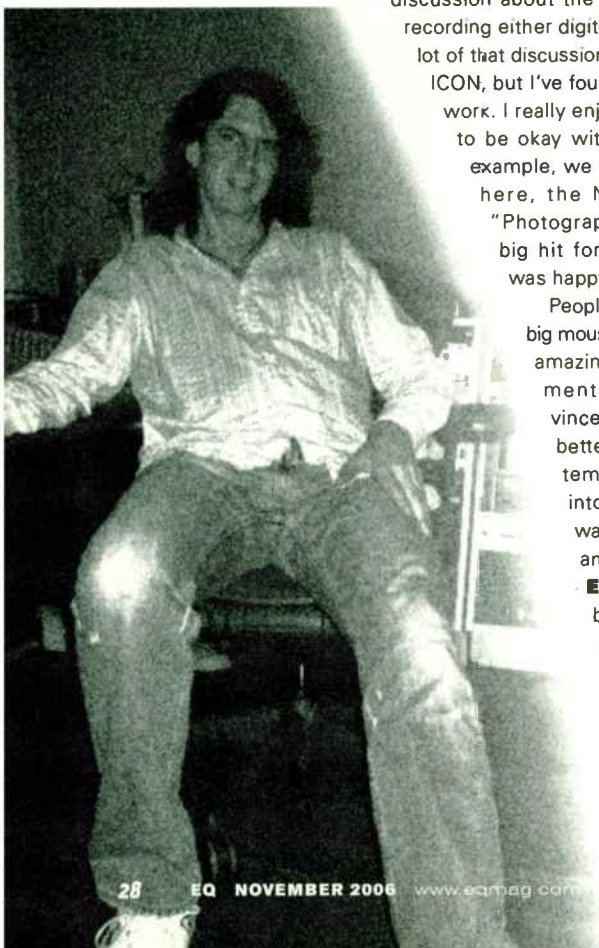
Mixing on the ICON, I use a new incredible plug-in that works brilliantly with Pro Tools called Source Connect. It's kind of like an ISDN line, but not. I can be listening at my studio in L.A. while someone is working in Nashville or the Holiday Inn in the Bahamas, anywhere with a Pro Tools rig, and everything can be heard in real time. I like to stay at home where I know I can get my sound; however with ISDN you have to work with a studio that has ISDN lines, and it gets very expensive. But with this plug I can send what I'm doing anywhere there's some version of Pro Tools, even to an A&R guy with an Mbox in his office, and he can say, "That's great, but can you make the kick sound a little different?" and we can solve the issues in real time.

EQ: Tell me about your running outboard. I hear you use a lot of vintage outboard compressors to compensate for the "cleanliness" of a rather digital-based rig. . . .

MS: The outboard gear is pretty much just compressors, as I'm not a huge fan of "in-the-box" compressors in Pro Tools. I have two Gates Sta-Levels, which I love, an RCA BA6A, a Crane Song STC-8, a couple of Empirical Labs Distressors and a FATSO, a Federal, a Euphonix, and a pair of dbx 160s. I like the fullness of the vintage compressors, especially for drum tracks, and the Sta-Levels in particular are great for treating bass, or even vocals.

EQ: When you are compressing bass, are you always relying on that Sta-Level?

MS: I always go for the Sta-Level; I love what it does for the bass, it's a magic box — it keeps the track beautifully solid with a tight bottom end. So many compressors will choke a bass out, but the Sta-Level doesn't do that. I find that if the player's pickup is sub-standard, one note will "out-boom" another, and though I'll first try to fix it with a dynamic EQ, it rarely works. So I'll just automate the track, dig in and find the frequency, and smooth the bass out. I can't

stand it when bass notes pop out real loud, and it's not always about compression, the problem often times lies in the bass itself. But as far as compression goes, I'm using the Sta-Level for the color — holding the bass "straight," and keeping it dynamically solid.

EQ: What bass tracks are generally provided to you? A DI, an amp, or a combination of both?

MS: At least a DI and an amp. There are usually a lot of phase issues with the tracks. You'll get a session where the bass sounds hollow, and you'll put both tracks up and mess with the phase to make the sound right, but generally I use both tracks in the end. If the chorus comes in and the bass needs a bit more grit, I might whack up the amp sound, but it depends on the sound they are going for, and which tracks sound best. Yesterday, the amp sounded right, the day before it was an equal combination of both, but it really comes down to "what's going to get the fattest sound?"

EQ: On to guitars. Are there any personal standards, in terms of approach, by which you abide?

MS: Nothing is standard — I'll tend to try them through any number of my outboard components before settling, as the only tactic for guitars is to carve out a space for them, EQ-wise, and make them sit well in the mix. I'll put the fader up and figure out why the guitar is masking another instrument and then isolate the problematic frequency and pull it out, because if everything gets bunched in the middle you get a very un-defined, two-dimensional mix.

EQ: Especially in your case, where you work with so many different types of music, this is probably a tough conversation in the making but . . . generally speaking, how do you approach drum tracks in the mixing stage? How many tracks do you prefer to have them on to play with? Is less more, or vice-versa?

MS: It really is different depending on the occasion, though, these days, it seems it's more than less. A project I'm working on now was engineered by Michael Beinhorn, and he's quite unconventional in the way he tracks drums. In this case, it was almost all room mics — six matched pairs of ambient mics and very little close miking, distorted to analog tape on 2" eight-track and a DSD system, to really get that classic '70s sound. It's one of those things that you just need to rip the faders up and get total sound, and that was really refreshing to me.
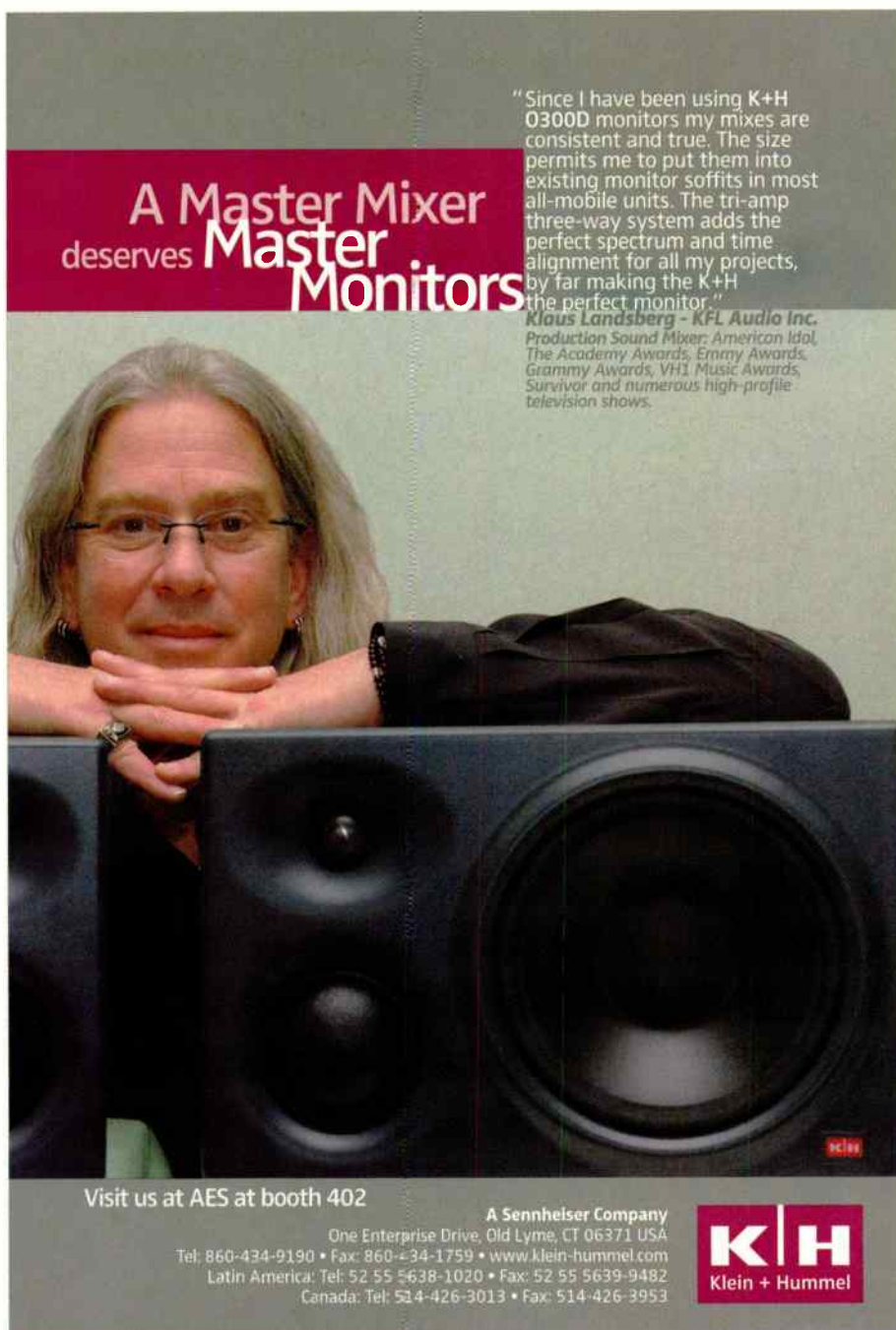
EQ: I can relate to that feeling. I've spent the last two weeks on a Latin Jazz project, and it's been a very different experience than recording most rock bands. What was, at first, a stressful nightmare got real cool the second I pushed the faders up and said, "Well, that's it!"

MS: Sometimes you push the faders up and go [gulp] "Okay, here we go, we're going to have to dig deep into this one to get the energy and depth of field that it needs." Sometimes you'll have to go for the blend — drop some samples in to complement what's been captured with the ambient mics, and just mix it until it sounds natural.

EQ: As you are working in Pro Tools, do you spend a lot of time working on phase and time alignment in the box?

MS: I always check the phase of what's recorded, and then flip the phase until it feels right. In most of the tracks that I get, I'll find something that needs to be flipped or else the ambiance is out. I'll put up the kick and snare and check all of the elements and the phase, because nine times out of ten there is a phasing issue in there somewhere.

EQ: I'm wondering about the recall in your studio — I take it most of your outboard

gear is plugged into Digi interfaces, so you can have a true recall of even your analog gear without patching?

MS: I print all of the processed tracks back into Pro Tools. We always print whatever we're using so that we can use a true recall.

EQ: I see . . . so you must be using hundreds of tracks, or have hundreds of tracks in your session anyway, even if they're on or off?

MS: Even if they're off.

EQ: How do you approach a mix? What are your first steps?

MS: I'll start by building the rhythm section, putting up the faders to see what individual tracks sound like so that I know what's ahead of me. Quite often what's on the rough mix is very different than what's on tape, and once I have a feel for what's on the tape as opposed to the rough mix, I will have made my mind up as to where I want to go with it. I'll start with the drums, but I'll only spend a certain amount of time with that before I pull up the rest of the rhythm section to ensure that the drums are being mixed in a way that's relevant to the other instruments.

EQ: When does automation come into play?

MS: Very early on, almost instantly, as there are almost instantly flaws — frequencies that need equalizing out right away so that I can build with the right feel. I'm getting sounds and automating simultaneously, from the get-go.

EQ: What format are you mixing down to?

MS: If I need to, I get a two-track [analog] machine, but for the most part I mix 96kHz into a separate Pro Tools rig using Crane Song converters, such as the HEDD.

We do summing outside of Pro Tools with a line mixer; I've got different mic pres at the end of the chain. If the signal is passive,

it needs to be bumped up to around 40dB, so I have a choice of a Crane Song Flamingo or a pair of Calrec EQs. Of all the records I worked on in England in the early days, the Calrec consoles are by far my favorite. They have a really fat sound to them, a way of changing the color again. Because depending on the source material, or what the sound is, [that] has almost as much to do with what mic pre you use, or how much you push it. . . .

EQ: Do you apply compression pre-mastering, when you're mixing down?
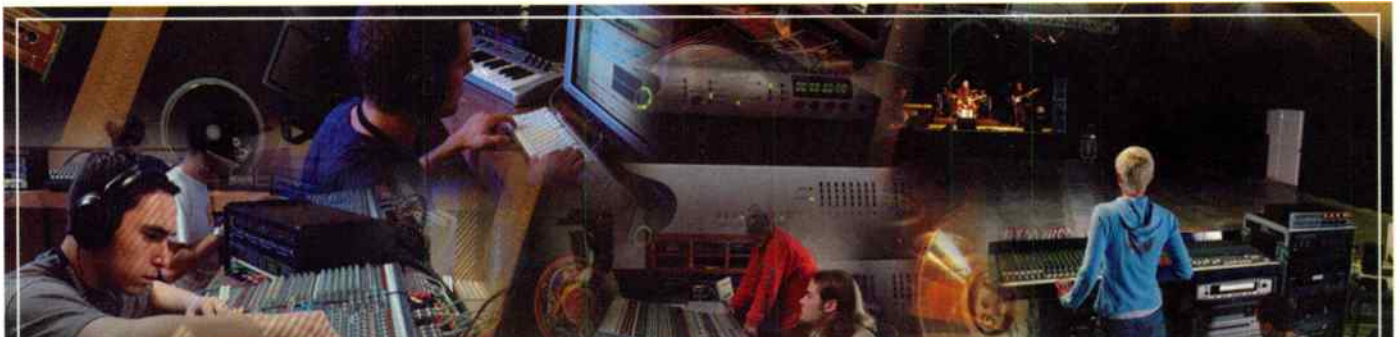
MS: I'll use the STC-8s. A lot of people smack the crap out of the stereo bus, but the way I was brought up we never compressed the two-track mix — we compressed individual channels and rode [levels]. I know with all the stuff I worked on with Mutt Lange, for example, if we reached for the compressor at that stage it would change the bottom end too much. So why bother? There are enough people down the line that are going to do that anyways!

EQ: I think the standard for "smashing" during mastering is getting a little better. . . .

MS: I hope so. It's not much fun these days to get a record back that you've mixed. . . .

EQ: A mix where you've built in all of these beautiful dynamics, and then they're gone.

MS: Okay, maybe they need to put some peak limiting on the mix — that makes sense to me in certain situations, but I can only put it down to the fact that so many clients want their music so damn loud, they are saying "more level, more level" so the engineers are just used to smashing instead of talking them down. I don't care if it's a dB overall volume less, I'll just turn the volume up in my car like everyone else does. It's not a competition. EQ

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FOCUSRITE LIQUID MIX

Trick out your tracks with dynamics and EQ control — while being kind to your CPU

by Craig Anderton

Using hardware DSP to assist your computer isn't new, going back to at least Digidesign's Sound Accelerator card. Creamware, E-mu, TC Electronic, Universal Audio, Waves, and others have all come up with hardware solutions designed to do one thing: provide massive amounts of DSP power that doesn't load down your CPU.

Unlike native processing, hardware-based DSP also offers a fixed amount of power. With a computer, the amount of CPU drain fluctuates based on several factors, so you always need to leave some breathing room. But you can pretty much "red-line" DSP without worries.

Focusrite's Liquid Mix is an external FireWire 400 device (currently Mac PPC/Intel/OSX compatible, but a Windows version is slated soon) charged with one task: deliver an array of vintage EQ and compressor effects. So is it indeed cool? Let's find out.

INSTALLATION

I hooked up a FireWire cable to the cute hardware box, and installed the software. And of course, I went to the website to check for an update and of course, there was one — so I installed that too.

And a guest is no good without a host, so Digital Performer 5 took care of that (which also has some tasty EQs of its own, which I figured would be good for comparison).

I opened an electro-type piece I'd been working on, went to an insert, selected Liquid Mix, and a warning appeared that it couldn't work because the buffer size was larger than 1,024 samples. But I checked in DP, and the buffer size was 256 samples. Huh? Then I realized I was too eager to try the thing out, and hadn't rebooted after installation. Oops. Once I did, everything worked flawlessly.

APPLYING THE LIQUID MIX

You can have 32 mono "channels" (or 16 stereo), with both EQ and compression, at 44.1/48kHz. Raising the sample rate decreases the available number of channels, but an expansion card (not included with the review unit) helps compensate for this. For example, at 88.2/96kHz you can run eight mono channels or four stereo channels and at 176.4/192kHz, two mono or one stereo channel; adding the expansion card doubles those figures.

Also note that you can't expand the number of channels by daisy-chaining units. Of course I wanted to see if I could max it out, and indeed, after 16 stereo channels at 44.1kHz, it politely informed me that was it.

Is this a problem? I don't think so. The convolution-based models of the various compressors and EQs have "character," and I think you would tend to apply them to change not only EQ and dynamics, but impart a bit of personality to a sound. If all you need is a simple bass or treble shelf, your host EQ can handle that. And if it can't, premix the track using the Liquid Mix to free up resources.

GO WITH THE WORKFLOW

One of several outstanding Liquid Mix features is that Focusrite has grafted a nifty control surface onto their FireWire box. It's compact but not cramped, and has a very high-end vibe thanks to sculpted knobs, a striking blue display, four 12-step LED meters, buttons that illuminate green when pressed, and 11 knobs.

Although you can adjust the parameters on-screen, it's well worth putting the control surface in a prominent place on your
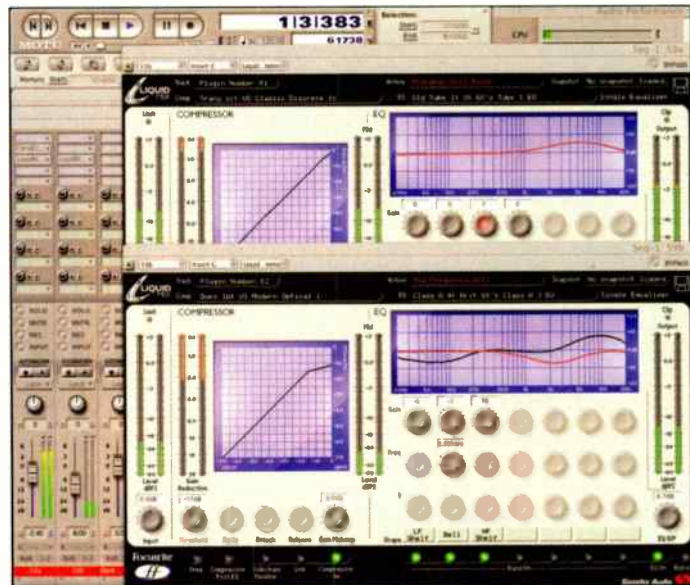
desk, and using that instead. As not all models utilize all controls, an LED on the control surface indicates which controls are present, and settings are confirmed in the LCD (which also indicates which parameters are and are not available). About the only awkward aspect is that selecting a strip on-screen does not call up the equivalent strip in the controller: You have to select the strip you want to work with manually. However, if you have lots of instances running, I actually found it easier to use the controller.

One final controller advantage is that a couple dozen plug-ins take up a fair amount of space on-screen. So you can minimize them with impunity, then run them off the controller. Bottom line: The hardware is a definite workflow enhancer.

THE MODELS

There are 20 EQ and 40 compressor models, spanning a variety of "vintage" and modern devices. (Focusrite says there will also be free, additional models available online in the future.)

The model descriptions are not too helpful — you'll see names like "Huge Tube: US Modern Tube 6 EQ" and "Brit Desk 5: Brit Modern Desk 3 EQ." (At least I figured out that the number in the second part of the EQ description was the number of active bands.) This means you have some auditioning ahead to figure out what each of the models actually does.

It's a bit easier to figure out the compressors because as soon as you call one up, you see a transfer curve in the compressor graph. For example, "US: Vintage Tube 1" has a steep knee that gets pretty flat pretty fast, and omits ratio and attack controls — it's pretty obvious this is your "fun with squashing" compressor. On the other hand, the "London: Brit Boutique Tube 1" has a gentle, sloping knee; if you bring up the threshold to a reasonable value, this does a fantastic job of bringing up a sound in the mix, but without any obvious dynamics effect.

If you instead want a track to scream, "I'm being compressed," the "Big Green: Brit Classic Optical" model has a transfer curve that flattens out, but then starts climbing again. The result is that there's a lot of squashing going on, but the peaks get a little extra lift.

Also cool: a "free" button for the compression, which with vintage models, makes the full set of controls available even

if a particular control was not available with the original emulation. You'll also find a link control in stereo mode (so it operates as a stereo compressor or dual mono). The enigmatic sidechain monitor button seems out of place; you can't actually feed a sidechain signal into the compressor, so it ends up monitoring what's at the input. Then again, there are rumors floating around that future plug-in specs will include true sidechain options and if so, this is going to look like a pretty prescient button someday. Finally, you can place the compressor pre- or post-EQ (thank you!).

I found it a little bit harder to differentiate among the EQs, because their complement of controls follow the emulations reasonably closely. This made it hard to do direct comparisons, as one might have a stepped midrange control that covered only certain frequencies not covered by other types of EQ. Still, the differences in "character" were obvious. For example, some midrange peaks had a "meatier" sound, while others were more subtle.

But there's more: You can load individual bands from different EQs into each of the available seven bands, so you could have the low shelving band from one EQ, a midrange boost from another, and so on. Amazing.

Difficult to manage, right? Not really; when you find favorite settings, you can save these as snapshots for later recall. This of course is in addition to the fact that settings are saved within your host program.

CONCLUSIONS AND OPINIONS

I was skeptical when I started this review. Sure, Focusrite makes great stuff, and Liquid Mix looked promising. But the world of compressor and EQ plug-ins is not exactly under-populated, and there are fine external DSP solutions. Besides, for $1,100 you can buy a lot of native plug-ins.

Then things started falling into place. First, it installed easily and was obvious to figure out. Second, despite piling on the Liquid Mix plug-ins in Performer, the CPU meter kept reminding me that none of this was affecting the CPU much. And while Focusrite is upfront about the FireWire aspect, adding about 2–3 ms of latency, I didn't notice any significant lag while adjusting the controls.

As I got deeper into using it, two things stood out: The sound quality is just plain wonderful, and so is the interface. It didn't take long to become familiar with

the control surface, and yes, it does speed things up.

My only real concern is that it takes a while to really learn the "sound" of the various models. Until you do, it's a temptation to try *everything* on a track to hear what works best. That can be a time sink, so my advice is simple: When you find a sound you like, stop there. Sure, maybe you could find something even better, but the object is to create recordings, not tweak forever. I've always maintained that just because a piece of gear has a particular feature doesn't mean you have to use it, but Liquid Mix really puts one's self-restraint to the test.

It's also worth noting that although there are some "plain" models, there are also plenty of models that add real character to the sound, almost like a signal processor. That's one of the factors that makes Liquid Mix cool: Those various models really do give you an arsenal of sounds that would formerly have taken a truckload of rack mount gear.

Given that one of the weak links in host software is often EQ (because you sometimes need a lot of instances, but they can't take up so many CPU cycles that you end up killing your processor's performance), Liquid Mix gives you great EQ and compression while leaving your CPU free to run CPU-hungry instruments and processors like convolution reverb. I don't know if I'd consider this an essential purchase for those with seriously constrained budgets, but if you can spare the bucks, Liquid Mix will definitely help take the sound of your native-based DAW setup to the next level. Maybe even the level after that, too. EQ

Product type: DSP processor for software-based hosts.
Target market: Native-based studios wanting higher-quality compression and EQ without loading down the host computer.
Strengths: Includes great control surface. Plenty of models, with lots of character and variations. Easy to understand and use. Sounds really good. Does VST/AU (and RTAS via a wrapper). Adds only a couple milliseconds to system latency. Includes power supply for use with four-pin FireWire ports.
Limitations: Selecting a plug-in in your host doesn't change it in the controller. Not expandable for more channels at 44.1/48kHz.
Price: $1,099.99
Contact: www.focusrite.com

SONY CINESCORE 1.0

Instant soundtrack: Just add software. Really.

by Craig Anderton

When it comes to workflow, soundtracks represent their own unique challenges. In a lot of the audio-for-video work I do, the budget and/or deadlines won't allow hiring musicians. Rather than break out a music library, which never seems to provide *exactly* what I want, I've taken to creating soundtracks using a combination of sample CDs and parts I play. With a bit of work, it's possible to have the music ebb and flow with the video, speed up and slow down to accent moods, overlay hits and transitions, and so on. Using this approach has saved me hours of time, and produced satisfied clients (always a good thing).

So when Sony claimed it was possible to basically automate this process, I was skeptical. After all, I put a lot of work into creating these soundtracks; how could a piece of software do the same thing?

WHAT IT IS

Cinescore consists of three main parts:

- An expandable sample library of "themes" (20 come with the program), with multiple variations (typically 20 or more) per theme.
- A workspace for assembling video and audio.
- An algorithmic composition generator that dips into the sample library, and creates music according to your specs.
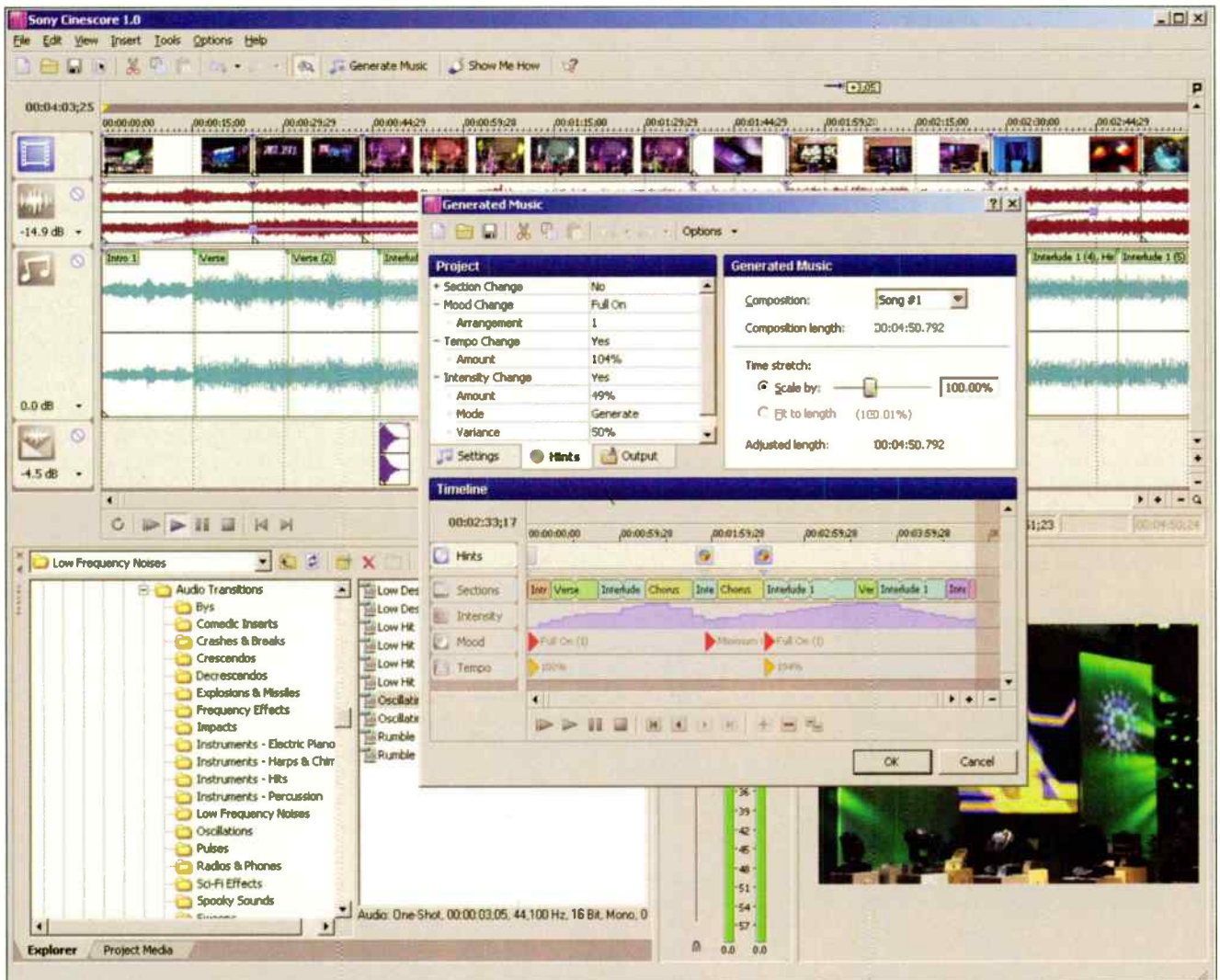
The Theme Chooser window is where you set the basic Score parameters; the Generated Music dialog allows previewing and fine-tuning the score. The four tracks are toward the upper left, and the video preview window toward the lower right.

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- :: arrange
- :: edit
- :: process
- :: mix

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SONY CINESCORE 1.0

Additional optional-at-extra-cost themes have already appeared, and I think it's a safe bet we'll see more in the future.

The workspace has four tracks: video, audio (e.g., the audio associated with the video, such as camcorder audio, narration, mixed dialog, etc.), generated music, and audio transitions (hits, sound effects, etc.; over 300 come with tne package). Although you wouldn't edit a video with Cinescore (for example, you can't add plug-in effects to the video tracks), you can cut, paste, copy, group, move, crossfade, do ripple editing, etc. This

an almost dizzying array of options, into the timeline to change the course of the generated music. A graph below where you do the edits shows the various "sections" that make up the score, and displays how some of the changes affect the overall flow. For example, you can see the arrangement of the sections change as you alter the "randomness" parameter. "Variance" causes more changes to happen, again, reflected by the names of the sections shown on the timeline. It's also possible to change the mood and most

The ability to key musical changes precisely to video puts Cinescore in a class by itself.

makes it possible to assemble multiple video clips together, then create the audio.

But the big deal here is the Generated Music track, as that's what holds the soundtrack. (Note, however, that WAV and AIFF audio files can be dragged into any of the audio tracks, so technically there are a total of three audio tracks available.)

APPLYING CINESCORE

First, you define the length you want the score to cover, then choose a theme. You can filter themes by instrument, tempo, genre, artist, keywords, etc. Next, you choose a variation on the theme; the audio preview button helps a lot with respect to finding you want.

An "advanced" option allows customizing the variation. You can choose a different intro, arrangement (changes the instrument sounds that are used), tempo, mood, and most interestingly, "intensity." This increases the vibe from more minimal to adding more instruments, more notes, effects, and the like. We'll revisit these later, because you can change the intensity and other parameters over the course of the score.

After setting up your basics, the Generated Music dialog box appears. Here you decide about the ending: fade out, normal (ends in a musically relevant way, but may change the length a bit compared to what you specified; in that case, invoke the "Fit to Length" option), abrupt (matches length exactly), and loop (for kiosk videos and the like that repeat).

But this is also where you hit the heart of Cinescore: Hints. You drop hints, with

importantly, the intensity, as well as specify variations within the intensity changes.

Still with me? If not, don't worry; you can just take what Cinescore gives you, and have something pretty cool. But the ability to key musical changes precisely to changes in the video is what puts Cinescore in a class by itself.

For example, when looking at the "Intensity" graph, because I'd inserted a hint to increase intensity, it kept peaking until the end. But the score I'd generated was intended only for the first half of the video; I planned to generate another piece for the second half. As I didn't want the video to peak too early, I inserted another hint about 2/3 of the way through the first section to bring things back down a bit. Cool.

Finally, you can audition the entire soundtrack, or portions of it, to make sure all the changes are as desired. This offers the usual play, pause, stop, etc. controls, as well as the ability to jump to subsequent and previous hints for editing. Once you click OK, the music goes on the generated music track.

You can also edit or recreate the generated music if you change your mind later on. The score is saved as a WAV file, so you can process, edit, re-use, or master it. However, Cinescore saves *everything,* so prune away the stuff you don't need if you don't want your Cinescore 1.0 music folder to get bloated.

CONCLUSIONS

Cinescore is amazing, but not miraculous. For example, I just did a video where

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there was a solo guitar player, and it was
decided that the soundtrack needed
drums playing along with the guitarist. So,
I dragged a bunch of individual hits and a
few loops from the Discrete Drum
libraries into Vegas, and matched the hits
up with the guitar notes. There's no way
Cinescore can generate this level of
detail, nor is it designed to.

What it *can* do, though, is shift gears
very ably to match moods, and do so on a
dime. This isn't "dumb" algorithmic gen-
eration, but has intelligence thanks to the
"hints" you provide. When you specify
that the soundtrack get more intense, it
really does get more intense. Being able
to drop back to something more minimal,
then build up to something more com-
plex, is an incredibly powerful technique
(as is stretching tempo).

The secret weapon is the transitions
track, as you can line up hits and sound
effects very precisely with any visual
cues. This furthers the sense that the
soundtrack was crafted specifically with
the video in mind.

Maybe Cinescore won't handle 100%
of my scoring needs, but it *can* come sur-
prisingly close — like when I need music
now. You can generate a soundtrack in
minutes (literally), and if you're willing to
put in a little more time, you can easily
take a soundtrack to the "next level"
where it truly fits the video like a glove.

In the short time it's been out,
Cinescore has proven to be a real time-saver
and workflow enhancer that definitely
exceeds expectations. If it wasn't so friendly
and obedient, it would be scary.

Product type: Algorithmic scoring
software for Windows XP or 2000 SP4.
Target market: Studios doing audio-
for-video work that need to produce quality
soundtracks despite low budgets and
tight deadlines.
Copy protection: Serial number
and registration.
Strengths: Actually does what it
purports to do. Easy to use, but you can
get pretty deeply into editing if desired.
Limitations: Audio plug-ins not sup-
ported. Limited to the themes that come
with the program, and whatever add-on
themes appear in the future.
Price: $199.96 list (boxed ver-
sion); check website for download-
able versions.
Contact: www.sony.com/mediasoftware

AUDIO-TECHNICA
ARTIST SERIES MICROPHONES

Ch-ch-ch-ch-changes meet A-T's Artist Series

by Phil O'Keefe

Audio-Technica has revamped their entire Artist Series of stage and studio microphones by redesigning several previous products from the ground up, while adding some interesting new mics to the roster. I was sent all eight mikes in the new series for evaluation and review. This month we'll take a look at four of them, and hit the remaining four in a future review. (And by "hit," I don't mean setting them up too close to the snare drum.)

ATM650 HYPERCARDIOID DYNAMIC INSTRUMENT MIC

Well, it looks like a snare mic, so I thought I'd try it there first. I positioned it in my usual starting point: on the side of the snare shell,

about 3" from the wood. I find this technique usually gives me a good balance of attack and snare rattle, with a warm, balanced sound — provided the mic is up to the task. If you're going to try this technique, be careful you don't aim the mic at one of the snare's vent holes, or you'll get a nasty blast of air directly to the diaphragm every time the drummer whacks the snare.

So how did it sound? Nice. It was somewhat similar to the "old snare standard" mic. There's a presence peak centered at around 5kHz that really helps things to cut through. The attack transients are excellent for a dynamic microphone, and off-axis coloration

was fairly minimal. Rejection off-axis is very good, which certainly helps. By pointing the mic at the snare, but angling it a bit so that the null point (135°) was aimed toward the hi-hat, I was able to reduce hat leakage into the snare mic by a significant amount — a lot more than would have been possible if I was using the old standby.

On guitar cabinets, it performed as I suspected it would after my snare drum experiences. If you want a mic to help a guitar cut through the mix, the presence peak on the ATM650 will definitely get you there. On the bottom, it rolls off about 6dB/octave below about 100Hz, and the low midrange comes across as fairly flat — both of which mean it may not be your best option if you're working with drop tunings or are trying to capture corpulent amounts of beef down low. It's built like a rock and looks like it should withstand the occasional drum stick hit without caving in. Overall this is a crisp, defined, and well-built microphone with a nice hot output, due to the neodymium magnet.

ATM450 SMALL DIAPHRAGM CONDENSER MIC

This has to be one of the more interesting new small diaphragm condensers I've come across in a long time. It looks like your typical pencil condenser, but instead of being end address, it's actually a side address microphone — "honey, I shrunk the C12!" On a stand, it made me think of a mini-me version of the C12, as it's 1/8th actual size. This is a surprisingly small microphone.

Small size and side address means you can sneak this mic into really tight spots with ease. It also means that it presents a smaller target for drum sticks.

Condenser mics on toms can be a beautiful thing, but most of us resort to dynamics due to the large size and difficulty with placing most condensers up close around a drum kit, and due to the cost of replacing them if they get hit. That's not an issue with the ATM450. I was able to have it peek just a couple of inches over the edge of the drum rim, angled about 45° towards the sweet spot, and it gave me a very natural and accurate-sounding image of the drum. With "heavy hitters," the onboard 10dB pad kept levels and overload in check. I wish I had a pair on hand so I could have tried these as stereo overhead mikes, but my experiments with close and distant miking of hi hats, cymbals and hand percussion, and tambourine (all of which sounded excellent) suggests that they'll probably perform well in that application.

With acoustic guitar, I found the ATM450 to have a tonality similar to my AT4041s — clean, crisp, and articulate without hype or flatter. The onboard −18dB/octave @ 80Hz high pass filter helped to keep the timbre from getting too "boomy" when the microphone was at "up close and personal" distances from the source, and works well at reducing room rumble.

The AT8471 stand mount is a wraparound rubber design with a tightening thumbscrew. A-T calls it an "isolation clamp," and while I didn't find it to be quite as effective as a spyder/elastic type shock mount, it was definitely more effective than most clip type stand mounts at reducing stand-borne vibration. It also held the mic quite securely, while still allowing for easy adjustment of angle and position. A foam-type windscreen is also included, which can be useful for reducing wind noise, but had A-T not included it I really wouldn't have missed it.

This mic really isn't at its best when used for vocals, and the foam windscreen dulls the extreme top end a bit, as expected from any foam windscreen. Unless you really need to use it — for example, when taking the mic outdoors on a windy day — I would suggest foregoing it and allowing the mic to breathe naturally.

But minor quibbles aside, I suspect this is going to be a very popular microphone due to the good sound and excellent placement options, thanks to the small size and

Intelligent I/O - Integrated F/X

Not Just Another Audio Interface

Konnekt 24D is a groundbreaking 14/14 I/O audio interface that integrates TC's world famous effects in a state-of-the art design. Developed for recording and performing musicians seeking the highest possible sound quality, Konnekt 24D ensures that your analog recordings make it to the hard disk in perfect shape.

konnekt 24D

Expected features

- 14/14 I/O: 2 mic/inst/line, 2 line inputs and 4 line outputs, 8 ADAT and 2 S/PDIF (optical and coaxial)
- Full feature stand alone mode
- Link up to 4 units to get more inputs, outputs and effects channels
- DICEII digital interface chip with JET™ Jitter Elimination Technology
- FireWire bus powered
- Low latency drivers: WDM, ASIO and CoreAudio
- 24-bit/192kHz sampling rate
- Sample accurate MIDI
- Bundled with Cubase LE

Unexpected features

- Built-in real-time DSP effects; Fabrik R reverb and Fabrik C channel strip based on MINT™
- IMPACT™ mic preamps
- True Hi-Z guitar inputs
- Front panel light ring control of internal mixer parameters
- Intuitive control panel with automatic input detection
- Analog volume control for perfect integration with powered speakers
- Dual headphone outputs, one with auto speaker muting
- Assimilator Konnekt native fingerprint EQ included
- 3 DSP programs for total recall of internal routing, mixer and effects settings
- TC NEAR™ 1394 FireWire based network for full feature direct monitoring – even between units

FABRIK R
INTEGRATED

FABRIK C
INTEGRATED

ASSIMILATOR KONNEKT
INCLUDED

konnekt 8
ALSO AVAILABLE

t.c. electronic
ULTIMATE SOUND MACHINES

side address design. This is definitely one of my favorite mics from the new lineup.

ATM250 HYPERCARDIOID INSTRUMENT MIC

This one has a big, bold, and punchy tone that takes hot source levels and begs for more. This mic just wouldn't overload for me no matter how hard I slapped it sonically I

placed it about 4" above the floor tom and angled slightly inward; the result was a big and deep tom tone, with a ton of chest thumping bottom end beef. I would describe the sound of this mic to be almost "pre-EQed," with a flat midrange and two bumps in the response — one centered at about 80Hz and the other at about 3–4kHz. It is nearly perfectly optimized for toms and kicks.

I am usually not a fan of pre-EQed mics, and normally prefer to dial that in myself, but this one really works well, and Audio-Technica did a good job of selecting the frequencies to boost and dip to make the source sound great I could see where live FOH folks will love this mic; you can set it up in a hurry and get a great sound out of it with minimal muss and fuss. In that respect, it reminded me of an improved ATM25 (which in my opinion has always been a very good kick and tom mic),

but the 250 has a bit more fullness down low and a touch more extension up top.

This mic also sounded very good on guitar cabinets, especially if you're looking for a little more beef down low than you would get from the ATM650. The only downside I can see to this microphone is that it is somewhat larger in physical size than the old ATM25, which means placement on rack toms may be more difficult than with the older microphone.

ATM250DE DUAL DIAPHRAGM MIC

This is very similar to the ATM250 in several respects, and when using the dynamic capsule in the ATM250DE, they both sound pretty much identical to my ears. Both use similar bodies, although the ATM250DE's grille is a little bit longer. However, the ATM250DE adds a second condenser element inside the same housing. The two diaphragms are perfectly aligned so that there are no phase differences, which means that with one stand you can capture two different tonalities from one mic, and still not have to worry about flipping phase at the console or mic preamp. Each mic's output is sent out via an included 16' cable that terminates to two separate XLR connectors, and each is clearly marked so you know which plug is for the dynamic element and

which one will need phantom power (11–52V) applied to power the condenser element.

The condenser is similar in tonality to the ATM450; it has a –10dB pad and a –12dB/octave low frequency roll-off switch. Unlike the hypercardioid ATM450, the polar pattern of the ATM250DE's condenser element is cardioid, and the wider pattern allows it to pick up a bit more of the shell resonance

when placed inside a kick drum. Both the ATM250 and 250DE include the same AT8471 isolation mount that is included with the ATM450, and they did equally well at securely holding even these larger and heavier microphones where you put them.

CONCLUSIONS

All of these new A-T mikes are built very well, and look like they would weather the storms of live touring or busy studio schedules without giving up the ghost at an inopportune moment. All four mics impressed me a great deal with their no-nonsense performance and quality sound, and the prices for them seem quite reasonable given their level of quality. While there are a lot of instrument mics on the market, these are definitely worthy of serious consideration; Audio-Technica's Artist Series has been a hit over the years, and the revised edition make the lineup only that much more potent. EQ

Product type: ATM650, ATM250: Instrumental dynamic mics. ATM450: Small

diaphragm, side address, fixed-charge back plate permanently polarized condenser. ATM250DE: Dual element condenser / dynamic instrument microphone.

Target market: Recording studios and live sound engineers who need high performance, rugged, and cost-effective microphones that are suitable for a variety of live performance and instrumental recording applications.

Strengths: Good off-axis rejection and minimal off-axis coloration on the ATM650. Unusual side address design of the ATM450 makes placement extremely easy, even in the tightest of locations. ATM250 and ATM250DE pre-EQed specifically for toms and kick drums, which makes getting a good sound fast and efficient. High maximum SPL handling, good sonics, and excellent build quality throughout the entire product line.

Limitations: Nothing significant at these price points.

List prices: ATM650 $169, ATM450 $369, ATM250 $329, ATM 250DE $549

Contact: www.audio-technica.com

World Radio History

APOGEE BIG BEN

Do you know what time it is? Big Ben does.

by Jeff Anderson

As a studio owner, I'm always looking for that next great piece of gear that will help make my end products sound better. As my facility is analog-based — from console to outboard — with ProTools being used in the context of a tape machine (the only digital interfaces are those within my TDM rig to send a signal from the converters to the digital mixdown recorder), I had never thought much about adding a digital clock to my system. I always figured that the few digital boxes we did use weren't very complex, and that surely, adding a clock to control a mere five pieces of gear would not yield much improvement. Right, guys? Right? Wrong.

THE JITTER PROBLEM

To grasp how a master clock can result in a drastic improvement in sound quality, we have to understand the "jitter" problem. For example, suppose you have a piece of digital gear with a 44.1kHz sample rate. Essentially, this gear is taking a measurement of the sound's waveform every 1/44,100th of a second. But with so many samples being taken so often, inevitably a sample ends up being taken either a little too soon, or a little too late. This tiny, yet ultimately noticeable, timing error constitutes "jitter."

While the jitter is typically in the hundreds of picoseconds to the lower nanoseconds, it will make a noticeable difference in sound quality that's especially evident in frequencies below 5,000Hz. This is because the timing errors in sampling will result into frequencies generated above and below the frequency being sampled, creating the so-called side bands which we perceive as distortion. Jitter can also "collapse" the stereo image somewhat.

What's *truly* disheartening is that every piece of digital equipment in your studio is going to be plagued by jitter. What's worse is that when using a digital recorder, there's jitter when running the signal into the computer, as well as back out. Before long these timing errors start to accumulate, and greatly diminish the overall sound quality.

APPLYING THE BIG BEN

Apogee has been widely respected for their C777 clocking technology (a staple of many of their devices, and the heart and soul of the Big Ben). Developed using Direct Digital Synthesis technology (DDS) and DSP-based digital filtering, C777 clocking is considered much more effective than the traditional phase lock loop circuitry (which lacks dynamic adaptability). Bottom line: DDS basically regenerates a stable, low-jitter signal that replaces the original clock signal. It can act as a precision master clock, regenerate its own clock signal, or accept signals from external sources at sample rates of 44.1–192kHz. These sample rates can then be varied (or "VSOed") by –44/78%, for ±999 cents.

Big Ben accepts the following inputs: two AES (XLR), S/PDIF (RCA), SPDIF or ADAT/Smux (optical), word clock (BNC), and composite video. Big Ben's outputs consist of two AES, S/PDIF, S/PDIF or ADAT/SMUX (optical), and six word clock outs, four of which transmit regular frequency clocks while the remaining two can also transmit up to 256 multiples of the clock signal. A video sync (re-) generator board is available as an option.

Integrating Big Ben into your studio is fairly simple, as the unit offers warning lights for over- and under-termination in your clock signal chain. (Termination minimizes reflections going back along the clock line, which can be a source of error.) This is incredibly useful during installation, as the indicator lights for each clock out alerted me to whether or not I was terminating in the correct location, and if the line was properly terminated.

Apogee's SureLock technology also assures that the Big Ben's outputs will remain stable even in the face of an external clock failure — which is super handy when the odd cable pops out in the middle of a session (this actually happened while I was testing the unit). Another helpful Big Ben feature that will have users wearing out buttons is its ability to convert signals in real time from SPDIF to AES to ADAT (and vice versa) with ease.

CONCLUSIONS

Even at one tenth of a nanosecond, jitter adds up and can quickly become noticeable in your recordings. Big Ben, though not the only solution, is indeed a fine one — and I like the additional real-time format conversion feature. Being able to regenerate external clock signals truly adds to the efficiency of your studio, and the "SureLock" technology means you can rest assured there will be a constant, glitch-free clock at all times.

Centralizing your clocking signals for system configuration changes is now a breeze, and being notified if the clock line isn't properly terminated makes adding the Big Ben into your chain relatively stress-free. What began as a simple trial run, with the overriding question of "do I really need this," has ended with Big Ben becoming an integral piece of my rig. EQ

Product Type: Digital word clock.
Target Market: Individuals using multiple pieces of gear with internal word clocks.
Strengths: Exceptional jitter reduction and format conversion/support. Multiple outs. Clock signal maintenance despite input loss. User-friendly installation.
Limitations: None.
Price: $1,495 list
Contact: www.apogeedigital.com

ackmount Power Tools! ℠

FURMAN AR-15 SERIES II

Here's something for real power freaks

by Jeff Anderson

Voltage sags, brownouts, spikes, and overvoltages — such are the nemesis of every studio's prized collection of gear. Were you to monitor the power that's pouring out of your wall outlets, you would discover readings that can vary greatly from facility to facility, depending on factors ranging from service providers to how your studio is wired. If you have not implemented some sort of voltage regulation to control the power feeding your equipment, you are leaving yourself open to a world of hurt.

Thankfully, there are devices, such as Furman's AR-15 Series II AC voltage regulator, to protect our vulnerable gear from power discrepancies. Technology-wise, it uses Linear Filtering Technology that, unlike some more traditional AC conditioners, ensures linear filtering and no leakage to the ground — which means a quieter, more stable signal for your equipment. The Series Multi-Stage Protection + (SMP +) technology is a great asset, as it safely handles damaging transient voltages without sacrificing the entire unit or requiring a trip to the repair shop; another safety feature, Extreme Voltage Shut Down (EVS), shuts off incoming power to the unit in case of accidental connections to 208 or 240VAC. Excellent.

This single space rack-mount unit boasts eight outlets on the back, and one on the front. All are regulated, spike-suppressed, and filtered against RFI. The 10-LED bar graph meter on the front of the unit displays input voltage ranging from 75–150V, and an LED displays "In Regulation Status" showing that the output is within ±5V of 120V. In addition, an indicator LED for "Protection" displays the unit's working status, while "Extreme Voltage" indicates if the unit is shut down due to protecting against a 145V or higher jolt. To top it off, there is the requisite on/off switch as well as an optional on/off for a gooseneck light, which can be attached to the back via BNC connection.

IN USE

Nearly slipping all of my discs out of place attempting to lift the AR-15 II out of the box, I got to work. I've had a previous power conditioner for quite some time, set in line to protect nearly 30 pieces of outboard gear, but had never given much thought as to what degree it was suiting my needs. But oddly enough, within minutes of installing the AR-15 II, a serious problem cropped up that let it be put to the test. I noticed that, somehow, the air conditioning unit for my machine room was pulling current from the same breaker as the equipment rack — every time the air conditioner would cycle, the Furman would display a huge dip in current. It was so bad that I could clearly hear the AR-15 II sounding off as it compensated for the power drain in order to provide the regulated 120V to the rack.

After hammering out the air conditioning issues, I decided to do a test to see if there was a noticeable difference in the sound quality of equipment protected by the Furman. So, I routed a stereo audio signal through 12 different compressors, EQs, and effects in then rack, then onto two tracks in Pro Tools. After setting all of the processors so that there was a "legible" sound, I tracked 30 seconds of a song with the AR-15 II regulating all of the equipment. Directly thereafter, I bypassed the Furman, plugged the entire rack into the wall, and tracked the same 30-second chunk of music. I heard a fairly notable difference — not huge per se, but a definite difference. During the 30-second sample that was recorded Furman-less, two "clicks and pops" appeared that were absent from the treated tracks, and the low end seemed a bit more apparent too.

CONCLUSIONS

After spending tons of money building a studio, it seems truly negligent to forego voltage protection, thus leaving all that great gear in a compromised situation. Within just five minutes of adding the AR-15 II into my rig, I found (and rectified) a potentially serious problem that would have otherwise flown under the radar. Saved hides aside, the unit reduced the noise floor and actually improved my overall sound, which is always a good thing. Basically, if you've yet to acquire an AC voltage regulator, there is no time like the present — and the AR-15 II is a great piece to start. EQ

Product type: AC voltage regulator/stabilizer.
Target market: Small to medium size studios that need to protect their gear from electrical problems.
Strengths: Multiple protection technologies. Handles overvoltages elegantly. Improves overall audio quality.
Limitations: Weighs a lot.
Price: $755 list
Contact: www.furmansound.com
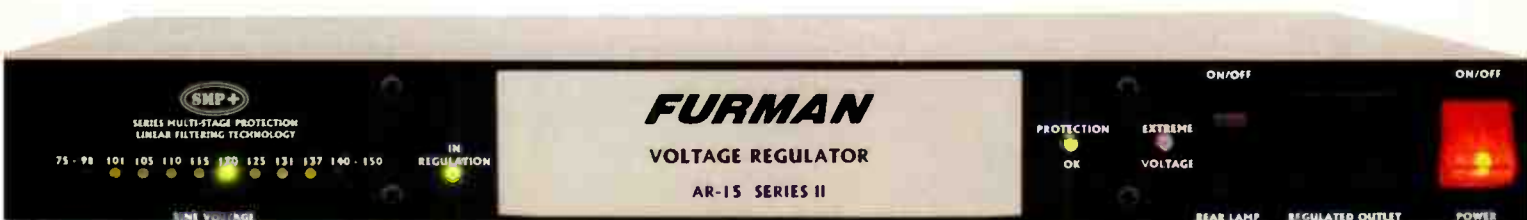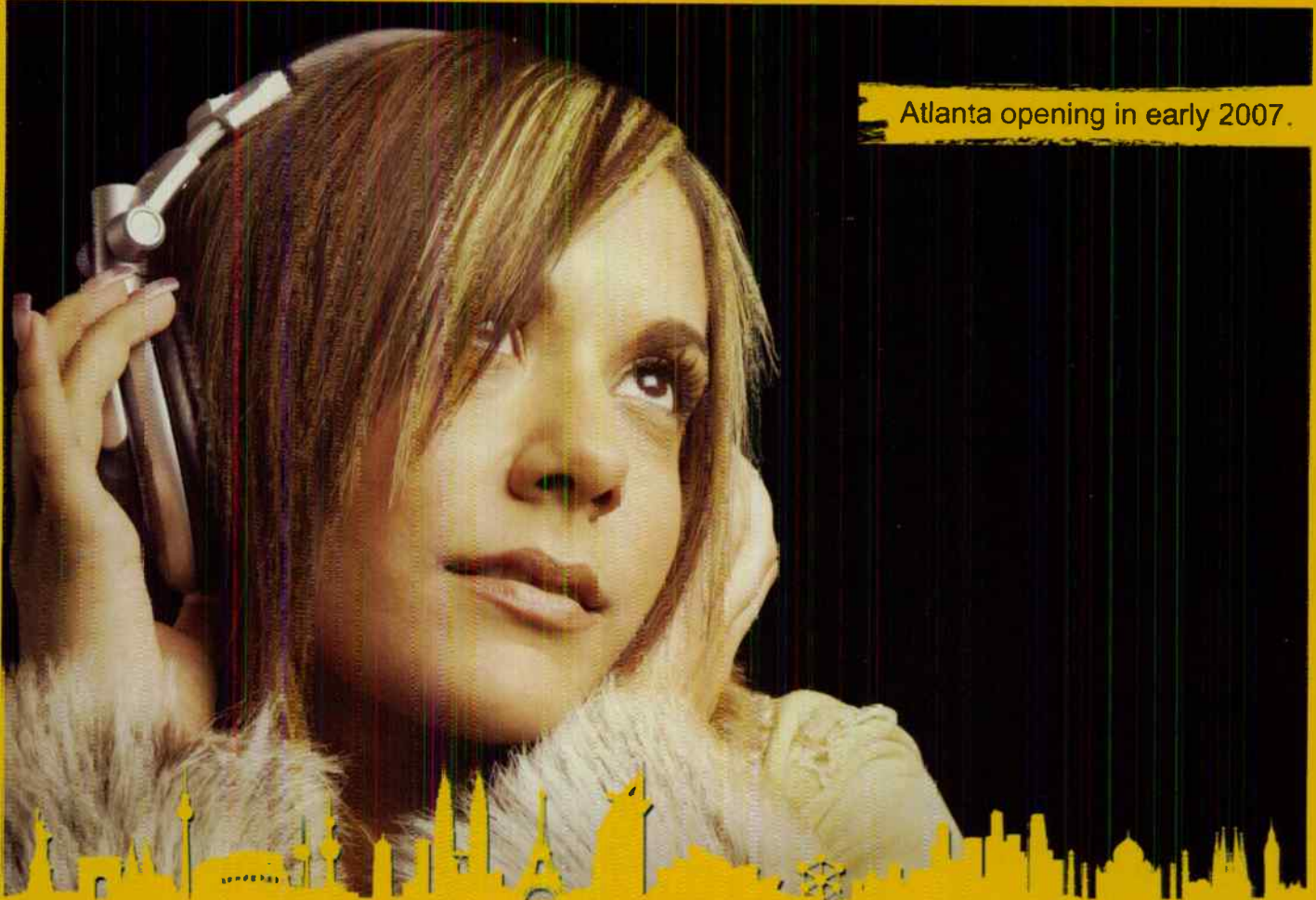
CHARTEROAK ACOUSTICS E700
MULTI PATTERN CONDENSER MICROPHONE
East meets West, and the winner is . . . you

by Roy Stein and Tony Gross

This past decade has witnessed an invasion of mics that have been tough on tradition. Chinese-manufactured, large-diaphragm studio condenser mics that copy both the circuitry and the external housing of their Western counterparts have found their way into many recording studios, as they're cheap in comparison to their erudite ancestors. However, in many cases the similarities end at "sharp looks," and with so many different brands flooding the market, one wonders which of these mics is wise to buy . . . if any.

Perhaps we should take some advice from Aristotle and Lao-Tzu, and embrace the middle ground — which is where the CharterOak E700 lies. This robust, dual-diaphragm FET condenser mic shares some characteristics with the import competition to keep costs reasonable. Sure, some of its basic components, including circuit cards, metal work, and the flight case, are manufactured at facilities that work with other rebranders of microphones, but that's where the cloning around stops and individual design and quality control steps in.

Starting with a piece shipped in from China in "rough assembled" form, the mic is then biopsied and reassembled by hand with Xicon, Mial, and WEMA polystyrene and polypropylene capacitors to optimize the audio path. Before shipping, each mic goes through a final testing phase before it is stamped with its *coup de grâce:* a lifetime warranty.

CONSTRUCTION
We were informed that if we took the mic apart we wouldn't find any cheap hookup wire, and the wiring would be neatly in its place. Way too late for that; within hours of its arrival we had that sucker pithed like kids on a frog in Biology 101. The split circuit board design was nesting on a rugged but cleanly machined platform, and only Teflon-coated wire is used to make the connections between the capsule and head amplifier to ensure maximum noise rejection and to faithfully reproduce detailed highs.

When it comes to design and quality control, CharterOak appears to be demanding. For example, originally the E700 employed a Chinese output transformer. However, only 11 out of 40 of those transformers passed the company's quality control standards. Starting at s/n 0012, it was replaced with an American component. Also, beginning with s/n 0040 CharterOak has replaced the 3-position pad and 3-position pickup pattern switch with a heavy-duty 3-position toggle, rather than the earlier model's slide switch. Alright, it's all sounding sweet . . . in theory.

APPLYING THE E700
An organic-sounding rock band was recording in Studio A doing the standby drill: vocals, drums, bass, guitar. The clock's tickin' off billable hours, so it's a trial by fire for the E700. Typically on a kick drum we remove the front head and use either an EV RE-20 or an AKG D112 on the inside of the shell placed slightly off axis about 8" from the beater, as well as a Soundelux E47 about 8–10' outside the shell (all through API 3124 preamps), as this usually captures spectacular results. So we decided to throw the E700 solo about 4' outside the front of the kick, through the API 3124 — it was so good, we left well enough alone. Recording the kick became a single mic/mic pre chain,

which freed some prime gear for other sonic territory.

Up next, acoustic guitar: a Gibson Hummingbird, known for its full-bodied, warm, velvety tone. But in a rock band competing for prime sonic real estate, it can easily get lost in the mix or, at higher levels, muddy up the track. We set up the Soundelux E47 on the low end of the guitar, off axis, along with a Røde K2 on the top end to capture the 12th fret nuances, which sounded great. Then we replaced the K2 with the E700, and had the band members take a blind test. They all chose the E700.

To cut vocals, we set up a Soundelux Elux 251 and the CharterOak E700. We ended up going with the Elux for the main track, but were not at all disappointed by how the E700 held up (a good thing, given the price difference between the two). The E700 was characterized by detailed highs, without being brittle, and an open bottom end. And by adding just a touch of compression on playback, the vocals sat nicely in the mix.

CONCLUSIONS
"East is East and West is West." Not in 2006. By combining U.S. quality control and the selective integration of high-end components with low cost manufacturing available in China, CharterOak offers an excellent buy for the buck. The E700 would be a welcome addition to any rocking mic cabinet, or a smart choice as a first upper-level mic. EQ

Product type: Dual-diaphragm FET condenser mic.
Target market: Higher-end studio and mid-level applications with cost restrictions in mind.
Strengths: Detailed highs along with clean, open low end. Excellent shock mount and flight case. Lifetime warranty.
Limitations: Hefty, so be careful with mic stands.
Price: $1,199 list
Contact:
www.charteroakacoustics.com

SONY GLOBAL GROOVE
International Hip-Hop Flavor

Contact: Sony, www.sony.com/mediasoftware
Format: CD-ROM, Acidized WAV format, 44.1kHz/16-bit, 524 files (loops and one-shots)

First things first: This is "global" only if your sense of the globe is focused on the Caribbean. Semantics aside, though, this CD steers clear of the trend toward a more Middle Eastern flavor in hip-hop, and instead appropriates the intoxicating sounds of zouk, soca, dance hall, reggaeton, and other genres of "island" music.

However, this isn't necessarily the carefree music you hear while sipping a drink with an umbrella in it by the beach. The vibe is darker, harder-edged, and more minor — in other words, more hip-hop. The recording is stark too, laying down muscular beats and sparse, angular melodies.

There are 20 construction kits, plus a folder of promotional loops. While the emphasis is on percussion (with drum mixes, individual drum loops, and a folder of one-shots per kit), there are also melodic elements like guitar, synth, piano, and effects. As with other Sony libraries, the acidization is superb, so mixing and matching is a done deal. And although the promotion hits hard on hip-hop, a lot of these rhythms are valid accessories to just about anything where you want to add a slightly heavy Caribbean vibe.

At first, I was taken aback because this isn't the kind of light, soaring island music I love (e.g., Kassav). But once I got past any preconceived notions, it became clear this CD offers a unique, equally valid take on Caribbean music — and has the ability to partner with other types of music as well. Check out the audio example at www.eqmag.com, and you'll hear what I mean. —*Craig Anderton* EQ

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BIG FISH AUDIO
Acid Jazz City

Contact: Big Fish Audio, www.big
fishaudio.com
Format: DVD-ROM with 30 construc-
tion kits in three formats: WAV, REX, and
Apple Loops; 24-bit/44.1kHz
Price: $99.95

Acid Jazz City . . . sounds fun, and word
on the street says it has an excellent bar
scene. So I stumbled into Steve's Place,
a smoky south side lounge — lured by
the Hancock-like pianos, Miles-type riffs,
and funky percussion floating through
the cool night air.

After a couple scotches I sidled up to
Lola, looking gorgeous as always. "What
you got for me tonight, doll?" I asked.
"The usual," she laughed. "Kits, each with
a mix file and its various elements . . .
about 36 loops per kit."

Kits: I should have known — it's a
Big Fish thing. But could a kit convey
the improvisational nature of jazz? "Chill,
baby," Lola cooed. "What I got is riffs.
Short riffs, long riffs, acidy riffs, even
some trad kinda riffs. You provide the
solo, and do your improvising . . . I'll pro-
vide the bed for it."

The way Lola says "bed" is enough
to distract anyone, but I still had a job to
do. "That flute player is pretty good,
what's the deal?" Lola smiled, "For the
leads, there are lots of little bits and
riffs. You can actually put together a
pretty convincing performance, if it
doesn't go on too long. But why not just
go to www.eqmag.com; there's an audio
example there that shows exactly what I
mean. I'll even go there with you," and
she winked.

Well, that was music to my ears. And
so is *Acid Jazz City*. —*Craig Anderton*

Power App Alley
by Craig Anderton

PRO TOOLS 7 |HD, LE, or M-POWERED

Want a really good, free workstation RTAS instrument? Of course you do!

OBJECTIVE: Improve workflow when tracking by using a general-purpose "workstation" plug-in with Pro Tools. Parts can be replaced later if needed.

BACKGROUND: Digidesign kicked off their Advanced Instruments Group by offering a free RTAS workstation plug-in. While it's said "there's no free lunch," this one is actually very tasty — it's well worth obtaining and installing.

steps

1. Go to the DigiStore and order the Xpand! CD. There's a small handling charge. If you already have the Xpand! CD, download the latest update.

2. Insert the CD, double-click on "Xpand! Setup.exe," and follow the instructions.

3. Create a new Track by going Track > New. In the New Tracks dialog fields, enter 1, Stereo, Instrument Track, and Ticks, then click on "Create."

4. In the Mix window Inserts section, click on an insert and go Multi-channel plug-in > Instrument > Xpand! (stereo).

5. Turn up the mixer fader for the Instrument track, play some notes, and assuming your MIDI setup assignments are correct, you should hear the default patch.

6. Xpand! is not multi-timbral, but you can layer multiple parts (instrument sounds) on one channel. Click on the part field's downward-pointing arrow, and choose the desired sound. Turn parts on and off by toggling the button to the right of the part letter (A, B, C, D).
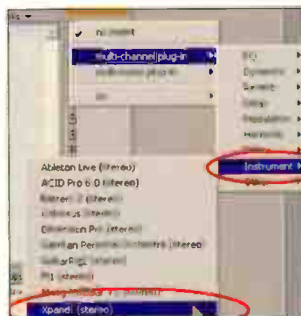
tips

■ The six Xpand "smart knobs" edit useful parameters in each part. Experiment!

■ To assign a knob to a MIDI controller, right-click (Windows) or Ctrl-click (Mac) on the knob. Choose "Assign" then assign a controller. Or, click on "Learn" and move the controller you want to assign to the knob.

■ Note that the same controller cannot be assigned to the same knob in multiple parts.

Power App Alley
by Craig Anderton

CAKEWALK SONAR

There's more to editing than cut, paste, drag, and copy

OBJECTIVE Do video-style "ripple editing" with Sonar.

BACKGROUND With ripple editing, deleting part of a track causes material past the end of the edit to move ahead in time (toward the left), thus filling in the deleted portion of the track. For example, if a solo section goes four measures too long, you can delete those four measures across all tracks, and the rest of the song will "close up" to fill in those four measures.
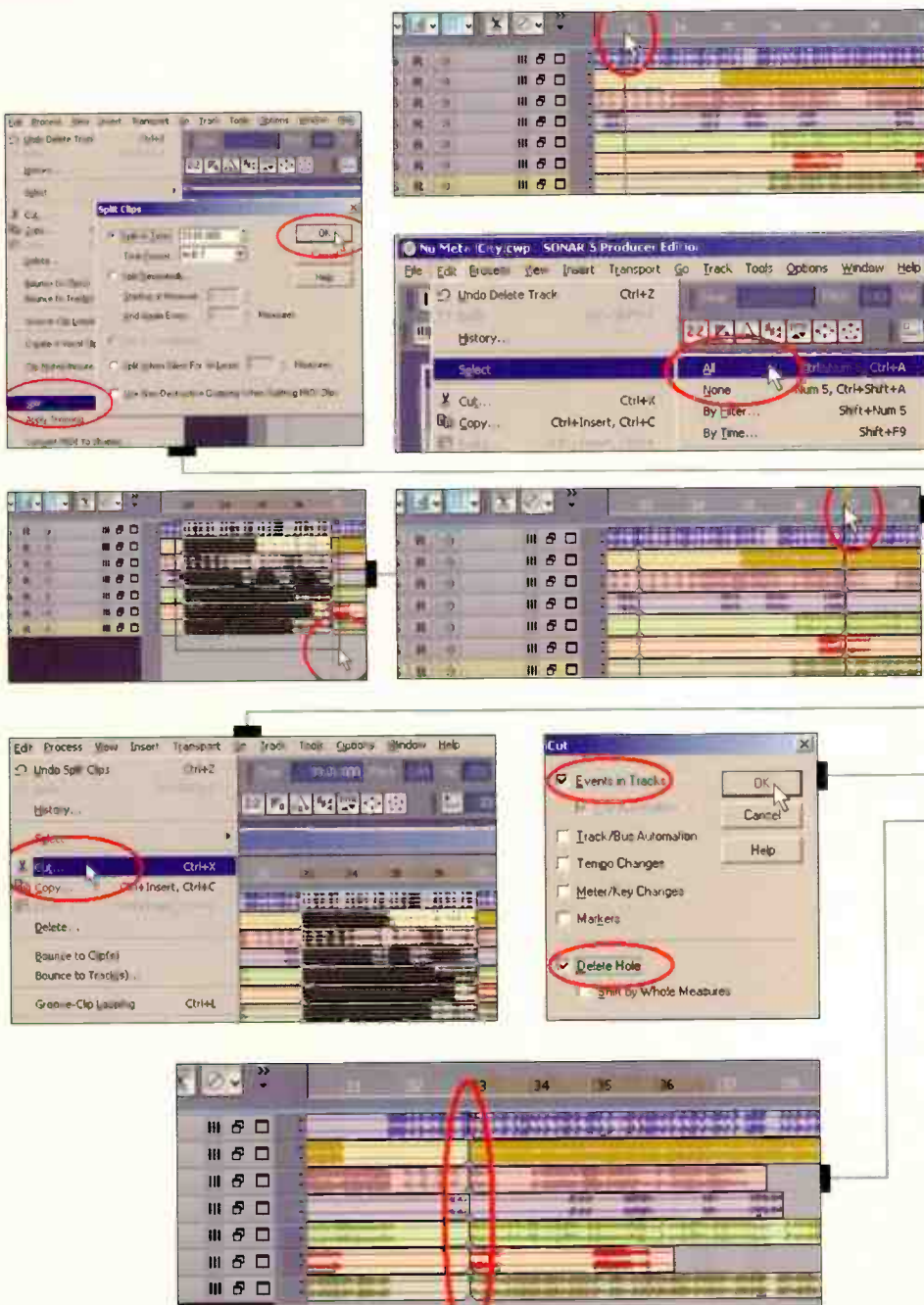
steps

1. Click on the timeline at the beginning of the region you want to cut. The Now time now sits at the beginning of the region.

2. Go Edit > Select > All (or type Ctrl-A) to select all tracks.

3. Go Edit > Split (or type "S") to split all tracks at the Now time.

4. Click on the timeline at the end of the region you want to cut, then repeat steps 2 and 3 to split all tracks at the new Now time.

5. Draw a marquee around the region defined by the two split points.

6. Go Edit > Cut (or right-click within this region and choose "Cut" from the pop-up menu).

7. When the Cut dialog box appears, check "Events in Tracks" and "Delete Hole," then click on "OK."

8. The selected regions will disappear, and the region that starts at the later split will move over to the left so it butts up against the first split.

tips

If you check "Shift By Whole Measures" in the Cut dialog box, the region that starts at the later split will move left to the closest measure after the first split. For example, if the first split occurs on a measure boundary, the region will shift left to that boundary. If the split occurs a quarter note before a measure boundary, the region will still shift left to that measure boundary.

There is no undo on your mic preamp.

Input channels can magnify the signal by more than 1000:1. There's no "Undo" after this point.

Coloration or distortion become part of the signal. Lost information can never be recovered.

Data compression algorithms can't distinguish signal from noise, so preserving absolute signal integrity through the recording and production process is critical to communicating your intention and individuality.

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In the Studio Trenches
by Phil O'Keefe

KEEPING IT CLEAN
IN THE DIGITAL DOMAIN
The art of deleting dirt never ends

Digital means clean, right? Well, I wish . . . but there are plenty of digital gremlins who are ready to rise up and bite you if you're not careful.

ONCE A DIGITAL, ALWAYS A DIGITAL

One way to help keep a digital track clean is to stay in the digital domain once the track is recorded. Each time a signal passes through converters and into outboard electronics, the sound changes — however subtly — and some noise hitches a ride. If an outboard processor adds to the track, or if you prefer to mix via analog summing, go ahead: But know what you're doing and why, listen to it both ways, and make an informed decision. If your signal path can live without an extra stage, *don't use it.*

NOISE REDUCTION: *NOT* A PANACEA

Suppose that for the most part your recording is clean, but there are a few remaining noise issues. Do you grab a noise reduction plug-in? I avoid them unless absolutely necessary, and when I do need noise reduction, I use the bare minimum. Noise reduction works by subtracting energy in particular bands, and when pushed too hard, they may subtract energy

Fig. 1.

you *do* want. Besides, the sonic artifacts from even the best noise reduction algorithms can sound more distracting than the noise. If the noise is *really* bad, it's re-recording time.

"COPY AND PASTE" NOISE REDUCTION

Not all noise is electronic. While tracking one of the songs on Julie Day's CD (www.julieday.org), we could hear the upright bass buzzing. First we thought it was from a humidifier that was accidentally left inside the bass and sympathetically vibrating as the instrument was played, but even after removing it, there were still places where the bass would buzz, and we couldn't locate the problem to fix it. As we had several other musicians waiting and the clock was ticking, we just went ahead and tracked anyway, figuring we'd fix it later.

Well, it wasn't easy. When we found an objectionable note, we searched the tracks until we found the notes we needed (with the proper pitch and inflection but without the buzz) from elsewhere in the song or from alternate takes,

and flew them over to replace the bad notes. There were so many spots with buzzes on that song that the majority of notes had to be edited. Doing that takes forever, and you have to be pretty good with your editing chops if you are going to sneak that past your listeners. Of course, if things are that bad, retracking is usually the better answer, but sometimes you have to do what you have to do.

MORE CLEAN MIXING TECHNIQUES

Here are some other tips that can help at mixdown.

- Listen to each instrument and roll off any unnecessary bottom with a steep high-pass filter. If there is nothing but room rumble and air conditioning noise down below 100–150Hz on a track, filter it out. This one step will make a huge difference in cleaning up a muddy mix, as well as reduce any subsonics that snuck their way into the track.
- Some soft synths have high-resolution or "low aliasing" modes that suck more CPU, but reduce artifacts. Use them! If your computer can't handle the load, freeze the track.
- If an instrument doesn't have much going on in the upper frequencies but noise and hiss, try a low-pass filter or a bit of high frequency shelving EQ to pull back the noise. Remember that most EQs can be automated, so you can open up the response if highs are present.
- Although you can trim individual regions to "remove silence," you can also use mute or fader automation to turn off tracks when they're not needed. In Figure 1 there's noticeable headphone bleed from the vocal track during a sax solo, so I used volume automation to bring the faders down in the spots where there weren't any vocals to mask it. However, don't go too far — some breaths and "humanisms" should be left in the mix, but *control* them and their levels as needed. (You can hear a "before and after" example of the soloed lead vocal in Figure 1's highlighted section at www.eqmag.com).
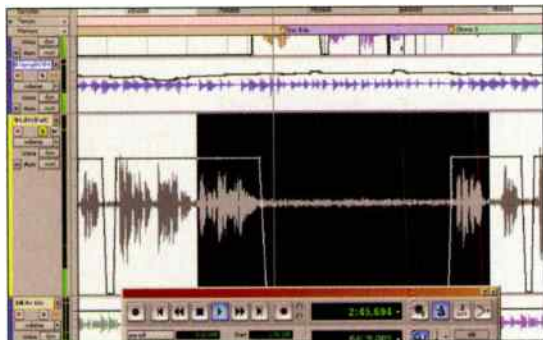
The bottom line is whenever possible, *avoid noise issues at the source rather than try to correct them later.* It usually takes less time and provides better results. But don't go too far over the top and kill the music; a little dirt once in a while isn't going to hurt anything — but a lack of vibe most certainly will. EQ

Phil O'Keefe is a producer/engineer, and the owner of Sound Sanctuary Recording in Riverside, California. He can be contacted at www.philokeefe.com, or via the Studio Trenches forum at www.harmony-central.com.

Guitar Trax
by Michael Molenda
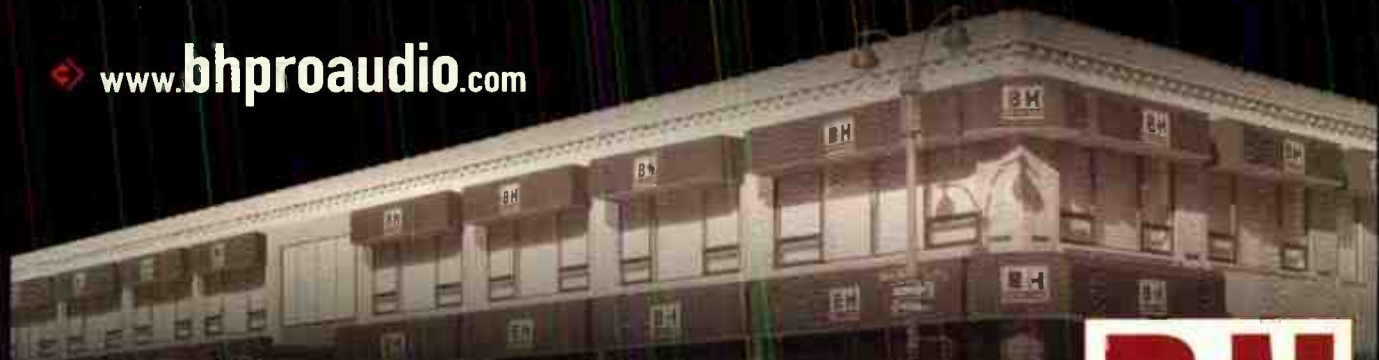
LAUNCH A GUITAR ARMADA!
Basic Layering Techniques for Massive Textures

For some blissfully uncomplicated guitarists, getting a sound is as simple as plugging a guitar directly into an amp, cranking the volume and tone controls, and wreaking havoc on those six strings. I love those guys, and, sometimes, I am that guy. But I'm also a proud member of a vast community of guitar players who employ signal processing and studio magic to conjure all sorts of fabulously bizarre and beautiful sonic textures. After all, it's fun — as well as very inspirational — to think of a guitar as a paintbrush that can dip into myriad colors and spew wonder across a spectral canvas.

It's also commercially smart, as many of today's young and successful guitar-oriented bands utilize density to great effect. Big guitar sounds are all over the charts (well, at least in those instances where guitars are on the charts these days), and the act of crafting textural vistas doesn't have to be limited to players inspired by Evanescence, Avenged Sevenfold, or Snow Patrol. Even if you're a blues or jazz cat, or a funkmeister, or a folkie, savvy applications of guitar layers can add drama and intensity to your recorded works.

Of course, "savvy" is the operative word, and that word may resonate quite differently depending upon the music you play. For example, a metal dude might construct a towering inferno of multi-layered rhythm guitars to drive home a chorus, whereas the same treatment might be a tad scary for an introspective acoustic song about how much your first puppy meant to you. So, although there are no rules commanding truly creative music, taste is still a good quality to bring to your audio-production values.

Ironically, the other main danger zone of layering pertains to the very goals that layering achieves: producing impact, dimension, and density. You see, by overdubbing guitar parts, you can actually diminish impact if you carelessly muddy the layers with parts that are too thick, too processed, or too woofy. Here, then, are three basic layering ideas to help you increase punch without adding paunch.

CHORDAL TSUNAMI
Layering chords to make a chorus sound as ferocious as a starving lion armed with an Uzi is an easy task — as long as you juggle the sonic elements. For a rock track, I'll typically layer four parts for each stereo channel (eight parts total): distorted humbucker tone, overdriven single-coil tone, direct (clean) single-coil tone, "wild card" (humbucker or single-coil through modeling processor, tiny amp, or other bizarre tone generator). However, each side will be populated by different guitars. For example, if my left-channel humbucker tone is a Les Paul, then my right-channel humbucker tone might be a hollowbody. Likewise, if the left-channel single-coil is a Strat, then the right-channel single-coil could be a P-90-equipped guitar. The secret to this recipe is mixing all kinds of disparate flavors together — and those dashes of goodness are not limited to different guitars. You should also mix different

miking techniques, different amps, and different modeling processors or plug-ins.

Finally, EQ the monstrosity to ensure that the low frequencies aren't "weighing down" the barrage (cutting a few dB at around 80Hz or 100Hz usually does the trick). Don't be afraid to try drastic EQ tweaks, either, as a tone that sounds truly disgusting by itself might just add the right zing to the layered sound.

ACOUSTIC ARMIES
An acoustic guitar's ringing harmonics can really tank the clarity and punch of layered textures. Mixing bassy, midrange-y, and sparkling chord inversions is a big help, and some producers even add a "Nashville-tuned" guitar (where the low-E, A, D, and G strings are swapped for a 12-string guitar's high-octave strings and tuned an octave above standard, while the high B and E strings remain unchanged) to the blend. EQ is usually imperative, as you must tame any floppy bass frequencies or low-mid mud. Care must also be taken with miking, because too much room sound may obscure each guitar's attack even further. As with electric layers, I often mix miked acoustics with piezo acoustic-electric guitars and Line 6 Variax Acoustic models.

SMILE LINES
A fun trick that works with either acoustics or electrics is using an EBow to play sustained, single-note lines that follow the song's chord progression. Well, you don't even need an EBow, as you can simply pick or trill the notes, but it's a gas to hear those yowling, bow-like sounds the device simulates. The single notes help clarify layered chord progressions — depending upon how high you use them in the mix. My favorite ploy is to make the lines barely audible, so that listeners can perceive the melodic movement without having the roar of the chordal bombast audibly diminished. Quickie reading comprehension test: Should you double or triple the single-note lines? (Please tell me you answered "yes"!)

LAYER UP!
These are just three ways to add guitar textures to your productions. I also recommend inviting fuzz boxes, delay units, reverb, tremolo, phasing and flanging, and other signal-processing toys to the party. Gather all your options, and like a fine chef blending flavors and spices, stir that stuff around and cook up a marvelous feast of guitar tone! **EQ**

Michael Molenda is a seminal San Francisco punk, multimedia artist, and producer who has recorded tracks for everyone from NASA to Paramount Pictures to various major and minor labels to hundreds of bands you've never heard. He currently co-owns Tiki Town Studios with producer Scott Mathews, and is signed to MI5 Recordings.

The Rock Files
by Lee Flier

OLD SCHOOL TRICKS FOR THE NEW TRADE
Making your DAW go with the flow

Most of us record and mix on computers these days. Many younger engineers have never done it any other way. We all know the benefits of working on a DAW as opposed to the "old days" of dealing with finicky tape decks and consoles. But there are also some often overlooked drawbacks — ways that working on a computer can hinder the creative flow as opposed to working the "old school" way. Fortunately, you can bring a little old-school mojo to your DAW sessions; all it takes is some awareness of how you work.

EARS BEFORE EYES

One of the greatest blessings in working with a DAW is also one of its greatest curses: the visual waveform. It's great for editing, and for getting a bead on any problems with nasty transients or DC offset. But sometimes looking at a waveform is actually a detriment to the recording process. Humans are very visually-oriented creatures — unlike dogs and many other animals, we tend to trust our eyes before our ears. This means it becomes all too easy to judge the way something sounds based on the way it looks
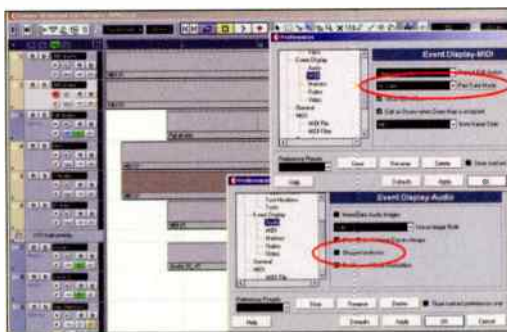
Fig. 1: Cubase SX lets you choose whether to display audio and MIDI data. Go Edit > Preferences; under "Event Display — Audio," uncheck "Show Waveforms" and under "Event Display — MIDI," choose "No Data" for Part Data Mode.

(and this certainly explains the success of some artists on MTV). Working with tape and consoles, we had no choice but to trust our ears because, apart from VU meters and the occasional spectral meter, we couldn't "see" the music.

Many artists, and even producers and engineers, suffer from "waveform overload" and become obsessed with how the music looks — from zeroing in on every little pop or click, to making everything snap to a grid. Therefore, when listening back to a take, try turning off your computer monitor so no one can see what's going on, or disable the waveform display (Figure 1). If there's nothing that sounds bad, it certainly doesn't matter if it looks bad. If everybody's happy with the performance, you can then check the visuals for any obvious technical problems that can't be cleaned up later . . . and if there are none, move on before the lead singer spends the next hour saying, "Whoooaaam, dude, trippy waveform!" and insists that you comp together each syllable from 27 takes.

In mixing, too, the "visual overload" can be an even bigger factor. I vastly prefer using some kind of control surface compared to mixing with a mouse, because again that takes the eyes out of the equation and leaves decisions up to the ears and hands. Mixing by feel, sometimes with your eyes closed, can really make a mix come alive as a "performance" in its own right. Sure, it's not quite like the days before automated consoles, when the whole band would have their hands on the board doing synchronized fader moves and mutes. But using a control surface does help to restore the proper balance of auditory, tactile, and visual.

TIME TO UNWIND

Another difference between recording to tape and recording to digital is that with DAWs, there's no rewind time. This may seem to be an obvious benefit, and for some artists, it is. Others actually welcome rewind time as a means of taking a little break and getting "geared up" for the next take. Again, there's nothing that says you can't do this even if you're not working with tape. I usually just ask the artist whether they prefer instantaneous re-takes or like taking a few seconds in between, and if they don't know, we try it both ways. I won't go so far as to suggest that you use your DAW's jog function to emulate the sound of tape rewinding, as opposed to going straight to a marker — but I will confess to having done this more than once.

COME ON, FEEL THE NOIZE

Very few people have ever been fond of tape hiss. But oddly enough, adding a bit of noise to the environment during a mix session can help you identify what stands out and what doesn't. When you think you have a solid a mix happening, try just adding in some low-level pink noise (or even a "tape hiss plug-in") to the master channel and listening back. It'll give you a new perspective on your mix. Of course, you have to remember to turn the noise off when you're done, unless for some reason you've really, really grown attached to that lo-fi sound.

In short, with a little awareness and ingenuity, it is possible to combine the strengths of old and new school, resulting in sessions that flow musically as well as functionally.

Lee Flier is a guitarist, songwriter, engineer and producer based in Atlanta, Georgia. Her band, What The...?, is a fixture in the Atlanta area, has released two independent CDs, and of late has been performing in other states and countries. She can be contacted via the band's website at www.what-the.com, and also moderates the "Backstage With the Band" forum at www.harmony-central.com.

by Gus Lozada

STUDIOS GO MOBILE

Portable mini-studios — for beginners only, or pro rig for the road?

When the new generation of external audio interfaces appeared, they provided access to recording for a new breed of recording enthusiasts. But how good — really — are those small, feature-packed boxes, and how can they be integrated into an existing studio?

ONCE UPON A TIME . . .

When you needed to record a stereo source on the road, you had to bring a battery-powered DAT recorder (or something like a Nagra) and mics. But that was only about tracking. You then had to go into the studio, transfer the tracks, and continue from there.

Today, it's possible to create a true "mobile studio" that runs entirely within a laptop computer, and can take a project from first take to final master. Even relatively inexpensive laptops allow recording and editing multi-track

Fig 1: A mobile recording studio can be as simple as an audio/MIDI interface keyboard and your computer. A cell phone with Bluetooth and MP3 player helps to transfer files.

Even "big studios" can benefit from having a mobile, "satellite" studio.

audio, adding MIDI tracks with external or virtual synths, FX processing, sync-to-video, mastering, and CD and/or DVD authoring. Some system configurations are even fully bus-powered, so you don't even need to bring a separate power supply: Just add speakers.

A mobile studio is a blessing for performing musicians, composers, and producers willing to be available 24/7 for their customers. One of my selling points as a producer has always been *availability*. They call, I go, get the takes, and/or produce tracks within the customer's location and finish the entire work onsite — or return home (or go to another studio) to mix the session in a controlled environment, "out of the box." All the previous work, such as the takes we recorded, the produced MIDI tracks, and added effects are used in the final production; even "big studios" can benefit from having a mobile, "satellite" studio.

THE COMPONENTS

The interface options vary from a basic two-channel USB interface to a full-blown system with 32 channels of I/O. The audio quality has varied over the years, but the good news is that some of the bigger names have released portable preamps, A/D and D/A converters, and software versions adapted for smaller systems. Consider Apogee's Mini Me or Millennia's TD-1, for instance; these are totally pro devices that deliver awesome results.

FireWire and USB 2.0 interfaces have less latency than USB 1.1 interfaces, because the bus cycle (the time it takes to send a signal in and/or out of the system) for USB 1.1 is far slower. The bus speed also defines the maximum simultaneous track count and recording quality: Four

channels maximum I/O for USB 1.1 @ 16-bit/44.1kHz, compared to up to 32 channels for FireWire 400/800 and USB 2.0 @ 24-bit/48kHz. Another option is to use a PCMCIA interface, such as models made by E-mu and RME. These are high-speed interfaces that don't tax your FireWire or USB bus.

I strongly recommend that you *not* chain audio interfaces to add track count, as a FireWire bus is already packed with those 18 or 32 channels. Adding an extra interface would mess with the bandwidth, and you might not be able to add a FireWire external disk. As a rule of thumb, if you need more than, say, 16 simultaneous I/O recording channels, consider something like a stand-alone hard disk recording system.

PRO TOOLS OR NOT PRO TOOLS?

Myriad software packages are suitable for mobile computing, but many people wonder about a solution that will integrate with an existing, desktop-oriented Pro Tools system. Fortunately, for mobile computing, the M-Powered version of Pro Tools is ideal: You can choose between two different host-based Pro Tools versions, and over 20 different interfaces that are compatible with higher-end |HD systems.

If you are running something other than Pro Tools, you can choose among over 100 interfaces from several manufacturers, and your software/interface of choice. Computer, audio interface, MIDI controller, your choice of built-in preamps or external, more dedicated units and *voilá*. Isn't that great? Get mobile! EQ

Gus Lozada fronts the band WoM [www.wom.com.mx, which recently debuted under the Universal Latino label in the U.S. and under Sony in Latin America), hosts clinics around the world about music production in computers, and moderates "Nuestro Foro," Harmony Central's Spanish-language community. His email is gus@guslozada.com.

World Radio History

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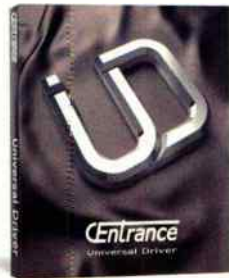
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by Matt Harper

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DAW: Pro Tools |HD3 running version 7.1
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Software: Ableton Live 2, Finale 2003, Logic 7.2, MasterWriter, MOTU Digital Performer 5.0, Pro Tools 7.1, Reason 2.5
Audio Interface: Digidesign 192 I/O, MOTU 828
MIDI Interface: Digidesign MIDI I/O, Emagic Unitor8 and MT4
Monitors: KRK V88, M-Audio BX5, Yamaha NS-10M
Recorders: Pioneer PDR-D5, Sony PCM-R500 DAT, TASCAM DV-RA1000
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Mics: AKG C 2000B, D112, SE-300B, Blue Blueberry, Kiwi, Earth Works QTC1MP, Neumann U67 Ai, 184 matched pair, Røde NT-2, Royer R-122, Shure SM-57, SM-58
Outboard: Antares Microphone Modeler AMM-1, AutoTune ATR-1a, Apogee Big Ben, Behringer UltraPatch Pro, CM Labs Sixty Four Patch Bay, Dangerous Music 2-Bus Summing Box, DigiTech RP100, Focusrite Red-2 EQ, Line 6 PODxt Pro, Monster PRO-2500, PRO-3500, Rosetta 200 (AD/DA) converter, TC Electronic D-Two, M-3000, Universal Audio 6176, LA-2A
Plug-ins: Antares Auto-Tune 4, Spectrasonics Stylus, Atmosphere, Yellow Tools Culture, Synthogy Ivory, Arturia ARP 2600V, Digidesign Reverb One, Focusrite d2/d3, Line 6 Amp Farm, Native Instruments Komplete 3, Sound Toys UltraFX Bundle, Universal Audio Compressor Bundle, Waves Platinum Bundle (TDM)

There once was a time when creating music barefoot was a patented jam band-specific endeavor. But nowadays with the emergence of Boston's own Studio LSE, one of New England's more prolific hobbyist-spaces-turned-real-deal studios, it appears this hippie-esque ethos has been applied to a greater demographic: namely, any artist who walks through the front door, only to be met with a smiling, yet strict "no shoes" policy. The department of health may have originally been made a bit nervous, but Jaw Maran, former Berklee student, internationally recognized musician, and LSE Intl. founder, allays our fears: "We like to keep the studio clean — which keeps your head clean, too — then ideas just flow uncontrollably. We want to create an atmosphere where musicians can comfortably come together, share ideas, and network. We are more than a studio — we are an entertainment community."

And how: Studio LSE is one of four divisions of Living Sound Entertainment, Intl., which also includes production, publishing, and record label branches. Sometimes it seems LSE Intl. is clearly planning a horizontal takeover of the entertainment industry, and their A and B rooms are clearly capable of such an undertaking — with Studio A's live room being large enough to house a small orchestra (which it has), and the key players being adept in the arts of post-production, mobile recording, and mastering at up to 192kHz at 24 bit into a variety of formats from CD to DVD. And with a queue of Grammy-winning producers and songwriters at their immediate disposal, it's no wonder that Studio LSE is quickly becoming one of the most sought-out facilities for rising musicians.

Yet they aren't looking just to turn a quick profit by running a recording studio; quite the contrary, actually. As Maran explains, "Studio LSE and LSE Intl. aren't like other studios and labels. We are only interested in projects that are worth listening to — when a musician has real talent, we do our damnedest to make sure the world hears."
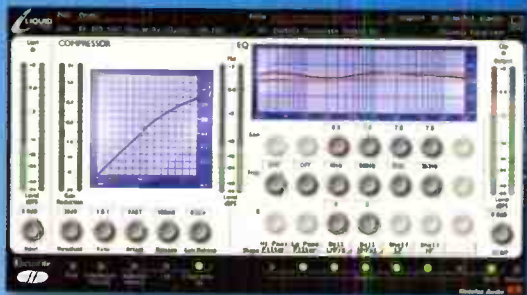
And thank you, kind sirs, for that. EQ

World Radio History