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IN THE STUDIO WITH

Young Love, Jeff Glixman, Mice Parade, Robin Morton, Dennis Ferrer

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FEATURES

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Auralex AudioTile ShockWave photo courtesy of Auralex Acoustics

Talk Box

SPEEDING UP EVOLUTION

I read every article that appears in *EQ*, but have yet to see the following exchange in any interviews with big-name artists or producers:

Question: "So, how did you get that particular sound?"

Answer: "I'm not going to tell you, it's a secret."

In fact, the only time I see "secret" mentioned is when the person being interviewed says "Well, the big secret to that particular sound is . . ." and then proceeds to make sure that it's a secret no longer — if indeed it ever was.

There are many reasons for sharing knowledge. One is because, well, it really *is* better to give than to receive. But even if you're the most selfish person in the world, sharing makes sense: There's something about giving knowledge away that causes more knowledge to come back to you. This seems to be an immutable law of the universe, just like "it only rains if you don't bring an umbrella." Some people call it karma, some people say "you reap what you sow," while others simply have seen too many episodes of *My Name is Earl*.

But there's another slant on this. Many years ago, I was a staff synthesist for a dance company called Group Motion Berlin. One of their founders was into the concept of "speeding up evolution," and it's an idea that's always stuck with me. Basically, he felt that if we learned something, it was our *duty* to pass it along so that people wouldn't have to spend time learning the same lessons we did, and could therefore learn new things that would get us all to the "next level" more quickly.

Any success I've had in this life can invariably be traced back to people who lived the concept of "speeding up evolution," regardless of whether that's what they called it. And there are more such people than the cynical might think. Take the article we ran in the May issue, where a bunch of readers contributed their favorite miking tips. They were all willing to share what they knew for the benefit of others (they certainly weren't doing it for the big bucks!), and I learned a lot from it.

So keep an eye out for any options that present *you* with a chance to "speed up evolution." One good thing about improving the world is that we get to live in the improved version.

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1987 Cakewalk for DOS introduced; 256 tracks

Punch In

Mice Parade

by Richard Thomas

Anyone who's relocated and reassembled their studio space knows the fountain of creativity that flows from having new acoustics and potentially a bigger, better means of tracking their music. For Adam Pierce, the multi-instrumentalist and producer that is Mice Parade, this change came about when he moved his modest bedroom studio from the outskirts of the city to the lush forest of Highland Mills, NY, for the recording of the brand new release *Bem-Vinda Vontade*. The wooden floors and high ceilings were a welcome change from the stifling, 5' 10" clearance of his previous space, but the most exciting addition was a vintage, two-inch, Sony/MCI 24-track tape machine.

"It was on loan from a local engineer friend of mine named Alan Douches," says Pierce. "What I didn't know when he gave it to me was that it was very broken, and getting a broken dinosaur up to speed isn't just a matter of one thing here or there."

Unfortunately Pierce was unable to get the behemoth tape machine repaired in time for the recording of this album, so he did something he'd never done before: tracked into a computer.

To appreciate the gravity of this transition, you have to understand Pierce's previous methodology. Mice Parade albums revolve around a mix of exotic instrumentation, non-Western scales, and odd time signatures. A drummer by trade and a former student of ethnomusicology, Pierce injects a dynamic energy into his songs that is both loose and calculated. To retain an organic feel, Pierce would record most of his parts in one take. Songs were born through a series of mixing board mutes and spontaneous effects.

"I had a half-inch, 16-track Fostex, but three of the tracks were broken," says Pierce of his previous machine. "I would fill up nine or ten tracks, maybe bounce the drums down to two, and keep bouncing down to stereo channels until I filled up the machine. If I felt I needed a few more channels, I would dump the 13 tracks onto ADAT."

Furthermore, extreme pitch-shifting and melody layering — a favorite tactic of Pierce's — was next to impossible to do without tape. On "Swing" and "Circle None," two tracks on the new record that feature multiple guitars, everything had to be done live by hand. "That's a clear example of a technology that the computer can't handle in any way," admits Pierce. "It was the worst part of recording this album."

To help with the record, he enlisted the help of Peter Katis (Interpol, Spoon) on the mix, and Jeremy Backofen as engineer. Regardless of the tracking inconvenience, hallmarks of the Mice Parade sound are still intact: airy keyboard passages, furiously strummed guitars, and fuzzed out drums. The relocation was particularly advantageous for the recording of the latter. On "Tales Of Las Negras," a swinging, Bonham-esque groove provides the backbone, kicking off the track with nothing more than a distant room mic before folding in the rest of the kit.

"I used an Electro-Voice RE-20 for the kick drum and floor tom, a Røde NT1-A for the snare, Shure SM57s for toms, Neumann KM 85s for overheads, and Audio-Technica 4033s along with some old broken toy mics for the room," explains Pierce. "I take the 'toy' mics and usually distort them, then add them into the mix."
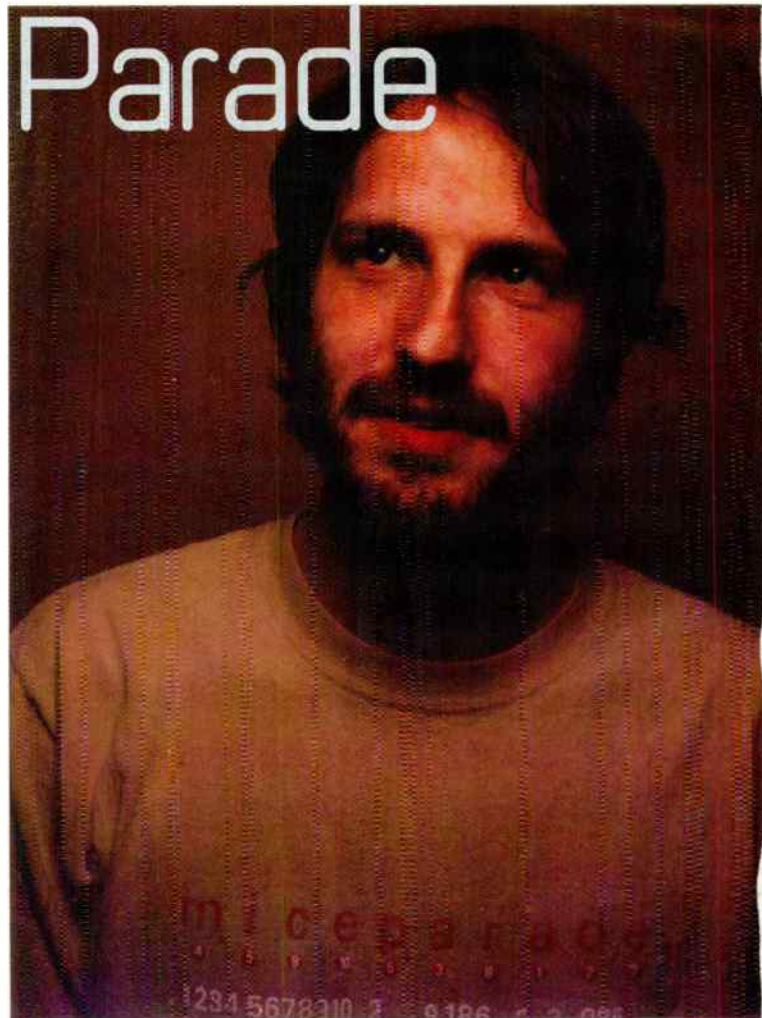
Pierce's guitar tone is achieved with the help of two primary weapons: a handmade Spanish guitar from Ruben Flores and a classic Fender Jaguar. Inspired by flamenco rhythms, Pierce puts the nylon front and center in the mix, using a Teletronix LA-2A to boost the sound without making it too harsh. For Pierce, a little bit of distortion goes a long way. Tracks like "Sneaky Red" and "Snow" get their edge by running through multiple channels of extreme compression and EQ; pedals and effects processors need not apply. Along with a pair of Alesis 3630 compressors, Pierce uses the ever-popular Urei 1176 to add a bit of sparkle and edge, as well as a pair of Avalon VT-737SPs which, as Pierce tells, are used for the mix EQ.

To cap off the signal path, everything passes into a new Soundcraft Ghost 24-channel board, while the monitoring is done though a combination of old Tannoys and a pair of Yamaha NS10s — a step up in terms
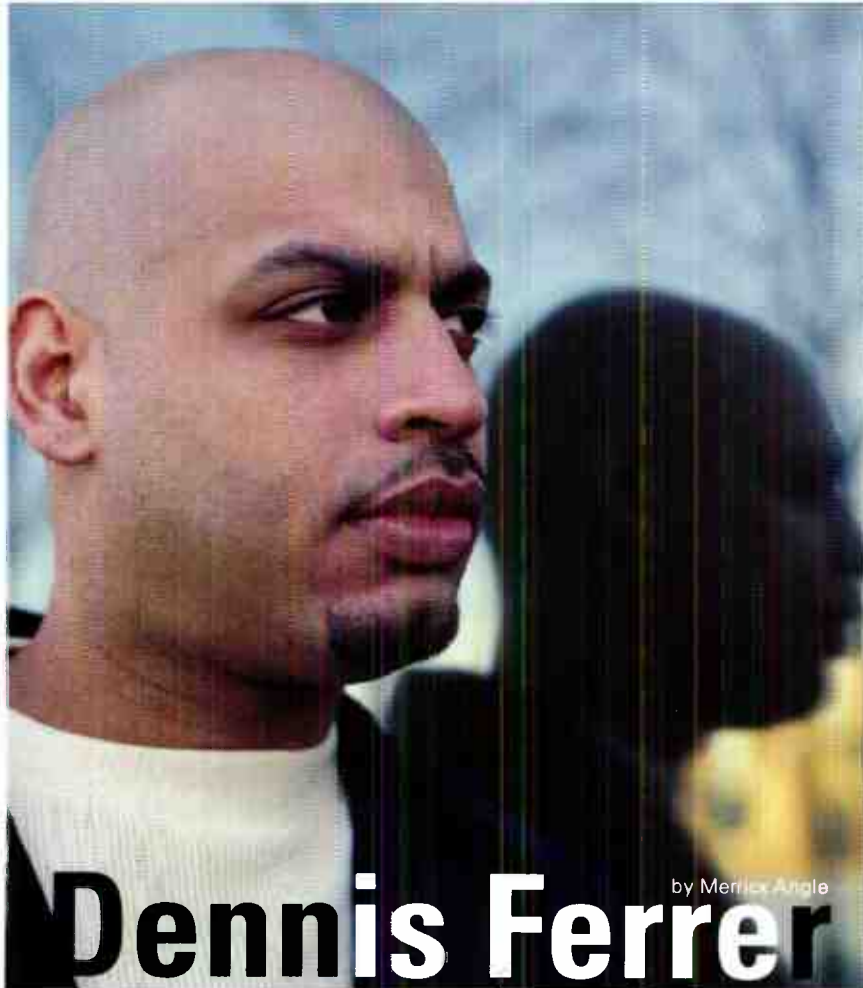
Dennis Ferrer

by Merrick Angle

New York-based deep house DJ/producer Dennis Ferrer — the artist widely known for the production prowess evident in his work on *Sandcastles* and *Underground Is My Home* — has delved further into his long-running dance floor odyssey on the brand new album *The World As I See It*. Intrigued by what we heard, we decided to track down the man behind the mix for a brief chat about building home studios, the pros and cons of going digital, and how to keep ahead of the game when you've already been on the cutting edge for the past 15 years.

EQ: I hear you decided to go the home studio route.

Dennis Ferrer: Commercial studio time is really expensive [in NY], so you have to make do with what you've got. We asked our landlord if we could rent the garage, and he said, "Give me a $150 a month, and do whatever you want."

EQ: Uh-oh!

DF: Next thing you know there's shipments of panels, 2x4s, and wool fibre flying in. We did the whole thing properly: Built a room with in a room, floated the floors, installed double doors — all for a 500 sq ft. studio in a garage. The next time the landlord came over he nearly fell flat on his face!

EQ: With what kind of gear did you outfit the studio?

DF: When I used to work at a music store, I had a lot of gear — every keyboard you could conceivably want. But this time, because of the space, I went mostly digital — working with software and plug-ins, but with really good "front end" equipment.

EQ: Like what?

DF: For mics: a Soundelux E47, Neumann U67, and a Shure SM7B. I love that mic [SM7B]; it sounds great on male vocals. I used it on *Underground Is My Home* because Tyrone Ellis has a real raspy voice. . . .

EQ: And it smoothed everything out? In this genre, a big, warm vocal is key.

DF: Every time we put that mic up, it sounds better than everything else. Anyhow, I went completely digital using a [Apogee] Rosetta 800 at the front end teamed with a Big Ben. Inside, I have every plug-in I could possibly need. I mainly use the UAD-1 for my EQs. The 1176LN and the LA-2A emulations are amazing as well. They are dead on. The only thing that I want right now is something I know won't be emulated, or at least emulated for some time: the [Roland] Juno-106. So I kept that.

EQ: There were no problems going "into the box" other than not having a 106 soft synth?

DF: Not for 3–4 years, when my CPU becomes worthless [*laughs*]. What happens when it doesn't all work with the 25th version of Windows?

EQ: So you're writing primarily in that studio then?

DF: Yeah, but when I'm on the road I'll take the battery-powered [Akai] MPC 500. Everybody laughed at that little thing and said it wasn't for pros, but when I'm on the plane all I need is six AA batteries, and I'm banging away on the next record.

EQ: Would you say your work as a DJ directly influences how you operate in the studio?

DF: If you had asked me that question five years ago, I would have said "no." But they really work together. When you go into a club you watch people's reaction to records, you work out pretty quickly what's going to work, and what isn't. You develop that and then, in the studio, it works like a sixth sense. I think you have to DJ out in order to get a sense of what's really going on. Otherwise you get too caught up in yourself.

of sound quality and clarity from Mice Parade's fifth album, *Obrigado Saudade*, which was mixed on a Mackie 24-8 without the help of the Avalon pres (or any mix compression or EQ for that matter).

Once again Pierce steps into his role as the primary singer for Mice Parade, but a duet with Múm vocalist Kristin Valtysdottir on "Double Dolphins On The Nickel" is a vocal highlight. Pierce says, laughingly: "She needs extreme and heavy compression because she makes no attempt to give you an even level or consistently face the microphone."

As with previous Valtysdottir collaborations, this session wasn't without its idiosyncrasies. "It was nighttime in the fall," remembers Pierce, "and the crickets were so loud outside that she decided to open up the door, pull the mic as far as the cable would reach, and let 'em sing."

It was hardly the optimal way to track in a vocal, but Pierce exemplifies anything but a traditional production style. Says Pierce of the process: "It's about not caring enough to redo things 20 times along the way."

Put Dan Keyes on a stage fronting his band Young Love, or solo behind an acoustic guitar performing songs from their album, *Too Young To Fight It*, and he's fine. Try talking to Keyes without those defenses and you might get some airy responses unrelated to the questions you're asking. Or you might get some jaw cracking yawns. Or you might just get disconnected altogether.

But this is not at all due to flakiness: Working hard is not something Keyes shies away from, especially in the studio. From his Austin, TX, roots in the band Revolver onward, Keyes has recorded numerous "types" of music before deciding to strip everything down to its simplest elements, only to rebuild singer/songwriter-crafted tunes into electronic-laden symphonies of sound.

Yes, on *Too Young*, Keyes decided to start unplugged and track upwards — translating the purely acoustic into the ethereal electronic. "I've been making guitar music since I was a kid," says Keyes. "I wanted to approach things differently. I got super into Daft Punk, Air, M83, Phoenix, weird French music, bands that use a lot of electronics."

Transposing strums into synths (Keyes lists Korg's MicroKorg and Roland's JV-1000 as his two go-to hardware synthesizers), sequencing and sampling on the grid via Pro Tools by way of the Digidesign Digi 001, and then further manipulating these sounds with the ProSoniq Orange Vocoder and SampleTank 2 plug-ins, Keyes and partner Rory Philips reinterpreted material tested live with little more than guitar and voice through a technological screen. "[Keyes] would bring in something on acoustic, but already know the sound he wanted," says Phillips on the subject of translating while tracking Keyes' songs on the pre-*Too Young* demos. "Usually the attack or mood of the parts reflected that

YOUNG LOVE

by Lily Moayeri

sound in some way so it was easy to translate. It wasn't a matter of him bringing in a folk song and me turning it into a dance jam. He was writing dance hits on acoustic. It was his writing apparatus. I was translating them into what he was already hearing in his head."

"Almost any traditional instrument would go through some processing," says Phillips, making mention of the infectious guitar solo on the high energy hit "Discotech" — a signal manipulated by sending an acoustic guitar through a Moog modular. "We both come from rock backgrounds. Doing more straightforward guitar recording wasn't in either of our interests. Lots of filtering, effects, editing helped to keep us from feeling bored and in that rut again."

But after painstakingly electrifying every song, once it came time to record the proper version of *Too Young*, Keyes decided to shift back. "Live, I didn't want it to be standing there singing to a prerecorded track," Keyes says. "The demos were so electronic it would have been really difficult to reproduce without working with a computer on stage. I didn't want to go that route. Live I wanted it to be more like a rock band, so I had to learn to track this like a rock band."

Calling upon Young Love's Robert Mann, Keyes did some quick scratch tracks all on "traditional instruments" — tossing the drum machine aside and recording all the bass parts dry, and putting the Moogerfoogers in the closet before head-

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Punch In

ing off to Moles Studio in Bath, UK, to work with producers/engineers Steve Osborne and Damian Taylor.

A duo matched for Keyes' multiple musical personality disorder, Taylor matched the old-school training Osborne underwent while recording some of the earliest rave and dance music known to man with his new approaches as a cutting-edge programmer. Says Taylor: "Most often we'd work against [Keyes'] vocals and any strong character parts from the original recordings, like a melody line or a riff. The songs were in pretty good shape as demos. [Osborne] would react to what he heard, follow his gut, and start trying stuff. He'd often do a lot of heavy treatment through the SSL 4056 E, severely EQing and compressing some channels while gating others and then running them back out through synths like his [EMS] VCS3 or my Korg MS-20."

Sounds a bit reckless? Over every song Keyes' vocals go through various echoes, distortions, and what sounds like harmonizing with himself along with tracks taken from the original demos with Phillips, which were cut with a Sony C37P into an API 512 Discrete Mic/Line Pre (with limiting courtesy of Digidesign's Bomb Factory Fairchild 660 plug-in *before* the mix).

Enhancing the live demo tracks soon became the name of the game, regardless of whether they were effected or not, with Osborne and Taylor bringing out a Minimoog ("for the bottom end"), a Roland Jupiter-8 ("chords, pads, arpeggios"), a Doepfer Modular, and Taylor's prized MS-20 — whose external trigger saw a good share of the action in firing rhythms.

This is most noticeable on "Discotech" where the washy pads in the chorus were taken by Osborne, played through the Roland Jupiter-8 and Doepfer Modular, then added together with the Orange Vocoder strings on top before being bused together and compressed through a sidechain input. According to Phillips, "By sending a click on the downbeat to the sidechain it created an upbeat swell effect that really helped tighten things up but still had the synth-y wash we wanted."

For "Tell Me," which started as an R&B tune centered around a piano, beat, and vocal, things progressed a bit differently. "We tried recording some drums and an electric guitar playing the chords but it was pretty difficult to find the extra direction for the track," says Taylor. "I grabbed a few of the guitar and drums parts [Osborne] had liked, gave them a little tweak on timing and placement, then pulled the original elements of the recording into Ableton Live. I started by replacing the bass end of the piano with some synths and a couple of samples so that I could turn off the piano and open up the possibilities for the track, while still keeping its harmonic structure intact. Then I pretty much threw the kitchen sink at that track, taking the approach that I'd give [Osborne] a huge library of sounds to work with.

"When we all got back together everything moved incredibly fast. [Osborne] did a bit of a rebalance of the live drums and the guitars, then dived into everything that I had lined up for him. He very quickly programmed up the analog drum machine type part you hear on the verses, then he just instinctively auditioned and chose the different parts I had given him. It was one of those classic cases of finding the right backspin for a track, giving it that little twist to set it apart. [Osborne] was able to really enhance the mood of the track, give it the right dynamics,

through his selection of all the elements everyone had contributed, from [Keyes'] vocal and the original chords he did with [Phillips], to [Mann's] guitars and drums, my weird bits, and his extra beats."

Keyes credits the duo with really helping him flesh out the album creatively. As he reflects: "The demos sounded like cheap Daft Punk with vocals, but with not as much money behind them. You need people to help fill in all the stuff that I would not know how to do personally. One day maybe I'll know how to program and engineer and work with all those crazy boxes. Right now, [I'm] trying to focus on the vision."

MOSES EXPOSES:
How the Ashanti royalty dispute may impact every producer/artist relationship
by Moses Avalon

It started out like any other spec deal in 1997. Genard Parker, an emerging producer, completed demos for an unknown 16-year-old artist, Ashanti Douglas, through his small Atlanta-based production company. He paid for them as a favor to her mother, Tina, whom he was dating. Men may do all kinds of things to impress women, but they also know that business is business. He had Ashanti sign a contract with Mom as guardian. Ten years later, Parker was suing the multi-platinum "hip-hop princess" for $2.6 million. I was his key expert witness.

While most of us are not working with stars of this level every day, this situation created case law that will impact producers and artists at all levels in odd, annoying ways. This is especially true when one considers that sloppily drafted spec deals get signed everyday of the week.

A few months after Parker completed his demos, a Sony label, Noontime, wanted to sign Ashanti. Tina asked Parker to release her daughter from their deal so she could trade up to Noontime, which had name-brand producers and a hit-making track record. She offered Parker a "go-away" package that included royalties on Ashanti's future "Noontime recordings." Parker agreed and Ashanti took a six-figure advance from Noontime. But shortly thereafter, she bailed on them too. This time her trade-up would make history.

Irv Gotti of Murder Inc. (who was not yet the star producer he was to become) got her a deal on Universal Music where she went on to sell six-plus million units and win a Grammy. She was all of 19. Yet, despite the millions paid to her, Parker never got a dime.

He eventually filed a lawsuit claiming $600,000 in back royalties. To pay his lawyers' retainer, Parker sold his Ashanti demos to a foreign label for a few grand; the ones featuring her performance without the benefit of, well . . . more experience in the studio, we'll say. Ashanti filed a countersuit for releasing the old demos. It was war.

Now, it's probable that Ashanti and Tina believed that they didn't owe Parker. After all, how many times have rising artists left former producers in the dust? Plenty of times. Plus, in this case, the

by Jeff Anderson

Debate: To Rent or To Own

A former employer once asked me, "Why on earth would you want your own toys when you can play with someone else's for free?" Before following up with the sternest warning I've ever encountered: "Make sure you don't get G.A.S."

No, he wasn't talking about collecting G.I. Joes or eating Mexican food; he was referring to owning a studio instead of renting one for your clients and commenting on the dangers of catching G.A.S. (Gear Acquisition Syndrome) if you decide to go with the former plan of action. And even though my goal at that point in time was to build the hugest studio possible for myself — and his claim seemed like little more than seasoned-pro complaining — I look back now, as an engineer with a huge overhead, and realize that he just may have been speaking words of wisdom all along. I mean, it's not as if owning a studio is necessary to work in the world of recording, and these days it can be quite a dangerous investment.

That said, there are some pretty obvious advantages to studio ownership. First, your potential profit is hypothetically greater; you're not just charging for your services, but for the space utilized. Second, your potential workload is certainly higher, as you don't have to bum around other people's facilities day in and out as a freelancer. Third, having your own studio available at your every whim is just about the coolest thing in the world, and worth paying for out of the pocket.

But on that note, it's a lot less rock and roll running a business than it is just showing up for sessions and hanging with artists — bookkeeping, financial planning, and marketing (should) have little to do with recording music. Plus, the sad fact of the matter is that recording studios are far from the first choice of investment for any entrepreneur with a half a brain (hint: That should be, could be, you). Owning a studio under the pretense of turning even a moderate profit is getting harder by the day, and your passion can soon turn into a pain when you're forced to allow the local Dave Matthews cover band to book your spot for $50 an hour (including engineer) just to make ends meet.

I've seen the scenario play out plenty of times in the past: A talented engineer establishes a business around his/her reputation — opening a studio and hiring on employees simply to keep up with the demand. Unfortunately for this guy/gal, the more successful they become the less likely it is that they are actually going to be spending their time doing what they love: recording! Before long, they are overseeing studio maintenance, session schedules, and accepting or denying payroll advance requests. Trust me, it's depressing.

This is where "playing with other peoples toys" starts to sound real good. I mean, as long as you don't break anything you are largely without liabilities, and you're free to focus solely on recordings without "business pressures." Furthermore, working in various studios is a great way to be a life-learner, and thus a versatile engineer. If you get hired by an already established studio, that's all the better as well: You won't be out trudging the streets for work. And if you're good enough, you won't even need your own place — your clients will follow you . . . and doors will open for you everywhere, with hungry studio owners salivating for your business.

MOSES EXPOSES:

contract specified renumeration only from "Noontime recordings." None were released. This seemed like a non-starter. No matter, Parker's lawyers had a brilliant, and never-before tried legal theory: If Noontime dropped Ashanti — so be it. No money for Parker. But, if she "quit" before they had a chance to make her a star, then quitting should not mitigate her obligations to Parker. Otherwise, every artist could weasel out of paying their original producer — who contributes the most risk capital — simply by quitting their next deal.

Many artists leapfrog producers as they climb the ladder, often leaving a wake of fledglings with little more than "I knew her when" bragging rights. Did Ashanti owe Parker even though his recordings were only demos that never saw the inside of a record store and were never heard by the executives who signed her?

To everyone's surprise, the jury said "yes," and based on Ashanti's royalty statements awarded Parker $630,000. Champagne time, right? No. Parker would end up seeing only a fraction of the award because another sloppy but common practice in the music business reared its ugly head.

The judge reduced the award to $50,000 — Parker's out of pocket expenses. He said Parker failed to show that the theoretical Noontime/Sony album would have achieved the same results as the albums actually released through Murder Inc./Universal. To get the other $600,000, Parker would have to prove that had Ashanti stayed with Noontime/Sony she would have sold at least the same number of albums.

Here's where I came in. I looked at sales reports from several of the top female R&B singers on both Sony and Universal. What I discovered surprised everyone. Artists of like type on Sony sold an astounding 270% more albums during those years. Parker therefore could be entitled to over $1.7 million in damages by this logic. That would be my testimony. Sadly, the jury would never learn any of this.

Minutes before I was to take the stand, Ashanti's lawyers dropped a bombshell: It turned out Parker, like many other small to mid-sized producers, had never properly incorporated. Parker's entire award could be tossed on this technicality.

For many home-grown production "companies," the expenses are so steep that most don't file for a business license but rather opt for just a DBA for about $100, which provides no real layer of corporate protection. Aside from not getting the $600,000, without the proper license Parker would be personally liable if he lost Ashanti's countersuit. In light of this, Parker settled for a number far south of what a verdict could have brought him.

Seriously, as producers and artists, can't we all just get a song?

But being a producer/gypsy can get hard on the soul, especially when you need "just one more hour" and you're being denied. When you're working for someone else's business or renting a space, you are at the mercy of the studio's management, and there's nothing quite like autonomy on a hot summer day. I can't even count how many times, before I owned a studio, where I just needed a bit more time to finish my mix and was forced to tear everything down — only to spend four hours restoring it — due to the next booking beating on the doors.

But not all of us are in this to be businessmen; some of us just want to have a place to play, and we've worked hard to allow ourselves that luxury. So why shouldn't we indulge, even if it may cost us? And who is to say that you can't open a studio tomorrow and live off owning it these days? I see it all the time. As with all things, it depends not so much of what you do, but how you do it.

Nonetheless, it sure would be nice to go work a session at the Hit Factory right now. . . .

SESSION FILE: THE UNDERDOGS
Making old sounds new and new sounds old

by Dan Daley

Motown met Midtown on the silver screen late last year when *Dreamgirls*, the hit stage musical, migrated to the movies — where it replicated its Broadway success. While the domains of pop music and film have increasingly cross-pollinated over the years, the creation of their sound tends to reside in very different camps. But *Dreamgirls* was meant to have a pop soundtrack that could stand on its own, and director Bill Condon called on the Underdogs (Harvey Mason, Jr., and Damon Thomas) — who have crafted hit records for artists including Mariah Carey, Britney Spears, Whitney Houston, Brandy, Jennifer Lopez, and Justin Timberlake — to give that record an edge.

The Underdogs took the project on as a challenge as much as a gig: Create 61 individual pieces of music, of which 44 would be full-length songs, giving each some element of the period it needed to represent (the show

and Ricky Lawson. "Every musician has his own sound, something you're not going to get out of a box. We purposely didn't use drum machines on this project except as part of the demo process. Instead we picked and chose the players to create bands that reflected the periods of the music."

LABWORK

Working at UnderLabs, their eight-room factory in Los Angeles, and in the main room with its 96kHz SSL C200 console and a Pro Tools HD system, Mason and Thomas would use their bare-bones demos, recorded in Logic, as the map for the finished song. "We had timecode running on both systems [synced using a Lynx], but one of the nice things about using live musicians is that the time and feel will move a little here and there, which really breathes life into the track," says Mason.

Mason and Thomas immersed themselves in hits of the periods the film covers, a form of research they say was critical to getting it right on the soundtrack. A key result of this is identifying and incorporating an instrument or technique that encapsulates the era, which gives the listener a subliminal audio cue as to what period in time the action on the screen is taking place. For example, to evoke the 1980s, they used a set of SynDrums; they listened to Donna Summer records to find the essence of late '70s disco era, which could be suggested by something as simple as adding a cowbell to the track, or asking the singer to deliver the lines with more of a breathy sound.

On the engineering level, UnderLabs staff engineer Aaron Renner recalls some of the other tricks. "The technique we used for miking the grand piano was not too out-of-the-box but had great results: We set up AKG 460Bs approximately 9" to 15" in from the sides, 7" above and 3" to 4" behind the hammers. We also placed a Sony C800G at the bell of the piano, at about the same height, around 8" to 9" in from the edge depending on phase considerations, positioned between the 460s and angled depending on whether a more ambient or direct signal was preferred. I use either Avalon M5s on the pair of 460s and a Neve 1073 on the Sony, or a combination of 1073s and Neve 108s on all three microphones."

Aaron Renner running the SSL C200 at UnderLabs during the Dreamgirls sessions.

follows the career of a girl vocal group from the 1960s to the 1980s) yet still keeping fresh enough for drive time on Hot 102.5 tomorrow.

The project would eventually stretch out over 15 months, and the sessions were commensurately huge. One song in particular, "Stepping to the Bad Side," ate up over 20GB and 128 tracks. However, the first order of business was not terribly technical at all, but very soulful. "We picked the right musicians," says Mason, including his father, one of jazz's most famous drummers, Harvey Mason, Sr., as well as players like Ed Campbell

IT IS THE WAY IT WAS

Getting the vintage "live band feel" called for classic gear and classic cats. "Harvey Mason, Sr., got a few

guys together for a jazz instrumental called 'Big' and they killed it," Renner recalls. "Contrary to the way we approached the other sessions, we tracked piano, upright bass, drums, and sax live with no overdubs. Piano, bass, and drums were set up in the main live room, and the sax was isolated in the sound lock. All the instruments except the sax were baffled in but we retained eye contact among the musicians. We used a Neumann U87 and a DI on the bass. On the drums I used the beyerdynamic M88 on the kick placed nearly outside the drum, with no secondary mic. On the snare I used a Shure 57 top and bottom, as well as an AKG 451 on the snare top, to capture a brighter, more present top end when the drummer was using brushes.

"I used almost exclusively Neve 1073, 1072, 1081, and 31102 preamps, except for the sax; for that I used an sE Z5600 microphone and an Avalon M5 preamp to an Avalon 2044 compressor with the lowest possible ratio setting and a maximum threshold setting. This allowed for additional output gain with a low input gain setting on the M5, to prevent clipping on input — which is not pretty on an M5! This Avalon chain is very clear and transparent, good for brass. While we normally take the time to clean up our drum tracks, we left this session very much as the players performed it, and I really tried to keep true to the performance during the mix, with virtually no dynamic processing, and very little as far as automation."

The fact that the Underdogs tried to adhere to non-synthesized instruments meant that their engineers had to deal with some acoustic instruments not found on most sessions these days. "On [Beyoncé's] 'Listen' and 'And I'm Telling You' [sung by Jennifer Hudson] we recorded a harp," he recalls. "The approach was to treat it like a piano standing on its belly, as a harp is basically like the frame of a piano. We used a pair of 460s or SE3s and the Sony [C800] once again. We placed all three mics on the same side of the harp — the Sony on the centerline at the bottom of the harp and the 460s or SE3s near the edges above the player's hands, in an approximate 'V' shape, aimed slightly in toward the strings obviously with adjustments for phase. It sounded great."

VOCALS: THE BIG PICTURE
In an age of widely used samples and presets, vocals are perhaps the signature of any producer. Mason and Thomas are ardent fans of the Sony C800G through an Avalon M5 preamp, with a touch of compression and EQ from the console or outboard Neve unit. After that, it's all about the performance.

"You have to know the person you're working with, know about their history, who they are, what their range is," says Mason. "You have to get a sense of how far you can push them. And they have to know they can trust you. They can't perform if they don't."

But it's not all vibe; the Underdogs' engineers discovered a cool vocal trick to get a Jackson 5 sound. Using the pitch-shifting capability of the Pro Tools system on the song "Perfect World," they recorded the male lead vocal to a track pitched two steps down, pitched the
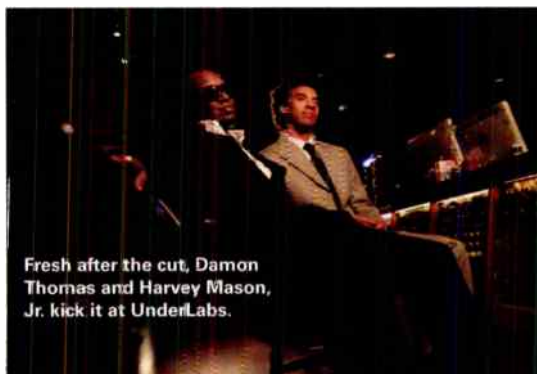
Fresh after the cut, Damon Thomas and Harvey Mason, Jr. kick it at UnderLabs.

vocal back up and then flew it into the track at the original pitch. "They couldn't start principal shooting until the songs were recorded, but as the production progressed a character's mood might turn out to be considerably different than it was imagined at the time of the session," Mason explains. "So we would use pitch change to bring the song into a key closer to what the scene called for."

I REALLY TRIED TO KEEP TRUE TO THE PERFORMANCE DURING THE MIX, WITH VIRTUALLY NO DYNAMIC PROCESSING, AND VERY LITTLE AUTOMATION.

"Then there's how things sit in the mix," says Thomas. "The vintage vocals tend to sit back in the mix a little bit more than today." Historical reality, though, collides with technical necessity at points like this; the movie is first and foremost a musical and thus needs the lyrics to convey and support the narrative. Again, this is not a challenge that's solved with a preset on a plug-in. "The way to deal with that is, the director told us to mix it as through we were mixing any other record, then to go through it again listening to the lyrics and to pull the vocal up if needed to get the story across," Mason explains.

While the Underdogs created stereo mixes that would serve as both the soundtrack album and as references for the film's own sound mix, they also delivered "stem" mixes — major elements such as drums, guitars, keyboards, and background vocals mixed to their own stereo pairs. This is the standard configuration that film music is delivered in for 5.1 surround layback to picture, but is also the way music mixers will deliver "music-minus" mixes for artists using backing tracks in live performances. This allows the film's music mixers to adjust relative levels to fit the on-screen action.

"Dreamgirls was a real challenge, but a dream project for a producer," says Mason. "You rarely get to cover so many genres with so much great talent for one project." EQ

SUCCESS STORY: ROBIN MORTON

Fiddling around at Temple Studios

by Jeff Touzeau

Just south of Edinburgh, Scotland (in what's affectionately referred to as "The Old Country"), one's ears need only follow a few bagpipes before being led to Robin Morton engineering and producing Scottish folk-oriented albums for his Temple Records label. Best known for his work with the Battlefield Band (a traditional Scottish ensemble that has been touring internationally and making records since the early '70s), Morton's latest project has been producing the four-piece ensemble's latest offering *The Road of Tears* — a powerful new release chock full of exquisite musicianship that serves to tell emotionally-charged stories of forced emigration, of families being unwillingly displaced all over the world.

Recorded in a converted church that was originally built in 1832 (and now houses Temple Studios facility, office, and residence housing), *The Road of Tears* was brought to life not at the hands of an overly technologically savvy engineer (Morton humbly states that he is anything but a gearhead) but rather is the result of an incredibly well-rehearsed band recording in their element. Of course, with all the years and production credits under Morton's belt, he's quite the apt pupil of the recording arts — having learned through trial and error which ingredients are essential to making great bands a recorded reality. Still, he credits the success of *The Road of Tears* sessions largely to the experienced musicians behind the collective . . . and his refusal to work with anything less. "I want [my artists] to come in the studio knowing what they're going to do," he says, without a hint of apology in his voice. "If they don't know what they're going to do, then they need to rehearse it and rehearse it and renearse it until they *do* know what they are going to do."

ALL IN THE ROOM

Thankfully for Morton, he was working with the very best from day one. From the first spin of *The Road of Tears*, it becomes evident that each respective player (Sean O'Donnell, vocals, guitar; Alasdair White, fiddle; Mike Katz, highland pipes, small pipes, various whistles and bass; Alan Reid, keyboards and vocals) is virtuosic in his performance. But this naturally creates barriers in the studio, Morton claims, stating one of the biggest challenges he experiences in dealing with very accomplished musicians is helping them recognize the importance of performing as a single unit: "They need to get past that stage of listening to themselves so they're listening to everybody else. That is the nearest thing you can get to a live situation before you can begin to sweeten anything up."

This was especially important when working with the Battlefield Band — a group known for their electrifying live performances; performances of which Morton strives to capture the essence in each recording. But though it has taken some time to figure out the proper environment, both externally and internally, for the band to transfer their live energy into a recorded product, Morton feels he has achieved a nice physical balance by utilizing the Temple Studio space to situate the players appropriately, thus maximizing results. "Just beyond the front door, we have a room that is about 35 feet long by 15 feet wide, ceilings about 14 feet high," he says when asked to describe the room he used to track the band. "The room has got a live end and a dead end, and I tend to put the guitar and fiddle on the live end. I'll situate the fiddle player, the guitarist, and the keyboard player in kind of a triangle, and I have the bagpipes out in a hallway, which is between the studio and the control room." This is done, Morton says, because working in the hallway allows for proper isolation of the bagpipes yet the proximity to the live room lets the members maintain visual contact — a primary concern when tracking a band live. "You couldn't do it with everybody, but I can certainly do it with Mike and Alistair. They know each other so well, and of course they are hearing each other on the foldback system. The hallway is really small and it gets bloody loud in there, but it works."

PIPE DOWN

Having recorded folk music in his native country for over three decades, Morton has learned a thing or two about how to capture bagpipes and other instruments used in traditional Scottish compositions — and how to marry those elements with more non-traditional components. "For the Battlefield Band, I'm recording a fiddle, highland bagpipes, small pipes, various whistles, accordion, acoustic guitar . . . and electronic keyboards," he relates. But in the case of the Battlefield Band, these rather disparate instruments work incredibly well in a traditional Scottish arrangement, leading to a mood-evoking and powerful experience that transcends culture and national heritage. And while the nature of the music may strike many listeners as a bit "foreign," and the instruments applied as "exotic," Morton's preferred tools should strike any engineer as familiar. For instance, when it comes to recording bagpipes, he uses the traditional, standby Shure SM57: "I mic it right on the chanter with the 57, because it's kind of forgiving. For the drones, as overheads, I'll often do a cross pair of [Beyerdynamic] 201s, maybe a foot above the drones. Everyone says that bagpipes are so hard to record — but if you listen carefully to them, they are very loud but very level."

Morton holds that it is very important to have the mic on the chanter fairly close at all times, otherwise the result is an unwanted bleed into the overheads, which reduces his ability to control sounds individually at the mixing stage. But when overdubbing he says there is flexibility in proximity, and he sometimes moves back away from the

instrument to add an extra dimension: "If I'm doing the drones alone as an overdub to lay against a chanter that's already been recorded, I'll give them a couple of feet to get a bit of air in there."

STRINGS 'N THINGS

Another instrument that is featured prominently on *The Road of Tears*, and in traditional Scottish music as a whole, is the fiddle. As with many acoustic instruments being recorded in a "live" scenario, Morton says the fiddle can prove to be tricky to record unless you have just the right mic, with the right response, on the right spot: "I like the Audio-Technica 4033 mics to record fiddles. They have a nice edge to them, with a bit of brightness at the top end. I'll put the 4033 on the fiddle mostly toward the bridge, toward the bass strings. Once again, I'm right on it really, because you've got another guy six feet away playing the guitar."

Morton also uses the 4033 for tracking the acoustic guitar, though he confesses to being the type to switch out mics just for the sake of doing so — oftentimes substituting a Neumann U87 for the 4033. "Sean [O'Donnell] has a light, very percussive guitar style so I like to get more bass than he's giving me. Therefore I'll tend to put the mic up behind the sound hole, but close to the bridge and on the bass end. I'll use a DI as well [for the electric-acoustic] and blend the signals in the mix."

Lastly, there is the small harp (or Scottish harp), which Morton prefers to record with a distinctly non-classical approach. "I'm not keen on making this instrument sound too ambient, and I rather like the finger noise," he states, informing us that he uses a pair of Neumann KM 184s in an XY pattern, fairly close to the soundboard of the instrument, coupled with a U87 towards the bass strings — a tactic he says gives the recording more presence and makes it sound less clunky. "I want this instrument right in your face, and to sound as it sounds to the musician playing it."

WIND DON'T BLOW

While most of the Battlefield Band's tracks contain either a bagpipe or a whistle at the forefront, both of which go along naturally with the tones achieved from the fiddle, guitar and synths (which, of course, are recorded direct), Morton says that the difficult-to-record whistle oftentimes serves as the bane of his engineering existence. He elaborates: "If it's a breathy sound, I tend to go back to the 57 because it's, once again, very forgiving. Whistles sound so different, even ones from the same company. Maybe there will be overtones, buzzings, or other unwanted sounds coming from the mouthpiece. The 57, as it's a dynamic, tends to keep this stuff out. So, again, I mic fairly close, positioning the 57 towards the center of the sixth note."

Border pipes — another wind instrument that can prove to be difficult to track — also make occasional guest appearances throughout *The Road of Tears*. Similar to the

Robin Morton studiously approaching the mix for the Battlefield Band's *The Road of Tears*.

Irish-oriented Uillean pipes in that they are elbow blown, Morton says the Border pipes sound sweeter than their larger cousin: "Border pipes are not as brash and harsh as the bagpipes, and certainly not as loud, so I tend to use ribbons to capture their gentle sound."

THE SIMPLE LIFE

Morton adheres to the "natural = best" rule when making albums, going so far as to forego the use of VU meters or other sound measurement apparatuses in favor of his ears, which he says are his ultimate gauge. Preserving the immediacy of the performance is of penultimate importance to Morton (which would explain why all of the vocals on *The Road of Tears*, sung by keyboardist Alan Reid, were cut live in the same room using just a Rode Classic II), and this mindset is amply demonstrated in his overall gear selection, which is, to say the least, largely utilitarian.

"If you're not recording all the time, which I'm not, my attitude is to keep it simple because otherwise you have to re-learn your setup every time," he says. He records into a Mackie HDR 24 through a TASCAM M600 console, which has served him well through the years: "It's a great old budget desk that is very quiet. It's only got 32 inputs and 16 out, but people generally like the sound of it." Furthermore, Morton employs as little processing as possible, though he does rely on an Aphex Compeller from time to time. "It is an intelligent compressor and it's wonderful; you can actually level things with it. It's especially good for controlling peaks on a fiddle or whistle."

It's an old adage, but it holds true: Despite all the gadgets and devices available in this day and age, technology is mostly just a means to an end, and an engineer needs to listen first and foremost. And Morton cautions against those who save the magic for the mix: "It is really about the live performance — there needs to be some kind of tension there. I've learned time and time again that things get very flat without that tension." EQ

Jeff Touzeau is a regular contributor to EQ. His latest book, Making Tracks: Unique Recording Studio Environments *is available at www.studioenvironments.com.*

TECH BENCH
Cable Construction for the Terminally Lazy

by Craig Anderton

I make a lot of my own cables: I look at the price of manufactured cables, realize how much I'd save by buying the parts separately, warm up the soldering iron, and get to work. But afterward, given the amount of time it takes, I often wonder if maybe I shouldn't have just bought the darn things . . . although I console myself with knowing I got *exactly* what I wanted, at the *exact* right length.

Recently, though, I found out about an almost hassle-free alternative for those who want to make their own cables. While more expensive than doing it completely from scratch, you'll still spend less than you would on high-quality commercial cables; and again, you get exactly what you want (as long as it's phone or phono jacks — there's currently no XLR option).

MEET THE CABLE STATION
The Cable Station is a DIY-oriented product from Planet Waves (www.planetwaves.com). The basic concept is they make cables and connectors designed for custom cable construction. Connector options include 1/4" plugs (standard or right angle types, with or without a "Circuit Breaker" switch, as well as standard thin profile types), and RCA plugs. The Circuit Breaker plugs (Figure 1) are particularly useful; pressing the recessed switch shorts out the plug, so you can plug or unplug without pops or other buzzes.

Cable types (Figure 2) include instrument cable at a street price of about $1 a foot, RCA cable at about 70 cents a foot, and dual RCA cables (yes, you can make those too) at about $1.30 a foot. Plug street prices range from $8 for the thinline 1/4" types to $11 for the right-angle plug with Circuit Breaker switch.

Fig. 1: Note the recessed "Circuit Breaker" switch toward the left side of this 1/4" plug.

Fig. 2: The Cable Station cables have printed markers every foot, making it easy to cut a cable to the desired length.

Fig. 3: Slide the cable into the cable cutter, then close the "jaw" for a clean cut. The blade is recessed, so it's virtually impossible to cut yourself accidentally.

Fig. 4: A right angle plug connected to the cable.

"Packages" are available too; for example, there's an RCA single cable kit with 12 plugs and 12 feet of cable, as well as a "pedalboard" kit with 10' of cable, 10 right angle plugs, and a mini cable cutter.

Bearing in mind that these are quality components (24K gold-plated plugs, 24 gauge wire with oxygen-free, double-insulated braided copper shield and metalized polyester foil inner shield), a comparable Planet Waves 20' 1/4" mono instrument cable would run $55. Do it yourself, and with thinline plugs, you're looking at around $37. Over 10 cables, the price difference adds up.

MAKING THE CABLE
In addition to needing cable and connectors, you'll also need a cable cutter to cut the cable to the desired length. To avoid distorting the cable's round shape, you can get either a mini cable cutter (Figure 3) for $6 or a combination cable cutter/cable tester for $26. Bulk cable comes with a mini cutter.

The process of making a cable is really easy. You loosen the plug's setscrew (in Figure 1, note the setscrew in the upper left corner), then slide the cable into the plug shaft as far as possible, until it stops. Finally, you tighten the setscrew until it won't tighter any further . . . no solder required, and you're done (Figure 4).

THE TUG TEST
I was surprised at how well the connector stood up to repeated tugging. I really, really tried to pull the cable out of the connector, but couldn't. I can't imagine a situation where pulling on the cable would break the cable before it pulled out of whatever it was plugged into.

This is good stuff; I've never found it so easy to make quality cables, while saving money in the process. If you're a DIYer at heart but find that time is at a premium, this is a *great* solution. EQ

World Radio History

Digidesign Studio Reference Monitors

The **Reference Monitor Series** (RMS), developed jointly with PMC, include the **RM1** (5.5" two-way; $1,249) and the **RM2** (6.7" two-way; $1,749). Both monitors incorporate a 48-bit, fixed-point DSP engine that manages tasks including the monitors' complex frequency crossover, speaker positioning EQ settings, trim level, and bass port emulation. www.digidesign.com

ASK Video DAW Keyboard Overlays

Editors Keys Keyboard Key Sets (currently available for Pro Tools, Cubase, Nuendo, Logic, Reason and Sonar; $17.99) transform a standard keyboard into an audio editing keyboard. Each Key applies easily to your standard keyboard, and contains the program shortcut icon, key letter, and text reference. www.askvideo.com

WK Audio EDIT DAW Controller

Designed for Steinberg's Nuendo Media Production System, **EDIT** — derived from the ID Controller — shares an almost identical layout and construction but its individual function areas are optimized for editing and working with tracks, rather than being a complete mixing desk. www.wk-audio.de

Disc Makers Medley DVD/CD Disc Publisher

The **Medley** DVD/CD publisher ($1,799) is an automated disc printing and burning system that connects to Windows PCs via USB 2.0. Graphics software allows creating full-color images directly onto the disc face. The Medley also integrates with iTunes and Windows Media Player to automate audio disc ripping. www.discmakers.com

Berklee Launches Internet Radio Station

The **BIRN** (Berklee Internet Radio Network) is a credit-earning course, complete with DJs, producers, engineers, and promoters who are untethered from corporate programming or institutional playlists. www.thebirn.com

Zaxcom Deva 5.8

Providing recording, mixing, and effects within one package, **Deva 5.8** features 8 integrated hardware faders, an internal DVD-RAM drive, and records to hard disk or flash media. Additional features include EQ, notch filter, compressor, and delay on each channel, and support for external FireWire recording drives. www.zaxcom.com

Toontrack EZplayer Free

EZplayer Free (Mac Universal VST/AU/RTAS, PC VST/RTAS) is a MIDI bridge, organizer and wrapper between different drum sampler formats. It also allows importing and using MIDI in EZplayer Free from any of the drum sampler formats supported by the software. www.toontrack.com

Antelope Audio Isochrone 10M

The **Isochrone 10M** is an atomic master clock device with stability of 0.03 ppB (parts per billion), which means practically immeasurable jitter. All Isochrone clocks are compatible with any audio converters that can lock to an external clock source using either word clock BNC, S/PDIF, or AES connectors. www.antelopeaudio.com

NI Spektral Delay 1.6, Vokator 1.2

Native Instruments' **Spektral Delay 1.6** update and **Vokator 1.2** update ($29 each) provide Intel-based Mac compatibility. www.native-instruments.com

Sound Devices Software for 7-Series Recorders

New firmware for the **7-Series** family of file-based digital recorders adds several important features including FLAC lossless audio compression, MP3 recording, and multi-unit linking enhancements. FLAC is an open source encoding algorithm used to data-compress sound files. No audio information is lost, while compressing files more than 50%. www.sounddevices.com

Abaltat Muse Soundtrack Composer

Abaltat Muse software represents a fundamental departure from traditional post-production tools, as it exploits artificial intelligence to measure select elements of a moving picture, and then composes music to match those elements. Music is played back using a sample player and virtual instruments, with technology
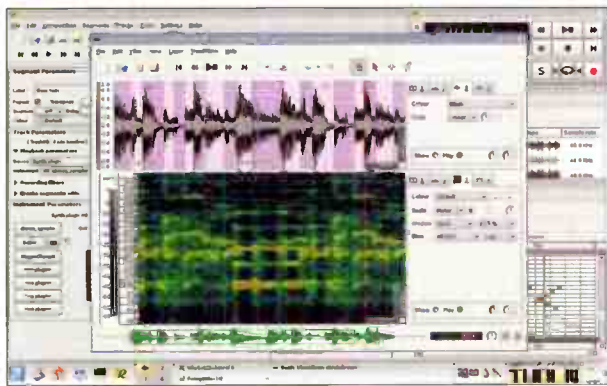
from Native Instruments and Garritan Libraries.
www.abaltat.com

Fervent Software Studio to Go! V2.0

Studio to Go! V2.0 provides Linux audio and MIDI tools for composition, audio recording, and production (Rosegarden sequencer, Ardour digital audio workstation, soft synths, loop base sequencing, sampling software, mastering, CD/DVD burning, notation support, Sonic Visualiser, and over 60 Fervent-specific packages).
www.ferventsoftware.com

IK Multimedia T-RackS Universal Update

T-RackS 1.3 (free for registered users) provides compatibility for Mac Intel Systems, as well as full Pro Tools 7 compliance and Cubase 4 compatibility.
www.ikmultimedia.com

Apogee Stand-Alone Mode for Ensemble Interface

Stand-Alone Mode firmware for **Ensemble**, Apogee's digitally controlled, multi-channel audio interface, allows Ensemble to function without being connected to a Mac. With stand-alone mode, control panel, routing and low latency mixer settings are configured in Maestro once; the settings then remain operational despite the absence of a host computer. All front panel controls remain active.
www.apogeedigital.com

Blue Cat Audio FreqAnalyst Pro

FreqAnalyst Pro V1.1 (Windows DirectX/VST plug-in, $61.25) brings major performance improvements and user interaction enhancements to this spectrum analyzer. The plug-in can be purchased separately or within Blue Cat's Analysis Pack; the upgrade is free for registered users.
www.bluecataudio.com

Amie Street Music Platform for Independent Artists

Amie Street, an online music community that brings together undiscovered artists and prospective fans, is the first site that lets the members of the community set the price of songs — these start out free, and rise in price based on sales.
www.amiestreet.com

Nady USB-24M Dynamic USB Mic

The **USB-24M** connects directly into any computer with a USB 2.0 (or higher) input port for plug-and-play operation with most DAW software. The dynamic neodymium element provides optimal sensitivity with cardioid pickup, and the A/D converter has a 24-bit, 96kHz sampling rate. www.nady.com

BBE DI-50x Active DI Box

The **DI-50x** is an active DI box with −15dB input pad switch, ground lift switch, electronically balanced XLR output, 1% metal film resistors throughout, recessed switches to reduce potential damage, and a non-slip bottom pad to prevent slippage.
www.bbesound.com

Brauner Phanthera Studio Mic

This acoustical principle pressure gradient transducer mic features a cardioid polar pattern, equivalent noise of 11dBA (IEC651), FET-based circuit design, maximum SPL 142dB SPL @ 0.3% THD, and power supply with +48V phantom power.
www.braunerusa.com

Cakewalk z3ta+ Version 1.5

This free update to owners of the **z3ta+** soft synth provides native Vista support, x64 binaries for use with 64-bit Windows operating systems, stand-alone mode, MIDI file import into the arpeggiator, MIDI learn functionality, and more.
www.cakewalk.com

Akai MPK49 USB/MIDI Controller

The **MPK49 USB/MIDI Controller** combines a 49-key, semi-weighted, aftertouch-equipped keyboard with 12 MPC-style drum pads that feature the MPC Note Repeat function and Swing parameters. Other features include a built-in arpeggiator, eight 360° rotation pots (each with 3 banks), 8 sliders, 8 assignable backlit switches, and custom LCD display. www.akaipro.com

All prices are manufacturers' suggested retail price. Toolbox material is provided courtesy of Harmony Central, Inc., and is used with the express written permission of the publisher.
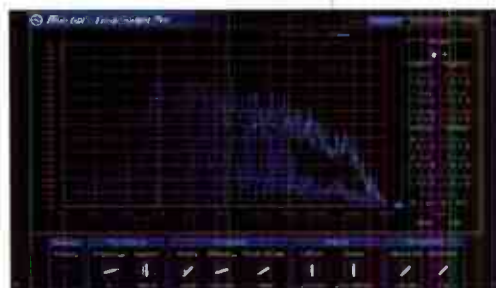
Jeff Glixman's résumé is, to say the least, pretty damn astonishing. Having sat behind the board for everyone from **Bob Marley** to **Kansas** to **Eric Clapton** to **Ludacris**, Glixman has truly been there and done that — with both "there" and "that" summing together to equal "over 30 million records sold." It's true: Jeff Glixman is one of the most sonically savvy recordists of the last 30 years — and he has the Grammy nominations to prove it.

Having done stints in studios all over the world, nowadays Glixman resides in the quaint locale of Bethlehem, PA, where he practically lives in **StarCity Recording** along with cohorts Jim Gentile and Lily Salinas. Together, the group operates a truly world-class studio, working in all areas of music production from engineering basic tracks to mixing in 5.1. So we decided to pay Glixman a visit to talk tech about all things recording, get a literal boatload of choice tips and tricks, and find out how some of the hottest up and coming acts find themselves spending . . .
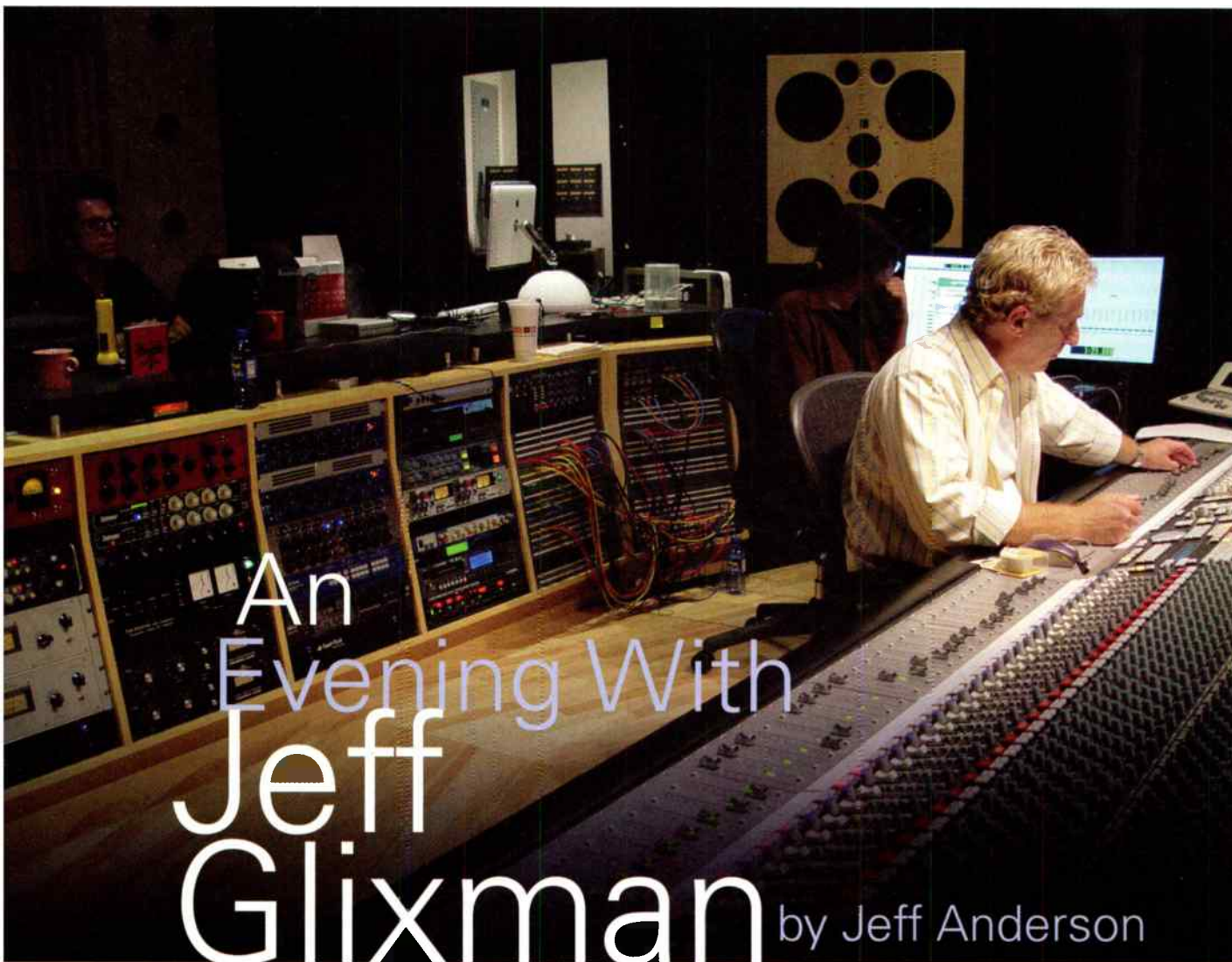
An Evening With Jeff Glixman by Jeff Anderson

EQ: So tell me about this new studio [StarCity Recording] I keep hearing so much about.

Jeff Glixman: The studio's managed by Lily Salinas, who has worked with me since I was in charge of the reconfiguration of Universal Music Group's East Coast studios

seven years ago. When Jim [Gentile, president and CEO of StarCity] and I decided to put this all together, I knew I didn't want to be bogged down with running a major facility, so Lily was brought on board to handle the management for us. We have an SSL 9000K in one room and an Axiom MT Plus, which was formerly Mutt Lange's, in another

control room that we just completely refurbished because, even though the facility is just a few years old, I wasn't satisfied with the sound or the feel of the room. All our live spaces, which run from medium to large, sound really great; and we're just finishing building a performance stage in the theater. We have a smaller production room

as well, which is also a surround sound room. And as we have multiple rooms at multiple price points it allows us to still do our productions while being available for public projects.

EQ: How is business for you out there in Pennsylvania? It seems to be a bit off the beaten path.

JG: So far, so good. I'm not sure that we could be anywhere near this price point if we had 14,000 sq. feet of production space in New York. Also, it's just a really nice, peaceful environment. Everyone who's come here has enjoyed it thoroughly. We're about an hour and fifteen minutes out from New York and Philly, and there are lots of accommodations available. Plus, gambling was just legalized here.

EQ: What current projects are you working on?

JG: Well, there are three. I'm working with the Rhythm Devils, which includes members of the Grateful Dead and Phish to produce a kind of documentary/performance DVD/CD that has really just started. Also, Tyrone Vaughan, which I'm working on with Zak Rivizi, who is our chief engineer and co-producer on several of my projects. It's a great, fun project that's really special to me — we recorded it all analog. I like the commitment by the artist to do this in analog instead of in the cut-and-paste world. I enjoy the process; I'm willing to make a lot of decisions early on, and I like the way you conceptualize and move forward in the linear domain. You don't have unlimited tracks; you decide what you like and want to keep and what you don't. Also, Nathan Lee Jackson, our first signing StarCity Production, is in the midst of 22 tracks with Zak right as we speak.

EQ: For Tyrone's project, what kind of deck and tape are you using?

JG: I'm using a Studer A827 Gold Edition with Quantegy GP9 tape — it's the machine I use for every analog multitrack project, but the tape was just what was available when we started tracking. Lily has just located the distributor/manufacturer for the BASF 900, which now goes under another name. They purchased all the gear and moved the operation to Holland. I've just taken a big delivery of that and it appears to be just like the old 900. I love that tape.

EQ: I ask because I just got off the phone with a well-known mastering engineer who was complaining about how all the new tape is crap.

JG: It's funny: I've had other people say new tape is crap, but I've still had good results. I don't know, maybe it's the batch? Mike Spitz at ATR Services is also manufacturing tape, and he dropped off a sample of his beta tape; it sounds excellent. I had absolutely no problems with the quality — no shedding at all.

EQ: Is it still around $200 a roll?

JG: Actually, it is close to $300 now. It makes for an expensive project.

EQ: How many takes are you keeping of each song?

JG: I have no restrictions on it, I'm pretty good at saying this is right and that's wrong. But there'll be a multiple of two or three; it's rare if you only have one. For the whole project, including the B-reels, I'm using about 20 reels.

EQ: Which room are you using?

JG: I cut Tyrone's project in the large studio A, our major space with the SSL, and then the overdubs are being done in the Axiom room where we have a lot of external gear available, a lot of class A mic pres.

EQ: What are you using for the drums? What mics and what configurations?

JG: I usually use three Neumann U87s — about 12 to 15 feet back — to carry the bulk of the drum sound. I use the kick drum as the center point to make sure they're all exactly the same distance. This project was no different; I moved the particular kick and snare until I loved the way it sounded in the room, and then miked it up from there. I also used two Neumann M149s in omni as more distant room mics that were back 20–25 feet from the drums, in a room that's 40 by 38 feet. I don't like the sound of overheads on drums, but I'll close mic on cymbals, like using a Neumann KM83 on the hi-hat — along with the Sennheiser 421s on the kick and toms and a [Shure] SM57 on the snare — so I can add them in if needed.

EQ: So did you have to dampen or deaden the room?

JG: We have some rather large gobos; 12 feet tall and 4 feet wide, double-paneled, used soft-side out — that I had behind the drummer because he was right in front of a glass wall. There was just a little too much high-end coming off of that. So that wall was kind of semi-dampened, but the rest of it was just open in the room. It's a really balanced, terrific sounding room, which is the key to getting good drum sounds. Besides that, the drummer has to be able to mix himself. My recording style that I'm using to capture his kit is dependant on this, as I can't go back and balance the toms with the mic levels later and still maintain the ambience. As you bring up the floor toms, that shifts the image of the kick a little bit, and I like to get a very solid, clear image on the drums. For example, if he does a great take and he was just a little heavy on the hi-hat in verse 2 he'll say, "let's do it again." I love that commitment to cut in this fashion, it makes for a fun project and when it's all there, it's just all there. You've got all the instruments at once so, even though we may go back and replace guitars, you can always tell where your song is.

EQ: How much of the initial, live tracking did you really end up keeping?

JG: I ended up replacing a large part of the guitars, probably 80% or so — strictly because we wanted to really narrow in on the different tones for each song, not due to the performance. Some of the bass I tracked for the same reason, just looking for tone.

EQ: So did you end up using DI/mic combination for the bass?

JG: All mic. A Sennheiser 421, I believe. We were using a '64 Fender P-bass into a '68 Sound City 50 watt amp with a 2x10" cab. It's a real fat combination. The amp is not loud, but it has incredible tone. It's very similar to the tone John Entwistle had on "Behind Blue Eyes." We also mildly compressed the bass to tape through a [Urei] LA-2.

EQ: Did the guitar setups vary at all?

JG: The guitars vary. I rarely compress guitars at all, to or from tape. I usually get exactly the sound I want out of the amps, of which we have a nice selection: early Sound Citys, a '65 Ampeg Gemini, a Tyrone's heavily modified '67 Bassman, a bunch of early '70s Marshalls, Orange amps, and a

beautiful Matchless that Phil Jamison custom made for us. It's just about everything you could want.

EQ: Sounds good. What goes in comes out.

JG: It sure does. When you create this good organic foundation everything just fits together so well. The boys are just playing their guitars right through their normal effects and into the amps.

EQ: What about the vocals? What was the signal path like?

JG: For Tyrone, the match was a Telefunken U-47 tube mic into a Jeff Tanner GTQ 2-A pre through an [Urei] 1176 limiter right to tape.

EQ: I take it you are mixing on the 9000K?

JG: Yes. That's a beautiful console, and it sounds amazing. I've always liked the ergonomics and flexibility of the SSL and enjoyed mixing on them.

EQ: Do you feel you're using the board to get any kind of color, or is it just clean and you already have the color you want?

JG: It's a pretty transparent console if you choose to use it in that fashion. The thing about the 9000K is that you can get it really punchy, like an old G+ SSL will get punchy. You know, even when we're doing a bunch of our Pro Tools sessions we split it all out into the SSL, and we do our processing through the SSL. I love the channel compressors on the SSL. If we're working in Pro Tools we're really using it as storage or smart storage; there might be a few level rides and some cleanup work but it's still our big tape machine.

EQ: So you're obviously not a big plug-in type of guy.

JG: It depends on the project; some of them sound real good. I'm not real big on plug-ins but that's because I'm fortunate enough to have other gear that gets me what I want. For the Tyrone Vaughan project, I wanted not to be fooling myself — living in the old world — so I went to great lengths to record this to Pro Tools at

> # We're really using Pro Tools as smart storage . . . there might be some cleanup work but it's still our big tape machine.

88.2kHz and analog simultaneously without passing through either medium. I then returned to Pro Tools on one half of the console and analog on the other half. The difference is night and day. For my purposes, it worked way better in the analog domain. I was describing this to another person in the business that had a similar experience who really gave a description I wish I'd given: The analog return sounded just great, it sounded just like what I wanted to hear and expected to hear — and coming out of Pro Tools it sounded like a different song. It sounded so different we double-checked to make sure we were doing things right! I then transferred all the analog recordings in to Pro Tools in 88.2 and that sounded fine. Something in that initial capture is radically different and I don't know if it's analog noise and the other one is perfect or how it works. It's fine after the transfer, but the original just sounds two-dimensional compared to the analog, which sounds a mile deep.

EQ: So you think it's the tape compression, and possibly the sound is just not being duplicated?

JG: I really don't know. I don't cut hot to analog — not looking for compression — but when you try and mix the two, the analog mix goes together just instantly. You want to do something: to add EQ, compression, some kind of something to the digital track. However, I've cut loads of stuff in Pro Tools that I like too. I'm not going to go around and say you've got to use one or the other. I have great results with Pro Tools. I don't process much within Pro Tools but, like I said, I'm always fortunate to be in a

position where I'm doing it on the board with a lot of great outboard gear.

EQ: Why not go as high as 96kHz?

JG: I've done it in 96kHz. I really don't hear any difference between 96kHz and 88.2kHz, but when you do the math and get it down to 44.1 it just seems better to me to use 88.2. Bob Ludwig suggested I use 88.2kHz when I was working on an Allman Brothers project that he mastered. I don't know if it's easier math for the computer or more natural.

EQ: Yeah, I sometimes wonder where we're going to be in ten years, having 9000GB sessions from recording at 2304K. Anyhow, what do you typically mix down to?

JG: 1/2", Pro Tools, and, occasionally, DSD.

EQ: When you're doing a project, how involved are you in the mastering process?

JG: I'm very involved in it. We get mastering projects here at the studio, which I'll master. I just never would master my own mix. I need that objective ear.

EQ: So what do you use on your two bus?

JG: Typically a Neve 33609 Discreet or the SSL compressor.

EQ: What percentage are you mixing to half-track and what percentage to Pro Tools? Do you do both and decide later?

JG: We do both and then decide. Sometimes it sounds good one way, sometimes it sounds good the other . . . unless it's a budget project.

EQ: You mix in 5.1 a lot. Tell us what you do during pre-production. Are you planning tricks for the 5.1 mix out ahead of time?

JG: Unfortunately most of the 5.1 mixes I do are of already existing material. I've only been involved in one project where I got to record the source that I was going to mix in 5.1. It was a live performance of The Orange

Sky that I did in Trinidad. I spent a lot of time with their house mixer and engineer for dialing in the sound system. It was an old 1,400 person theater — we got a great sound in that theater from the front of house sound, and then I miked the room to anchor the surround position. It was the easiest thing to mix because it was all there from the beginning.

EQ: Tell me a bit about the mic placement.

JG: I had a stereo pair above the stage center behind the drummer, in a little balcony area. Midway back I had two mics off to the side and another two for the rear of the hall. In the mix, I had to do a little delay to just get the phase tight between the mid mics and the rear mics. It's pretty obvious and it gets real clear real quick when you're in the right place.

EQ: When you're doing a 5.1 mix you're adding reverb to make the space? How do you know the reverb times? Is it just a feel thing, or is there some heavy math involved?

JG: For me it's a feeling thing. Certain times there's an established prerogative that you're working toward — like with [The Allman Brothers] *Live at the Fillmore East*. That is an extremely respected and established performance that everyone is familiar with, so you want to follow the parameters of the original mix but also want to come up with something different. I was working with the content producer at Universal and he was looking for an audience perspective. The only audience mics were on the stage, so I really worked with delay to get the audience perspective — what I figured was about 20–25 rows back. I didn't really have a lot of house mix, but I was able to determine the reverb times by listening to certain parts of the performance. I used a Lexicon 480 to emulate the room and tweaked it for that, because you're trying not to have a different reverb in every corner of the room.

EQ: When you're mixing a live record in 5.1, are you physically moving tracks inside the box to adjust time or setting delays to make them line up?

JG: All of the above; they're all different. Like with the drums on *Live at the Fillmore East* there were only four tracks total for two drummers — so on that one I had to do a lot of track-splitting and simulation. Because of the processing and the separation, you need to apply certain delays; you don't want it to get phase-y or anything.

EQ: It sounds like a bit of a pain, but it sounds like you enjoy it as well.

JG: It's a blast. Like anything, it has its own set of problems and challenges but it also has its own set of rewards. EQ

HOME STUDIO BOOTCAMP

Fight bad project sound with these quick, cheap, and effective countermeasures.

by Merrick Angle

Everybody knows that the sound of the room(s) in which you are recording makes all the difference in the world. Having a sound space (no pun intended) to work in will affect the way you mic, monitor, and mix. But does upgrading your area really have to be so costly and painful?

Yes and no. Audio perfection is never attained through cutting corners; to work exactly like the pros takes resources. But for those on a budget who have to record in odd places like box rooms, cellars, and garages — at crazy times of the day and night — there are ways to improve your sound without draining your bank account. Whether you are tracking in a rehearsal space next to a cement works, or you're mixing in a bright box room that seems to be giving you tinnitus, you *can* make the best out of an imperfect envi-

ronment. And these quick tips and tricks can aid you in getting the best sound possible from a worst-case scenario.

#1 CLAP YOUR HANDS SAY NO

Before you make the mistake of nailing egg boxes or carpet to your walls, take a step back and be realistic about frequency control. As Rich the Tweakmeister [www.tweakheadz.com] points out: "Egg cartons don't work. It's one of those urban legends that people repeat over and over. They don't make any discernible difference at all." Ethan Winer, musician/writer/co-owner of Realtraps [www.realtraps.com] agrees: "Small room ambience is always bad ambience. So the only practical solution for a room that size is to make it as dead as possible. You can add a much higher quality ambience electronically during mixdown if needed, and that's the only practical approach with such a tiny space. Using only thin materials like 1- or 2-inch thick

acoustic foam or, even worse, carpet, is a recipe for disaster."

Lack of absorption of high and midrange frequencies is generally the cause of a room's undesirable "pingyness." But, thankfully, these issues are fairly easy to sort out. Start by clapping your hands and listening for reflections from the walls. If you get some pingy reflections, your best bet is to pony up and invest in some acoustic foam (ideally a couple of pieces that measure approximately 4' x 4'). Start moving these pieces around the room while continuing to clap until you identify the room's problem areas, and then place accordingly. [*Note: A good way to measure the effectiveness of your foam placement is when the "cracks" from your claps sound crisper and reverberate for a shorter period of time than they previously had.*]

Lack of low frequency absorption, unfortunately, is a bit trickier of a problem to solve. But Winer further warns against shoddy soundproofing practices, claiming that bad soundproofing is worse than no soundproofing at all when it comes to low frequency absorption. "Thin materials absorb only high frequencies," Winer says, "so the result is a room that's too dead — yet it's 'boomy' and 'boxy' at the same time. For DIY types, I recommend rigid fibreglass, 4' thick, wrapped in fabric, covering most or all of the room's surfaces including the ceiling." For folks with a sufficient budget, investing in broadband acoustic panels and bass traps is the best course of action (Figure 1). But for the truly broke, a rug on the floor, a bookcase on the wall, or even a strategically placed sofa between the source and the walls will help dampen the room.

No matter what your budget is, it's important to do something — anything — to help control these frequency demons. Rich the Tweakmeister reminds us:

Fig.1: Absorption panels, such as the Realtraps Minitraps (pictured), can be placed strategically to make an architecturally disadvantaged studio shine with sound.

"The most important thing for me, when recording of listening critically when mixing, is to have a good sounding room that enhances what I am doing. Let's not forget: Listening is for pleasure too, so your pleasure factor is your best guide. A bad room really grates on the nerves in a short period of time. Your ears get tired and you can end up with a headache. It's like eyestrain for the ears. Anyone who has ever painted the walls in an empty room knows what an extremely annoying room it sounds like. But as you start moving furniture back into the room, it starts sounding better. For your studio, you want to do this in a more exacting way, to make the room actually sound pleasant and friendly to the ear."

Fig. 2: Setting your monitors at 60° angles from your listening point is of penultimate importance when it's time to really listen.

#2 IT'S ALL IN YOUR HEAD

Headphones are generally scoffed at by pros, being tolerated only moderately when used by recording musicians for monitoring during tracking/overdubbing sessions, or for giving a different perspective while mixing. But for someone with a bad sounding mixing area, they can be incredibly helpful. Karl Coryat, author of *Guerrilla Home Recording* [www. backbeatbooks.com], suggests using them when you are forced to listen in a poorly treated room. "Every serious home recordist should have a good pair of headphones," Coryat says. "Investing in a good pair is, in my opinion, much more effective than spending the same amount on acoustical room treatments, or to upgrade your monitors. For one thing, in terms of monitoring, headphones are a way to take any bad aspects of your studio's acoustics out of the equation. I've also found them very effective to check out the finest details of your recordings, as well as to analyze pro productions figuring out how they're made and what makes them sound so good."

A decent pair of headphones will do three things. First, they will block out unwanted background noise. Second, they will provide you a complete stereo image of your studio's output regardless of where you are. Third, in the case of having a poorly soundproofed room, the

endless playbacks of that guitar solo/beat/or eerie synth pad will no longer have your neighbor looking for his shotgun. But beware: Don't rely too heavily on your headphones. As Coryat says, "The psychoacoustics involved with wearing headphones make it difficult to gauge the relative levels of elements in a

> BAD SOUND-PROOFING IS WORSE THAN NO SOUNDPROOFING AT ALL WHEN IT COMES TO LOW FREQUENCY ABSORTION.

mix, particularly in the low end, so you should always double-check mixes — in your car, on a boom box, and/or on consumer home speakers."

So that leaves us with the crucial decision as to what type of headphones to invest in. Some say closed-backed headphones compromise sound quality, but they block out much more external sound, making them a good choice for, say, monitoring a loud guitar band. Open-backed headphones, on the other

hand, may give you a better sound quality, but leak more audio into your immediate environment. Regardless of what you decide works best for your situation, Coryat stresses that you "look for headphones that aim for clear sound and true bass and avoid anything that intentionally colors the sound or attempts to improve the listening experience. For instance, if the box touts 'turbo bass' or something similar, steer clear."

#3 MAP, MONITOR, SURGE

It may seem like a no-brainer, but it's incredibly important that you set your monitors correctly in your mixing space (Figure 2). Winer recommends the following: "In a rectangular room, the loudspeakers should always be set up so they fire the longer way down the room. The goal is to put the mixing position furthest from the rear wall behind you because that's where the inevitable peaks and nulls are worst."

"The ideal mix position puts your ears 38% of the way back from the front wall you face while listening," Winer continues. "Then place the loudspeakers in an equilateral triangle. Anything outside that magic equilateral triangle and you could be throwing sound all over the place, really compromising the performance of a good set of monitors. If your room size is a problem, as a rule, monitors being too close together are better than them being too far apart. This will leave you

Fig. 3: We don't need no stinking iso-booth. Only to be tried at home, the 60 Second Vocal Booth (err . . . couch) can give you the necessary isolation to cut truly great tracks.

with a massive 'hole' making panning and placement difficult, to the point of being impossible."

Set up your monitors roughly at about head height; mount them securely and at the same level. Please don't be tempted to put the monitors on their sides either, unless the manufacturer recommends it; they won't give you a true, balanced sound. If you are in a box room, having issues with high end reverberations and overall clarity of sound, fixing absorption panels on your side walls at the same height as your ears will help iron out a lot of these problems.

#4 THE SOUNDS OF SILENCE

"I find it kind of funny that some people will spend thousands to treat their room

yet never quiet the stuff inside the room itself!" exclaims Rich the Tweakmeister. "You walk in and hear a noise coming from computer fans, hard drive whines, ZIP drives, SCSI drives for samplers, fans in samplers, fans in amps. This is no way to work on music or produce audio, as this racket masks other problems in the studio — like 60Hz hum at the console outs, or poorly setup gain on mics, synths, and other instruments."

It's true, all that gear that you have begged and borrowed could be a big part of this very problem. All you need is a computer with a dodgy fan and a few antiquated amps to mess up a recording. And, if you're recording every track in the same noisy area, you can count on your problems multiplying in the mix.

HOME STUDIO BOOTCAMP

So what can you do? "I tried putting the computers in a closet and ended up creating a second furnace for the house," Rich the Tweakmeister says. "It got way too hot in there! You could also get a sound enclosure box, but they are pretty much out of the home studio budget. For me, the solution was moving my industrial strength PC and Mac into the next room. What an amazing difference!

"The simple answer is to get all of that stuff outta there," he continues. "Drill a 4" hole above the baseboard going into the adjoining room. Make sure there is space in the next room for your computer and a rack unit. Then make an inventory of the cables you are going to need to pull this off. [Hint: If you have a FireWire or USB audio interface, this makes life much easier.] Get a few powered USB hubs for stuff like your mouse and keyboard. Perhaps the hardest are cables for the video monitors: VGA extension cables are easy to use, but avoid the cheap ones as they may cause ghosting on the screen. Digital video extenders are available too, but if you map things out carefully before you drill, you might find a way to get the stock six foot cables on most LCD monitors to make it into the next room and into the back of the computers."

If you simply don't have the space, you could always set up all the noisy gear on a trolley and when it's time to record, wheel all the gear outside the room, close the door, and get on with the work.

"This was, without a doubt, the best upgrade I have made in my studio since I started using hard drives," he concludes. "I can once again hear and pinpoint troublesome noise at my mixer and take steps to get rid of it. When I am doing sound development work, I don't have to crank the gain or wear headphones to hear subtle nuance. Thanks to the lower levels of monitoring I can compose and mix all night long without disturbing neighbors or roommates."

#5 SO CLOSE TO YOU

In the world of the truly acoustically awful, there is but one sure-fire last resort: Close mic it all. As Winer tells it: "The only viable approach to recording in a small room is to put the mics as close to the instruments and amplifiers as possible. This captures more of the instrument's sound and less of the room tone.

HOME STUDIO BOOTCAMP

But there are a few things of which you need to be aware. First, cardioid mics increase the level of bass frequencies when they're placed near a source, so you'll probably need to use the low-cut filter on your microphone preamp. Another possibility is to use omni microphones that don't have this proximity effect — but then the risk is that by being omnidirectional, they'll pick up more of the room tone you're trying to avoid."

Rich the Tweakmeister concurs: "My first advice is to fix the room. But if you can't, try using a quality dynamic mic rather than a condenser. Most dynamic mics are less sensitive to reflected sound. Condenser mics, however, can be extremely sensitive and can pick up every little sound in the room, including those outside your house if you crank the gain enough. The Sennheiser MD421 and Electro-Voice RE20 are both great mics given that you have a quality preamp to boost the signal . . . but so is the Shure SM57."

But, with some instruments, there is a problem recording only with dynamics. Cellos, violins — even acoustic guitars — require a different approach, as they have frequencies emitting in multiple directions. As these instruments need a little bit of space and some of the room's ambience to capture their full range, your best bet is to start close in and work outward from the instrument until you find a nice balance between the sound of the instrument and the room itself.

Another potentially problematic instrument is the bass, with its major low frequencies and window rattling waves. Coryat, a bass player himself, reckons the most direct approach is the best. "I can think of only one time that I miked an amp to record bass, and that was years ago. All other times, I've recorded direct — which completely eliminates the room factor. If I want distortion or color, I supply it electronically. If you record bass this way, it will sound the same whether you're at a million-dollar studio, at home, or in an old VW bus. With the tools that are available today, if someone hears your recording, it's unlikely that they'll know you recorded direct. They'll just hear a great bass sound."

"The same goes for guitars," Coryat adds. "For some of us a decent space to mic up a great sounding vintage amp is an unheard of luxury. A good direct box can get fantastic results, or, dare I say it,

> ## EXPANDERS AND COMPRESORS ARE LIKE LIVE UMPIRES: THEY'RE DOING THEIR JOB BEST WHEN YOU DON'T NOTICE THEM.

going directly into the desk or soundcard and using a good plug-in can sound awesome . . . and keep your neighbors happy."

#6 GONE IN 60 SECONDS

This seems ridiculous, but is simple, cheap, and very effective: Get a sofa or a futon and stand it on its side, placed in the corner of the room so it creates a hollow (Figure 3). Get in with your microphone and voilà, instant vocal booth. An engineer friend who wishes to remain anonymous (he's a pro that fears this sort of knowledge made public could end with him being burnt at the stake) says he actually recorded an entire album's worth of vocals for a successful hip-hop album this way. "I had no studio time, a deadline hanging over me, and this was the only way of getting the job done," he says, apologetically. "I got the MC in question to come over and we nailed it. Nobody has ever suspected that this was done in anything other than a bona fide vocal booth."

So you don't have an extra sofa just lying around? You've got to get the vocal tracks down now, and the room is making them sound like they were recorded by the girl down the well at the end of The Ring? Coryat suggests this trick: "My favorite secret weapon is a toilet paper tube. Anytime I want a filtered, bandpass 'telephone' sound — which is also a way to improve intelligibility — I hold the tube about an inch from my mouth while I sing, pointing the other end a few inches from the mic. This usually results in a somewhat

'peaky' sound though, as your voice tends to hit resonant frequencies in the tube; so some after-the-fact compression and EQ is going to be needed to keep the track seated in the mix."

#7 THINGS THAT MAKE YOU GO HUMMM

Whilst not being strictly an acoustic issue, buzz is something that, like poor acoustics, can hamper every stage of the recording process. Causes of the dreaded hum fall into two main categories, according to Winer: "One is solid state light dimmers, and it's especially a problem for apartment dwellers because the buzzing dimmer could be in a neighbor's apartment over which you have no control. Another cause of buzzing is ground loops that develop when using pieces of gear plugged into different AC power outlets. First, as long as the gear's current consumption doesn't exceed that of the outlet you want to use, you can plug all of your gear into the same outlet. You can use power strips if you have many devices to power. Second, if using one outlet for everything is hazardous or not practical, the next step is to use audio isolation transformers."

Other factors could include the proximity of the instrument to your computer. "For guitars and basses that buzz, don't sit too close to a CRT-type computer screen, and before you record, be sure to rotate your position while you play quietly to find the orientation that causes the least buzz," Coryat says. "When you find the position that's least noisy, take note of something in the room that the instrument's headstock is pointing toward, and be sure you're in that position immediately before every take.

"In a home studio there will always be a little buzz, hum, or hiss to contend with," Coryat admits. "I recommend running all but the quietest sources through an expander, set to a low threshold and a gentle ratio, between the source and the recording medium. Set the expander so that it sounds like it's only cleaning up the spaces between the notes; if you're hearing it do its thing, or if the sound is cutting in and out, use subtler settings (a lower ratio and/or threshold). In most cases, expanders and compressors are like live soundmen and umpires: They're doing their job best when you don't notice them." EQ

WHY ACOUSTIC REFLECTIONS MATTER

Has "reflecto-phobia" taken away something vital from our recordings?

by Arthur Noxon

The early days of recording were all about making live mono recordings of acoustic bands — a bunch of mics wired direct to tape. Next came the multitrack, and recording evolved into click tracks, iso-booths, post-processing, and mixdown sessions. Then the digital age showed up. Acoustic recording was out, sampling and DI was in, and everybody's cousin had a home studio.

Times keep changing, however. Home studios have evolved into more sophisticated studios, and the bands themselves have evolved. These days, there are more mic companies than ever because studios are doing more miking and less going direct. Studios and bands want to do live, ensemble recording. Even rapping is going in this direction. Today's studios are actually recording more real music, air breathing acoustic sound, than ever before.

But some things never change. Open up a mic, and we get two kinds of sound. The first is the sound we want to get: the direct signal from the talent. The second is the sound we don't want to get — the sound from the room. We usually end up needing a cleaner signal at the mic and so our goal, as recording engineers, is to figure out ways to boost the direct signal and cut the room signal.

By following in the footsteps of the last few decades of recording, we try to get an acoustically dry signal, as close to an acoustic DI as possible and then perform the familiar post processing on it to get it into the mix. To do this, we have to build acoustically dead spaces and in the process, kill all the reflections.

But in so doing, we are also throwing the baby out with the bath water. Who'd have guessed that some of those hated, hunted, and hammered room reflections actually help make real sound, sound real? Recording used to be full of this type of "real sound" and today, by implementing a few acoustic tricks, recording can once again sound real.

THE MIC IS A 2-CHANNEL ACOUSTIC PREMIX

In any room, the mic acts like a two track acoustic premix, as the direct signal is mixed acoustically in with the room signature: reflections, echoes, reverb, and general room noise. And most of the time, the room signature track is too loud. To boost the signal to noise ratio, we need to boost either the direct signal or fade the room noise; usually it's some combination of both. What we want is an "acoustic fader" but air faders, like air guitars, don't do much for sound.

We boost the direct by getting the talent to eat the mic and reduce the mic gain back down to zero VU. But now our talent sounds like a radio DJ and that just might not be the sound effect the producer wants. In addition, we lose control on dynamics, plosive, and proximity effects. To regain control we add the wind ball, dial in EQ, compression, and limiters, and hope for the best.

Another way to increase the direct to room signal strength ratio is to change the mic pattern (or position). Start closing it down and narrow the focus pattern of the mic, stopping somewhere between cardiod and shotgun. But the tighter the pattern, the more colored the voice, and we go back to EQ, compression, and limiters to try to doctor the track into a semi-real sound.

The other way to get a better SNR at the mic is to just dump the room and get pure sound flowing into the mic. Forget EQ, compression, and limiters; just set the mic up in a soundproof anechoic chamber, and one would think we have the ultimate recording space — essentially it's acoustic DI, all direct signal with a –80 dB noise floor. Later, this very dry signal can be revived by post processing, and adding some warmth and depth with a little delay reverb, as well as some sparkle with a spank from an exciter.

DRY RECORDING ROOMS

When working in dry rooms, any reflection is audible and likely sounds bad. All it takes is one reflection, and the sound we are trying to get picks up a hollow tonality called the Comb Filter effect. This is when the desired signal is combined with a lower level and time delayed signal — in other words, an early reflection. The

Fig. 1: Recording a duet in a living room. One large QSF lightly blends the vocalist and piano, while isolating them from the living room. Mics under the lid get QSF splash back.

Fig. 2: Here's Bruce Swedien, who's used QSF for 12 years, doing a mic seminar in a concert hall. Note the wide spacing; he's adding a light splash of QSF to a room that already sounds great.

Fig. 3: This early QSF setup (circa 1987) uses an overhead QSF "cloud" to help hold in upward energy.

combination imposes a harmonic set of cancels and adds onto the original signal spectrum, which sounds like the direct signal was recorded at the bottom of a drinking glass. Dry acoustic recording is very sensitive to the presence of early reflections (comb filter effect), late reflections (echo), fast repeating reflections (flutter echo), boundary loading, mode coupling, and finally, reverberation. Still, dry recording seems to be the primary tool for today's recording engineer.

The rule of thumb in a dry recording studio is "the best room is a dead room." Engineers are trained in engineering schools and the school of hard knocks to hate reflections. Today, "reflecto-phobia" is rampant. Music is proudly recorded in acoustically sterile environments. Fueled by fears of comb filter coloration, every single reflection, near or far, that might ever hit a mic has been systematically exterminated over the last 30 years. Many of today's so-called "live rooms" are "reflection-free zones."

HUMAN LISTENING

When people, in contrast to microphones, listen to sound, they generally just listen to what they want to hear and pretty much dial out the rest. People can be located pretty far from the talent, compared to a mic, and not even notice the room sound; they just hear the talent. People are able to tune the room out naturally, and focus in on the talent. The engineer with a mic has to work hard to tune the room out and focus in on the talent.

A person is a biological signal processor, not an electronic one. We use a different mechanism to hear than what is built into mics. A by-product of our hearing system is that we automatically mix all early reflections right into the direct signal and end up hearing one composite "direct" sound. Early reflections are those that arrive within about 1/30 second following the direct signal. It doesn't matter where those early reflections come from, they just add together (correlation signal detection) in a way that makes the perceived sound significantly louder than the direct signal. This sound fusion process creates a composite direct signal, which has easily more than twice the sound power than the direct signal alone.

Although it doesn't matter to the sound fusion process where the early reflections come from, we aren't confused by where the direct sound comes from because of something called the precedence effect. We cue in on the direction of where a sound comes from by tracking and locking on where the original sound signal comes from. The process of knowing where a sound comes from is called echolocation.

There is one adjustment to echolocation that we need to mention: the Haas effect. Very early reflections (those arriving within 5ms of the direct signal) will distract us from knowing exactly where the direct signal is coming from. The perceived direction of the direct signal is somewhere between the location of the direct signal and the location of the very early reflection

People like early reflections. In fact we're designed to hear direct and early reflections, and to mix them together into one "direct" sound. This process helps us hear more easily what is going on.

THE EARLY YEARS

In the 1950s, they had one, maybe two takes and then the session was over. The idea was to use a number of mics distributed throughout the group, adjust their position and gain and get a live, hard-wired mix down direct to tape on a mono track. Their goal was to capture enough signal to recreate the sound that was heard when sitting in the room. Those days are far from the idea of recording separate tracks in iso-booths at various times and in various parts of the country, then mixing them together a few months later.

A good example of the tail end of the early days recording technique was in the RCA Victor Studio B in Nashville back in the '50s and early '60s. This topic came up during an AES Sectional presentation on the Quick Sound Field (QSF) recording technique, held there in 2003 (for more information on QSF, visit www.ase-studio-acoustics.com/qsf.htm). QSF is a modern way to acoustically capture sound fusion at a mic; see Figure 1. Studio B had finally been renovated but it wasn't open to the public yet; the room was full of engineers — a lot of new ones who hadn't even been in the studio since it closed, and some old-timers who worked there when they were young. After the QSF presentation was over, the question and discussion time quickly led back to the recording

WHY ACOUSTIC REFLECTIONS MATTER

techniques that used to go on in that room Studio B is a shrine. It's enough to just stand there, inside that room and wonder upon all those hallowed vibrations — the ones that hit the floor tiles and bounced off, and those that lie buried still in the wall and ceiling tiles. So many early greats worked and played there: Elvis and the Jordanairs, Roy Orbison, Everly Brothers, Chet Atkins, and many more recorded in this old RCA Studio B.

The QSF lecture reminded the old-timers about recording in this room. They talked about the mic setups and how the band played all together, at one time, one song from start to finish, direct to tape. And that was how they made records.

This was all well before multitracking and mixing capability became available in recording. When multitracking came, in the '70s, Studio B accommodated the growing interest in this "new sound" of music. The room was deadened and hosted a small village of iso sound shacks lining the walls. Eventually Nashville was overrun with recording studios, and Studio B closed. Now it has been renovated back to the glory of its former years. All the sound shacks are gone now, and the room has been returned to its original, one big recording room, configuration.

Back in the early days, the room had a 3-mic, gain and mix to Ampex tape approach. Later, more mics were added. There were no isobooths. At best, there were gobos. In this environment, each mic got signal from every instrument. For example, if there were 12 mics and six talent sources, there would be at least one direct signal from each talent source arriving at each mic. That means that there were at least 12 different signal path versions of each talent source after mixdown. And then the early reflections have to be added in; floor bounce, glass bounce, other instruments and what not.

The net result after mono mixdown would be that each talent source would have at least 12 direct signals, with time delays ranging from 4ms out to 25ms, and levels ranging from 0VU down to –16dB on the track. And then there would be the reflections, filling in the mix with even more random time offset signals. In a 12-mic setup there would actually be captured up to 30 or 40 distinct time delayed signal paths for each talent source. That qualifies as a "Sound Fusion" effect recording.

VARIATIONS ON SOUND FUSION EFFECT RECORDING

During this early period, the ASC TubeTrap factory got a few calls from engineers who heard about the QSF sound (Figure 2). One had been doing a radio for many years. He said he developed a magic black box that was his trade secret: It consisted of a whole bunch of amplitude adjusted time delays inside the box. He fed his mic into one end, and got a synthetic QSF sound (direct + a whole lot of random time offset signals) out the other end. The time delays matched exactly the QSF window of about 25ms.

Another engineer contacted the factory and told his story how he had hooked 30 some mics up over the top of a classic opera singer. Each mic was located at a

different distance and angle from the talent. He just added them all together and ran it out to the house sound system. He said the sound was fantastic and used the technique many times. He effectively collected some 30 random-time off set signals, all within the 25ms time window. Each signal was basically the same signal except for the acoustic EQ due to the off-axis coloration of the voice. And, as the talent moved around, the sound package didn't change; the total sound remained the same, even though the signal fed into the different mics did change. The listener's brain can't tell which reflection is where inside the Sound Fusion effect time window.

At that time, digital reverb was starting to become affordable. The reverb plate was being replaced with a four adjustable delay/reverb returns. When ambience was set tight (300 to 500ms) and the delays set shorter (30 to 100ms), it produced a synthetic ambience, much like a room. By setting it even tighter and shorter, the Sound

Fusion effect could be generated. But the big advantage with the acoustic version, the QSF, is that it controls the presence of natural ambience in the room at the mic while adding close and natural flush of early reflections into the acoustic mix at the mic position.

THEORY INTO PRACTICE: SETTING UP QSF WITH STUDIO TRAPS

Although there are many ways to create early reflections electronically, probably the easiest approach is an acoustical one built around ASC's StudioTraps. These are tubular structures where the front half is reflective in the treble range, and the rear half is absorptive in the treble range. The entire surface of the trap is bass range absorptive. Originally, they were set up to absorb treble and create a sort of "instant vocal booth," where room noise was reduced, so it was possible to raise the mic gain. But as engineers experimented with it, they found that pointing the reflective surface of at least eight StudioTraps toward the mic

could create the effect of early reflections that decay rapidly.

In a smaller home studio, a typical setup is to set a tight QSF pattern. The smaller the room, the tighter the pattern, and the more intense the Haas reflections, which boost the live effect and at the same time, blocks the room even more. Typical small room recording does very well with only eight StudioTraps in a semicircle setup, 4 to 5 feet in diameter.

While working in QSF, the talent is free to groove to the music without causing a shift in the sound at the mic (Figure 3). As the talent moves, all that's changing is the arrival time of the various Haas reflections. However, the ensemble package of direct + early reflections remains at the same level and sounds the same.

In any event, regardless of how you add early reflections to your sound, I hope this article has gotten you thinking about why they're important and useful. Make them work for you, instead of against you, and you'll be rewarded with a more live, animated sound. =Q

THE SMALL STUDIO WORKSHOP

by Jeff Anderson

You've spent many restless nights dreaming about it, and now the time has finally come to build your own home studio. But how do you take a small space and turn it into a studio capable of turning out awesome sounding tracks? Worry not, dear readers: With the help of world-class studio designer Carl Tatz, we're going to walk you through the process of designing, building, and treating your home studio. Starting from the ground up, we have all the tips, tricks, and studio designing secrets you'll need to build your own modest session space — all without sacrificing your ability to get truly great sounds.

CHOOSING YOUR SPACE

After making the decision to build a studio in your home, the first and most important step is deciding which room it will occupy. "The primary consideration in choosing the room of your home to implement a studio would be symmetry," Tatz says. "A shoebox-shaped room is always best for a couple of reasons. The first is because of its simple symmetry, and the second is that with a shoebox-shaped room it will be easy to calculate the primary axial modes in the room for the speaker and engineer's placement."

But while the average room in a home is rectangular, chances are they also have some sort of closet and/or windows in them. As Tatz comments: "Windows are great but need to be addressed for two reasons: sound leakage, both going out and coming in, and their placement in the room. If you have a window centered on the front wall, in front of the console, you're golden. However, if the window is off to one side it will make the room look and feel wrong. You may have one monitor in front of the window that has its own diaphragmatic acoustic properties and the other in front of a wall, making the two speakers each have a different frequency response. The trick is to choose the end of the room that is most symmetrical from the front wall to at least a couple of feet past the engineer's position."

ONE ROOM OR TWO?

The next decision is whether or not you want to split your room into two separate areas, or if you want to have both your tracking and control spaces together in one area. "It's relatively easy to run a mic cable to another room dedicated to vocals and overdubs," Tatz says. "If budget and need allow, you can always build an iso-booth in the same room if it's large enough, but again, keep in mind that symmetry in the room is a virtue as you're making your initial plans. Some musicians are happy to deal with headphones and be in the same room as the engineer while recording — but it can get difficult to dial in tones without using a proper monitoring system.

"If you do decide to both track and engineer in the same room, I recommend that people put their computer and any other noisy gear that they have in a separate area, such as a closet," Tatz advises. "The last thing you want is a high noise floor while you're listening — or even worse, recording. A computer's fan noise put to tape can totally ruin a recording. So put your gear in a roll out rack in the closet and install an exhaust fan that will both suck out the warm air and draw in the cool air from beneath the closet door. I highly recommend the System 1 Exhaust Module from ATM [www.activethermal.com]. The other alternative, if you don't have a closet nearby, is to get a noise reducing cabinet, such as Sound Construction's IsoBox [www.custom-consoles.com]. Keep in mind that while these pieces work well, they can take up a lot of room in your small space."

DEALING WITH ELECTRICAL ISSUES

Other considerations lie in your studio's air conditioning and electrical systems. "Having an air conditioning supply and return in your room is a positive. For the electrical in your studio, have an electrician make sure that the grounding is good in your home. Typically for a home studio, you will need three extra 20-amp circuits, as home studios usually don't take up much power. Use one circuit for the computer, one for the outboard gear, and one for the monitoring system — sub woofers take up a lot of power! The Furmans, Monsters, Richard Grays, or Equi=Techs are worth having, but they only go onto one circuit and that can get

Constructing on the grid: The photo at left shows the grid system for floating floors, while the photo at right shows the dual wall construction (note the cable troughs).
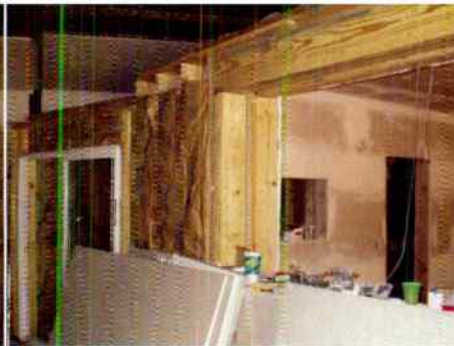
Room within a room construction: Sound Logic studios being built inside an old barn in the back yard. The photo at left shows the far wall to the control room being constructed. The middle photo shows the double wall design between the control and the tracking room. The photo at right shows the future control room, complete with two layers of sheetrock on each wall for solid soundproofing.

expensive if you purchase one for each of the three circuits. Amplifiers have large transformers so they are more resilient than other gear and don't always require an AC filtering/surge protection device like some power conditioners. Furthermore, it's very important that you do not have anything not related to audio, such as lights or air conditioning, on your three audio circuits."

TO THE WALLS

At this point it's time to address the issues of keeping sound from traveling outside of, and into, the studio, so you need to start planning out some ways to soundproof the room. "Mass is the best way to soundproof a room, which can mean layers of drywall, plywood, floating the floors, walls, and ceiling and if your budget can afford it — building a room within a room, which is by far the best way to stop sound," Tatz believes. "There are new products that can help you with the construction. Quiet Solution [www.quiet solution.com] makes a sheetrock product called QuietRock that can be more effective than regular sheet rock. GreenGlue [www.greenglue.com] also makes a self-titled vinyl polymer product that you sandwich between layers of sheetrock that offers similar results to the QuietRock. (Incidentally, Quiet Solution makes a very similar product to the GreenGlue called QuietGlue.) If you are on a tight budget, try using GreenGlue or QuietGlue to add another layer of sheet rock onto your existing walls. This is one of the most cost-effective ways to add mass to what you already have. One thing that I advise people is that someday you may re-sell this space, so don't go too crazy. Kinetics Noise Control [www.kinetic-snoise.com] is also a great resource for isolation products, so check them out as well."

ALL ON THE FLOOR

Once the walls have been treated, focus on the flooring. There are three reasons to float a floor: to make the room sound

better, to provide isolation, and to run cable through the space to service your studio. "To build a real floating floor, for sound isolation purposes, the best way to do it is to construct a frame in a grid shape with upright 2 x 4s," Tatz advises. "The trick to all floating floors is to make sure that they do not touch the walls. Try to keep the entire floor at least 1/4" away from the wall. Even once you've completed the floor, you want to keep the base-

BUILDING A ROOM WITHIN A ROOM IS BY FAR THE BEST WAY TO STOP SOUND.
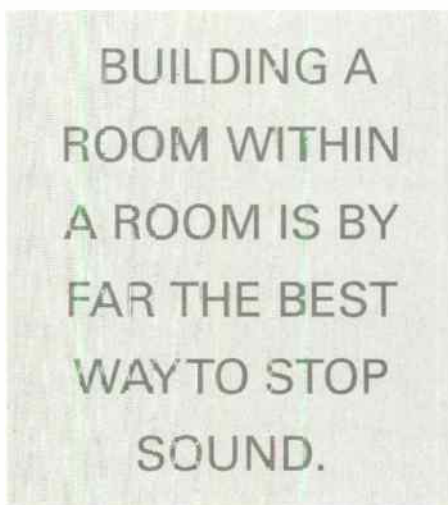
board trim at least 1/8" from the floor so that it is not transferring the sound from the walls to the floor.

"Once you have the 'grid' constructed, run your audio snakes and cabling through the grid and fill all of the empty space with 4", 8 lb. rockwool. Then place 1" x 6" strips of closed-cell neoprene, or Auralex U-Boat Floor Floaters [www.auralex.com], every foot underneath the 2 x 4s. On top of the grid, glue and screw 3/4" tongue and groove decking. You can add more mass to this to increase isolation by adding layers of 5/8" sheet rock and finish with more decking. Of course, as you plan this out, keep in mind that by adding mass to your floor you will be incrementally lowering your ceiling height.

"On the top layer, add your carpet or hardwood floors. Chances are that you will have a rolling chair set up in front of your gear. If you plan on using carpet for your control room, consider building a wood floor 'box,' or area on which your chair can roll around."

There is also a poor man's way of making a floating floor. As Tatz says: "Try putting down a layer of 3/8" Owens Corning Quietzone Acoustic Floor Mats [www.owenscorning.com]. Then put the decking right on top of that. I usually do this when there is a concrete slab. Again, as with all floating floors, be sure not to touch the walls with your floor. And be aware, the effect of building this floor isn't for isolating the sound from and to below, although it will help. Building this floor will make the room sound, and feel, much better."

DOORS OF PERCEPTION

Poor choices in door selection/placement can lead to serious sound leakage problems in a studio. Tatz recommends "installing two 1-3/4" thick solid core doors, with space in between each, opening in opposite directions" to combat this problem. "If you choose to, you can also use two glass doors that are made of 3/4" laminated glass," Tatz continues. "The cheapest way to soundproof the doors is to get an exterior door jam, with gaskets already on the top and both sides. On the bottom, you can get a sweep, or make your own jam so that you have a gasket all the way around. By using the external jam, most of the sound that does get through will pass through the gasket. However it's mostly high-frequency sounds, so if you have two doors it's really not an issue.

"If you can't manage two doors because of space restrictions, at least use one solid core door — and beef it up with an additional rubber screw-on door gasket. It's not perfect, but you don't always need perfect. With the extra gaskets you will stop at least some of the sound from passing through."

BOB BULLOCK CONTROL ROOM

AXIAL MODE CALCULATOR

Enter room dimensions in inches		1st Through 4th Axial Mode Frequencies			
		First	Second	Third	Fourth
Height:	96 in. = 2.44M	71Hz	141Hz	212Hz	283Hz
Width:	119 in. = 3.02M	57Hz	114Hz	171Hz	228Hz
Length:	155 in. = 3.94M	44Hz	88Hz	131Hz	175Hz
Room volume =	1,025	Cubic feet			
Enter speed of sound:	1131.0	Feet per second	(default: 1131 feet per second)		

HDS1on1@gte.net

Room Height = 96 inches

| >12in. | >16in. | >24in. | >36in. | >48in.< | 36in.< | 24in.< | 16in.< 12in.< |
| 283Hz | 212Hz | 141Hz | 283Hz | 212Hz 71Hz | 283Hz | 141Hz | 212Hz 283Hz |

Room Width = 119 inches

| >15in. | >20in. | >30in. | >45in. | >60in.< | 45in.< | 30in.< | 20in.< 15in.< |
| 228Hz | 171Hz | 114Hz | 228Hz | 171Hz 57Hz | 228Hz | 114Hz | 171Hz 228Hz |

Room Length = 155 inches

| >19in. | >26in. | >39in. | >58in. | >78in.< | 58in.< | 39in.< | 26in.< 19in.< |
| 175Hz | 131Hz | 88Hz | 175Hz | 131Hz 44Hz | 175Hz | 88Hz | 131Hz 175Hz |
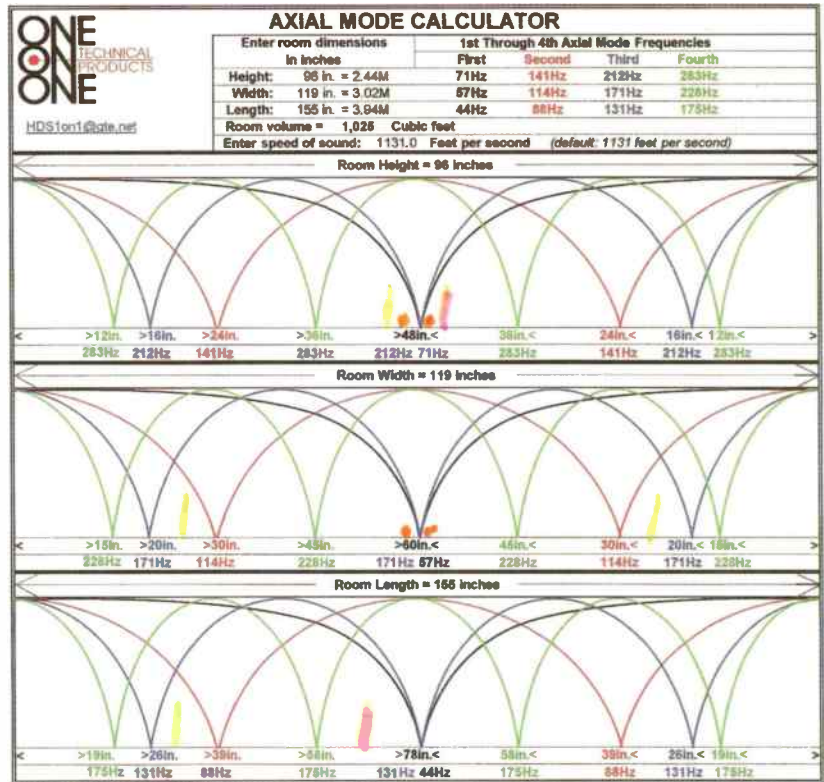
Engineer
Speaker woofer
Avoid

BOB BULLOCK CONTROL ROOM

PLAN VIEW

Scale: ¼" = 1'

H= 96"

W = 119"

L = 155'

World Radio History
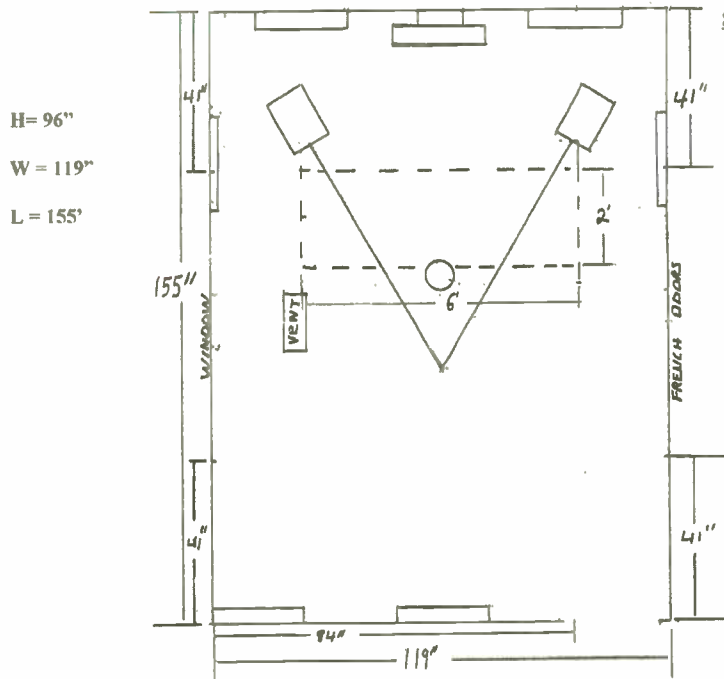
The top photo on page 38 shows the Axial Mode Calculator, used to determine optimal speaker/engineer placement for Bob Bullock's home studio. The bottom photo on page 38 is the napkin blueprint of Bob's room, showing speaker placement and mix position. The photo above is the end result: Bob's control room, all dressed up and ready to go. Photos courtesy of Carl Tatz and One On One Technical Products.

TREAT YOUR ROOM (AND YOUR MAMA) RIGHT

"The main issue to address in a listening environment is the first reflections from the monitors," Tatz tell us. "Sit in the engineer's position and have someone hold a mirror up along the sidewalls. Where you see your speaker's drivers is where you will place your treatment. Repeat this for the ceiling as well. The ceiling is the most overlooked spot and is very critical. If the first reflections are not treated you will have to suffer with comb filtering, meaning the monitor's direct signal will be combined with the reflected sound, causing phase and delay cancellations. This will cause destructive peaks and dips to the overall frequency response.

Once the first reflections and ceiling are taken care of, treat the back wall. You don't want sound coming back to you in the mix position from the back wall or any of the first reflection points. Once you've taken care of the back wall, treat the front wall with some evenly-spaced panels to attenuate the speaker boundary effects.

"The best treatment to use in small rooms is 4" of 703 fiberglass wrapped in a fabric panel. These panels will absorb down to 125Hz. Thinner panels will start to make the room sound 'honky,' because they are taking out only the highs and upper midrange frequencies. [Note on low-end absorption: Corners are where the energy builds up; ceiling corners in particular are where things will get funky. Don't think that sticking a foam wedge 'bass trap' in a corner is going to do anything below 125Hz. Auralex makes the CT45 corner panels that have a 45° angle on them. Real Traps (www.realtraps.com) offer low-end corner absorption panels (Corner Mondotraps). RPG (www.rpginc.com) also offers the Modex Corner. Try these or similar products instead of foam wedges.]

"Generally speaking, my philosophy is that absorption is your best friend in a small control room while diffusion and absorption are your pals in a tracking room. After you've done the above, you can tame down the flutter echoes along the parallel sidewalls by placing panels as you see fit. It doesn't take much. And don't worry about reverb time — you don't have any. This is small room acoustics, not auditoriums." EQ

THE SMALL STUDIO WORKSHOP

QUICK STUDIO DESIGN TIPS
by Hamid Aumfrey
(Acoustical Engineer for Auralex)

#1 IDENTIFY YOUR NEEDS

First of all, you need to decide what purpose your studio is going to serve. Are you going to record live instruments? If yes, what instruments? Are you planning to record several instruments at the same time? Do you want to record vocals? Do you simply need a nice control/mastering suite to work on your tracks? Answering these questions should help you get a better idea of what you are looking for in terms of size and number of rooms.

You also need to identify what level of quality you are looking for in your studio. Even though this is often limited by the money and space available, it is important to know from the beginning where you are going. For example, if you plan to produce professional-level recordings, it would be worth it to invest a little more money at the beginning of the project in order to do things the right way.

#2 RESEARCH, RESEARCH, RESEARCH

Whether you are planning to hire someone to help you, or are going to do everything by yourself, research the subject thoroughly. This will help you to avoid making big mistakes, and should prevent you from being fooled by some of the common misconceptions in the field of studio design.

The Web is a good way to start. Unfortunately, you'll come across both good and bad information on the Net — so try to match different sources that support a concept before settling on a plan of action. There are also plenty of very good books out there that you can buy (or borrow from your local library), and you can usually get some help from speaker or acoustical treatment manufacturers.

#3 FIND A SUITABLE LOCATION

Location is crucial, as it will weigh heavily on what exactly you need to do to optimize your space. For instance, if you choose to build in your home's basement, you may only need to isolate the ceiling.

As you may have noticed, I have already used the word "planning" several times — but that's because planning is essential. This includes looking at your budget, the space you've decided on, and your overall resources. You'll want to decide on what shape and size you want to build the room to maximize your sound. If your budget allows it, I would suggest hiring an acoustical consultant to help with the design of your rooms.

#4 CONSTRUCT IT

Once you have found the right place and put together some floor plans, you should be ready to get started with the construction of the room(s). This is a very important phase of the process because the level of isolation you can expect from your studio will depend directly on the quality of the construction of its walls, floor, and ceiling. Having a $2,000 "soundproof" door is useless if it is not sealed correctly and there is sound leaking all around it. It is usually helpful to think of sound isolation like waterproofing: If there is any leaking point, all the sound is going to go through this hole, no matter how good the isolation is everywhere else in the room. Beware of gaps around openings such as doors, windows, power outlets, etc.; in addition, HVAC ducts should be addressed in order to reduce sound transfer from room to room.

#5 FINISH YOUR SURFACES

There are different options for surface finishes — all with various pros and cons — and your choice will affect the way the room sounds. Carpet on the floor provides absorption in the mid to high frequency range, but it will be fairly reflective in the lower range (particularly if laid on concrete), resulting in an unbalanced room. The same type of problem happens with a drop-tile ceiling (which usually make rooms sound "boomy"). Hardwood floors are preferable, as they offer a much flatter frequency response. (In addition, psychological studies have shown that musicians perform better in rooms with hardwood floors, so that's reason enough.)

#6 CONTROL LOW FREQUENCIES

In small rooms (*i.e.*, under 10,000 ft^3), the dimensions are close to the wavelength of audible sounds, resulting in room modes (also known as standing waves) between parallel surfaces (wall-to-wall and floor-to-ceiling). Even though a proper room design can help by providing a good distribution of these modes among the frequency spectrum, you will still notice hot and weak spots at various frequencies within the room.

As all of the modes have their maximum intensity in the corners of the room, it is a common practice to use "bass trapping" to flatten the low frequency response of the space. There are different types of bass traps. Some are tuned to a narrow frequency range, whereas others offer wider band absorption. Again, some research will be necessary to figure out what will be the best for your specific needs.

#7 IMPLEMENT BROADBAND ABSORPTION

Having a certain amount of broadband absorption in your room(s) is also essential. Spreading it over the walls and ceiling in a recording space will help to make sure the recorded sounds will not be altered by unwanted reflections. In a mixing/mastering space, it will help to control early reflections that would mess up sound imaging. A good starting point would be to address the first reflection points on the sidewalls and ceiling. [*Keep in mind that it is not usually recommended to put too much absorption in a room, as it will make it sound too "dry" or "dead."*]

#8 PLACE DIFFUSORS STRATEGICALLY

Diffusion in strategic areas for recording/listening spaces is key, as it helps to spread out sonic energy while breaking up flat reflective surfaces. Diffusors can be used on the walls and ceiling to keep the room sounding natural and open. Placed above the mixing position in a control room, they can also widen the "sweet spot" for listening. The main point is that diffusors are your friends. If you don't have any, go out and meet some.

#9 TEST AND ENJOY

There are standard testing methods to measure if a room is "good" or "accurate." But remember that the best testing devices you can find are your ears and your brain. If the room "sounds" good and you feel like your mixes translate well, then you probably do not need to change anything. So give it a good listen, and then get back to the fun of working on your tracks.

RENEGADE ROOM TREATMENTS

On-the-cheap fixes to funky sounding spots

by Chris Mara

So you're at the point in your recording career where you've plunked down all the coins needed on that great gear that promises to "capture all the subtleties and nuances of a performance." The boxes keep showing up and, like most gear sluts, every time a new piece of gear shows up you call in your buddies to rush to your studio space so you can try out each new addition post-haste.

But when you plug in that awesome new mic into that high-end boutique pre, all those "subtleties and nuances" make it glaringly apparent that your performance area just doesn't sound that good. It's not like you can just build another studio or spend $10,000 treating and tuning a room — so what do you do?

DO NOTHING FIRST

Borrowing from the old medical adage: First, do no harm. Before you start assaulting your studio walls/ceilings/floors, take the time to really understand your tracking space and really pinpoint the problem areas. Unlike control room treatments (which focus mainly on creating a flat frequency response in the mix position), performance area acoustic treatments generally fall under the lines of absorption and diffusion. The prime objective is to get a nice, good sounding decay without such traits as flutter echo and slap back.

As in all scenarios, "good" is relative. The needs of your room depend on the room's projected uses— so take some time to rough cut some tracks using the instruments, playing in the particular styles you foresee yourself most often recording in your space. It's important that you document these experiments: Make notes of placement of sources and mics, so you can listen back to your test sessions and make some educated guesses as to where your room's issues lie. Trust me, this will save you much more headaches than just buying some overpriced "acoustic foam" and plastering it randomly all over your room.

TESTING 1 . . . 2 . . . 3 . . .

You know you have some issues to address, but before you commit to a long-term, costly fix, I recommend trying out some cheap, temporary solutions. Drape some blankets over the wall that might be causing you trouble, or stick an overstuffed chair in a corner that gets a little weird when you're cutting drums. Throw a rug or two down to deaden your floor somewhat if it's a bit too live for your taste. Remember, these are just to test out your theories; they aren't as effective as real acoustic treatments. Record with the same placements from your first test run, listen for improvements, and move these "fixes" around as you deem necessary.

WATCH AND LEARN

Phase three is to start working on implementing permanent solutions. As we're working under the assumption that you don't have an unlimited budget, we'll forego instructing you to call in acoustical engineers. While the above option is great (better than doing it yourself, that's for sure), lacking the monetary resources to call in help like a professional studio owner doesn't mean that you can't use your own two eyes and ears and learn from their successes.

Whenever possible, I like to learn from other people's mistakes. For example, my little brother taught me that it's not really a good idea to pound a nail into a spray paint can. I learned a valuable lesson from watching his mistake and watching the solution (washing his face with turpentine over and over). I encourage you to take a similar approach and study some studios in your area, taking note of materials used and methods implemented. Ask some questions: Did that area have flutter echo areas? Was that problem solved with diffusion or absorption? Take your notes and return to the home front, ready to apply the knowledge you gained to your own room.

IMPROVISE AND CUSTOMIZE

Take a cue from HGTV: Look for second-hand materials to use, and don't be
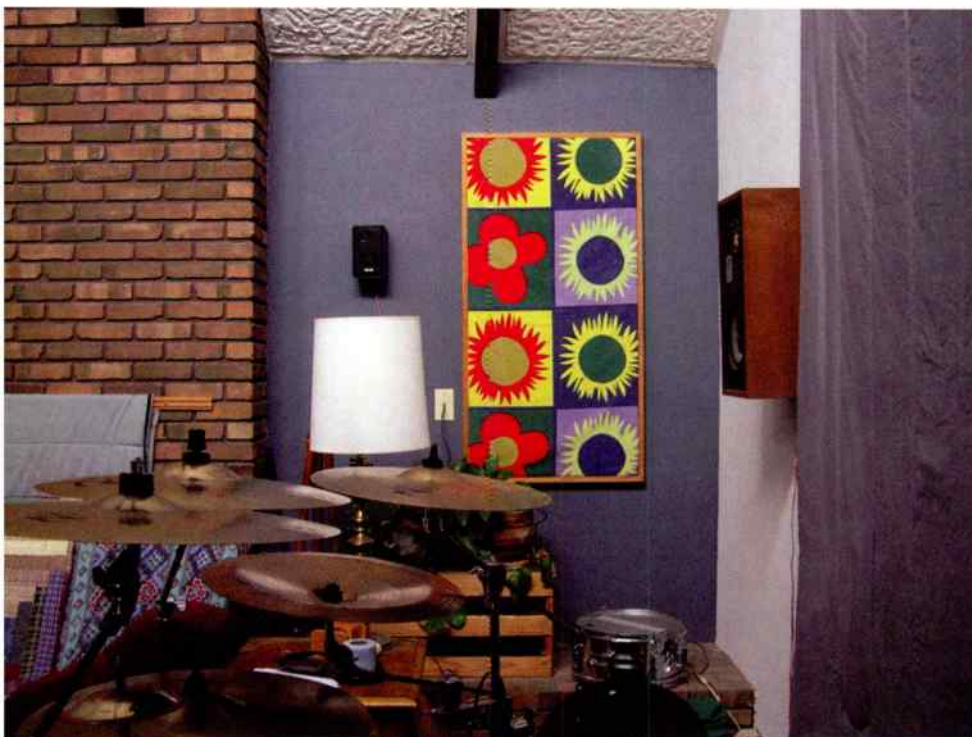
Fig. 1: There's no reason you can't get a little creative with your acoustic treatments, and give your studio a bit more vibe. (Photo by Marshall Green)

afraid to get creative in your presentation. Remember, artists tend to like recording in rooms that have vibe. Need to fix a problematic wall? Build a "shadow-box" style frame 2' wide by 4' long, slide several of those tube-shaped decorative pillows into it (couch cushions work as well), and hang it up. Not only is it effective; it's portable and can be slid around easily to fine tune your room. You can also frame a sheet of 703 insulation, cover it with canvas, and then paint a few designs on it (Figure 1) — cool looking and functional!

Sure, absorption is easy, you say, but what about diffusion? Another thing I've learned from home improvement shows is that room dividers work great as diffusers. I once saw an old episode of *This Old House* a few years ago where they made a room divider out of 1" galvanized pipes, attached from the floor to the ceiling about every three feet. They had cut 1/4" plywood into 2' by 8' strips and weaved

Fig. 2: You can attach acoustic treatment to a wall with hooks (see inset) so it's easy to detach and reattach. This also makes any panels easy to flip around if you have a different type of treatment on the other side, thus giving you two types of treatment with one panel.

these strips through the vertical pipes (much like a basket weave) — all for less than $80 in material costs. Not only is that a great idea, but you could expand

on it to suit your needs: Glue some soft fabric to help treat excess high frequencies, and even break out the paint brush again and customize it for your room. This will address your room's sonic issues and make your space look unique. As said before, vibe is an important component in getting great cuts out of the band you're tracking, so make your room look cool. It'll make your workspace much more comfortable to be in during those 16-hour sessions that stretch through the lonely hours of the night.

THINK FLEXIBLE
Another thing to keep in mind as you start building your unique acoustic solution is that you can construct project pieces that are versatile by just changing up materials used on different surface sides. After all, if you're like most studio owners, you work on a variety of different projects in any given month, and therefore you'll need to liven or deaden your room a bit from project to project.

For example, if you go with the picture frame/shadow-box treatment, adding a backing layer on the opposite side of the box that is more/less absorptive than the front can give you added control over your sound. If you're using canvas as your front, try denim on the back for more absorption, or cotton for a less absorptive side. This way, you can simply flip the panels around as needed, or even attach them to the wall via hinges or hooks (Figure 2) so you can swap textures quickly and easily.

NO SET RULES
Acoustic treatment, whether done via the DIY route or by hiring help, is all trial and error. I don't care what anyone says: You won't know if it works until you get it in the room and hit "record." Even professional acoustic designers with years of experience have to experiment to get a room just right. So try, test, and improvise your way through your project (Figure 3) — a few hardware store excursions and some quality time with your power tools is all it takes to fix up a room. If you keep an open mind, and bring a good dose of that creative spirit that got you into this business in the first place, you'll be surprised at the quality of results you can get. EQ

Fig. 3: Here's a great example of tweaking a room with a temporary solution even after some room treatment has been done. Not only does the quest for perfect acoustics never really end, the definition of "perfect" can change from project to project. (Photo by Marshall Green)

FIXING ROOM RESONANCES — AFTER THE FACT

Sometimes you really CAN "fix it in the mix"

by Craig Anderton

After all that's been said about acoustical treatment, you'd think that anyone who was halfway serious about recording would take some steps to improve their room's acoustical properties. However, recordings are still being made where unwanted room resonances are part of the tracks. In a situation where a mostly acoustic recording is done with multiple tracks recorded in the same room, and there are resonances, these frequencies may build up over the course of several tracks to add some serious peaks in the final mix.

You can try to fix this during the mastering process; you'll need a very precise digital filter to do so, and I use the Har-Bal program (basically a 8,192 stage FIR filter optimized for mastering) for this type of application. However, if you're handed tracks like this to mix, sometimes you can get rid of the resonances during the mixing process itself, which can yield better results. This involves finding where the resonances lie, then processing each track to get rid of those resonances.

SO . . . HOW BAD IS IT?

As you're setting up tracks to mix, some resonances might jump out at you if the acoustics are really bad. What's more, you may hear this on all tracks that were recorded in the room, unless they were close-miked or recorded in significantly different ways.

If possible, it can be helpful to record a sweep tone in the tracking room to give you a very rough idea of where the resonances might lie. Figure 1 shows one such sweep, which was then brought into Har-Bal for analysis. The green line shows average signal level, while the yellow line shows peaks — which are basically identical, as

we're simply looking at a sine wave, not a complex signal.

The major resonance areas are outlined in red. Granted, there are plenty of other peaks and dips, but these aren't as severe.

THE BIG FIX

Har-Bal is a stand-alone application, not a plug-in. Therefore, it's easiest to have linear tracks to work on, as opposed to lots of smaller segments. This may require exporting portions of the mix to create tracks that last the duration of the song. That way, when you re-import the tracks back into the song, you can insert them at the same

Fig. 1: A room sweep tone shown in Har-Bal. The most problematic resonance areas are circled in red.

Fig. 2: This curve compensates for the resonances that were highlighted (the original curve is shown in gray, in the background); the curve can then be saved and, as a point of departure, applied to other tracks recorded in the same room.

start time and know they'll sync up. (Some hosts will let you call up an editor to edit a particular track, in which case you can use whatever EQs you normally use to do surgical-type EQ.)

Then, before even thinking of doing any processing, look/listen for resonances the tracks have in common. But be careful, because lots of resonances are part of a characteristic instrument sound; you don't want to mess with these

Once you've identified the main peaks, work on each track individually. Har-Bal has a function where if you come up with a particular compensating EQ curve, you can save it and apply it to other tracks (Figure 2). If the curve is gentle enough, this can provide an excellent point of departure for doing finer tweaks, but *always* listen carefully to double-check — a track may have been recorded differently, and have different resonance characteristics. With other programs (I've also used this technique with iZotope's Ozone 3), you can almost certainly save any EQ preset you come up with and apply that as well.

TIME TO MIX

Now you can re-import the tracks into your project, and begin your mix. You may be surprised how much more easily a mix comes together if the tracks have been properly treated for resonances; the more tracks that were recorded in the same room, the more of a difference this will make. A side benefit of this technique is that by reducing built-up peaks, you'll have more headroom available for the mix, and will be able to get a "louder" sound without having to resort to as much compression or limiting.

So the next time you need to mix a project that was recorded under less-than-ideal acoustic conditions, try fixing the tracks before you mix — it might make the difference between "okay" and "yeah!" EQ

SONY CREATIVE SOFTWARE SOUND FORGE 9

Surround? New mastering effects? Sure . . . but there's more

by Craig Anderton

Just as some people bought an Apple II to run Visicalc, some people bought PCs in the mid '90s to run the Windows-only Sound Forge. It was more powerful than its contemporaries, yet had a clean and simple user interface — two crucial qualities when you're staring at a computer screen all day long.

Since then, Sound Forge (SF for short) has faced some stiff competition. While periodic updates have satisfied the faithful, in other respects it was falling behind. However, SF9 offers several important new features. Furthermore, it's Vista compatible, bundles in the formerly optional-at-extra-cost Noise Reduction 2 plug-in, includes the CD authoring program CD Architect 5.2 (also Vista-compatible), and is priced aggressively. As you can download a demo to check out all the features, we'll concentrate on what's new.

MULTICHANNEL EDITING

"Multichannel" doesn't mean an Audition-type multitrack window, but something more like Wavelab's "montage" feature — although montage is implemented as a separate mode, with limited support for importing or rendering multichannel media files. SF9 now accommodates up to 32 channels (16 stereo), which is obviously aces for surround project editing. But it's also ideal for those with, say, a couple ADATs who want to bounce all their tracks into a computer. Record them into SF9 then edit heads and tails, add processing, or whatever (Figure 1).

And if you export each track in a multi-track project, you can drag and drop them into SF9 to do DSP that's likely more sophisticated than what you can do in most host software. Furthermore, when you drag in sounds, they can be mixed and crossfaded (choose from 25 crossfade curves) with existing sounds; this also simplifies doing album assemblies with complex crossfades or layers. (Formerly with SF the only way to mix down multiple files into a final stereo mix was to copy each track to the clipboard and mix it into a destination file — but no more, thanks to the Channel Converter feature mentioned later.)

What about surround authoring? SF9 doesn't do DVD-A authoring, although given the format's status, I doubt that many people care. It can, however, export in multichannel WMA and WMV formats, and SF9 also includes the AC-3 Studio Encoder (although if you also have Vegas installed, SF9 can use its AC-3 Pro functionality). For stereo, you can save to the usual suspects (WAV, AIF, MP3, etc.) but also do AAC (MP4), Ogg Vorbis, Sony MXF, and ATRAC (my nominee for most underrated music compression format). But read the fine print: MP3 files can't be distributed in revenue-generating applications (you need a separate license from Thomson for that), and the rules for Dolby Digital (AAC, AC-3) are similar — Dolby says you have to use "approved Dolby Digital Professional Encoders" to create content intended for commercial distribution.

In terms of effects, there's nothing like Sonar's "BitBridge" surround technology that applies any effect across multiple channels; however, nearly all of the included effects (except the iZotope effects, Wave Hammer, Noise Reduction 2, and Acoustic Mirror) can work with multichannel audio files as well as mono/stereo files. Also, as of version 9.0a, multi-channel VST plug-ins are supported.

MASTERING EFFECTS

Aside from the addition a few years back of Wave Hammer and Acoustic Mirror (one of the first convolution reverbs), SF's lineup of plug-ins hadn't changed much. Now Sony has teamed up with iZotope, and while you don't get the full Ozone mastering suite (which costs $249.99, so it's not surprising it's not bundled in a $399.99 program), you do get four excellent DirectX effects — all of whose parameters are automatable, which is a Very Good Thing.
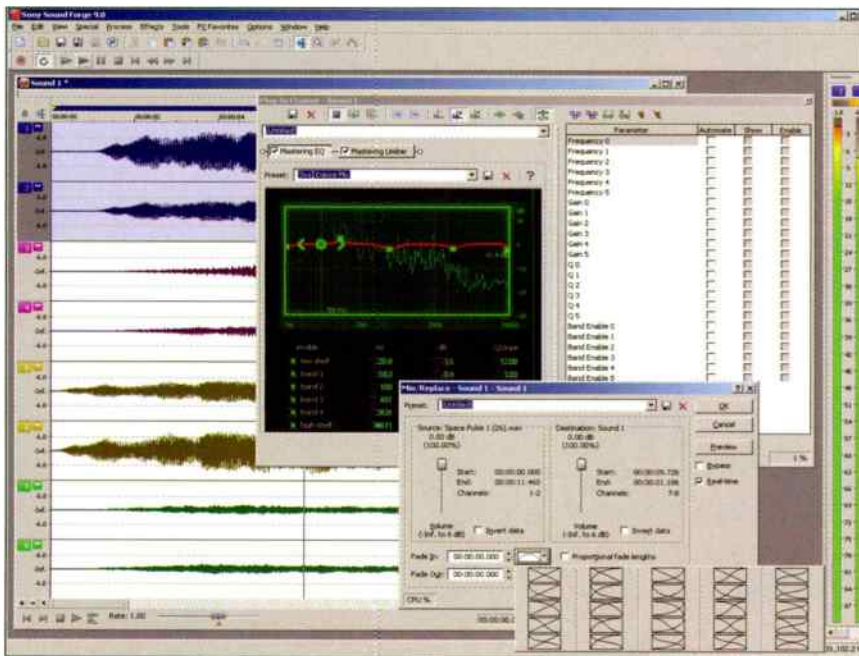
Fig. 1: Sound Forge 9 editing eight channels of audio. iZotope's Mastering EQ is toward the center (automatable parameters are toward the right); the dialog box in the lower right appears when you drag and drop audio on top of another piece of audio. Note the multiple crossfade types for fade-in and fade-out.

The Mastering EQ plug-in has high and low shelf (with adjustable frequency, gain, and Q), four fully parametric bands, and spectrum analysis in the background. Mastering Limiter has threshold and margin parameters, as well as a "character" control and (cool!) an option to prevent inter-sample clips.

The Mastering Reverb adds ambience to sources like narration, and performs other quick "gimme some space" applications. The final member of the quartet, the four-band Multiband Compressor (Figure 2), is very useful, sounds good, and looks good. These plug-ins are "keyed" to SF, and don't show up in other programs — even CD Architect, although Sony says CDA compatibility will be added in the next update (also, other Sony Creative Software DirectX hosts, like Acid Pro and Vegas, will be updated to show these effects).

The Noise Reduction 2 plug-in is also bundled with SF9. And to go along with the new effects, you can now alter the wet/dry amount more easily, but more importantly, fade an effect in and out over time using various fade-in/fade-out curves. I also like that you can choose whether "spillover" sounds like reverb tails can either stop at the end of the selection, get mixed with the file past the selection, or push the selection to the right to make room.

OTHER GOODIES

The Channel Converter seems tailor-made for multichannel projects, as you can downmix to various other formats (mono, stereo, quad, 5.1, 7.1). A dialog box lets you specify the percentage of each channel you want in the final mix, and whether you want to invert a particular output. For example, you could send one channel to the center channel of stereo mix by sending 100% of it to center and 0% to anything else, or add in a little "leakage" to left front and left right to have the center channel sit better in the sonic panorama. And, the Channel Converter upmixes: You can take, say, a stereo file and expand it into a 7.1 mix by putting various amounts of the original file into the eight possible channels.

SF9 also has new diagnostic tools. The Spectrum Analyzer (standard or sonogram display) can handle multichannel files, but one of the features I appreciate the most is the Mono-Compatibility Meter, which shows if mono playback will produce phase cancellation issues; a Phase Scope offers Lissajous and Polar plots (linear or circular).

Finally, there are workflow enhancements: More keyboard shortcut customization options, drag-and-drop pasting and mixing between channels, and more display

Fig. 2: The Mastering Multiband Compressor has a colorful, and comprehensive, interface.

preferences. Interestingly (wave of the future?), SF9 can link directly to Sony Music Studios Internet Mastering Services, making it easy to upload files for mastering.

APPLYING SOUND FORGE 9

I had just transferred a bunch of old songs from DAT, and wanted to do a "test" mastering job to make them a bit more consistent. I also wanted to create an audio CD of all the sample CD examples I've created for EQ over the years.

Many people swear SF is a dead-simple program to navigate, while others claim they don't "gel" with it. As one of the first pro editors, SF developed its own workflow and by this point, I'm totally comfortable with it. As just one example, one of my favorite features is that if you select a section to cut, and hit Play while holding Ctrl, SF does a pre-roll before the selection, skips the selection, then plays a post-roll after the selection This makes editing sooo easy. To me, this operation is second-nature; but those who don't know it's there might think it's hard to evaluate splices in SF without actually doing them.

The mastering plug-ins are extremely useful. In particular, the Limiter — if not abused — can give very musical results. About the only plug-in I miss from the full Ozone mastering suite is the multiband harmonic exciter plug-in, but admittedly, I don't use it much.

I also wanted to restore a couple cuts from LPs I'd done from decades ago where the master tapes could no longer be found. I had DATs from the house that had done a transfer, with restoration, from vinyl for a CD re-issue; but there were still a few vinyl "blemishes." One was particularly problematic: A major crackle in the middle of the word "stay." Getting rid of it killed the sibilance of the "s," but by zeroing in with the Audio Restoration tool on just the crackle and using

minimal reduction, it was as if the crackle was never there — check out the example at www.eqmag.com (as well as an excerpt from a song before and after mastering with SF9).

When it was time to create the CD, I opened up CD Architect and popped in 26 tracks of examples. I made a few subtle volume adjustments, but one cut was markedly lower than the others so I used a bit of Wave Hammer. My burner won't let me test and burn with CDA (I have to test first, then burn); while testing, CDA pointed out some possible digital clipping (who, me?), so I dropped the master volume down 0.1dB. I hit burn, and within a few minutes, the CD was done.

CONCLUSIONS

Upgrading is a no-brainer if you do any kind of multichannel editing, and many would likely spring for the upgrade just for the iZotope effects. If you don't do multichannel and already have Ozone, it's a tougher choice; however the extra test gear is great, and the workflow enhancements are helpful.

As to the Wavelab/Audition/SF question, much depends upon what you need. For example, both Wavelab and Audition have "spectrum editing" where you can edit specific frequency or volume ranges on a Sonogram display, but neither has the iZotope effects or some of SF's export options. Also, Wavelab lists for $699.99 — a big price difference — and requires a dongle. Fortunately, demo versions are available for both Audition and SF, so you can resolve that issue yourself.

Personally, I'd been using SF less recently, but SF9 is changing that. It's a fast, painless, efficient, and simple way to get work done that also represents good value — qualities I *always* appreciate. EQ

PRESONUS FADERPORT

And who doesn't want more control over a session?

by Craig Anderton

In the days of large multitrack tape recorders, one of the most common accessories was a remote box with a thick umbilical going back to the recorder. The Faderport (for Windows XP and Mac OS X Universal) harkens back to those days, as it's not a "control surface" for tweaking synths and effects; instead, it's optimized for machine control, with the added bonus of being able to write and punch automation moves for volume, pan, mute, and solo.

Faderport is small but substantial: heavy enough not to slide around on your desk, and with 24 buttons that could handle even the touch of a staff engineer in the '70s who hated his job. The 100mm moving fader with 1,024 steps of resolution has a smooth, "pro" feel (it's the same Alps fader used in Digidesign's D-Command), as does the detented 360° encoder knob for panning.

The Faderport connects to your computer via an included 6' USB cable, but also requires an AC adapter as the USB bus doesn't provide enough power for the motorized fader. If you're willing to forego the fader, the Faderport can be bus-powered.

GETTING STARTED

The package includes a driver CD, but as expected, you're better off going to the website and downloading the latest drivers. The FaderPort uses a custom Windows driver for Cubase, Nuendo, and Sonar, as well as for Digital Performer on the Mac. Otherwise, it supports Pro Tools (including M-Powered), Logic, and Cubase/Nuendo (Mac) as a HUI controller — but check the website for the latest compatibility info, including supported versions.

However, not all implementations are created equal. While the transport and automation functions work with everything, there's a User button with custom functions that doesn't work in HUI mode (nor does the Output button, which otherwise moves from selecting tracks to

selecting the output section). The User button can select key bindings in DP and enables/disables punch out in Cubase; Sonar has five custom mappings you can do with the User button and various combinations of other buttons.

APPLYING THE FADERPORT

There are four main elements to using the FaderPort:

- **Transport buttons.** These are the usual Play, Record, Stop, Rewind, and Fast Forward functions, along with Punch and Loop. However, a shift button allows some buttons to do double-duty, like go to Start/End, Redo, and even drop Marker or go to Previous/Next marker.
- **View buttons.** These call up various DAW views.
- **Undo button.** With shift, it also does redo
- **Automation fader and mute/solo buttons.** Fader mode buttons are Read, Write, Touch, and Off. Additional buttons select the channel (track) or bank you want to work with, and there are Mute, Solo, and automation Record buttons.

When working with automation, selecting tracks involves hitting arrow buttons. But as communication with the host is bi-directional, you can select something

at the host and the FaderPort will follow along (the buttons illuminate to show when something is selected). Because the fader is motorized, with Touch mode selected it will follow existing automation moves until you grab the fader and move it; otherwise you can use Write and just overwrite whatever's there, or make new moves.

Although I'm a big fan of key equivalents and use them often, I nonetheless found it a whole lot easier to have large, dedicated, lit buttons guiding my way. The FaderPort is good about letting you stay in "right brain" mode, and it takes minutes — not hours — to get comfortable enough to fly around the buttons. Even the shift button works logically and doesn't get in the way of efficiency.

CONCLUSIONS

Let's crunch some numbers. Assuming you buy the FaderPort for $200 and value your time at $40/hour, it pays for itself once it saves you five hours. If you're drawing envelopes for automation, you'll save that much in short order — and get more musical results, too (some might argue, but I appreciate the human touch). Even if you use it solely for transport control and window selection, you'll find that having a dedicated control surface simplifies your workflow — and having a sturdy, cost-effective one working on your behalf is even better. EQ

Product type: Recorder remote with motorized fader for automation.
Target market: Studios with hard disk recording systems but no controller.
Strengths: High-quality moving fader. Very solid feel. Saves time. Small footprint.
Limitations: Supports a limited number of DAWs. Can't reassign fader/pan to other controllers.
Price: $229.95 (list)
Contact: www.presonus.com

WAVES MAXXBCL

So this is what they do for an encore

by Garrett Haines

Many of us were con-
founded when Waves
announced they
would no longer make
the L2 Hardware
Limiter — the closest
to a "must have" unit
for mastering engi-
neers as anything pro-
duced in the last
decade. Was it
because of the people who abused it and
made it sound wretched?

But then the MaxxBCL was unveiled.
Taking the L2 Ultramaximizer/IDR dither
and adding the proprietary MaxxBass pro-
cessing, digital converters, and, for the
first time, a hardware implementation of
the popular Renaissance Compressor (RC
for short) — and a cornucopia of digital
connectivity options — the MaxxBCL is a
serious overhaul. So, let's see if this next
generation box can beat its predecessor at
its own game. . . .

OUT OF THE BOX

Before turning on the MaxxBCL, you must
configure the unit for your local power
supply (100, 110, 220, or 240VAC) using an
adjustable fuse unit — choose correctly
before proceeding.

Around back, the unit looks like a digi-
tal United Nations: Optical, coaxial S/PDIF,
AES/EBU, whatever — and just flick a tog-
gle for word clock termination. The analog
ins/outs use Jensen transformers (Waves
states that future models will be outfitted
with their own brand of transformers),
and can handle signals from 9 to 24dB in
3dB steps. Ground lift is also a toggle
switch option, as are balanced and unbal-
anced inputs.

The MaxxBCL, at 8.8 lbs., has a solid
feel. The input gain knobs and rotary con-
trols are stepped; illuminated buttons make
it easy to tell which mode is selected, and
the multi-colored meters are discernible in
low-light situations. Unlike the L2, Waves
moved the power switch to the front panel
— great, because no one wants to crawl
behind a rack to turn a unit on or off

THE MAXXBCL IN USE

Having always been a fan of the RC, I was
excited to drop the MaxxBCL into my
mastering chain and take it for a spin. In a
rather cool twist, the unit was designed
not only to allow the user to bypass any of
the processes, but to choose whether the
MaxxBass or the Renaissance
Compressor comes first. However, the L2
Ultramaximizer always comes last which,
given the dithering options at the chain's
end, makes sense.

The input gain knobs provide a good
amount of gain, and even at maximum
boost there is no hint of noise or harsh-
ness — something I attribute to the
Jensen transformers. The MaxxBass
works like the plug-in, as it boosts har-
monics to improve the perceived low end.
However, with the plug-in, I almost always
encounter internal clipping issues; thank-
fully, the MaxxBCL's 48-bit internal pro-
cessing handled nearly every source I
threw at it.

Tweak-wise, the RC differs slightly
from the plug-in. There are no release
controls, and the attack choices are fixed.
As most people use the RC in Auto Mode,
these controls are of little concern. But
for mastering I want as much attack and
release control as I can get. This is a
minor quibble, but worth noting. The L2
and IDR dither section work exactly like
the hardware version. If anything, the unit
is slightly more transparent when used as
a converter

Speaking of converters, the A/D and
D/A are stellar. They were more open,
alive, and believable than the original
unit. In all my tests, the MaxxBCL

sounded better than the
converters in our Pro
Tools HD rig, and nearly
as good as my Crane
Song unit. (I say "nearly
as good" because at the
highest levels there is
so much subjectivity; in
any event, the
MaxxBCL's converters
are in the top of the cur-
rent class, for sure.)

I used the MaxxBCL as an A/D on a
voiceover session (in front of the
DigiDesign Mbox 2 Pro) with great results.
From good mic to good pre, the output hit
the MaxxBCL's A/D and, from there, I
added a touch of RC and just enough L2 to
keep things tight. This also used the
MaxxBCL as an A/D for vinyl transfer
(from a dedicated phono preamp). I
bypassed the MaxxBass and RC, calling
on the L2 only to prevents overs. The
results were pristine.

CONCLUSIONS

The MaxxBCL is not the sequel to the L2;
it's a totally different tool. With incredible
connectivity choices, it's a unit for the cur-
rent PCM era. Having the Renaissance
Compressor and MaxxBass is a joy, espe-
cially in a unit that uses 48-bit double pre-
cision processing and boasts 125dB of
dynamic range. Add the reliability of the
L2 limiter, IDR dithering, and state-of-the-
art converters, and the MaxxBCL could
well be on its way to being the next
"must-have" unit for mastering engineers
. . . and more. EQ

Product type: Bass enhancer/com-
pressor/limiter and more.

Target market: Primarily mastering
engineers, but also tracking/mixing engi-
neers.

Strengths: Great sound. Loads of fea-
tures. Flexibility of use.

Limitations: No release controls.
Attack values are fixed in Renaissance
Compressor.

Price: $3,200 (list)

Contact: www.waves.com
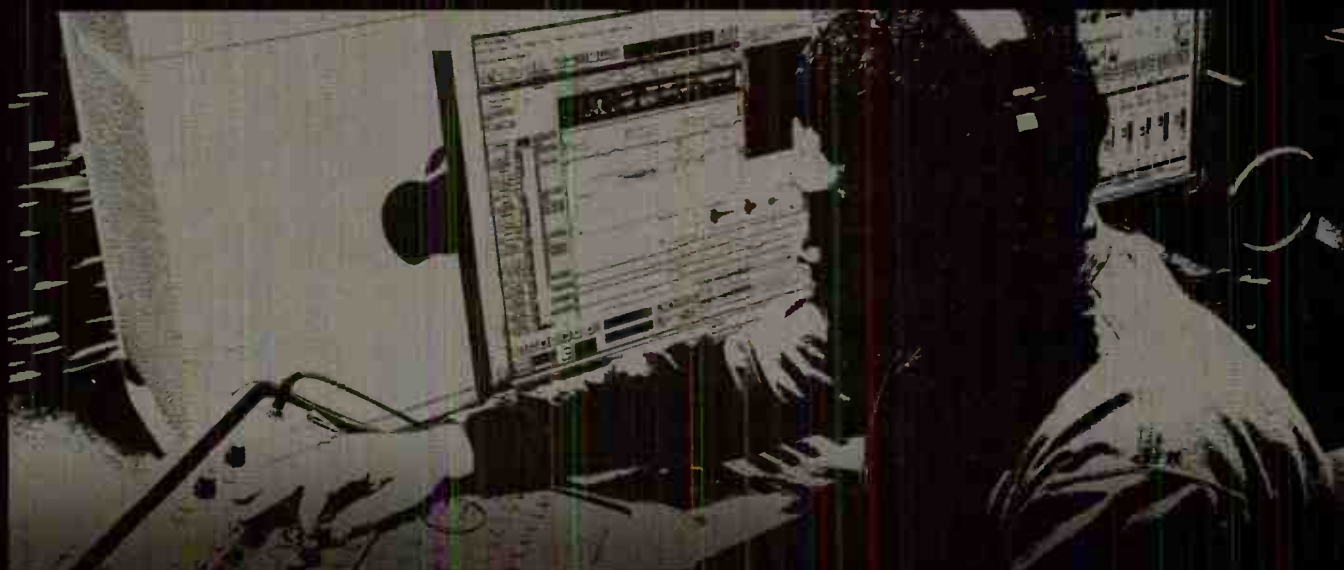
DIGIDESIGN MBOX 2 PRO

The Mbox line grows up

by Garrett Haines

In the last few years, Digi's famed Mbox line in particular has undergone numerous changes — growing from a single device to a whole family of portable LE interfaces. From the ultra-compact Mbox 2 Mini to the new Mbox 2 Pro, users have been offered a range of Mbox interface choices. But how does this new-fangled design hold up? What distinguishes this particular piece from the rest of the pack and, more importantly, how does this upgrade apply to you?

OVERVIEW

Presently, the Mbox 2 Pro is the most feature-rich (and physically largest) unit of the entire family. While the rest of the line uses the USB 1.1 protocol, the Mbox Pro 2 is FireWire-based — taking advantage of the bandwidth increase compared to USB 1 to support up to 24-bit/96kHz resolution. *(Note: For laptops or desktops that have 4-pin instead of the full 6-pin IEEE-1394a FireWire connectors, users will need to use the supplied 12 volt DC adaptor — test for this before taking it on location.)*

For most recordings (done in 24-bit 44 1 or 48kHz), the Mbox 2 Pro supports six simultaneous inputs and eight simultaneous outputs. Other new features worth mentioning

> **I was impressed by the unit's heavy-duty design . . . it stood up to some serious field testing — making the Mbox 2 Pro a great companion for remote assignments.**

are MIDI In and Out and a BNC word clock I/O. A surprising, and very useful, addition comes in the form of a set of RCA phono preamp connectors (see sidebar). This makes the Mbox 2 Pro a great choice for archival, sampling, or DJ work.

I was really happy to see the small security receptacle on the side panel. This is the

same lock slot found on laptops and other high-value electronic gear and while not foolproof, it's a significant deterrent against would-be thieves.

Above that, I was impressed by the unit's heavy-duty design. I had expected the knobs to get ripped off within two weeks of no-punches-pulled, on-the-road recording, but it stood up to some serious field testing — making the Mbox 2 Pro a great companion for remote assignments.

APPLYING THE MBOX 2 PRO

On the subject of remote assignments, I used the Mbox 2 Pro to record a live concert with Irish troubadour Mark Dignam. The first thing I noticed was that the two pres were pretty nice sounding: clear, relatively uncolored, and giving a good bit of gain before getting noisy. Likewise, the instrument inputs are solid, with a touch of midrange that proved to be flattering on Mark's Taylor acoustic (and in other tests with a Yamaha bass loaded with EMG pickups).

The headphone jacks seemed to be more powerful than the original Mbox, and were definitely less noisy at high volumes. Having individual adjustment options is a plus, especially when tracking multiple players. But having MIDI available with my LE interface was probably one of my favorite features. As an original Mbox user, I've always needed an additional interface to hook up sound modules. That's now a thing of the past. At about 10" wide and 9" deep (excluding rear connectors) the Mbox 2 Pro fits beneath most laptops, and helps declutter most MIDI setups.

DJs, remixers, and people who want a copy of their vinyl transferred to CD will be happy about the phono pres. There's even a grounding post — a prerequisite for phono work. We do a bit of preservation work at Treelady Studios, so I can say the Mbox 2 Pro doesn't have the rich sound of a $900 Acoustic Research phono preamp, but it's as good or better than the common $50 Rolls phono pres I see in many studios.

I also used the Mbox 2 Pro as a digital-only interface. Recording a voiceover spot, I used a Millennia Media TD-1 going into the new Waves MaxxBCL as my A/D. I connected

the S/PDIF out of the Waves into the Mbox 2 Pro, and added some BNC cables to handle word clock issues. The team worked seamlessly, but there is one thing to remember: The Mbox 2 Pro cannot be used as the clock source when choosing a S/PDIF input — so it's important to make sure your digital source is capable of handling clock duties, or you'll need a dedicated system clock.

The only real issue I had was a buffer *underrun* when recording sessions with a single mono track. Many of us have experienced buffer overruns; these happen when the audio data is coming in faster than what the interface/computer combination can handle. My problem was the exact opposite — which is fairly humorous when you think about it.

The Mbox 2 Pro is essentially saying, "I'm much too powerful for your silly project. Why don't you come get me when you need to track a band?" Fortunately, Digidesign's tech support was able to help me remedy the problem, so no real complaints here.

CONCLUSIONS

Mbox users have been waiting for quite some time for increased throughput, MIDI ports, and improved mic preamps; the Mbox 2 Pro delivers these, and a good bit more. From multiple lines out to the phono preamp stage onwards, this unit packs a lot of flexibility in a relatively small package. Original Mbox owners will want to look into this unit for the improvements in the headphone

amps alone. The short of it: This is a roadworthy, resilient unit that offers greater mobility and flexibility for Pro Tools LE users than its ancestors. My only complaint is that now I just can't wait to look at those 003 units. EQ

Product type: Hi-def portable audio/MIDI interface/production system.
Target market: Engineers looking for a multitrack mobile recording system.
Strengths: Roadworthy. Increased feature set including good sounding pres/headphone jacks. MIDI ins and outs. Built-in phono pre. BNC word clock in and out.
Limitations: Nothing significant.
Price: $799 (list)
Contact: www.digidesign.com

PHONO INPUTS AND THE RIAA CURVE

You may have seen gear with a "phono input" before. This is more than a marketing term: A dedicated phonograph circuit is needed for correct playback of vinyl records. (Sure, you can plug a record player into any line-level RCA jack, but the sound will be thin and tinny.) Since the 1940s, records have been manufactured using a "pre-emphasis" equalization curve. To reproduce the original recording, you need a way to apply inverse curve to undo — or deemphasize — the original equalization. The reason has to do with a problem of basic physics, and a devilishly ingenious way to overcome it.

WHY THE CURVE?

By the 1940s, the predominant consumer format was a rigid platter made from the secretions of the shellac beetle. These discs were played at 78 rpm (revolutions per minute), earning them the nickname "78s." (Actually, a lack of mastering standards meant playback speed ranged from 71.29 up to 80 rpm. Different manufacturers cut masters at different speeds.) Shellac discs were resistant to scratches, but very brittle, making them susceptible to breakage in shipping and everyday handling. In general, a 78 could hold one song per side. (That means you would need 8 to 10 discs for a contemporary 18 song hip-hop release.)

It did a record company no good to mail a disc to a radio station only to have it arrive in pieces. So vinyl records were developed for promotional, or "throw away," use. Unlike shellac discs, vinyl records could be shipped without fear of damage. And though they were less resistant to scratches, vinyl's physical flexibility quickly won over critics and consumers alike.

Meanwhile, some clever mastering engineers figured out a way to improve fidelity and fit more than one song on each side of a platter. First, the cutting process was adjusted to slow down the playback speed from 78 to 33-1/3 rpm. This extended playback time a bit, however something still had to be done about the groove size. Consequently, a pre-emphasis equalizer was applied to the cutting signal. The exact amount of boost and cut was not arbitrary, but nearly every company releasing vinyl had their own way

of doing it. EMI, HMV, Columbia, Decca FFRR, NAB, Ortho, BBC transcription, and numerous others developed individualized methods. (First the 78 speeds, then the repro curves! Notice a theme? No standards. Welcome to the music industry.) This proved somewhat problematic, so the RIAA (Recording Industry Association of America) established a standardized pre and post curve, earning it the name, RIAA curve.

HOW DOES IT WORK?

As audio engineers, we know that low frequencies are larger (have a higher amplitude) than high frequency sounds of the same perceived loudness. To cut a groove big enough to reproduce bass sounds caused a few problems. First, they took up more space, which decreased the amount of music that could be placed on the side of a disc. Second, large bass grooves made it difficult for the playback cartridge to follow, which would lead to high distortion, skipping, and perhaps even launching the stylus off of the platter. High frequencies also posed concerns. When cut at their standard size, high frequency grooves are very tiny. During playback, the physical contact between the stylus and the groove creates high frequency noise. This unwanted noise would leak into playback of treble and quiet parts, reducing the overall quality.

To resolve this problem, mastering engineers began to artificially lower bass frequencies as well as raise high frequencies during the cutting process. The smaller bass grooves allowed more music to fit on each disk. Upon playback, the curve is reversed, so that high frequencies are deemphasized and low frequencies are boosted. In a properly setup system, the result should be a reproduction of the original 'flat' master recording. This meant more running time because of smaller bass grooves. Meanwhile, the larger high frequency grooves yielded a better signal/noise ratio as the noise is reduced by the playback curve.

So next time you're at a party, throw out this little story about the RIAA curve. If the girl/guy you're talking to sticks around for the whole thing, you know you've found a keeper.

STEINBERG SEQUEL

Consumer toy? No — this program's a contender

by Craig Anderton

So why would any *EQ* reader be interested in a program that seems interested mostly in beating Garageband at its own game? Well, just read these words from the online FAQ.

"When you buy Sequel, you automatically purchase the right to use the contained sounds and loops, so you can use them without limit in any production."

And we're talking 4,500 loops (plus another 500 when you register), along with the ability to cut your own hard disk tracks, use loops from other sources, and play the 600 included MIDI instrument sounds from a MIDI keyboard.

If this sounds to you like a music library where you get to call the shots, well, you're right. And if it sounds like a big honkin' loop library, it's that too. If you need to throw music together fast — whether it's for backing tracks, songwriting, audio for video, or just for the sheer inspiration of it all — keep reading. And also keep reading if you're into groove-oriented music, because it has all the elements needed to put together totally credible groove-type projects. It may be priced like a "lite" program, but it most certainly isn't.

There are some familiar elements: It has a "one screen interface" like Ableton Live, a philosophy like Cakewalk's Project5, the panache of Garageband but with cross-platform operation, some Acid in terms of simplified audio stretching, and the Cubase lineage. Nonetheless, the end result is unique; Sequel weighs in at under $100, and seeks to stake out its own claim in the world of music making. It's arguably simple, but by no means simplistic: Go to www.sequel-music.net, click on Check Out Sequel, then click on Geek City for tech specs on import/export formats (you can even export to iTunes so it shows up in your iTunes library), resolution, and the like.

This marketing is pretty Eurocentric, and appears aimed primarily at the people who would rather *make* the music they hear at clubs than *dance* to it (even the graphics remind me of the flyers you see for underground dance clubs in places like Germany). As someone who spends a lot of time in that musical world, I was taken aback when I went to the Steinberg booth at Musikmesse and a Yamaha representative said "You've got to see this, it's like it was made for you." He was right. And it might be right for you, too . . . but maybe not for the reasons you think.

GETTING STARTED

You need an Internet connection to get Sequel registered; as it doesn't use a USB dongle (where you can take the dongle to a different computer) there's no offline registration option, so if your music computer never touches the Internet, you can turn the page now. Once you register, your code is toast, so choose wisely: desktop or laptop? I chose desktop, which in retrospect was a mistake, for a couple reasons. First, Sequel seems like it would definitely help pass the time the next time my flight's canceled and I have to enjoy the sights of scenic Hartsfield-Jackson airport until I can get rebooked. Second, it does have a lot of cool instrument sounds, but no way to rewire into host software. So, if ensconced in a laptop, you could treat the laptop as a keyboard instrument, and feed the audio into your host. Third, it doesn't put much strain on a CPU.

If you register the program, you can get another activation code to replace your original one should your hard drive die or you switch computers; but there's no way to get it working on your desktop and laptop without buying two copies.

Sequel comes on a single DVD, of which 4.77GB is content that's compatible with Steinberg's SoundFrame media browser. Yet you're not limited to the included content; you can drag in AIF and WAV files, and it (most of the time) manages to

Fig. 1: The main screen really does look quite cool; note the pop-up menu toward the left for automation options. The bottom strip shows the track inspector for a MIDI instrument and the MIDI editor, which includes nudge, step input, and of course, quantization.

get them to match the project tempo. And when it doesn't, you can tweak the file to get it to fit. It also imports Acid files with their tempos. However, there are limitations in terms of the instruments; Sequel is not an "open system" that accepts VST instruments, nor does it do rewire (host or client). Too bad — it would make a great rewire client.

But I'm getting ahead of myself. . . .

THE USER INTERFACE
As mentioned, Sequel is a one-window interface (Figure 1). Across the top is a control strip with transport and a toolbar with project, automation, and edit buttons. These have cryptic icons, so leave tooltips checked until you figure them out. The only traditional menus are for files (e.g., exporting, managing projects, new/open/save) and basic edit functions (cut/copy/paste/duplicate/split, add track, delete track, and the like). Better yet, though, the Smart Tool cursor, which changes the cursor's function depending on where it is in an event, lets you do most common editing functions (mute, repeat, resize, split) without the menus.

Below that is your track view, but with two extra strips at the top: One for creating arrangements (more on this later), and a transpose track that works like Acid's transpose markers. Tracks are pretty much like what you're used to with DAWs: Drag in audio or MIDI, or record audio or MIDI into tracks.

The bottom part of the screen shows one of six pages, as selected by a vertical toolbar toward the left. Again, the icons look cool, but it takes a bit before parsing them becomes second nature. The pages are mixer, track inspector, Media Bay (for browsing the available content), edit, arrange, and program settings. The latter is easy to explain: It's where you adjust audio settings, specify where content is stored, set the basic interface color, and choose from a few basic edit options.

LET'S CREATE A PROJECT
I'd just done a video for a website on all the cool lighting effects at the Frankfurt ProLight + Sound show, and wanted to create a club-oriented soundtrack for it. So this section is basically a diary of how I did it.

First up: the Media Bay page. If you're not familiar with this concept, it's a browser that "tags" all content with attributes, à la Kore. You can then narrow down the

Fig. 2: Media Bay makes it easy to find sounds and instruments.

Fig. 3: You can insert two track effects for each track, in addition to the compressor; note there are also tabs for equalizers, global effects, and output effects.

attributes to lead you to particular content. For example, in this case, I wanted to start with some drum loops. So I selected Drum&Perc, then "filtered" that to Beats (i.e., complete loops), Electronica/Dance, and Techno. I also selected Trance too, figuring they might work well together. A bunch of beats showed up on the right (Figure 2), ready to audition.

I found a useful loop with no kick — a good way to start, then found a couple good drum loops that would make a fine foundation. But I also wanted some breakbeat pauses; some FX-oriented loops worked well for that. Trimming loops to specific sizes, or repeating multiple times, is easy. It's also easy to split and cut loops, so you can do tricks like cut a specific eighth- or sixteenth-note section and paste it multiple times to get a stuttering effect. Before long, I had 64 measures of a drum part churning along at 133.33 BPM.

Stretching is handled pretty much transparently; drag something in from the Media Bay, and it matches the project tempo. Dragging in non-Sequel files is a bit different, although most of the time, it figured out what to do with the various WAV and AIF files I brought in. In a few cases I needed to do a little manual warping, but this was the exception rather than the rule. Furthermore, like Live, you have multiple stretch options: drums, plucked, pads, vocals, mix, and solo. It's worth experimenting with these, as sometimes the "wrong" settings give a cool effect anyway.

Next I decided to add just a single kick for the last 16 measures, but it needed

some distortion to stand out. By going to the Track Inspector page, you can choose two track effects (see Figure 3; there's also a compressor for each track as well as MIDI effects for instrument tracks, the send amount to two global effects, and settings for two Output Effects — Stereo Enhancer and a Maximizer). The effects are actually fairly decent: ping pong and stereo delay, two distortion types, dynamics (gate or maximizer), filtering, seven modulation effects, reverb, and stereo enhancer.

It's worth noting that you can mix and match different loops on the same track, but you can also break out sections to individual tracks, should you want to add effects to one particular set of sounds or loops. Furthermore, you can record your own audio tracks like vocals and guitar (there's even a tuner), and if you have a low latency ASIO interface, you can record through the effects in real time. When you're done, there are extensive automation options — including automation of various effects parameters — which is definitely unexpected, given the price.

GETTING MIDIFIED
I'm pretty picky about bass lines, so rather than look through loops, I figured I'd just load a bass instrument. It may seem like "instrument" is a bit much, as it feels more like you're loading a preset; but even there, you can call up the "Instrument" tab in the track inspector and have access to filter cutoff and resonance, drive, DCA attack and release, LFO frequency and level, and FX mix . . .

so you can get a fair amount of mileage out of one sound.

Sequel kind of takes you by the hand no matter what you do, so I used Media Bay to find an instrument and drag it into the track view. Record was already enabled ("You just dragged over a MIDI instrument, I suppose you want to record something with it . . . right?") so I started recording. And shades of Sonar, you can do MIDI looping really easily.

Instrument tracks allow pretty extensive MIDI editing, not just quantization. There are two MIDI effects, Chorder and Arpeggiator. I fell in love with applying Chorder to some parts of the MIDI bass line, so I cut those parts, dragged them to another track, and applied Chorder just to those parts. Way, way cool. It's fun to have an arpeggiator at your beck and call, as well.

ARRANGING

But we're not done yet, as one of the best features is the Arranger Page. It has two ways of dealing with arranging your

composition, live and programmed.

You create the elements you want to arrange using a separate Arranger Track by clicking and dragging over a range of the composition, which can be as little as an eighth note. This gets a letter from A to Z.

Now you have two options. One is to type a list ("chain") of letters, and each segment plays back in the desired order (yes Opcode Vision fans, you are having a *déjà vu*!). You can even move the letters around while the piece is playing for on-the-fly changes.

The other option is live playback using 16 pads. Each pad can be assigned to any of the letters (or stop), so clicking on a pad triggers the equivalently-numbered segment. (If you're averse to typing, you can also program the chain by clicking on the pads.) Furthermore, there are five start time quantization options: Now, 1/2/4 bars, 1 beat, or "end" (the next segment starts playing as soon as the current segment ends). You can also name each pad so it's obvious what each letter stands for.

The only real limitation here is that I wish you could assign the pads to MIDI note numbers — that would be great when using a pad controller to give more of an "MPC" kind of vibe, or for triggering from a keyboard.

TRANSPOSITION

And of course, you can transpose loops to match the project, either on an individual loop basis or with a project transpose "track" that affects everything playing at that point. If you use the loops included with Sequel, the program knows exactly what to do with them, doing a "smart transpose" (*e.g.*, ignoring drums, although you can transpose drums manually if you want to change the timbre). Unfortunately, it won't read key information in Acidized files but with whatever files you drag in (WAV or AIF), you can specify how much it should be transposed to match the project. At that point, it will follow along with any transpositions you specify in the transposition track.

CONCLUSIONS

I've seen a lot of attempts to come up with programs that would appeal to "potential musicians," but as I'm a bit past that stage, it's hard for me to be completely objective as to whether a newbie could really just open this up and start creating music. But if that newbie had a bit of background in terms of sequencing and synthesis, it's not out of the question at all. Yet that's a different subject for a different time. What about *EQ* readers?

Honestly, I didn't expect to get excited about Sequel; I figured it would be the kind of program I could appreciate from a technical standpoint, load on to my computer to evaluate it for the review, then remove it from my hard drive to save space — and go back to my "real" tools.

But no. After an initial few minutes of "What's going on here?," all of a sudden everything just clicked into place and I was making music on it as if I'd used it for months. Sequel is fast and efficient, didn't crash once (despite torturing it with numerous pilot error *faux pas* while learning the program — pretty remarkable for a V1.0.1 program), and had a great time. A couple hours after working with the program for the first time, I had the soundtrack for my video. And if probably would have been done in under an hour if I hadn't been learning the program as I went along.

Now, understand that my enthusiasm is in large part due to my affinity for this type of music, and how well Sequel handles it. If you're a meat-and-potatoes rock fan, Sequel may not seem appropriate, but keep an open mind: There's a lot of straight-ahead rock, hip-hop, and even country pop content; the program can do more than it may seem at first. But also, if you check it out, you might discover what's so appealing about groove-oriented music: You might think of Sequel as a "gateway drug" to various addictive forms of music. So while I can't recommend this to *everyone* without hesitation, if while reading this you've been going "Hey, sounds interesting," then you probably will enjoy it as much as I do . . . which is a whole lot! EQ

Product type: Audio/MIDI sequencer oriented toward loop-based music.
Target Market: Marketed to consumers, DJs, and groove musicians, but should also be targeted at those who need to generate music for videos or

commercials, songwriters, etc.
Strengths: Included content is a major plus. Cross-platform (XP SP2 or Vista, OS X Universal Binary). Stable. Versatile. Extremely good value and high fun factor. Deeper capabilities than expected. Included effects. Arranger track, chains, and "play pads" add a useful dimension for improvisation and live performance. Media

Bay simplifies browsing. Strong automation.
Limitations: Closed system; can't use other VST instruments, or ReWire into host program. Doesn't recognize REX slice information. Tempo changes can't be automated. At this price, any other complaints would look petty.
Price: $99.99 (list)
Contact: www.sequel-music.net

SONIVOX
BENT CIRCUITS

Contact: Sonivox, www.sonivoxmi.com
Format: DVD-ROM of loop files: 794
Acidized WAV, 794 REX, 794 Apple Loops,
375 Audio WAV; 24-bit/44.1kHz
Price: $99.95 list

Electro meets drum 'n' bass meets gabba
meets glitch meets twisted electronic
instruments, with sounds ranging from
environmental to frenetic (with more of
the latter). Produced by Ben Cantil, this
isn't a construction kit, but a collection of
loops that start at 120 BPM and work their
way up to 220 BPM. However, due to the
above-average quality editing for
Acidization and "REXing," you can pull
these puppies downtempo with decent
results; just don't expect to get into the
sub-100 BPM rap range.

The demos on the Sonivox site give
one slant on using these sounds, but
the example I put together at
www.eqmag.com shows another option
— an almost classic electro rhythm bed at
a locked-groove 133.33 BPM, with addi-
tional sonic anarchy thrown into the mix.
While you can put together cuts using
just the sounds on the DVD, the loops
also work well when you want to insert
mayhem in something that's otherwise
too tame.

One cool touch is the inclusion of the
original audio from which the loops were
cut (that's why there are less audio files
than loops). You can then extract bits of
these, or cut your own loops.

At first, the loops seemed almost *too*
insane; but after a little while, things fell
into place. If you appreciate something dif-
ferent (and put together well), *Bent Circuits*
offers purebred electronic sounds you won't
find elsewhere. —*Craig Anderton* EQ
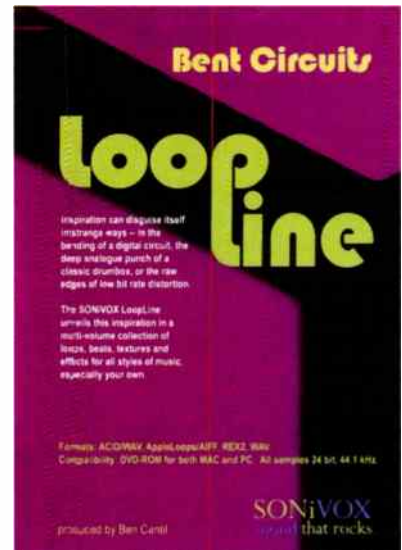
BIG FISH AUDIO
SYMPHONIC MANOEUVRES

Contact: Big Fish Audio, www.bigfish
audio.com
Format: 2 DVD-ROMs of loop files: 489
WAV, 489 Apple Loops, 237 REX; 24-
bit/44.1kHz
Price: $99.95 list

This isn't an orchestral virtual instrument,
but a collection of 30 construction kits
(2.36GB of WAV files duplicated as Apple
Loops, with 362MB of REX files). As this
is soundtracky stuff, it's not just orchestral
instruments; you'll also find drums,
synths, etc.

Each kit contains a main mix with all
parts, mix with traditional orchestral parts,
and looped separate elements. Also, extra
points for useful titles and file names,
which (among other things) indicate the
section of a song to which a loop belongs.

Three cautions, though: Individual
parts aren't isolated (most loops are
ensembles), many files don't lend them-
selves to the REX format, and mix and
match can be difficult — there's a good
selection of keys and tempos, but your
stretch algorithms better be up to the task
of changing them if needed. Also, some
loops have abrupt transitions or aren't cut
perfectly to the beat, leading to slight
gaps; an easy fix is copying a quarter note
from the beginning or end of a clip, sliding
it over an eighth note, and crossfading to
cover up the splice. Still, mix and match
isn't overly difficult, as the audio example
at www.eqmag.com — taken from three
kits — clearly proves.

This is a well-recorded live orchestra,
and the open-ended nature encourages
creativity. Put a little work into arranging,
and the results will be well worth the
effort. —*Craig Anderton* EQ
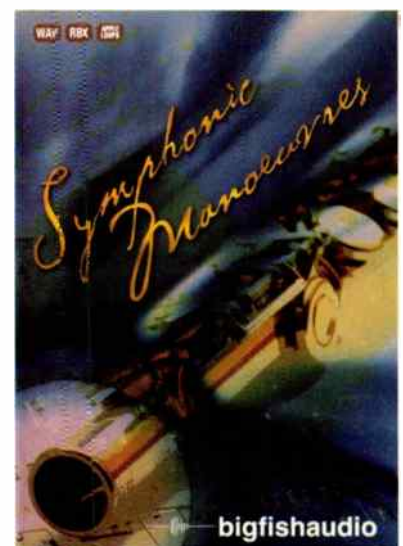
M-AUDIO
BEYOND FOLEY VOL. 2: ORGANIC

Contact: M-Audio, www.m-audio.com
Format: DVD-ROMs with 584 unique files, presented in 48kHz, 44.1kHz (both 24-bit), and MP3 (128kbps) formats
Price: $49.95 list

And who among you hasn't lusted for the sound of a bear rustling through garbage? Well, *Beyond Foley Vol. 2* has grabbed that sound for you — along with seven folders of sounds in total.

The Animal Kingdom folder is about two-thirds birds and one-third dogs (don't expect lions, cats, elephants, etc.). Outdoor Environment lengths are generous, with many hitting at least 30 seconds and some lasting several minutes; they're very good. Most are natural (except for some outside Central Park) and evocative. The Stones folder has ceramic breaking, stones dropping, stones falling, and other material that may not appear useful — until you treat the sounds as percussion samples.

Streams/Falls/Beaches/Lakes has some excellent natural water effects. Under Foot has foot Foley (cracking branches, leaves, and the like); it's okay, but certainly not something I need every day. Water Zone has splashes and drops that sound as expected although some can make way cool drum sounds, as can some of the various Wood Vibe sounds.

Bottom line: While not a comprehensive sound effects library, there are some useful environments — but also a lot of sounds you don't find in traditional libraries, and these are fertile ground for waveform editors. This wouldn't be my first call FX library, but the price is right, and you get a lot of sounds with considerable potential.
—Craig Anderton

Power App Alley
by Craig Anderton

CAKEWALK SONAR 6

Use Sonar to manage your master keyboard controller's MIDI data

OBJECTIVE: Filter incoming MIDI data to create splits based on pitch or velocity, as well as determine which splits should receive controller, pitchbend, and aftertouch data; and if so, how much.

BACKGROUND: Sonar accommodates MIDI plug-ins, and the MIDI Event Filter plug-in allows filtering out various elements of the MIDI stream. We'll use this plug-in to set up keyboard splits for either Sonar's internal soft synths, or external MIDI keyboards if the MIDI track's output is being sent to an external MIDI port.
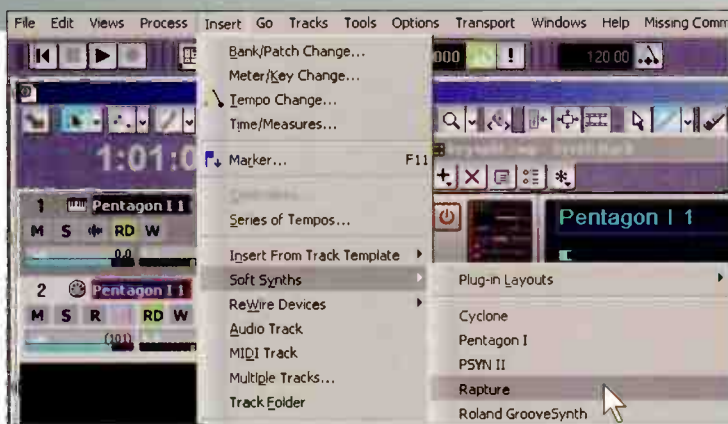
steps

1. Insert the synths you want to use for your splits. In this example, Pentagon I has been inserted, and Rapture is about to be inserted.

2. In the synth's MIDI track, right-click on the FX bin, and go MIDI Plugins > Cakewalk FX > MIDI Event Filter to insert the MIDI Filter. The effect's GUI will open automatically.

3. To enable the Notes filter, click on the Notes button in the Event Filter associated with the first split you want to create.

4. Adjust the upper and lower note limits by click-dragging on the handles for the Key parameter.

5. If you want to split based on velocity as well, edit the upper and lower velocity limits by click-dragging the Vel handles.

6. To prevent the wheel from affecting a split, click on the Wheel box and set both handles to 0.

7. Repeat steps 2–6 for the next split.

8. Play your keyboard, and the incoming MIDI data will be split as you just specified.
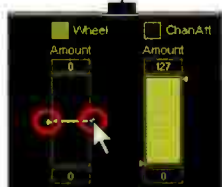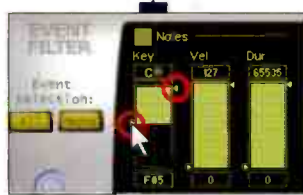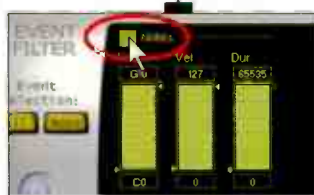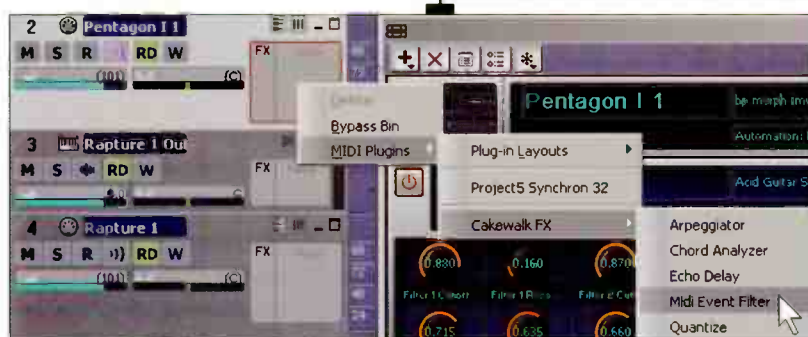
tips

In step 6, you can also limit the amount of aftertouch, key pressure, and/or controllers by checking the appropriate boxes and limiting the range as desired.

Also in step 6, if you double-click in the Amount field, it defaults to zero. In fact, all parameters in the filters default to their halfway points if you double-click on their respective fields.

The John Lennon

SONGWRITING CONTEST

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ABLETON LIVE

Use multi-timbral soft synths within any Ableton Live project

OBJECTIVE: Insert a multi-timbral VST soft synth into Ableton Live, and trigger sounds in specific channels from MIDI Track clips.

BACKGROUND: Although Ableton Live can accommodate multi-timbral synths, the process of inserting a synth and driving specific sounds from specific tracks takes a little bit of setup. Here's the procedure.
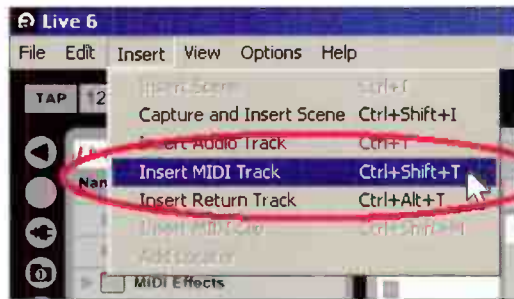
steps

1. Create a MIDI track by going Insert > Insert MIDI Track, typing Ctrl-Shift-T, or right-clicking in the Clip/Device Drop Area and selecting "Insert MIDI Track."

2. Insert the instrument (or instrument rack) by dragging it from the browser into the Track View Selector. The instrument now "lives" in the MIDI track you created, and the track has an audio output so you can hear it.

3. To simplify the upcoming track assignments, rename the MIDI instrument track by right-clicking on the Track name, selecting Rename, and typing in the name of the instrument.

4. Create another MIDI track (this picture shows what happens when you right-click in the Clip/Device Drop Area to create a track).

5. Assign the new MIDI track's Output Type ("MIDI To") to the instrument you named in step 3.

6. Assign the same track's Output Channel ("Track In") to the channel that corresponds to the sound you want this track to trigger.

7. Repeat steps 4–6 for each track that you need to drive a MIDI channel in the instrument.
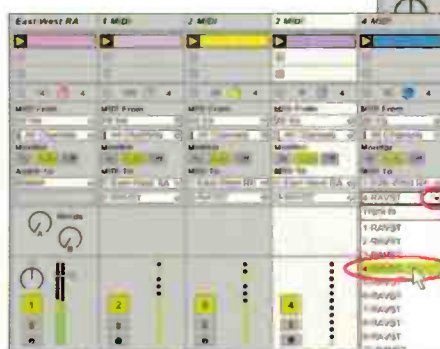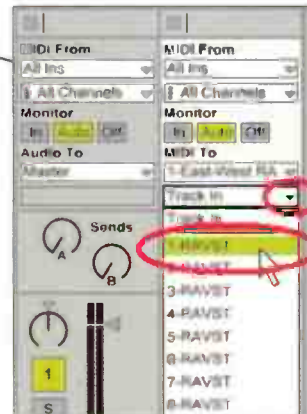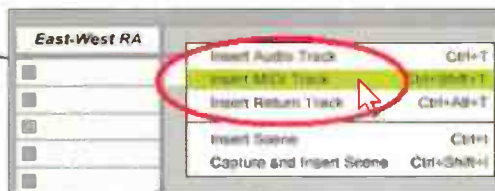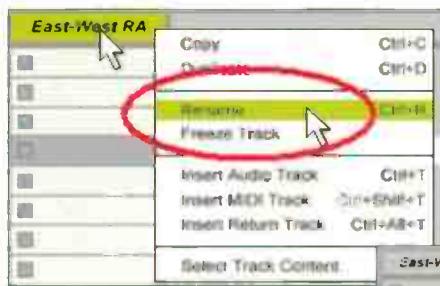
tips

In step 2, because dragging in an instrument causes the track to produce an audio output, this output does not get assigned to any instrument MIDI channel. The other MIDI tracks "feed" MIDI data to this instrument.

AdIndex

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

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World Radio History

In the Studio Trenches
by Phil O'Keefe

MID-SIDE RECORDING
The theory and practice behind this stereo miking technique

Stereo mic techniques are great for capturing and creating a natural, or even a hyper-natural sense of width and space. Probably the two most common techniques are the *XY coincident* arrangement, and a *spaced pair*. But the lesser-known Mid-Side ("M-S") mic technique has its advantages too . . . let's investigate.

THEORY

A lot of people are hesitant to try M-S recording, maybe because they read about decoders and math formulas and feel overwhelmed. Don't worry — we'll keep it simple.

You'll need two mics: one cardioid, and one bi-directional ("figure 8"). Ideally you want two similar mics, but this isn't essential; experiment with whatever mics you have that meet the polar pattern requirements.

M-S uses the center or "mid" mic in combination with the bi-directional mic to achieve stereo. As the cardioid mic points right at the sound source, it picks up the direct sound, while the off-axis bi-directional mic picks up the room ambience and reflected sound. M-S stereo "sum and difference" is just the center mic *plus* the side mic for one channel, and center mic *minus* the side mic for the second stereo channel, with the center mic being positive polarity and common to both sides. As the left and right sides originate from the same mic, but with the phase inverted, collapsing an M-S recording to mono cancels out the left and right sides from the bi-directional mic, leaving only the positive polarity signal from the center (cardioid) microphone. This significant advantage of M-S recordings insures perfect mono compatibility without any phase issues.

Fig. 1.

SETTING UP THE MICS

Aim the cardioid mic directly at the sound source. As with normal cardioid mic placement, adjust the "aim" to taste; but if you're a fan of close miking, try moving back a bit further from the source for M-S recordings.

Next, place the figure 8 mic so that the two lobes of the pattern are set 90° relative to the cardioid microphone. M-S is a coincident microphone technique, so you want to get the diaphragms of the two mics as close together as you can. Figure 1 shows a Soundelux E250 (bottom) and ELUX 251 (top) set up as a M-S pair. The cardioid E250 is pointed at the sound source (in this

case, the camera), while the pattern selector on the ELUX is set to bi-directional; it's picking up to the left and right, and its side null point points directly at the sound source/camera.

SETTING UP THE BOARD AND RECORDING

At your DAW, simply route each mic to its own preamp, and assign the cardioid mic to a single DAW track. You can either record the bi-directional mic to two identical, separate tracks of its own and invert the polarity of one of the two tracks later, or record the bi-directional mic to only one track and use a decoder plug-in, or clone the single bi-directional mic's track later and invert the clone track's polarity — your choice. I generally record the bi-directional mic to two tracks simultaneously in Pro Tools, labeled "SIDE+" and "SIDE-," and insert a Trim plug-in on the "SIDE-" track to invert the phase.

Now group the two "SIDE" tracks and pan them hard left and right. As you raise the level of the side mic tracks, the stereo width will increase; lowering them decreases it. Being able to adjust the amount of stereo information in the recording after the fact is one of the big advantages of M-S recordings.

FREE DECODER RING INSIDE!

So why would you need a decoder or hardware matrix box? The downside of simply cloning the "side" mic track and inverting the polarity is that you can't hear the final results as you position the mics — you have to record them, then flip the polarity and start playback. By encoding the sum and difference data from the two mics and recording the result, you can hear how the stereo field will sound before you track. PAiA has a hardware M-S matrix box schematic available on their website for you solder jockeys: www.paia.com/ProdArticles/msdec work.htm. For PC DAW users, Voxengo's free VST format M-S decoder plug-in works very well; download it from www.voxengo.com/downloads/.

M-S might not be used as frequently as some other stereo techniques, but the perfect mono compatibility and ability to adjust the stereo width at mixdown make it handy for broadcast production, live recordings of small ensembles, individual instruments, and small groups of background vocalists on multitrack music sessions. Give it a try! 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

BIG IS BETTER
Making the point with counterpoint

I recently enjoyed a few tremendously fabulous "wow" moments talking to Police guitarist Andy Summers in his Southern California office/rehearsal space. As everything was being uprooted and packed for the band's sellout arena tour, the cozy studio area was crammed with all kinds of gear, from Stewart Copeland's drums and percussion toys to Summers' guitars and custom Bradshaw effects-switching system. Hearing Summers perform solo bits of classic Police hits as he demonstrated his guitar rig was an intense thrill, but the real kick-in-the-head epiphany came as he shared the details of his craft.

"For me, interesting music has a lot of counterpoint," he said, "so I've always been very adamant that the other musicians don't play what I'm playing. Particularly in a trio setting, having three different parts interlocking makes for a much bigger, and more interesting sound."

Bingo! Duh! News flash!

Summers wasn't necessarily discussing recording, but his comment definitely offers the promise of a solution for anyone disappointed in the, ahem, less-than-mammoth stature of their recorded works in general, and their guitar tracks in particular. Too often, we shrink the impact of our mixes by piling on overdubs, effects, and textures. The temptation of the "more is better" approach can be great, but if "more" means more things doing relatively the same things, or more elements enhancing similar frequency ranges, then all the layers you're adding may just be creating something more worse.

In these instances, the guiding premise of Mr. Summers' strategy is brilliantly simple: *Do not record anything that apes, mimics, clones, or mirrors a part that already exists in your mix.* By devising different, rather than similar and supportive elements, you just might churn out guitar tracks that sound as big as all Montana.

Obviously, this theorem doesn't work for all styles — '70s-style punk comes to mind — but, purely as an experiment that might lead you to new discoveries about arrangement, engineering, and production, let's approach Andy's mandate in three easy steps.

STEP ONE: SET THE STAGE

Do you ever ask yourself exactly what your track is trying to achieve? Is it sad, happy, full of bravado, aggressive, sensual, bombastic, or light? It's good to know, because focusing on the end result can help lead you to decisions about appropriate tones, signal processing, and the placement of elements in the final mix. When I do recording seminars, I often find that musicians tend to develop a gaggle of cool tracks, and then try to fit everything together.

This is not a Summers-approved tactic. Summers and his Police mates work extremely hard to craft parts that enhance the meaning of the song. In this experiment, start by limiting yourself to the minimum number of parts required to effectively deliver your message. I recommend tipping your hat to the Police by using drums, bass, a couple of guitar parts, a lead vocal, and a few background vocals.

STEP TWO: FIND YOUR PLACE

Drums, bass, and guitar fill up a ton of sonic space, and the punch factor is going to be more intense if those instruments aren't fighting each other for breathing room. Think about making space, rather than filling it. First, employ the Summers Mandate by ensuring your drum and bass parts — and tones — aren't colliding with each other. Have the bass play off the drums, and don't EQ the kick drum and bass to sit in the same frequency range.

Work with the guitar the same way. Look for a rhythm part that drops into the holes, and watch your tone. Don't dial in low mids that obscure the toms, muddy lows that mess with the bass pattern, high mids that neuter the snare crack, or highs that veer too close to the hi-hat. Of course, you can decide to have the guitar be the chunky low boy — in which case, thin out the drums and bass a bit. Take care with effects. Don't dilute the punch with too much delay or reverb, and monitor the timbre of your processors, as well. A spiky delay can screw with the hi-hat almost as much as a thin guitar tone and a rhythm part that's locked to the hi-hat tempo.

STEP THREE: EMBRACE THE LOUD

One of the producers I worked with early in my career had a maddening rule about mixing solos. He believed a solo should only be as loud as absolutely necessary to ensure all the notes could be heard above the backing tracks. So he'd mix the guitar solos down until they weren't so much climatic explosions as polite musical fills. You don't want to be that guy. Use dynamics aggressively to accent critical moments. The manipulation of loud and soft parts — and *really* loud and soft parts — keeps the listener in a constant state of awe and surprise. It also helps your mix sound like a raging sonic bull.

YOU'RE HUGE!

This month's column is more conceptual than practical, but as I constantly hear the Summers Mandate not being addressed by so many recording zealots who desire bigger-sounding mixes, a little theory seems appropriate. Experimenting with disparate, interlocking parts may not gain you a hit, but it should teach you volumes about dynamics, tone crafting, and impact. Of course, if you're lucky enough to have a brilliant song on the board, you could end up with something as timeless as "Message in a Bottle." Good luck! 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.

WE SHOWED YOU OURS

NOW SHOW US YOURS

The Rock Files
by Lee Flier

BUT WHAT ABOUT THE TRACKING ROOM?

Why "Acoustically Great" doesn't always equal "Musically Great" — and what you can do about it

We all know the importance of acoustical treatment in our studios, but most of the focus for small project studios is on treating the control room. But what about the tracking room? Do the same principles apply there, and what effect does acoustical treatment have on the instruments' sonic character?

In terms of pure physics, you still have to deal with standing waves, early reflections and all the same issues. In fact, in the tracking room these problems may be amplified (literally), as you may be dealing with loud sound sources like drum kits and guitar amps. Many people opt to simply deaden the room as much as possible to kill all those reflections, covering the room with acoustic foam, then adding "character" to the sound in the mix using effects. While this approach works for some styles of music, it may not be the best solution for rock and other "organic" genres where you want a more "live" sound.

Moreover, an oft overlooked fact is that the sound of the recording room dramatically affects the tone of the recorded instrument, including vocals. In fact, you can think of the room as being like a giant equalizer, where the early reflections in the room combine with the original sound source to produce the overall timbre of the instrument or voice. You can take away those early reflections as much as possible to get an acoustically "purer" sound, but in the process you lose much of what enlivens a sound in the natural environment and gives it character. And of course, it's difficult to substantially change the basic timbre of an instrument in the mix.

So how to retain the character of early reflections, especially in a small room, without dealing with standing waves and other phasing problems? Some compromise is necessary. You can approach treatment of your tracking room a bit differently than a control room because after all, in the tracking room you aren't striving for "accuracy" so much as "musicality" — you just want the instruments to sound good in the room. And luckily you can measure that without complex calculations or expensive instruments; all you need is ears. So, here are a few tips to help you achieve a good balance between "sonic character" and "acoustical problem."

Don't deaden everything. Don't put foam all over the room. Leave some reflective surface on the walls, or cover two walls and leave each opposite wall mostly untreated. Same with the floor and ceiling: In most cases you want an absorptive ceiling and a reflective floor, so for best results, you want a wood or linoleum floor (not carpeting) and a drop ceiling or foam treated ceiling. If you have carpeting in your room, consider putting plywood down on the floor while recording; I've had good luck recording drums and guitar

amps this way in small rooms, tacking up foam on the ceiling. Unless you're deliberately going for a deader sound, live instruments tend not to sound great on padded carpeting.

Use bass traps. You'll achieve a much more solid low end by the liberal use of bass traps. Many bass traps, such as Mini Traps from RealTraps, are slightly reflective in the high end so that they don't completely deaden the room, yet they do an effective job of killing standing waves in the low end.

Build some baffles. Using baffles rather than treating all of the walls lets you adjust the location and number of baffles so that you can adjust the amount of "liveness" in the room, or on individual instruments. Put baffles around a drum kit, amp, or vocal mic for a deader, more close-miked sound; move them further away or eliminate them for more reflections. It's a great exercise to move baffles around while listening, so that you can hear the dramatic effect they have on the timbre of the instrument or voice. And remember, vocalists can turn in a much more confident performance when the sound of the room meshes well with their voices.

If you're in a small home studio where you track and mix in the same room, you can still employ many of these techniques. By using as many bass traps as you can and building moveable baffles, you can increase and change the reflections in the room when tracking, and deaden them when mixing. If you prefer a very dead room while mixing, treat one side of each baffle with foam and the reverse side with more reflective material. This gives maximum flexibility in changing the room's character to suit the task at hand.

Recording live instruments in small rooms is always a challenge, but there's no need to track in a lifeless room. Many classic recordings have been made in less than "acoustically perfect" rooms, gaining an unmistakable character of their own as a result. Even hallways, stairwells, tiled bathrooms, and other acoustic nightmares can work, depending what kind of sound you seek. So ditch the foam for awhile and feel free to experiment! That's the only way to find out for sure if you can defy the laws of physics. EQ

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, countries, and even virtual reality. 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.

21st Century Recording
by Gus Lozada

THE BIG SCORE: GETTING STARTED

Before you can make music for videos and films, you have to cut through the jargon

by Gus Lozada

Yes, once upon a time there were silent films. Scoring a movie meant having an orchestra, or at least a piano player, performing along with the projected picture. Many years later, scoring became easier but then we started mumbling strange acronyms like MMC, SMPTE, MTC, and SPP . . . say what?

THINK OUT OF THE BOX

Scoring can now be done totally inside of a DAW. This is helpful when you are scoring your own video or someone else's film, and all you have is a QuickTime or AVI/WMV video of the work. However, many studios still work with external hardware for scoring, so you need to know about the current proto-
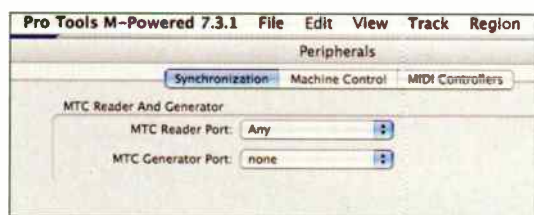
Fig. 1: Pro Tools can send and receive MTC, even in the simpler M-Powered Version. For more options, you'll need an LE or HD version as well as the DV Toolkit.

cols, or update your knowledge if it's been a while since you've worked with video.

Every computer system/DAW has an internal clock that keeps track of what events occur at any given time. Most of them can generate their own clocks, as well as receive an external source as a master clock. It's essential that devices expecting an external clock can sync to the master clock, or your timing becomes a train wreck.

Regarding sync protocols, the main choices are SMPTE, MTC, MIDI clock, and MMC. Which one(s) you use depends on your circumstances and gear, so let's check out each type.

- **SMPTE** ("Society of Motion Pictures and Television Engineers") is one of the oldest sync protocols. It is independent of musical tempo, as it marks events based on hours, minutes, seconds, and frames (H:M:S:F). With early MIDI systems, SMPTE code was recorded on a tape recorder's outside track (to minimize crosstalk), and a standard audio cable took the output to a MIDI interface. This decoded the signal to provide timing information to sequencers. SMPTE is still in use in film and video today, but is embedded in digital video files instead of recorded on an analog audio track.
- **MTC** (MIDI Time Code), the MIDI equivalent of SMPTE (Figure 1), is also independent of musical tempo and also displays elapsed time in H:M:S:F.
- **MIDI clocks** include no position or timing information, but simply provide a timing reference, like a metronome. Devices follow these clocks to sync to tempo, while additional commands tell a device when to start, stop, and

continue. This is perfect for syncing synth and sampler LFOs, delay times, arpeggiators, drum loops, and other time-based sounds to a master clock.

- **Song Position Pointer** (SPP) can be transmitted along with the MIDI clock signal; it keeps track of how much time has elapsed during a tune in bars, beats, and subdivisions of beats (called ticks or clocks, which are the finest resolution of the MIDI device). Sometimes all you need is MIDI clocks and SPP rather than MTC or SMPTE, unless you must lock certain events (e.g., sound effects) in a sequence to absolute time instead of musical time.
- **MIDI Machine Control** is a system of controlling (play, stop, etc.) external gear, such as video tape decks, via MIDI.

Which protocol is best to sync everything together depends on the hardware involved. Some devices will not handle MIDI clocks, while musical instruments may not recognize SMPTE.

Fortunately all this work can be frame-accurate, so if you get a video file to work with on your computer, you can create the score, bring your laptop to the studio, and sync your computer to the studio's system. This lets the main engineer use the video system at full resolution and your computer's music for the final mix — and you won't need to render a huge video file for hours along with all your audio tracks.

For us traveling road warriors, having a huge array of high-speed hard disks for our laptops is not a reality (yet!), so we need to face reality: Even though virtually all DAWs can import a video file for basic scoring, many of them cannot export to professional video formats for final delivery. So, we will be dealing with sync for at least a little while longer. Many of today's portable interfaces (e.g., MOTU's Traveler or UltraLite) can handle all needed protocols, including SMPTE, so you won't be alone in your "desert island" studio.

Making music for video is hotter than ever and we all need to be ready for it. Even better, we should work on making our own videos. Produce music in your project studio or a hotel room, send the files or take your computer to that huge studio with an Avid system . . . and get a check. See what I mean?

I'll cover the 100% digital counterpart of what we've covered (doing everything "in the box" instead of using external hardware) in a future column. EQ

Gus Lozada hosts conferences and clinics in Latin America about music production. He also moderates the "Nuestro Foro" and "KSS — Keyboards, Samplers and Synths" forums at Harmony-Central.com. His email is gus@guslozada.com.

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Room with a VU

by Matt Harper

STUDIO NAME: Rudy's Studio
LOCATION: Denver, CO
KEY CREW: Mark Obermeyer, Justin G. Preston
CONTACT: www.rudysstudio.com
CONSOLE: Digidesign ICON D-Command w/ Digidesign 192 Interfaces (3)
SUMMING: API 8200 (2), 8200A
COMPUTERS: Dual G5 Mac 2.3GHz; Pentium 4 PC 3.2GHZ
RECORDERS: Pro Tools HD 3 Accel
CONVERTERS: Apogee AD16X (2), DA16X (2)
MONITORS: ADAM S3A; Yamaha NS-10M, SW-10 (2)
PRES: API 3124, 3124+MB; ART TPS II (2); Avalon VT737 (3); Focusrite ISA 428; Grace Designs Model 801; Great River 1NV (2); Universal Audio 2-610 (2)
MICS: AKG 414 ULS (2); Blue Baby Bottle (2), Blueball (2), Blueberry (4); CAD Tube V2; Electro-Voice 868 (2); Neumann KM184 (4); Røde NT-2 (2); Sennheiser 421 (6); Shure Beta 52, Beta 56 (2), SM 7B, SM 57 (6)
OUTBOARD: A-Designs ATTY (2); Alesis MasterLink, Apogee Big Ben; BBE 882i; Empirical Labs Distressor; Furman IT1220; Manley Massive Passive; MOTU MIDI Timepiece II; Peterson Digital Strobe Tuner; Radial J-DI; SSL XLogic G Compressor; TASCAM DA-78; UREI 1178; UREI 535 Dual 10 Band EQ

NOTES: "We are obsessed with finding new production techniques, and creating really unique sounds," Mark Obermeyer says when asked about why he decided to start Denver, CO-based Rudy's Studio — a facility he co-founded with mastering maharishi and post production habitué Justin G. Pearson (of sister sanctum Dragon Studios) in 2000, following a stint at the University of Colorado's Recording and Music Program. "With our combined experience, we built a studio that is not only a great working environment, but also has the tools to help bands sound like more than just the sum of their parts — and we've been booked steadily ever since."

Comprised of three separate live rooms ("We capture natural reverbs from the various rooms, adding to the live feel of our recordings") that satellite a sizeable control room, Rudy's Studio is built around a Pro Tools HD 3 Accel system with a Digidesign ICON D-Command console at the helm, set to control the heart of the sun. But while the boys choose to work in Pro Tools due to the obvious benefits of utilizing the platform, they manage to achieve incredibly spirited sounds via a slew of high-end analog hardware — effectively integrating the best of both worlds: efficiency in workflow with the heightened audio capabilities only boutique equipment can produce.

And they keep expanding their arsenal at a rate rivaling most red giant stars. As Obermeyer comments on some of the recent upgrades to Rudy's Studio, "With the clean power courtesy of the Furman IT1220, the advantages of using the [Apogee Big Ben] word clock, working at 44.1kHz, and having a ton of sweet pres and analog outboard equipment at our disposal — not to mention the [ADAM] S3As for playback — really makes for a pleasurable workspace. The live room is nice and flat, and when you get back into the control room everything has a way of just sounding great."

But it's not just the impressive gear selection that is to blame for the great sounds that are constantly emanating from Rudy's Studio; it's the shared vision of Obermeyer and Preston — two audio engineers hell bent on translating their artist's intent truthfully to disc. "Retaining the live feel, natural dynamics, and overall energy of the artist's performance is what's most important to us," Obermeyer states. "The ability to work with so many amazing musicians makes us feel incredibly lucky. We really do have the coolest jobs in the world." EQ

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— Michael Giacchino
*Composer
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