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RECORDING AND THE "IS-OUGHT GAP"

One of the main questions of ethical theory revolves around the "is-ought" gap, as described in the writings of Scottish philosopher David Hume (1711–1776). Whoa! *This is* EQ *magazine, what's going on here?!?*

Well, this relates a whole lot to the world of recording and music, so bear with me. (Or if this is putting you to sleep, at least make sure you're not smoking in bed.) Hume's premise was that you couldn't deduce what *ought* to be based on what *is*. This is something that philosophers have debated over the past few centuries, but trust me, musicians and recording engineers already know this. For example, just because a session *is* composed of great players doesn't mean it *ought* to produce great music. Bad vibes, egos, equipment breakdowns . . . there are a lot of hurdles between is and ought.

So if we can't deduce a moral "ought" from a factual "is," what difference does it make? Well, musically or in terms of recording, "we is what we is" and that's the reality of the situation. Maybe you're a great player, maybe you're not, but you're playing music. Maybe you've engineered platinum records, or are using an all-in-one recorder for the first time as you record a band that's been together for a week — but you're engineering. And it's true, we can't predict what *ought* to be based on what *is*: The great player might be in a creative slump, while the band that's been together a week might make all the right moves and strike gold on its first outing

And that's what makes music and recording so fascinating. Trying to create an "ought" from an "is" is something that most philosophers seem to think is an impossibility. So, forget about what *ought* to be the result from your session, and just think about what *is*. Establishing expectations and goals — what *ought* to be — makes sense, and provides a sort of road map. But also understand that what *is* might imply a completely different result, and you need to be sensitive to any new, unexpected directions your music might take you.

So try to cut free from rules, plans, and the way things were done in the past. If you don't worry about what ought to be and instead concern yourself with what is, how something *ought* to turn out will take care of itself.

Craig Anderton

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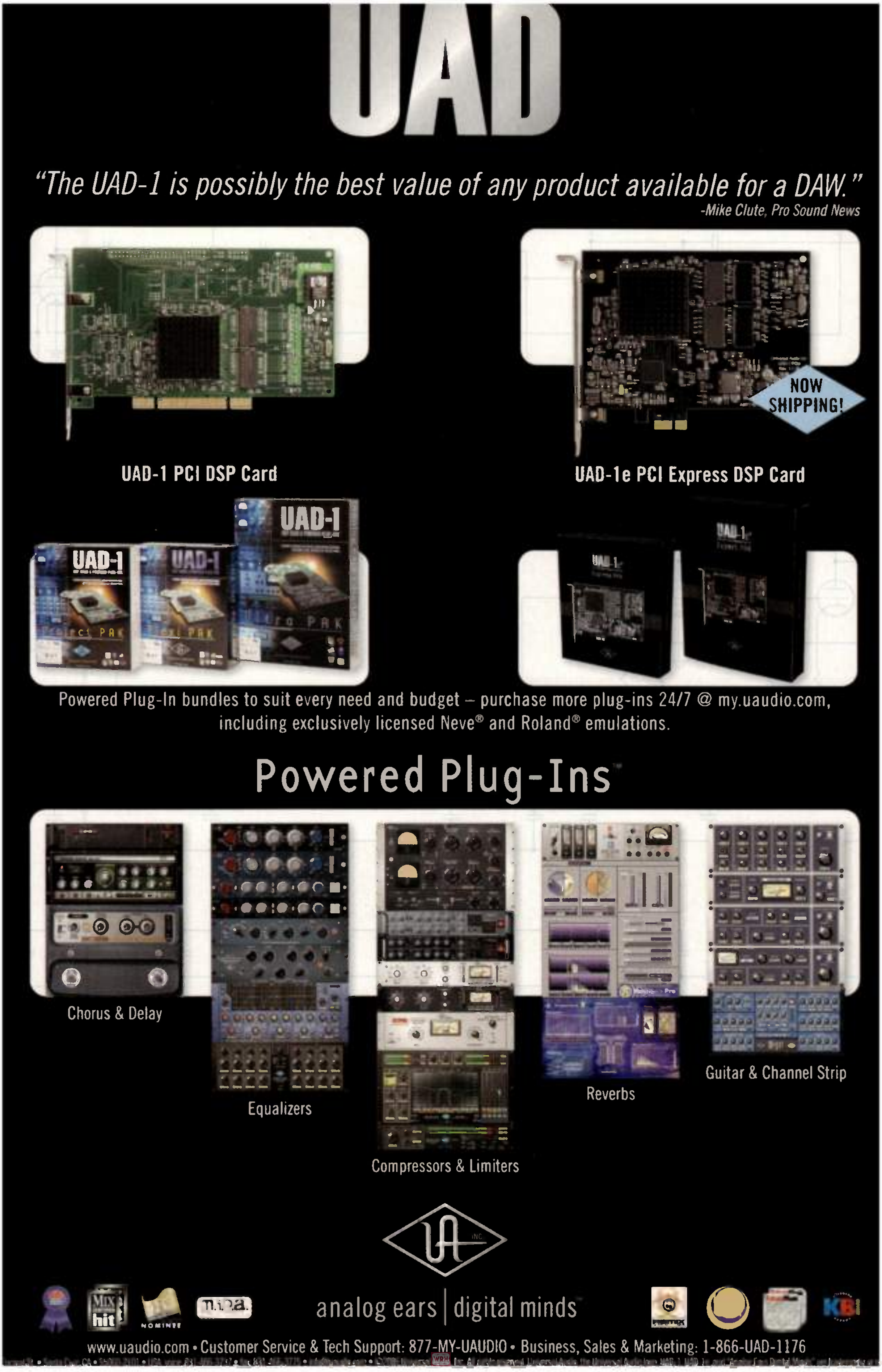
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Punch In

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

KASABIAN

by Lily Moayeri

Fresh out of Monmouth's own Rockfield Studios, where producer Jim Abbiss and engineer Andy Wallace provided the royal aural treatment, Kasabian is enjoying an inordinate amount of success — unveiling their second album *Empire*, the follow-up to 2004's eponymous-titled debut, shooting to the top of the U.K. *Billboard* charts upon debut and taking their message across the Atlantic to a new demographic eager to make the band's moniker a regular household fixture.

Though one could be quick to attribute this rising success to one of the many sonic changes Kasabian underwent in the past few years — such as the shedding of the electro-based skin inextricably linked to former member Chris Karloff (responsible for the past-characteristic machine-generated beats and blips) and the adoption of drummer Ian Matthews, who brings a decidedly more organic element to the group — perhaps it's the undeniable mojo of a more traditional production, in that the key players behind the console (not to mention the recording environment itself) have played host to all the usual suspects to which a band like Kasabian eagerly affixes credit for inspiration, that is making the band one of the most widely embraced rock acts of recent memory.

But while there is a relative familiarity that the listener experiences almost immediately after tossing *Empire* into their stereo, Kasabian is anything but orthodox in their approach to their craft both in and out of the studio. In fact, they are nothing if not a strange hybrid of classic rock charm and indie-minded experimentations in sound. And this is more than aptly reflected in the techniques the band, along with Abbiss and Wallace, employed during the Rockfield sessions — effectively merging old-school approaches with a forward-thinking studio ethos that has resulted in perhaps the most interesting-sounding rock album of 2006.

One of the first things one notices upon encountering *Empire* is the vintage character of the drums, something vocalist/guitarist/keyboardist Sergio Pizzorno attributes to Wallace's "George Martin-esque" approach to tracking the set: Throwing an Electro Voice RE-20 on the kick, one Neumann U87 (set omnidirectional) approximately 2 feet dead-center above the snare, practically eye-level for most drummers, and then crushing the two signals with a Fairchild 670.

Likewise, one glaring component of the album is the noticeably lo-fi sounds that seem to weave in and out of tracks — particularly keys. This can be attributed to Pizzorno's "pre-production" practices wherein his fleet of synths — most noticeably a GEM Pro 1, a Moog Modular system (for it's Clockwork Orange sound — "It's about ten foot high and you need a physics degree to get a sound out of it," he laughs), an ARP Odyssey ("for it's skuzzy sounds") and MicroKorg — are run through a Digi 001, cut to and mixed at home in Pro Tools 5.1 LE, where they were simply integrated into the final mix at Rockfield months afterwards.

"It tends to give it an atmosphere, which is vital," says Pizzorno of the "demos" from which the rest of the album is molded around. "When you get to the studio, the edge can get taken out of your music. We keep those elements in there so it doesn't just sound like we've gone to Abbey Road; it still sounds like a band that records themselves. It's that beautiful ugliness, and we always want to keep that."

But perhaps it's the implacable sounds that randomly clash with the thoughtful consonance that make *Empire* much more than just your average rock-fest. Sources as unlikely as neighborhood lawnmowers and tea kettles, captured with mics that have been shoved into gas masks and tin cans, are introduced, albeit tastefully, throughout the entire album; this leaves *Empire* an effectively enjoyable active, as well as passive, listening experience — depending on the mood of the listener.

Or maybe not.

"There's a place on 'Stuntman,'" Pizzorno announces, "where we put the guitar through every pedal we own; we needed something really ugly and evil. We ran through every pedal, patching into 20 channels on the [MCI 500] desk. It overloaded it. We ended up with the most hideous sound you've ever heard . . . but it's a great sound.

"The last record was bits and bobs, sketches and scraps, which is great, it's brilliant for that. [This record] sounds a whole lot more human."

Photo: Danny Clinch

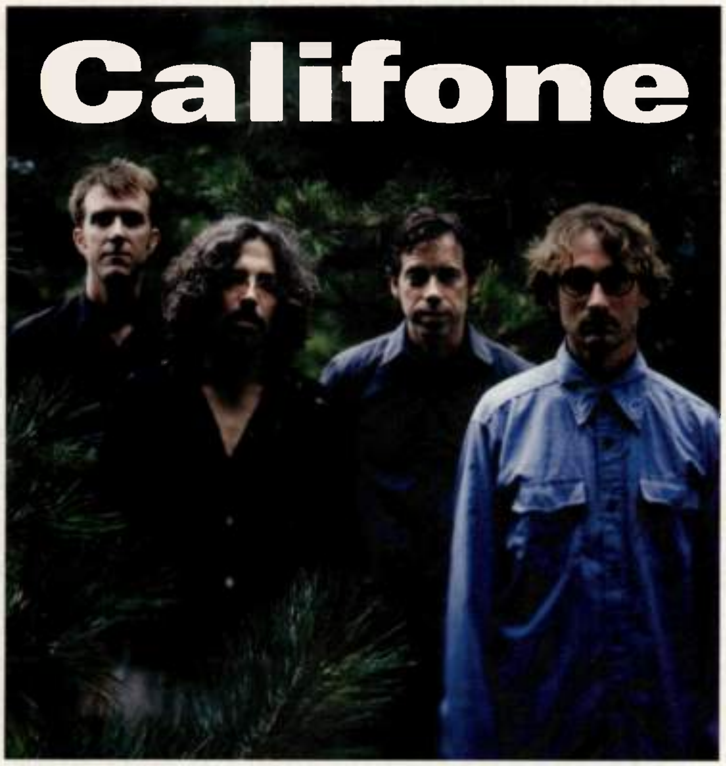
METAL SUPER-PRODUCER ROY Z ON SYNTHESIZING CYMBALS

EQ: So . . . using e-cymbals and replacing drum sounds, huh? What are you working with?
Roy Z: I was using Digidesign's SoundReplacer, but I'm using Drumagog now. As far as eCymbals: eRides you can get away with, same with crashes. But eHi-hats? Forget about it. A hi-hat is so dynamic, with so many nuances. I've never been happy with the sounds from electronic hi-hats.
EQ: When you're replacing cymbal sounds, are you replacing original sounds from real cymbals, or recording the eCymbals and then replacing the sounds with samples in a library?
RZ: What we've done in the past is record the songs without the cymbals, and then add them later. That way we'd have more flexibility. But not every drummer can do that.
EQ: I remember first hearing about that technique with Slayer's first album. What is that process like?
RZ: What we do is set up something that represents a cymbal, that the drummer can hit — like a towel on a stand — something that makes practically no sound so it won't bleed over into the mics. But the real good guys just play without cymbals.
EQ: And even to go back afterward and only play the cymbals, I can't imagine. . . .
RZ: You need a really good drummer, a guy that arranges his routines and writes them out. The best guy I know is [Halford drummer] Bobby Jarzombek. He maps out everything, including what cymbal he's going to hit and when. He also times how long each cymbal's decay is, and won't hit a cymbal again until its totally decayed.
EQ: I respect his doing that. But if you listen to a lot of metal records, the cymbals are definitely held back in the mix from what their true, live sound would be. . . .
RZ: Especially in metal, you need to pick the cymbals that are going to cut through the guitars. And that's hard. I really like the Paiste Signature sound. The cymbals are real musical; they have a beautiful decay. They're a little bit warmer, so when a drummer bashes them for metal, you don't get a brassy, overblown sound.
EQ: At what point do use the Drumagog program in the mixing process?
RZ: I just started getting into it for replacing, or enhancing, drums and cymbals. If I'm mixing something and I can't quite hear the ride, I'll make a trigger track of what the ride is doing, try to build some dynamics into that, and go ahead and use the Drumagog to replace the trigger track.
EQ: Are you miking each cymbal separately?
RZ: Yeah. You have way more control that way. That way you have real balance before you bounce it down to two tracks.
EQ: So then you isolate whatever track you find problematic and then use Drumagog to replace? Do you then re-amp the replacement track into your room?
RZ: Yes. I set up a couple of Yamaha NS10s in the live room, or even in a bathroom or garage, and run the track through those to get a little room ambience. Then you mix that in with the trigger sound, add a little reverb. . . .
EQ: But the cymbal hits still sound all the same!
RZ: Then you change the pitch, and the volume, for each hit. The pitch of a cymbal will change depending on how hard it's hit.
EQ: Manually? Doesn't that take forever?
RZ: Yes. It takes a few hours.
EQ: So you mic every last piece of the kit, then replace *specific* hits?
RZ: Say I get a session I didn't do myself; they didn't mic the crashes properly. I'll make the trigger track, go in and physically cut one of the cycles in the cymbal. Then I use that to trigger and replace within Drumagog or SoundReplacer.
EQ: I've dabbled a bit with Drumagog — mostly for replacing kicks. I've found it works a little too well. I wanted all my kicks at the same strength, but unfortunately, I'm not a perfect player, and the program was layering in a sample based on the actual force of my hit. . . . [*Editor's note: Drumagog offers the option to trigger with a constant velocity.*]
RZ: You can bypass the sensitivity settings with SoundReplacer. But you really need to make trigger tracks if you want to get that effect. I know guys like Andy Sneap, using [Steinberg] Cubase or Cakewalk [Sonar], do it all via MIDI — making MIDI triggers of each

hit. What I do is actually cut each drum hit, and make a trigger track. It takes a long time.
EQ: Okay, but if you're miking each cymbal individually, doesn't it make it really difficult to replace any single one of them, as the rest of the cymbals are ringing in each of the other cymbals' individual microphones?
RZ: Isolation. . . .
EQ: How are you isolating each cymbal?
RZ: Cardboard. Once you have your mic in place, you can sort of go around and shield the area that you don't want bleeding in with the cardboard.
EQ: So what are you throwing up on those cymbals?
RZ: It varies, but I like using [AKG] 451s and 414s. We tried Langevins on the hi-hat and it worked great! We've had trouble getting a good hi-hat sound — getting good isolation — so we'll use a clip-on, where it's mounted just above the cymbal so the two are almost touching.

It's just trial and error. That's what's so great about the whole vibe now: Everyone does their own thing. There's no "right way" to do it. At the end of the day, all that matters is if it sounds good.
—Roberto Martinelli

Califone's new full length *Roots and Crowns* is truly a testament to the sonic versatility afforded by integrating analog and digital recording technology. Tracked over a course of many months in various homes, warehouses and studios across the country, *Roots And Crowns* is a glance into a world of sound where tape loops wrestle with banjos, and kid's toys are meshed together into an orchestral barrage of sound. We caught up briefly with Califone head honcho Tim Rutli, mixing engineer/tracking engineer Brian Deck, and producer Michael Krassner to hear about how, sometimes, it takes an entire village to raise a record.



EQ: So, if I understand correctly, *Roots and Crowns* was largely unwritten until you started recording?

Tim Rutli: I had loose ideas for about half the record — sometimes only a chord progression. We started in the fall of 2005, writing and recording as we toured around the country, dumping everything we had recorded onto my external hard drive at friends' houses, from Chicago to Phoenix, into Pro Tools at a warehouse Michael [Krassner] rented in Long Beach, during our week long "main" tracking sessions.

EQ: I take it the warehouse wasn't a proper studio environment; what gear did you bring in to work with?

Michael Krassner: I just brought ten Shadow Hills 8-channel GAMA pres and a bunch of mics in — from Beyerdynamic M160s to RCA BK5s to the Avenson's STO-2 Omnis, which I used for all the drums — and then we cut the basic tracks live.

EQ: So the basic tracks were cut live, but they were made to fit with previously recorded material culled from remote sessions. What did you end up mixing the album on?

Brian Deck: A Pro Tools Mix Cube system, though it only had 16 outs, so I summed through our console, an AMEK BIG.

EQ: In addition to mixing the album, you also tracked a lot of the overdubs, and a good portion of the vocals. What did you use to capture Tim's vocals?

BD: A Peluso Limited Edition Long-bodied 22 47 and a Shure SM57 set up together and tracked simultaneously. They complemented Tim's voice, and each other, greatly — the 57 was very present in the mids, but the 22 47 filled in the highs and lows nicely.

EQ: There are literally dozens of strange instruments responsible for hundreds of strange sounds on the record. . . .

TR: Everything from junk lying around the studio to things we picked up at thrift stores. Almost anything we could get our hands on found a place on the record. A friend of mine came back from China with a bunch of cheap instruments he bought for a total of $10, and those *all* got used. For example, we took the keys off a vibraphone, placed them on the head of a kick drum, and recorded a counter rhythm/melody by miking inside the shell, the drum/keys being struck with a mallet, and then sped up the tape for more effect. We recorded some of the vocal harmonies through lapels. . . .

BD: At one point we tried to imitate the bass you hear from those huge woofers in people's cars. We pushed the bass cabinet up against this old furnace, downtuned the bass, struck the note and slowly tuned it up to pitch — miking the furnace for the rattle.

EQ: You're all nuts. It makes sense why you decided to work together.

BD: Yeah, we're all suffering from the same illness. —*Shane Mehling*

EQing Life Into POD Sounds

Guitar and bass amplifier emulators like the Line 6 Pod, Vox Tone Lab, or Behringer V-Amp can sometimes lack the true tones of the amps they model in their presets. I've noticed that many times the low end richness of some of the original amps are what's most unfaithfully reproduced — but there's a fix: EQ an increase between 50Hz to 120Hz to add more deep low end, while reducing in the 150Hz to 300Hz range to combat muddiness. If you need more "bite" to your tone, reduce in the 700Hz to 4kHz range; and if you need to bring out more midrange movement, simply reduce between 175Hz to 400Hz.
—Glenn Bucci

Addition

In reference to the October issue on Podcasting, the buy-out music library is another option for using music legally in podcasts. While the costs of individual CDs can run as high as $129 (or as low as $29), the buyer has the right to use the music as underscores in podcasts for 99 years. Libraries are available from companies like CSS Music, The Music Bakery, The Production Garden, and many others. (Thanks to reader Marshall Such for the information.)

A TOUCH OF CLASS

by Vincent Miraglia and Rich Tozzoli

We've all heard the term "Class A" used as a sales attribute for certain pieces of gear. But what exactly does that mean, and how does Class A design ultimately affect your sound?

Amp classifications are based on a circuit's *topology* — an arrangement of components and how they connect together. Circuits classified as A, A/B, and B are defined by what percentage of the input signal is used to turn on the amplifying device. Typically, vacuum tubes (valves), transistors such as JFETS (Junction Field Effect Transistor), MOSFETS (Metal-Oxide Semiconductor Field Effect Transistor), and ICs (Integrated Circuits) are used as the amplifying devices in the design of audio amplifiers.

What makes a Class A amp great, you may ask?

- The sound quality is pure.
- The design is inherently linear, because the output varies in direct proportion to the input.
- It introduces no crossover or zero-crossing distortion (see below).

And on the downside . . .

- It's inefficient because the amplifying device is always on.
- Class A amps run very hot. Because of this there can be a question of reliability, due to higher current demands and shorter tube life.
- They have relatively low output power.

Breaking it out even further, there are a couple of amplifier circuits that relate to Class A design.

SINGLE-ENDED

The single-ended Class A amp, a basic and straightforward design, originated before the push-pull amplifier. The signal flows from input to driver to output stage.

To illustrate single-ended Class A more clearly, visualize a cellist playing a note. The cellist must extend the bow across the string and then retract it. This is similar to a Class A single-ended amplifier. Regardless of whether the signal alternates positive (extend) or negative (retract), this *single* output device (cellist) handles the entire output waveform. Just remember that as the design is linear and there is no crossover distortion (see later), the resulting sound quality is pure.

PUSH-PULL

Another variation is the push-pull Class A amp. This circuit is more complex then single-ended, but its design is still relatively straightforward. To illustrate the push-pull Class A concept, let's imagine a second cellist grabs hold of the bow at the other end. Both cellists push and pull the bow across the strings equally. This is similar to a Class A push-pull amp. Again, regardless of how the signal alternates positive or negative, the *two* output devices equally handle the entire output waveform.

Class A amplifiers use one or more vacuum tubes or transistors as the output devices. These will conduct electricity for 100% or 360° of the input signal and therefore reproduce the entire waveform as a single, continuous entity. The fundamental point here is that the output devices are *always on* and have a constant flow of electricity, compared to other amp classes.

Now let's look at a couple of Class A example products, starting with the popular Vox AC30 guitar amplifier. The AC30 uses a quartet of EL-84 tubes in a push-pull configuration. Being that the AC30 is Class A, the tubes are always running, which helps provide its unique sound. It has an immediate response with a defined high end. This is similar in class to a Fender Champ amplifier, which is single-ended Class A with that distinctive Fender clean tone. On the other hand, a higher-powered guitar amplifier such as Marshall uses EL34s or 6550s in a push-pull configuration, and typically operate in Class A/B. This leads to — you guessed it — a different kind of sound.

Next, let's explore a typical Class A mic preamp. These can use vacuum tubes, transistors or ICs as their amplifying devices. They commonly employ robust power supplies, incorporate short signal paths, and can utilize transformers (or not).

Some high-end Class A preamps use discrete circuitry, which means having separate electronic components and no ICs used in the signal path — again providing pure, clean tone. However, sometimes there is a misconception that Class A cannot use ICs. This is not the case, as there are plenty of designs that use them.

OTHER CLASSES

Class B amplifiers conduct 50% or 180° of the input signal. This means each output device is on for exactly one-half or 180° of the input signal, so Class B amplifiers require two output devices to operate. This class of amplifier runs cooler, is more efficient, and offers more output power than Class A. The main drawback to this type of amplifier is degraded sound quality — hence you don't see marketing departments trumpet Class B designs as they do with Class A. This is mainly due to *crossover distortion*. Crossover distortion, or zero-crossing distortion, is the degradation in a musical waveform within a push-pull amplifier where one tube or transistor hands the signal over to the other — the point in time where one tube turns on while the other one turns off.

Class A/B amps conduct more than 50% or 181° to 359° of the input signal, so Class A/B amps require two output devices to operate. Class A/B amps possess advantages of both Class A and B: They approach Class A sound quality while having the efficiency of Class B. Note that there are also two sub-classifications of Class A/B, which are Class A/B1 and Class A/B2, developed for vacuum tube design.

And what are "Class D" amplifiers? Briefly, these convert the input signal into an *analog* pulse-width modulated square wave. Transistors do the work here, and their function is to turn on or off (like a switch), and thus represent positive or negative values. Higher-quality Class D amplifiers have been steadily working their way into subwoofer amplifiers used in the pro audio industry.

Soft Compression For Harsh Keys

You'll find with many keyboards, particularly cheaper models, there will be a noticeable lack of warmth, an overall "electronic" vibe, that just may not suit your needs (*i.e.*, they will sound just a bit too synthetic.) In the case of slightly harsh sounding keys that could benefit from subtle compression, employing a soft knee compressor with the ratio set to approximately 2.5:1 will help smooth out the sound of the keys and give you the silky sound for which your mix may be crying out. *—Glenn Bucci*

DEBATE:
IS MYSPACE STEALING YOUR MUSIC?

A new urban legend has begun to grace our inboxes. It goes something like this: MySpace owns everything that is posted on it and if you clicked "yes" to their user agreement and then posted your pictures or music to your page then you've given up your copyright to that material.

I've received several panicked emails wondering if this rumor about giving up rights to music in exchange for use of the service is true. In my gut I knew it could not be, so I asked my list of industry lawyers what they thought.

The first issue to address is the click-to-agree "user agreement." Can you actually transfer a copyright in this way without any exchange of money? The simple answer, according to my attorney-affiliates is, theoretically yes, but in reality, probably not.

By law, a copyright, like most real estate, can only be assigned in writing. While recent cases have shown that clicking an on-line agreement is the legal equivalent to "getting it in writing," this is not absolute. Also, there is another aspect to the assignment of copyrights — if there is a dispute over two parties claiming rights to an assigned copyright, the one who offered "reasonable consideration" generally wins over the one who just had it on a "handshake."

So, for example, if you had music on MySpace and you wanted to grant rights to that same music to a major label and MySpace then came out of the woodwork to assert ownership over the content, could they get away with it? To do so, they would have to make the argument that the service they offer is "reasonable and valuable consideration" and that you fully understood that when you innocently (without the advice of a lawyer) clicked "yes" to the user agreement, you *knew* that you were assigning them your music in exchange for their "service." [Section 205d of the Copyright Act.]

Seems like a stretch. Unfortunately, because "dumb" is not a real legal defense, the "I didn't know" excuse is a serious gamble that will hinge on how good your lawyers are, and/or how sympathetic your judge might be.

My personal take is that even the late Johnnie Cochran would have to argue his ass off to get a Federal judge to coldly agree that giving up valuable (or even valueless) copyrights is a fair exchange for a free Web page. But some of my legal watchdogs warn that I should not give the impression that money *needs* to change hands in order for you to *accidentally* click away a copyright. However absurd it might seem, you can assign a valuable song without *any* exchange of cash, even though it happens very rarely.

The good news is that there is great misunderstanding about what the MySpace language user agreement says. Here's the clause: "You hereby grant to MySpace.com a non-exclusive, fully-paid and royalty-free, worldwide license (with the right to sublicense through unlimited levels of sublicenses) to use, copy, modify, adapt, translate, publicly perform, publicly display, store, reproduce, transmit, and distribute such Content on and through the Services."

Nothing in there implies a permanent assignment of *exclusive* rights. Therefore, if a major label wants to pay you for the recordings posted on MySpace, you can still take their money in good faith. (Such is not the case with certain digital distributors who take exclusive rights just for the privilege of delivering your songs to iTunes.)

Conversely, the language in the MySpace agreement is very similar to what one might find in a standard release for MTV, which keeps a catalog of unsigned music that they use for filler and background in promos. It's great exposure for an unknown act; the artist gets no money for this license and MTV gets to exploit the music in any way imaginable, *but they don't own it.* So if you're upset about the MySpace situation ask yourself this: if MTV wanted to use your music in a national add campaign for little to no money, would you say, "yes?" If you would, then you have nothing to fear with MySpace.

A third and final factor to consider is that even if MySpace could legally claim they owned your music just for posting it on their site, would they do it? While no one knows the mind of another, I think I can say with reasonable certainly that they would not. The ill will it would create would utterly destroy the $100 million dollar value of the overnight sensation. MySpace was recently bought by Rupert Murdoch. Rupert is a pretty smart guy. While his taste or style is something that you are free to criticize, this does not change the fact that like most billionaires, Rupert likes a nice profit and probably does not feel like throwing $100 million out the window . . . because $100 million here and $100 million there, pretty soon we're talking about real money. Even one lawsuit about this matter would spread like a massive virus.

In fact, MySpace recently responded to the controversy, eliminating all doubt about assignment of copyrights. They amended the user agreement with "MySpace.com does not claim any ownership rights in the text, files, images, photos, video, sounds, musical works, works of authorship, or any other materials (collectively, 'Content') that you post to the MySpace Services. After posting your Content to the MySpace Services, you continue to retain all ownership rights in such Content, and you continue to have the right to use your Content in any way you choose."

Post away. *—Moses Avalon*

SESSION FILE: THE FLAMING LIPS

Goin' On: Recording Mystical With The Flaming Lips

by John Payne

The Flaming Lips' latest release, *At War With the Mystics*, is but one more adventure in a long string of sonically inspiring and surreally rocking musical experiments; one more recording that serves as a great maze of ever-shifting textural gambits that the Oklahoma City trio has turned out in their 20+ year existence. A conglomerate of some of rock's most technically adept and creatively inspired musicians, the Lips (comprised of singer-guitarist Wayne Coyne, multi-instrumentalist Steven Drozd, and bassist Michael Ivins) recorded and mixed the album readily ignoring convention — keeping their eyes on the prize while simultaneously being as sonically ambitious as having a Warner Bros. sanctioned budget (and full access to long-time producer Dave Fridmann's state-of-the-art Tarbox Road studio) would allow them to be. As a result, *At War With the Mystics* is a collection of uniquely characteristic tracks that benefit from pure, unbridled experimentation with a veritable mountain of odd instruments and non-traditional techniques.

"We don't know where we're going, that's the problem," says Steven Drozd, laughing. "That's why it takes us so long to make a record. Through the whole process, we're always second-guessing ourselves, or thinking that we've gone too far — because we're *always* going too far."

YOSHIMI BATTLES THE ORTHODOXY

"A lot of bands will record all their songs, and then mix the album," says Drozd. "We record the song and *then* mix the song, one song at a time. It's a process that enables us to figure out what we're trying to do."

This is quite the antithesis of what's generally regarded as "efficient" album-making, but hardly a surprise given the Lips' collective predisposition towards doing things the wrong way for the right result. And ringleader Wayne Coyne considers this Multiple Personality Disorder approach as being integral to the Lips' overall goals of being a band that, much like the Beatles before them, can go from varying extremes in sound from track to track — but in a way that, ultimately, makes total sense to the listener.

"Where we're really jumping from almost different personalities within ourselves, we try to track only two, maybe three, songs per record," says Coyne. "So for us, from one song to the next, we can sound like an utterly different ensemble."

HOW WOULD WE KNOW?

Sounding like an utterly different ensemble is the Flaming Lips' critical goal toward which their choice of instrumentation, effects, and engineering schemes strives — an ethos that, particularly on *At War With the Mystics*, leaves the listener not entirely sure of what band, let alone what sound source, he/she is hearing.

"We've never used an *actual* guitar synth," Drozd interjects. "A lot of those sounds come from a guitar signal running into a Line 6 DL4 delay, envelope filtered through various pedals and, finally, into a sawtooth wave generator."

Which is not exactly an unfamiliar approach to many; a large number of musicians have learned to string together plenty of different gear to achieve sounds that were previously deemed out of their price range — and that formative stage troubleshooting is something that has influenced Drozd's method, most notably in his choice of cheap keyboards which, as he tells, is his secret weapon for the Lips' psychedelic emissions.

"On this last record," he says, "I was buying as much cheap, weird gear as I could find, stuff that I knew no one else had or *would* use — like my Radio Shack ConcertMate MG-1 Keyboard which, ran direct, sounded real unique. It's funny: I bought a $1,700 Nord Electro 2, and I think I've used that twice since I got it three years ago; but this Radio Shack keyboard, that I got for literally $11, was used all over the record."

THE SOFT (SYNTH) BULLETIN

Luddites, however, the Flaming Lips are not. On tracks such as the ambient-synth laden "The Wizard Turns On," Drozd opted for working with soft synth programs such as the multitimbral Absynth 3, employing the Meta-Physical setting for creating the song's netherwordly soundscapes. Similarly, the prevalent Mellotron sounds that permeate many of the *At War With the Mystics*' tracks were synthesized primarily in Reason 2.0 or performed on Drozd's trusty Roland EP-707.

"You know what?" Drozd inquires. "We've never used a real Mellotron, ever. Years ago I was going to lay out $10,000 for one, and now I'm glad I didn't because I think we ended up with unique Mellotron string sounds just by using the EP-707, which we've used from *The Soft Bulletin's* "Race for the Prize" onwards. But for this album we also ended up using Mellotron M300 and M400 ReFills in Reason, which have great sounds."

Which could be considered for the best, as recording with a real Mellotron is notoriously difficult, but Drozd assures that it wasn't about cutting corners as much as it was a choice made simply in tonal regard. In fact, the Lips had a Mellotron at their disposal, as Drozd tells: "At one point, Dave had another band up there recording, and they'd bought a Mellotron, and brought it to the studio. We didn't like it; we thought it didn't sound very good. For a lot of our Mellotron parts, the more lo-fi it is, the better it is. You start getting really pristine . . . and then it just dies. We didn't want *that* sound — we wanted that creepy "Nights in White Satin" sound."

TRANSMISSIONS FROM THE STUDIO HEART

For the distinctive wah-fuzz sound that can be found on many of the *At War With the Mystics'* tracks, Drozd nods towards the DeArmond Thunderbolt Wah, a relic of Tarbox Road for which Drozd has since been scouring eBay. But his vintage guitar proclivities also came to the forefront on the *At War* sessions, perhaps most notably the 12-string Gagliano that can be found on tracks such as "Pompeii Am Götterdämmerung"—a chime-y, thin, and tinny-sounding guitar track which Drozd claims was inspired by the Beatles' "Fixing a Hole."

Equipping the guitar with an acoustic guitar pickup, Drozd and Fridmann sent a mixture of direct and miked signals (sometimes so close-miked that, as Drozd tells, "the mic was put directly against the body of the guitar") to tape via Tarbox Road's Otari Concept Elite 40 x 24 console, into a Studer A80 VUII 2-track recorder. But for songs such as "The Wizard Turns On" and "The Sound of Failure," Drozd credits the characteristic guitar ambience with Fridmann's ingenious room miking techniques, tactics employed to a similar end on many of Drozd's drum tracks.

"If we're going for that big room sound," he says, "sometimes we'll have to literally just throw up two mics at the opposite end of the room — no close miking at all." But this is dependant, as all things are with the Flaming Lips studio endeavors, on mood and intent — such as on "The W.A.N.D." where Drozd, outfitted with his trusty Dixieland-style drum setup ("just the cheapest little 18" kick, a cheap little popcorn snare, no toms — that's my optimum set"), was given the full close-miked treatment throughout the set by Fridmann. "For that song," Drozd continues, "we got a sound that I love. I was playing kind of heavy, but kind of open, and by taking the ambient mics out we ended up getting something that was super tight and full of attack."

THE VOCALS TASTE METALLIC

With all the various instruments performed on by Drozd — and Fridmann's numerous approaches to capturing a band that seems to change by the moment — it's vocalist Wayne Coyne's "non-singerly" voice that manages to pull the sometimes disparate/always erratic songs together, conveying a cheerful melancholy that firmly embeds itself in the listener's mind. But while Coyne's voice may be the glue in the Lips' sound, his approach to recording vocals is as strange as one would expect given the band's penchant for experimenting at every turn while in the studio.

"I'm always demanding Dave to change tonal texture," Coyne says. "So on 'The Sound of Failure,' we miked the bottom of the bell end of a big brass record horn running from the end of a P.A. speaker that my voice was coming out of, to give it that sort of tinny quality, almost like a Billie Holiday song."

Yet some of Coyne's most varied vocal takes, he feels, were achieved recording outside of Tarbox Road, particularly inside his famous "giant plastic space dome" at his Oklahoma compound.

"We'd track in this round, plastic room, and the reflections would have as much to do with the end sound as the source and the mic. We would put the mic right in the center, and then I would run to the back, and to the side, and to the back to perform. You could almost understand the unpredictable reflections by moving around in the dome — but every time you thought the sound was bouncing away from you it would come jumping right back at you."

FLAMING RHAPSODIES

For all the talk of the Beatles both inside and outside the band, perhaps it's Queen that played the role of primary inspiration for the *At War With the Mystics* sessions — something that is evident in the album's culmination: An online buyer's exclusive cover of Queen's "Bohemian Rhapsody." This homage is most appropriate given the typically untypical Lips production ethos of almost far-fetched amounts of layered vocals, guitars and effects — a sound that has been described as operatic in end result, and thus not exactly a sacrilegious adoption. "We had already been doing a bunch of stuff in the studio that we thought reminded us of Queen's more complex stuff, so it wasn't out of our range to do it," Coyne says.

But the Lips had no intention of taking the "easy" route of merely re-creating the song, deciding instead to deeply examine the recording techniques of Queen producer Roy Thomas Baker in an attempt to crack the code of the track's lush harmonic and opulent textures. "We picked it apart so thoroughly," says Coyne, "We discovered that Roy Thomas Baker recorded two drum kits — that's why Roger Taylor sounds so clumsy sometimes."

So the Lips, paradoxically, innovated in part by recreating some of Queen's original production techniques, with Fridmann recording Drozd's drum tracks on two separate kits and elsewhere, along with Coyne, patching discretely EQ'd guitar tracks phrase-by-phrase, much as guitarist Brian May did on Queen's seminal *A Night at the Opera*.

But while their cover of "Bohemian Rhapsody" may very well be the last song a listener experiences on *At War With the Mystics*, it's placing as the first track recorded for the album's sessions is telling as to why the seemingly difficult process of constructing the album in the studio was much less frustrating for the band than one would expect.

"Once we tackled that behemoth, we felt unstoppable," Coyne says." Getting inside that song just really showed us that new level of ridiculousness to which we could go." EQ

SUCCESS STORY: ANDY JOHNS

From Leading Zeppelin to Tuning Television, Andy Johns Did it — and Saw it — All!

by Dan Daley

Andy Johns — engineer, producer, raconteur — was born in the U.K. circa 1950 almost genetically predispositioned to making great recordings. Watching his older brother, Glynn, rapidly move up the ranks as a producer in the 1960s, Johns decided to follow in those familial footsteps, landing a gig as Eddie Kramer's tape op in the early 70s and moving on to amass a discography that would include key records for Led Zeppelin, Jethro Tull, Television, and the Rolling Stones. Establishing himself as a leading engineer in the mid-'70s U.S. rock scene, Johns soon turned more towards "production," eventually becoming a fundamental influence on the sound of '80s modern hard rock. And he's not slowed down a bit in recent years; in fact Johns has followed the advancement of technology into DVD productions (for acts such

as Godsmack) while still finding time to produce records from his base town of Los Angeles.

The aforementioned latter-day projects notwithstanding, Johns remains happily defined by his experiences with seminal rockers like Eric Clapton and Jimmy Page during the U.K. music revolution of the late '60s, which was as much about the culture as the music — a milieu of sex, drugs, and rock & roll. It permeates his recollections, and serves as a blueprint for how he continues to work (minus perhaps the whole ducking from a drawn shotgun wielded by an inebriated and irate Keith Moon bit). Johns' life in that period was truly a front-row seat to the madness that created the world's most enduring music since Mozart. "Figuring up the brain damage, I have an IQ now of about 75," he jokes, in his raspy voice. "But, God, it was worth it, eh?"

BONHAM'S SECRET

A bass player when away from the console, Johns has a natural, shared empathy with other rhythm section players, particularly drummers. Such kinship is apparent in his recordings, with one of the most notable pairings being with a young John Bonham. Commenting on the Olympic Studios sessions for *Led Zeppelin III* — an album that was among the first to benefit from 16 track capabilities — Johns says, "I put his kick drum on two tracks. John was always saying there was not enough 'thrush' on the kick. It was a pretty dead room, so I asked him if he had another kit. He brought in this green sparkly thing, and the bass drum had a massive sound. I put one D-30 on the outside of [the shell] and another deep inside; you can actually hear the pedal squeak. An M-160 was on the snare and one Neumann U-87 as an overhead — very sparse miking, really. The sound was all in the drum, the drummer, and the moment."

But that sound was also attributed to careful "tuning" on Johns' behalf. From early sessions to the current projects he undertakes, Johns' still relies on "tuning" drums to their resonant frequencies: He removes the heads so he can bang directly on the shell whilst striking notes on a piano, as he looks for the note with which the drum most closely resonates. Placing the heads back on the shells, Johns then "tunes" each drum to its respective note.

GUITAR TRICKS FOR THE MIX

A protocol Johns uses for recording guitars, he tells, is this: "If you're working with, say, a 4x12 cabinet, get a pair of [Shure] SM57s and point one dead on the middle of one speaker cone, backed off by about half an inch. Then, put the other 57 at a 45° angle to the center of the cone, placed for perfect phase with the first mic. Then bring them up on the board: The 'dead-on' mic will have the top end, and the 'angled-on' mic will have the bottom. Then add a [AKG] 414 or a [Neumann] U47 FET on another speaker, and mix it in. It always works."

THICKENING VOCALS

When recording vocals, Johns adheres to a classic recording philosophy in terms of gear. "I still use either a [Neumann] U87 or an AKG 414 with a touch of a [Urei] 1176 as it's being recorded; compressed again in the mix," he explains. "I'll also add a very short delay — 28 to 32ms — on one channel and another, longer delay — about 250 to 600ms — on another. This adds a lot of depth and dimension to the vocal. You can use a lot less reverb as a result and get the same effect." But Johns warns that this is not a replacement for a properly doubled vocal, as it further reminds him of one of his main grievances with the digital age of recording. "It's nice to be able to fly stuff around, but it also makes people very lazy," he growls. "I have people now complaining to me when asked to double a part; they'd rather make a digital copy. But it's not the same thing!"

SMALLER STUDIOS/SIMPLER SESSIONS

Johns does not work out of a home studio; he doesn't even own one . . . nor does he have much in the way of personal equipment. Being stationed in Los Angeles, he notes happily that there are

> **If you're working with a 4x12 cabinet, get a pair of SM57s and point one dead on the middle of one speaker cone, backed off by about half an inch. Then, put the other 57 at a 45° angle to the center of the cone, placed for perfect phase with the first mic.**

plenty of great rooms — and great consoles — nearby, stating that he's perfectly content working out of various facilities, much as he's done for the entirety of his career.

But that doesn't mean that he necessarily prefers to work out of the huge L.A. studios that have survived the digital revolution. Quite to the contrary — Johns claims a serious affection for smaller studios. In fact, he almost sentimentally relays stories of using London's Morgan Studios circa 1968–69, where he worked on records for Free, Spooky Tooth, Humble Pie, and many others. It was there that Peter Frampton, then a member of the Herd, contacted Johns about doing a collaborative record with him and Steve Marriott of the Small Faces. "Steve was on a 'Mr. Natural' trip," Johns remembers. "No production — just set up and play . . . with no overdubs! Then go into the next room over, and leave the door to the control room open just a crack for playback. If, in the other room, it sounds good to you then you know you've done it right!"

"It was very primal and fun," Johns continues when questioned about his excitement over recording so recklessly. "Morgan had this Scully 8-track deck that gave you a bad bubble below 100Hz because of the way the tape went across the heads. But it never broke down, and that was important when you wanted everything done on the first take. Back then, we'd mix entire records in a day."

BIGGER BUDGETS/VINTAGE VIEWS

But when Johns made the hop across the Atlantic and decided to set up shop in L.A., it didn't take him long to find a comparable, in terms of comfort, studio to work out of. United & Western Studios, built by Bill Putnam, was every bit as legendary as Johns' beloved Olympic, and lent itself as a good home for Johns innovative approaches. In fact, it was at United & Western where Johns recorded the first two Cinderella records which, after selling three million copies each, started a streak that saw him working with some of the '80s biggest hard rock acts, including Ozzy Osbourne and Van Halen. And it was in this genre that Johns was given the opportunity to breathe life back into drum sounds — one sonic aspect of popular '80s music that was being either too heavily processed or, perhaps, downright neglected by navel-gazing engineers with one-too-many crutches. "At least 50 per cent of their sound was coming from the heavily gated echo returns," Johns explains. "Massive AMS! And it sounded like it was being recorded in a racquet-ball court down the hall."

So for Van Halen's *For Unlawful Carnal Knowledge*, Johns went back in time a bit — to the days of getting the most from each mic on a kit *à la* tracking

It's nice to be able to fly stuff around, but it also makes people very lazy.

John Bonham. Alex Van Halen, who had been used to nearly three dozen mics being utilized at any given time for his behemoth kit, had to be knocked down to just a spare few in order to escape the arguably cluttered, if not criminally overdone, modern sound. As Johns tells it, he took Van Halen out of the United & Western's main live room, and set the drums up in a nearby storeroom — utilizing only a couple Neumann U47s and Telefunken 251s, set up as stereo pairs for overheads, instead of miking each individual cymbal and/or tom. Though taking the approach of olde didn't make for the easiest sessions, as Johns explains, "It took us a couple of weeks to get each song; nearly a year for the entire album!"

It's an older way of working, stemming from an older way of thinking, but Andy Johns is fine with that. For as annoyed as he can be at the notion of resting on laurels, he takes no small satisfaction in having been there when the world of recording music underwent its first huge change. "When we were making records in the '60s, there was a passion that came about, a sense of discovery, which we don't have now. There was a sense then that we could rule the world! Every day we would wake up and believe we had a shot at making the best single in history. And it wasn't just us. And it wasn't just the music. It was the times." EQ

TECH BENCH
Build Your Own "WAA" Boxes

by Craig Anderton

Fig. 1: A-B Switch Box photo.

WAA stands for "weird-ass adapter." It's something you won't find at Radio Shack or your local music store, but it's incredibly useful. The only catch: You're going to have to build any WAA boxes yourself. But don't panic! The process is simple, and doesn't take much time (or effort) at all. You do need to know how to solder; if you don't know how, commune a bit with Google and the Internet, and all will be revealed.

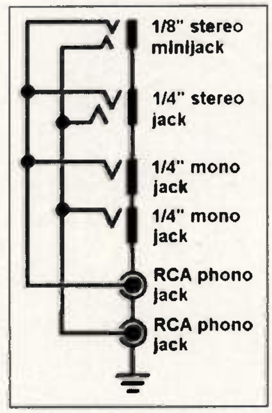


Fig. 4: The Do-All WAA Box.

WAA TOOLS

You'll need some tools, but you may have some of them already.

- Variable speed electric drill with a selection of bits (1/16", 1/8", 1/4", and 3/8" are particularly important). When drilling a metal box, you'll need a center punch to create a small indentation prior to drilling. With a plastic box, unless you can find some bits for drilling plastics, begin with a 1/16" hole, then enlarge slowly by using ever-larger drill bits.
- Vise grips, crescent wrench, and/or nut driver for tightening nuts on jacks and switches.
- Small needlenose pliers for bending and working with wire and component leads.
- Diagonal cutters for cutting wire.
- Wire stripper for removing insulation from hookup wire.
- An assortment of screwdrivers, including Phillips head and jeweler's types as well as regular flat types.
- A small vise to hold parts for soldering.
- 60 watt, small-tipped soldering pencil or (for those who like to go first class) a temperature-controlled soldering station. Wear eye protection while soldering; sometimes the rosin can spit out. Also, solder in a well-ventilated area.
- 60/40 rosin-core "multicore" solder intended specifically for electronics work. Never use acid core solder! It's for plumbing.
- Hookup wire; #22 or #24 gauge stranded works fine.
- A small metal or plastic box in which you can mount the parts. Hey, some people use coffee cans . . . whatever works.

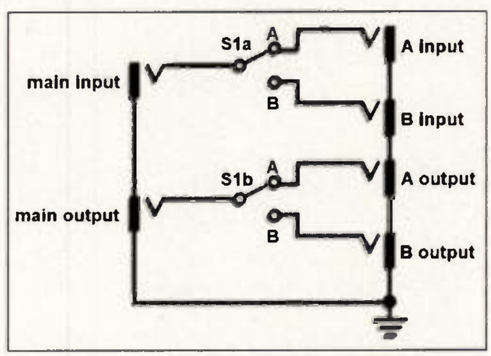


Fig. 2: Schematic of the A-B Switch Box.

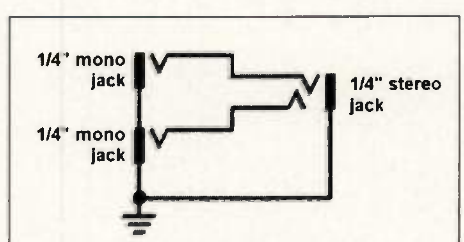


Fig. 3: Stereo/Mono Breakout Box schematic.

So now that you have your tools, what can you build? Here are a few quick examples.

A-B SWITCH BOX

Figure 1 shows a photo of the front and back of an A-B switch box. This switches a main input and output between two different effects or other devices; Figure 2 shows the schematic. S1 is called a "double pole, double throw" switch. That doesn't look so hard, does it?

STEREO/MONO BREAKOUT BOX

The stereo/mono breakout box (Figure 3) provides an easy way to get mono gear to relate to gear with stereo insert jacks.

Wire one mono jack hot lead to the stereo jack tip connection, the other mono jack hot lead to the stereo jack ring connection, then connect all the grounds together.

To use the box, patch a stereo cord between a device's insert jack and the breakout box's stereo jack, then patch the mono jacks to your signal processor's in and out connections. If you don't get the tip and ring connections right the first time, reverse them and you should get signal.

THE DO-ALL WAA BOX

Figure 4 shows another real simple, but invaluable, box. It just has a bunch of jacks wired together: stereo phone jack, two RCA phono jacks, stereo minijack, and two 1/4" phone jacks. As one example of an application, this is really useful for laptops. Run a cable with two mini plugs between the computer's audio out and the stereo minijack, and you now have a breakout box: Use the stereo jack for headphones, or use the two RCA or 1/4" phone jacks to feed a stereo system or mixer.

These projects can take less than an hour if you don't care too much about looks. Besides, it's winter . . . a warm soldering iron will help heat up your room! EQ

Tool Box

AKG Acoustics Perception 400

This multi-pattern, large-diaphragm condenser microphone features selectable cardioid, omni and figure-eight pickup patterns with switchable 10dB pad, switchable bass cut filter, shock mount, metal case, and two separate 1" diameter vibrating surface capsules with gold-sputtered foil diaphragms.
www.akg-acoustics.com

McDSP Updates Project Studio Bundle

McDSP now includes Analog Channel LE and ML4000 LE in the cross-platform Project Studio bundle ($495). Analog Channel LE contains the tape emulation algorithms from the AC2 configuration of Analog Channel, while the ML4000 LE contains the look-ahead brickwall limiter from the ML1 configuration of the ML4000. Upgrade for original Project Studio customers is $95.
www.mcdsp.com

Eventide Anthology II Bundle for Intel-powered Macs

This Mactel upgrade (free to current Anthology II customers) allows users to take full advantage of Apple's newest computers, including the Mac Pro with quad-core architecture. Anthology II includes the best Eventide effects from the past 35 years.
www.eventide.com

Arturia Minimoog V1.6

This update offers Universal Binary support, is VST 2.4/Cubase 4 ready, and includes tweaked factory presets and some bug fixes.
www.arturia.com

Soniccouture Synthi AKS For Kontakt 2

This 24-bit instrument for Kontakt 2 explores the legendary sound of the vintage E.M.S. Synthi AKS from 1972. The library features detailed multisamples, a selection of hits and effects patches, sequenced loops, and convolution of the famous Synthi Spring reverb.
www.soniccouture.com

Yamaha MSP STUDIO Monitors

The top-of-the-line MSP7 STUDIO Powered Monitor Speaker incorporates a 6.5" polypropylene cone woofer powered by an 80W amplifier, and its 1 titanium dome tweeter employs a 50W amp. The frequency ranges are divided by an electronic crossover built into a one-piece molded enclosure with a rounded baffle. The MSP5 STUDIO is a two-way, bi-amplified bass reflex system with more compact dimensions. It features a 5" cone woofer driven by a 40W amp and a 1" dome tweeter driven by a 27W amp. Other than woofer size, enclosure size, and amplifier power, the MSP5 uses the same technology as the MSP7.
www.yamaha.com

SM Pro Audio M-Patch 2

The M-Patch 2 is a compact desktop/rack-mountable passive volume attenuator and patch control device. It provides an ideal stereo level control solution for monitoring or distributed sound applications, and passively attenuates stereo signals from soundcards, CD players, preamps, and mixers. The unit features balanced combo XLR/TRS input jacks and XLR outs, RCA and 3.5mm jack inputs, mono/stereo summing switch, mute switch, A/B output pairs, and a built-in stereo headphone amp.
www.smproaudio.com

Bremmers Audio Design MultitrackStudio 4.2

Version 4.2 of MultitrackStudio (MultitrackStudio Pro Plus, $119; MultitrackStudio Professional, $69), the audio/MIDI multitrack recording program for Windows, is Vista compatible and offers several improvements, enhancements, and optimizations.
www.multitrackstudio.com

Steinberg WaveLab Studio 6

WaveLab Studio 6 ($399) for Windows provides mastering, editing, and CD burning tools for musicians and project studios at a lower price point that the standard edition of WaveLab.
www.steinberg.net

Obedia Launches Four Software Training Plans

Obedia has released four training plans, which can be purchased from select Guitar Center stores nationwide, for most major software products on the market. Now, anyone can access a personal trainer any time of day or night. Each training session is customized to meet the goals of the customer; an Obedia trainer will personally help the user plan out and then accomplish their goals.
www.obedia.com

Minnetonka SurCode for Pro Tools

Minnetonka Audio Software announces ProTools plug-in versions

($795) of their award-winning SurCode for Dolby Pro Logic II (Mac and PC), allowing Pro Tools users to encode Dolby Pro Logic II in real time. This allows game developers, broadcasters, video producers, and audio professionals to encode surround sound into stereo delivery formats.
www.minnetonkasoftware.com

Korg K61P USB-MIDI Studio Controller
The K61P ($450) combines controller functions with 24 onboard keyboard sounds. Assignable performance controllers include two knobs, two switches, slider, and a footswitch jack. They can manipulate numerous functions, allowing hands-on hardware access to virtually any software instrument parameter. Korg's proprietary ClickPoint controller can function as an X/Y joystick (with Hold function), or as a single-button mouse for computer control. Users can assign separate MIDI control messages to be sent by the X-axis (left/right) and Y-axis (forward/back) for control over two simultaneous parameters.
www.korg.com

Dangerous Music D-Box
D-Box ($1,400) is a 1U rackmout multi-purpose hardware product for DAW users. It combines eight channels of analog summing technology, integrated programmable monitor control section with analog and digital inputs, onboard 24-bit/96kHz D-to-A converter, talkback, dual headphone amps, speaker switcher, and more.
www.dangerousmusic.com

Magma PCI Expansion Products
Mobility Electronics' Magma PCI Express to PCI Expansion Chassis (compatible with Windows, MacOS, and Linux) offers a simple expansion solution that allows users to upgrade to the newest PCI Express computers without losing their investment in PCI hardware. The family of products features plug and play installation, offers configurations with different numbers of slots, and supports 5V only, universal, and 3.3 volt PCI and PCI-X cards.
www.magma.com

Two New Applets For Snowball USB Mic
Blue Microphones announces two new freeware Applets for their Snowball USB Microphone. Download the low gain applet for loud sounds (like drums, electric guitars, louder singing), and the high gain applet for quieter sounds like speech, podcasting, internet telephony, or recording sounds from a distance. You can also switch back and forth as desired.
www.bluemic.com

Disc Makers Introduces Custom Apparel For Indie Musicians
Disc Makers is now offering custom apparel and accessories for music artists, including free design services and low minimums. The new program is called Merch and helps independent musicians to outfit their fans in a variety of stylish, customized garments emblazoned with the artist's name, logo, or design. Merch is easy to use and doesn't require high minimum orders like many apparel companies. For example, a short run of 24 custom T-shirts start at just $6.99 each.
www.discmakers.com

Millennia Media HDOE/HROE Output Expansion Options
The HDOE and HROE output expansion options for the HV-3D and HV-3R eight-channel mic preamps provide an additional two buffered outputs per mic channel, for a total of three outputs for each microphone. The extra outputs are ideal for providing splits for live, broadcast, and recording feeds without resorting to transformers. They can also be used when tracking to record a clean version, a version sent through processing, and safety copy at approximately 12dB below the standard level to prevent digital overs.
www.mil-media.com

Rain Recording LiveBook and LiveBook Studio Laptops
Both LiveBook and LiveBook Studio feature Intel's Core 2 Duo technology, and approach the speed, HD capacity, and performance of a desktop workstation. The completely redesigned Rain Recording LiveBook offers a built-in web cam, Bluetooth, and internal SATA hard drive as well as the integrated Azalia audio chip; the LiveBook Studio features a new 17" widescreen display with built-in web cam, DVD/RW Super Multi drive, external SATA port, and Bluetooth.
www.rainrecording.com

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Salvage Your Drum

Wimpy snare? Muffled kick? Lousy drum part? There's hope!

by Craig Anderton

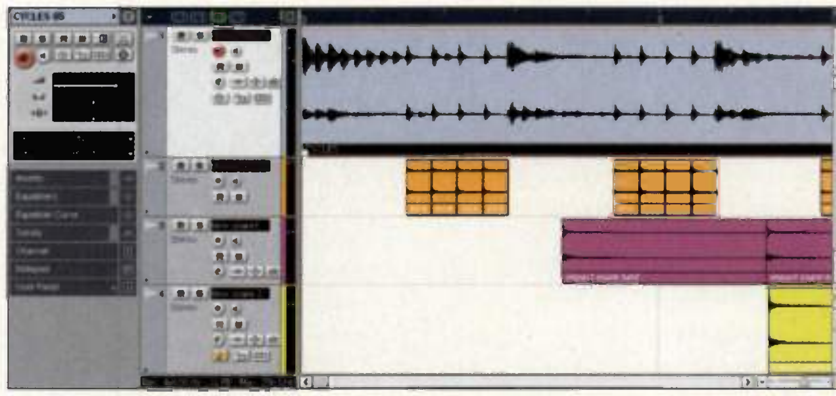


Fig. 1: Dragging over sampled hits to mask existing drum sounds can improve the sound of a drum track or even mixed drums.

Fig. 2: Two instances of Drumagog are inserted in Digital Performer; one is replacing the snare hit, and the other, the kick. The front instance is running in Visual mode, which makes it easy to set the sensitivity and resolution for optimal triggering.

If a pop/rock/dance track doesn't have a great drum sound, you're in trouble. Big trouble. The Drum Replacement 101 solution is to have a drummer come along and play with the original track, then erase the old drums. But this doesn't always work, and if you try to add drums after the fact to a track that doesn't have them, the process becomes even more complex.

This was brought home to me when I produced an album by Michael Kac and Linda Cohen that had a tasty harpsichord/classical guitar duet. After one of the tracks was done, we decided it would sound great with drums . . . oops. This was years ago, so I worked for hours doing digital audio stretching to get some loops to line up.

Then there was a cover version I did of Julian Cope's "When I Dream" that was mixed with a scratch drum track. I wanted to resurrect it, but no longer had the original tracks, just the mix — and the drum part was useless. Oh well.

But today's tools can salvage those kinds of tracks, and more. Here's how.

REPLACING DRUM HITS VIA DRAG-AND-DROP

Sometimes a tune is basically okay, except perhaps for a questionable kick or snare sound, or an electronic cymbal that would sound far better as a real one. In this case, you may be able to just drag in samples to "double" what you're hearing.

Figure 1 shows a Cubase project with a two-track mix in the top track. Below it, in orange, are kick hits dragged over from a Discrete Drums sample CD. The magenta clips are snares, and the yellow, "accent" snare hits for flams and such. Because these are lined up exactly with the existing hits, they "mask" the bad sounds with good sounds.

Replacing all hits in this manner can be tedious, but is likely less time-consuming than bringing in a drummer to play over the track, then moving some of the hits around so they fall exactly on the existing hits.

DRUM REPLACEMENT SOFTWARE

Several software solutions can analyze a file and generate triggers

Tracks

for alternate samples. SoundReplacer, part of Digidesign's Music Production Toolkit or available separately for $395, is an AudioSuite (non-realtime processor) that can pick transients out of a file, and split them into three velocity zones for triggering multisamples that get mixed back into the file.

An RTAS realtime option, TL Rehab ($495), works as an insert and allows realtime audition-ing of samples. Its principle of operation is similar to Drumagog, which is a more universal solution as it's a cross-platform plug-in that works with VST, AU, and RTAS systems. Drumagog is designed to be used with multitracked drums, where you have separate tracks for snare, kick, etc. It includes a set of excellent samples you can use to replace existing drum sounds, or you can create your own drum sample sets to work with Drumagog. (Drumagog Pro lists for $289; a lite version costs $199.)

To replace a drum sound, you insert Drumagog as a plug-in, then adjust its Sensitivity and Resolution controls for reliable triggering. A visual mode (Figure 2) shows the incoming drum hits to be replaced, along with a dot that indicates triggering; the height of the dot indicates the level (Drumagog's triggers not only follow hits, but track the level as well). This is vital when triggering drums multi-sampled at various velocities. There's also filtering if you need to isolate the drum from bleed — I've even used this feature to pull a drum sound out of a mixed track with some success, although the results depend on how "buried" the drum is in the mix.

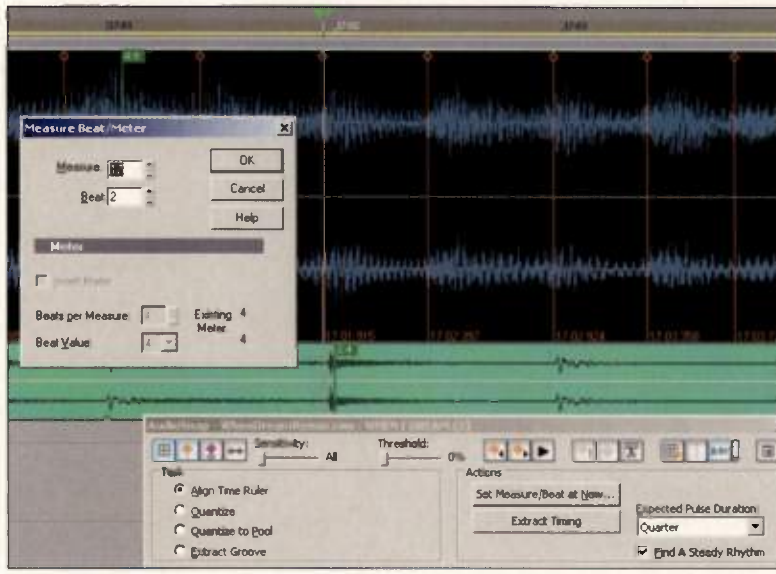


Fig. 3: The transient indicated by the Now time should be sitting at measure 17, beat 2. So, in the Measure Beat/Meter dialog box, measure 17, beat 2 is being "pinned" to that transient. After hitting OK, the tempo will shift so that the specified beat lands exactly on the transient.

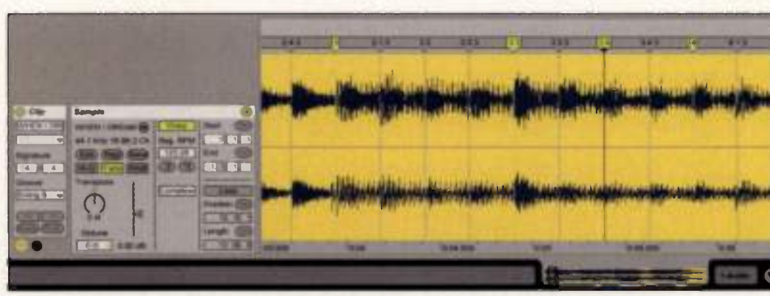


Fig. 4: Live's Warp markers have chartreuse "handles" that show the particular beat, and can be moved to line up with a file's transients. This pins the transients to measure or beat boundaries.

In addition to replacement, it's also possible to alter the sample pitch and the blend of the original and replaced sound. Latency is low — 3 ms worked fine in my system — but you can always bounce the track and slip it forward a bit to line up with existing drums.

Drumagog is surprisingly effective and also offers some advanced features, like a "ducking" option (*e.g.*, your "old" snare sound remains in an overhead mic track; duck the snare sound in the overhead track when the "new" snare hits). Drumagog can also generate a MIDI out if you want to trigger a soft synth or hardware synth with drum triggers. (Note that there's a downloadable 14-day demo at www.drumagog.com.)

MATCHING PROJECT TEMPOS TO CLIPS

Another approach is to use loops to "cover up" a bad drum part, like I ended up doing with the cover of "When I Dream" (check out the audio example at www.eqmag.com). This was recorded on tape over 20 years ago, and the original multitrack couldn't be salvaged, so all I had to work with was a 2-track 1/4" master tape with a less-than-stellar drum part. So I transferred the song over to Sonar, and got to work.

Because drums tend to hit on the beat, adding a drum part on top of the file with a drum part already mixed in tends to mask the original sound. So, you bring the original file into a host program, use the host's tools to match the project tempo to the rhythm of the file, and then you can insert loops or other drum parts into other tracks. Following are examples using both Cakewalk Sonar and Ableton Live.

Sonar 6's AudioSnap feature can analyze a clip, and place markers at transients. Generally, transients line up with beats; we can then "assign" a beat to a marker, thus causing the tempo map to conform to an existing piece of music. Here's how to do this using Sonar's "Set Measure/Beat at Now Time" option; the "Extract Timing" command seems better suited to clips with regular, well-defined transients, not program material.

1. If there's an existing tempo map, clear it.
2. Import the clip into a Sonar audio track.
3. Right-click on the clip and choose "AudioSnap Enable."
4. If the AudioSnap palette isn't showing, type Shift-A.
5. Set the tempo as closely as possible to the file's tempo. This isn't necessary, it just makes life easier.

Salvage Your Drum Tracks

6. Either place the Now time *exactly* on a transient marker that should line up with a particular measure and beat (*e.g.*, measure 3, beat 1), or better yet, use the AudioSnap palette's "Go to next transient marker" button or "Go to previous transient marker" to "park" the Now time on the desired transient.
7. Click on "Set Measure/Beat at Now Time."
8. In the dialog box that appears, enter the measure and beat you want to "pin" to the chosen transient.

Basically, you're creating a tempo map (Figure 3). If the tune has a fairly constant tempo, and you've set the project tempo pretty close to the song's tempo, find a transient marker that's close to the end (making sure you know the measure and beat to which it corresponds!) and pin the correct measure/beat to it. If you're lucky, this will "distribute" the correct tempo throughout the song. If not (or if the tune contains significant tempo variations), slog through the file a measure at a time, or a beat at a time if absolutely necessary, and you'll eventually get the entire tune mapped.

At this point, you can start adding in loops; assuming they're "acidized," they'll fit along with the tempo. If not, you can grab a loop's edge and while holding down the Ctrl key, "slip-edit" the length to fit a particular number of measures.

Ableton Live's "elastic audio" feature allows matching clips to tempo very easily; even better, Live does most of the work for you. The process starts by bringing a long file into Live's arrangement view. Live analyzes the file, and creates "warp markers" at rhythmically important places within the file (Figure 4). Live's ability to find these is uncanny — in many cases, you won't have to adjust the markers at all, and the file will slip right into the tempo.

However, suppose you're bringing in a file that was played by humans, and the song's tempo varies around 120BPM. By matching warp markers to transients that indicate the start of a measure or beat, you can "tweak" the tempo to match the file. You do this by simply dragging the warp marker to the transient; if Live hasn't put a warp marker where you want one, double-click on the beat in the timeline to create a warp marker, then line it up with a transient.

After you've used warp markers to define the desired measure boundaries, if you then insert loops into the song, they will automatically time-compress/expand so they fit into what the markers define as a measure. (To hear an excerpt from another song called "Modern World" that I updated using drum loops from Discrete Drums, and was also transferred over from an ancient two-track tape, go to www.eqmag.com.) Live's on-line help gives an example of how to "elasticize" a piece; even if you don't read manuals, you'll master the process a lot faster if you follow along with the documentation.

The bottom line: Never give up just because the drums don't make it — many times there is indeed a fix. EQ

Selective Beat Detective

Do multitrack rhythm correction in Pro Tools LE

by Rick DiFonzo

Pro Tools Beat Detective has been a boon to many engineers and DAW aficionados, and has recently become invaluable to me in my work as producer of drum loop products. When I upgraded to Pro Tools 7 LE, I was pleased to see that it included Beat Detective. But I was also dismayed to see that one key function was available only to TDM users: You cannot apply the same time correction to a group of tracks using Pro Tools LE. Or so they would have you believe. . . .

This work-around takes a few steps, but once you become accustomed to the workflow, it's a breeze and works very well (we'll assume you're familiar with how Beat Detective works).

First, create a group for all of the percussive material you want to process. (If you use the key command to enable and disable the group, things will go much faster.) Next, listen to the segment against a click, and pick the track that you feel will work best as the source track. If there are problems with the snare being early or late (or both), select that track. In the Region Separation window, use "High Emphasis" for a snare or other relatively high frequency material, and "Low Emphasis" for the low stuff. Adjust the "Analyze" sensitivity until you get markers where you need them. This is an important decision. Chances are that the drummer did not stray a lot between each beat, so try chopping things up in larger chunks — you can always undo it all and try again.

Once you have the source track chopped up, enable the group, and click in the first region with the grabber tool. Because the group is enabled, this selects all tracks (Figure 1). Use "Separate Region at Selection" (Mac Cmd-E, Windows Ctrl-E). This applies the cut point from the source track to all of the others. Repeat this process down the line, and you'll have chopped all of the tracks at the same point, allowing you to conform them all. (As the segments will be chopped front and back, you can save steps by choosing every second region.)

Fig. 1: Enabling a group selects all tracks.

Next disable the group, and select all of the newly created regions of one track (*e.g.*, snare); see Figure 2. Select "Region Conform," dial in the quantize strength, and click on the "Conform" button. Repeat this with all of the other tracks.

Unfortunately, Pro Tools LE requires that you perform these steps on each track individually. However, it does allow you to smooth the edit points for all of your tracks at once. Re-enable your group, select all of the regions, and perform the Edit Smoothing function. I would not recommend using any cross-fades, as they can compromise a sharp attack such as a snare.

When all of this is done, check each track for artifacts (flams, clicks, etc). You may need to move some splice points around and apply a crossfade manually. I sometimes slide the region start earlier (in Slip Mode) to lose an early snare hit that causes a click or a flam. A small crossfade at the new splice point fixes any problems.

The final step should be to make one contiguous file for each track using the "Consolidate Region" function.

IT'S THE LEAST YOU CAN DO

My general philosophy regarding any sort of time (or pitch) correction may be a bit old school, but I firmly believe that in *most* cases, the less you do to a performance, the better. When using Beat Detective, I almost never use 100% correction, and I usually use a low resolution when analyzing rhythmic content. If you chop up a drum segment in too many places, and use a strong correction value, it might as well have been programmed. This is not to say that there aren't times when rhythmic material should line up perfectly with a grid; that's a judgement call. Just consider this: Just because you *can* do something, doesn't mean you *have* to. Choose wisely.

Rick DiFonzo has toured and/or recorded with the famous and near-famous, is the co-founder of Discrete Drums, and is working feverishly on a new music library (www.difonzomusic.com).

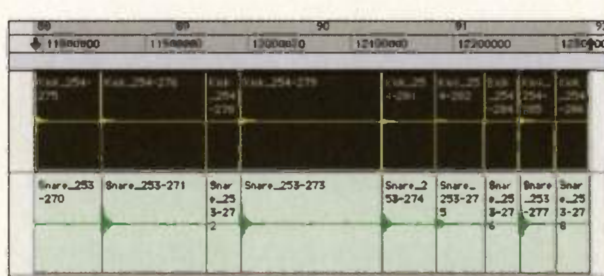


Fig. 2: Select all regions in each track, one at a time, then conform them.

Pre-Mix Vox Trix

Prep your tracks before the mix hits the fan

by Jeff Anderson

In a perfect world, mixing engineers would simply push the faders on the console to zero, press record on the two-track, and receive copious amounts of platinum records for all their hard work.

Back to reality.

Many times you will be handed less than perfect vocal tracks from the tracking engineer, and have to make do with what's been passed on to you. Or maybe the singer just flat-out sucks. Surely you can't just let these terrible tracks stay as they lay, so what can you do to make a less-than-stellar vocal performance sound nailed?

TAKE OUT THE GARBAGE

Fixing vocals is, like the AA mantra says, a "one-step-at-a-time" ordeal. Extra space is a breeding ground for sonic undesirables, so start by cutting away any extraneous sections and dead space out of your vocal tracks that may have been left to you after comping — leaving only a nominal amount of space as "bookends" on the takes. This ensures room for flexibility (like adding a fade-in or fade-out to "slide" into any residual noise level or "air").

NUKE THE NOISE

One of the main problems we must deal with in editing vocals is the presence of unwanted noise. The origins of these problems are numerous — perhaps a tracking engineer neglected to place a pop filter in front of a mic during a session as a way to combat high frequency loss, or the performer wanted to handle the mic while recording (a huge no-no, but something we've all had to deal with sometimes). Or maybe the vocalist performed too "straight-on" with the mic, or maybe it's just a prominent noise floor. Regardless of the genesis, vocal take(s) tend to be riddled, to varying degrees, with noise such as plosives (consonant sounds such as "pa" and "ba" that occur when the vocalist expels a quick burst of air), sibilance (exaggerated "s" and "sh" sounds), noise floor gremlins, and, in the case of grab-happy vocalists, general handling noise. While comping "best take" tracks, you may find yourself having to choose the noisier tracks for their performance over less noisy, but also less well-performed, takes. So whaddya gonna do?

For sibilant sounds, a de-esser is your best bet, cleaning up with automation later if needed. You can also cut some of the highs, but whether that will work depends on the voice's character; and, if you've taken some lows out to add clarity to the vocal sound, you can really screw with the natural overall dynamics of the vocalist — so avoid this unless the vocalist really needs a general hi-cut EQ job.

The best way to deal with the remainder of these sonic annoyances, while maintaining total control of the sound, is to simply isolate the track and crossfade into and out of the problem area(s). Zoom into the track you wish to edit, find the zero crossing (where the waveform meets the center 0dB line), highlight said area, then crossfade accordingly so as to maintain a natural sounding edit. Shorter fades are key to this, particularly if you're editing in the midst of a bunch of vocals, taking into account the natural amount of space in between vocalizations. This can be time-consuming but by going for crossfades rather than cuts or "butt splices," you'll mask the noises without sacrificing the delivery, or clarity, of the vocalist's lyrics.

But what's most critical is that, prior to these edits, you apply whatever effects you need to get the sound you are going for — especially compression. You want the perspective of what the effects are going to do to your unsavory sounds *before* you attempt to rectify them, so settle on your sound and *then* do your fixes. For example, consider the overtly breathy vocal track. Compressors have a tendency to really bring out the "breathiness" of vocals, so what may be unnoticeable pre-compression may be glaring afterwards. Attempting to edit out noises prior to compressing is likely to just send you back to work after the fact, so apply your compression (and other critical effects) to the vocals first, then edit accordingly.

WORD!

If a vocal is great except for one frustrating mistake, like a single off-key word, you may not need to comp: Try to find the same word in a different part of the song, copy the good word, then paste it over the off-key word.

GETTING GOOD TAKES AFTER THE FACT

A common practice nowadays if a take is too noisy, or just generally bad, is to take the "word!" technique even further and copy an entire "good" take so you can use it to replace the "bad" take. This should be an absolute last-ditch effort unless the goal is to have each respective part sound "same-y." When pasting vocal tracks all over the place, it is imperative that you have the necessary perspective to keep yourself from making the song sound redundant. It's easy, when honing in on one particular problem area, to lose sight of the big picture; so make sure you are always listening a good ten seconds prior to the "mistake" before you make the decision to chop and shop. After all, sometimes the feeling is "in the flat."

However, if you do find yourself with only one good take of a chorus and you absolutely have to fill the holes, try massaging the copied tracks with some smartly placed delays — applying them to one word here and one word there — to give the illusion that you aren't merely sequencing, and sterilizing, vocals.

THE TAO OF PITCH CORRECTION

Pitch correcion is an extremely efficient and powerful tool, but also way easy to overdo. Much like merely cutting and pasting vocal lines, pitch correction is a beast that should be approached with a fair amount of trepidation, as it can really lame out your overall sound. But if you *must* use vocal pitch correction, make sure to record the corrected output onto a separate, new track so you can preserve your source, muting or deactivating the corrected version when it's not in use.

Assign the output of the corrected vocal track to an internal bus; then create a new track and assign its input to the same bus (*i.e.*, the output that you are monitoring). In real time, you can record onto the new track while manipulating the pitch-correcting processor on the original track. Then, if you mess up, you can simply "punch in" over the mistake — and you don't have to worry about losing audio quality, as the routing is all being done internally.

TRACK ALIGNMENT FOR A JUST WORLD

There are many times that you will need to align backup vocal tracks to the lead vocal track, or make the lead vocals' "stacked tracks" tighter and more uniform. First, so that you can work on a section-by-section basis, break ("split") the audio track into discrete pieces. To do this, select quiet spaces before and after each problematic segment and delete them — making sure you aren't trashing critical sounds that may not be strong enough to register visually in a waveform view. This will isolate a segment from the remainder of the track, allowing you to then pick the section up and move it to sync up with correctly occurring tracks.

But be careful with how perfectly you align your doubled/stacked tracks. To make the vocals sound thick, there must be some disparity of timing to avoid sacrificing the chorus effect. If you initially want to sync the tracks precisely for organizational purposes, you can always apply a little bit of delay afterwards to achieve the desired effect.

For backing vocals, first align the backing tracks with themselves, applying the same theory as stated above regarding the chorus effect. Performers are going to be a bit off with one another — therein lies the charm of the sound. Don't sync to the millisecond; just keep them close enough with one another so they don't sound cluttered.

For backing track alignment, the general rule is that backing vocals should start and end after and before the lead track respectively, so as to not draw too much attention away from the showcased performance. So start by grouping your backing vocals together, for the purpose of uniformity, and fade them in and out as a whole just before your lead vocal ends. This will help preserve the "backing track" quality in the midst of heavy track relocation and alignment.

GROUPING FOR THE MIX

Once you have the vocals cleaned up, tuned, de-noised, de-liced, and aligned properly, group the tracks together for global editing. If you are mixing inside of the DAW, set the level for each of the tracks that will be grouped together. Once the individual track volumes are set, group all of the necessary instruments together inside of your workstation — and watch the stereo bus meter for your stereo pairs instead of simply relying on your ears. Solo both the right and the left vocals tracks, and raise the volume of each until the levels are equal in the stereo bus meter. This will save your mix from sounding inconsistent and wavering, and the mastering engineer will love you for it. EQ

Amperage

Forget everything you knew about re-amping . . . then read this

by Chris Mara

Let me clue you in on a little something that the fat cats in Washington don't want the public to know: *Guitar amps are probably one of the coolest things ever made, period.* Few things invoke the true American spirit like a distorted power chord rumbling out of a super-sexy tweed cabinet. This statement shouldn't be taken lightly, as I consider myself to be quite the connoisseur of "the true American spirit." I dig all the really cool American stuff like motorcycles, fast cars, four wheel drive trucks, guns, guitars, belt buckles, and jet planes. (Actually, the *most American thing* is the common denominator of all of these items — *chrome*. But I digress. . . .) If it wasn't for guitar amps I would've hung up this music biz thing a long time ago and signed up for a "real job" with a 40-hour workweek, steady paycheck, and eventually, the obligatory receding hairline. This begs the all-American question: Why the @*^%$ should guitar players have all the fun?

Now maybe it seems that an article on re-amping should be geared towards the practice of re-amping guitars and bass in order to change the tone, cut down on bleed during tracking, etc. Although this is a useful and popular practice, not only was it covered pretty thoroughly in the Dec. '03 issue of *EQ*, I found it to be a fairly presumptuous subject for me to write about because . . . well, I never, ever do that — ever. Instead, I ultimately decided to take the advice of countless (other) great American writers and write about what I know: creating unique sounds and performances by amping and re-amping anything (or anyone) I can put in front of a microphone, or figure out how to get its signal down a 1/4" cable.

So let's take a walk down the dimly lit hallway that leads to the tiny, mold- and

Fig. 1: They look innocent, but this toolset of toys is the key to twisted re-amping.

mice-ridden room that houses all of my tips and tricks. (It's a scary place, I know, but it'll be OK.) I'll start by filling you in on the "tools of the trade" I use (Figure 1) and why I use them, then we'll take a look at some creative ways to utterly mistreat them in all sorts of applications.

TOOL TIME

I like to use **small amps.** Five, 10, and 15 watt amps are awesome to use in re-amping situations, for several reasons: They break up at lower volumes, they fit anywhere, they're easy to carry around, and they're cheap. You can walk into any pawn shop and easily find one for $30–$75 to get you started. (If there aren't pawn shops where you live, move to somewhere that has them. You are missing out on a huge part of life, trust me.)

Small amps have a lot of character, regardless of whether they're tube or solid state. I really dig the ones that have the built-in amp modeling and effects, especially the Vox Valvetronix series. These allow you to toggle through some settings and get vastly different sounds in a matter of seconds . . . and who cares whether it really sounds like a Mesa Triple Rectifier? I don't, and you shouldn't either. The best part of using small, cheap amps that you own is you don't have to walk up to the guitar player and say "dude, we got a killer distorted drum sound . . . by the way, someone blew up your Fender Bassman." To sum it all up: With small amps, no one gets hurt.

The whole re-amp thing kinda goes against nature, so you'll need various **audio adaptors, XLR turnarounds, and 1/4" barrels.** If you're scared of some amp noise . . . well, this isn't for you. One of the most valuable tools I own is a Radio Shack XLR to 1/4" adaptor with a built-in isolation transformer. Get both the male and female versions, they're $15 each and worth 10 times as much (part #274-016 and #274-017 respectively). These help maintain your ground potential without all the ground hum problems. If you're looking for some high-fidelity re-amp options, you may want to look into buying a Reamp box ($199 at www.reamp.com). Little Labs also makes a box called the Redeye Recording Tool ($250 at www.littlelabs.com), and other companies like Radial Engineering make re-amp-friendly devices. As I'm not typically looking for a high-fidelity re-amp situation, I haven't used these boxes (yet). I feel that working with some of the sonic and technical issues that accompany my "guerilla re-amping" style is part of the process — some days a good part, some days a bad part!

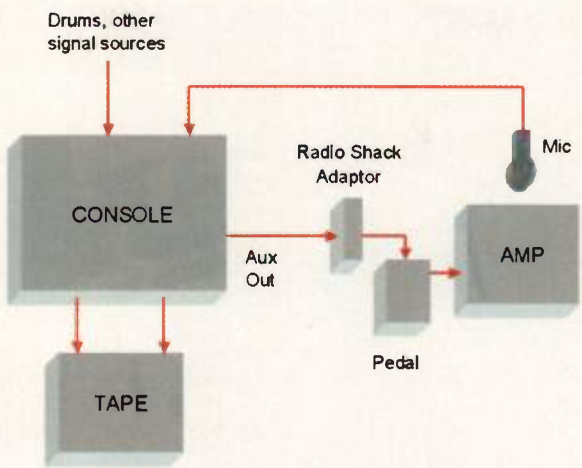


Fig. 2: Block diagram of a basic re-amping setup.

Another way to add one more dimension to the joys of re-amping is to **incorporate oddball microphones** by either miking the source instrument or the amp itself with an oddball mic. A well-placed lo-fi mic will get a surprisingly good blend on a drum kit — just place 'em low, you'll have enough cymbals once you amplify the signal. Keep a watchful eye at garage sales, estate sales, and army surplus stores for these little gems. A trip to your local electronics store may happen on your lucky day, too. Often times CB radios will break, but the handsets are just fine.

However, don't pay more than $20 for any of these oddball mics; they may not work, and if they do, they *will* sound like total crap. Most of these are high-impedance, so slap a 1/4" plug on the end and either use a DI before your mic pre, or slam them directly into a guitar amp to begin your fun-filled adventure! I have about a half-dozen of these chrome-laden beauties, and wouldn't part with them for the world. (I've already voiced my opinion on chrome. . . .)

Guitar pedals are your friends but put that distortion pedal down, you probably won't need it! Instead, go for some phaser pedals, delays, or flangers (and anything else you can think of) and put them in the signal chain right before the amp. Results may vary depending upon the pedal and the setup you're using; sometimes they make dealing with the gain stage tough . . . but the results can be worth it.

Get a **Mackie 1202**. Do it now. I personally think these should be given to everyone the first day of kindergarten. I can't imagine life without mine, and often my more intense tracking/re-amping sessions will require two or three of these bad boys. They're a great way to record things like acoustic guitars effortlessly without effects (via the main out) and also send the signal to one or two guitar amps via the aux sends. That way you can end up with one track "dry" and one track "amped." Same goes with vocals. If you're recording by yourself, just set it where you can get to it, and adjust levels as needed.

GETTING FULLY AMPED

Now that we have our needed tools, let's have some fun by exploring various ways to use amps during the recording process. It's important to note that contrary to popular opinion, I really, really like to do most of my amping during the actual performance phase of the session, not during mixdown. I feel it helps to tap into the creative energy of the session, and can actually shape the feeling of the entire song.

I'll typically set up two or more amps on any given tracking and overdub date. I make room by tucking them away in unused iso-booths, closets, hallways, and even bathrooms. Then I'll mic each amp with a close mic (usually a 57) and a room mic — something omni, or even a vintage ribbon. Usually I'm going for something that will be vastly different than the close mic.

The best way to be ultra-flexible with the amps is to feed them via an aux send or bus output directly from the console (Figure 2). If the console doesn't have bus output trimpots, strap a cheap compressor across the output — something with gain make-up. You'll really need just a

Amperage

Fig. 3: Adjusting level to amp with help of my "spotter." (Photo: Anthony Arrigo)

fraction of the +4 level to drive the amp! If you're working on a DAW, the same theory applies: Just create an aux send that feeds a spare interface output. Most studios have mic patching, and this is a great way to feed the amps in the iso booths, etc. Just patch "up stream" from the aux send or bus out to the mic panel, then use an adaptor from the panel to the amp. The mics you have on each amp can be bused together, and ultimately each amp will have its own track.

Now you're set up to send whatever instrument you want to whatever amp you want, and record it with a blend of close and room mics. It's a good idea to have an amp "spotter" hang out next to each amp for a minute or two to make sure you're not kicking the snot out of it. Once you have established the amp's "comfort zone," you can drive the amp harder or softer via your master aux control, or bus output pot. You'll have a virtual cornucopia of sounds at your fingertips. Cool huh?

INTO THE FORBIDDEN ZONE

Now you're set to go wherever your creativity will take you, so here are a few basic ideas to get you started.

- Try sending a kick and snare blend to one amp, then another amp to hear the difference. Now try getting a good drum mix through the amp, or try sending the kick and snare to separate amps. Then add some pedals!
- You can send an acoustic guitar to an amp for a beefier sound, or the vocals for that "small" sound desired for the intro verse, for example.
- If the artist is recording with a click track, try using a drum loop instead and send it to an amp as well — you'll be surprised to hear how cool it can make even the cheesiest drum loop sound. (It's important to check phase compatibility, especially if you're going to blend sounds together to one track.)
- Enlist an intern (or bored bass player, they'll do anything if they're bored enough) to tweak the controls of the amp or pedal(s) to help create tension or a release in certain parts of the song (Figure 3).
- Don't be afraid to set up some amps during mixing as well — all of these signal paths will work just fine, and when inspiration strikes you'll be ready. When the client asks for that "messed up drum sound" on the intro, you'll be a knob twist away from freaking him out.

LIFE BEYOND PLUG-INS

I think the fundamental reason I've come to use amping and re-amping so much is it really helps to combat the sameness of the "plug-in *du jour*" syndrome. I've had the not-so-fun experience of using a new plug-in to get a fresh sound, only to hear it on several other records just a few weeks later. My original idea suddenly appeared as if it was a major label ripoff (that really made my day). Using tools such as guitar amps and oddball mics ensures that you can create original sounds that can't be duplicated by anyone, probably including yourself!

Having the musicians tweak the amps and pedals while you're recording keeps the whole process performance-based; it really "speaks" to listeners, as these days they're getting a bit burned out on all the transparent post-production "tricks." It's also a great way to get new sounds out of existing gear that you have lying around. Run your standard keyboard patches through an amp, try running both sides through different amps to get a unique keyboard sound — something that you helped create for a specific song. Guitar amps on aux sends are also a fun way to create some interesting "out of the box" reverbs during mixing: Just stick a guitar amp in a hall, put an omni mic several feet back, and feed it something that needs just the right reverb to make it shine.

All these ideas should help stimulate the creative atmosphere when things get a bit stale. Running a rain stick through a Marshall half-stack may not always be the right answer, but it sure will be a fun distraction and get everyone working toward the same goal again. I'm half-tempted to purchase that '70s board game Twister and mark all the dots with the artists' names, instruments, pedals and amps; spin the spinner and get a "Jerry will play the floor tom through a phaser pedal into a Fender Champ" game going when the proverbial creative juices have stopped flowing. It sure beats digging through your plug-ins looking for something "new." EQ

Chris Mara is a a full time independent record producer/engineer based out of Nashville, TN. Check out his website at www.chrismara.com*, or his MySpace page at* www.myspace.com/chrismara.

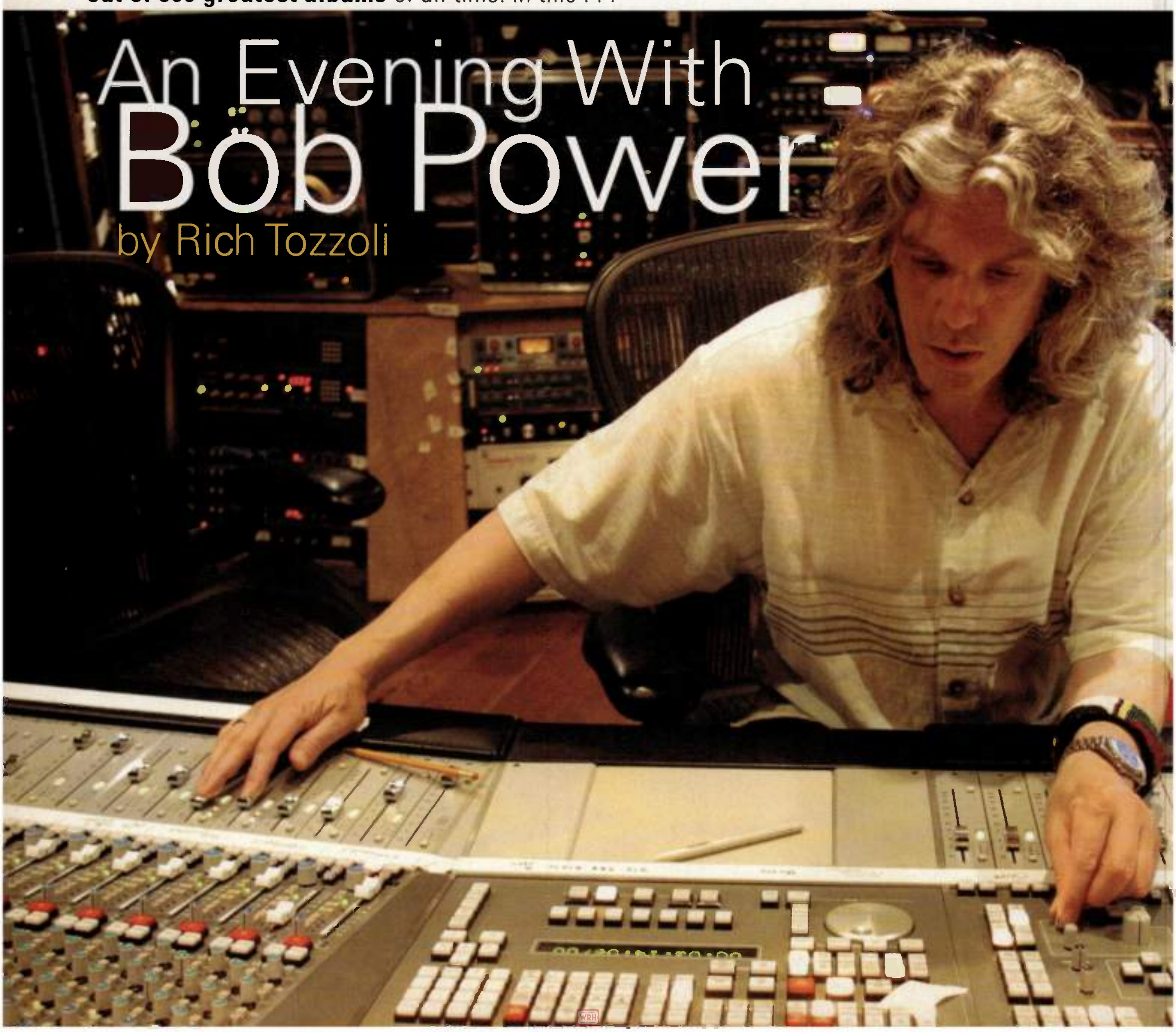
Grammy Award-winning producer **Bob Power**, whose long list of credits includes the likes of **Erykah Badu**, **The Roots**, and **Miles Davis**, has been behind over 50 charting records, and has received more than 20 gold or platinum records — securing his position as one of the leading producers/engineers in the worlds of hip-hop, R&B, jazz, and soul.

Taking Power's résumé into ample consideration, it's virtually impossible to pinpoint the catalyst moment in his rise to the top of the production world, as it's hard to imagine a time when Bob Power was anything less than . . . well . . . "Bob Power, man." But one of his more career-defining achievements was, inarguably, in producing a "then-marginally-popular" Queens, NY, collective known as **A Tribe Called Quest's** sophomore release ***The Low End Theory***. Simply put, *The Low End Theory* broke new ground for rap in the early '90s. Fusing the talents of **Q-Tip**, **DJ Ali Shaheed Muhammad**, and **Phife Dog**, the intelligent, sophisticated sound combined elements of jazz, laid-back samples and smooth lyrics with funky grooves. Critical acclaim followed, establishing the term alternative rap, and securing Power a seat in the pantheon of great hip-hop producers.

So we caught up with Power late one afternoon to pick his brain about what went into what has become widely regarded as *the* album that ushered in the "second wave of hip-hop." And he delivered — far exceeding our original hopes. So kick off your shoes, turn off the phone, and read on as Power offers a piquant look into the making of what ***Rolling Stone*** has dubbed album **#154 out of 500 greatest albums** of all time. In this . . .

An Evening With Bob Power

by Rich Tozzoli



EQ: *The Low End Theory* was mostly captured in a variety of New York City studios. Given that you didn't work in any one set location, on what consoles did you do your primary tracking/mixing?

Bob Power: The record was recorded primarily at both Battery and Calliope studios, which were the scene of many innovative things, in terms of recording, happening in hip-hop at that time; though we did a bit of mixing at Soundtrack and Greene Street studios, as well as a few overdubs. The console in Battery K2, which at that time was known as Studio B, was an old Neve 8068 that came over from England and later went to the Battery Studio in Nashville. Everything was tracked through the Neve, from vocals to samples to scratches. We did most of the mixing at Battery as well — although we did the lion's share of that on an SSL 4064 G. The one or two mixes we took to Soundtrack were done on an SSL 4000 E; and Greene Street had an early, rare, and quick-to-go fully automated Amek APC 1000 that we used as well. I had a very severe sinus infection during a lot of the mixing of that record — complete with some serious stuffed ears — that led to a lot of balancing problems, but looking back it's like "who knew?"

EQ: Was it all cut to tape?

BP: Yes, to two-inch 24-track on a Studer A-800, 827, and also a slightly modified 3M M79, using Ampex 456 tape — 30 ips, no noise reduction. Pro Tools was at a pretty primitive stage at the time, and certainly sounded rough, so it wasn't much of an option. Things didn't really start happening sonically with Pro Tools, for my money, until it went 24 bits.

EQ: There are a lot of breaks on the disc — did you have automation?

BP: Yes. Again, we were mostly mixing on the 4064 G, with some 4000 E, which had what I called the "Pacman interface": Green type on black — incredibly primitive! I was still in the midst of a big learning phase with SSL automation at that time, and I learned a lot from my assistants. It's funny though, we actually edited a lot of the interlude material in Pro Tools, though the full songs were all mastered from tape. Prior to that, around the time of the first A Tribe Called Quest album, we were mixing to 1/4" — either 15 or 30 ips, depending on the budget. No joke.

EQ: Did you actually fly all the samples in manually?

BP: Most of them were sequenced, and the rest either DJ Ali or Q-Tip would cut in directly from the turntables. A lot of the material was sampled from turntables during the sessions, into an Akai S900. My, how far we've come. At that time, I was way into the Atari [computer] and [C-Lab] Notator, which later became Logic. Tip would often come in with a beat sequenced and sampled into an E-mu SP-12 and Ali, bless him, had already started to get into Macs at that point, and would bring in tracks sequenced with Vision, as well as samples loaded on the S900. In many cases I would dump those sequences into Notator, go into another room for 45 minutes, and just push the stuff around 'til it felt right. Remember, this was the beginning of the '90s and reliable sync was still dodgy. Many people used the Roland SBX-80, as Mac interfaces were not terrifically consistent. But one of the nice things about the Atari/Notator combination was that, if you had a Unitor [their giant dongle that read timecode], sequencing and sync happened in the same box, which was a big plus.

EQ: "Check The Rhime" contains music from the Average White Band's "Love Your Life," as well as some really grooving keys. Were the tracks all sampled, or were there performances by Tribe included as well?

BP: As far as I can recall, it was mostly all samples and loops. I *do* remember, however, that was around the time where we started supplanting loops with "helpers," playing the same thing to give it a little more dimension, so sometimes we would play the existing bass line an octave down on those wonderful sub-bass sounds on a Roland Juno-106. You didn't hear it as an octave, but it gave the impression of massive bottom end on loops that really didn't have it. If you tried to EQ into that same thing, you mostly ended up with mud. Some of the Rhodes element we doubled as well, but it was largely samples and loops.

EQ: There also seems to be many layers of drums on that tune that cycle in and out. How did that come about?

BP: That happened on a lot of Tip and Ali's beats at the time, though the specifics are a bit hazy. A really large part of that was the fact that even if the sequenced drumbeat stayed the same throughout the tune, the addition of the different loops *over* the beat made a sort of combinative beat, so the different sections of the song sounded as if the beat elements had changed.

EQ: The snare on "Jazz (We've Got)" has serious pop. Did you layer the samples and if so, how do did you then sync them back?

BP: The recording of that album occurred right around the time I really got into layering samples. I'd never heard of the concept, but from time to time I'd think, "Gee, this snare could use some more attack." So I started doing things like layering in the attack portion of a cowbell, for instance, with the snare sample to give it some "hurt." Of course, the sample's tuning also had a lot to do with the overall effect, and I would do this all in the S900. We used a lot of different things — sometimes drumsticks clacked together — and one time I actually had Tip hit a Sennheiser 421 against a tabletop, and we used that behind the snare.

I learned that if you leave a little "airspace" at the beginning of the sample, you can quite precisely slip the timing of the different samples by adjusting the start point, all while the sounds are playing from the sequencer. Also, if you assigned the different samples their own MIDI note, or channel, you could slip them around in Notator. If we recorded another element to the sound after the first was on tape, again, I would move it around on Notator while the track was playing until it felt right. A great trick that I learned from Chris Julian Irwin, who owned Calliope and is a brilliant producer/engineer, was this: If you were using the E-mu SP-12, which a lot of people were, the trick was to offset the SMPTE start time of the song slightly early in the SP-12, and then feed the timecode from tape to the SP-12 through a digital delay — usually a Lexicon PCM 42. Then, as the song was playing, you'd change the value of the delay time, thus "slipping" the feel of the overdub coming from the SP-12 in real time.

EQ: Do you remember what you used to record guest bassist Ron Carter on "Verses From The Abstract"?

BP: Yes, very well. It was a funny evening when we recorded him; Tip, Ali, and I were all slightly terrified. At times, Ron can seem like a very prickly fellow so, because of his stature, I made sure to converse with him in the booth about both how I was planning to record him and see what he was comfortable with. I ended used a Neumann Tube U47, and took his pickup direct, as well. I'm pretty sure I used some stand-alone Neve mic pres that were in an outboard rack and then balanced them on faders on the 4064 G console, bussing the combined signals to one tape track. I had planned on using a Neve 2254e compressor/limiter on him — a piece I was very familiar with. But when I asked him if he was comfortable with that, I got the quickest and most resolute "no" I've ever heard. That said, I learned a valuable lesson that night from Ron about working with "heavy" players in general. If they are serious professionals, and if you speak music with them, they respond in a very "pro" way — quickly . . . and very musically.

EQ: What was the overall music climate at that time in the hip-hop world?

BP: I'm probably not the best person to answer that, as I was much older and from a very different background than the artists I was working with. That said, it was what I consider the meat of what I call the "second wave" of hip-hop. Beats made exclusively from 808s, or other drum machines, with their straight eighth note "tat-tat-tat-tat" hi-hats were quickly becoming passé, and there were a whole slew of artists coming up who were influenced by many different musical genres, and the atmosphere was one of very open and unfettered creativity. There didn't seem to be the overwhelming need to be "down," just engaging. And Calliope Studio was ground zero for much of that. It's mind-boggling to think of the people coming through there at that time; A Tribe Called Quest, De La Soul, Black Sheep, Stetsasonic. . . .

EQ: What was the "must have" studio gear at that time for recording this "second wave of hip-hop"?

BP: Samplers coming into their own were an absolutely huge fulcrum in the progression of the "second wave." Akai S900, E-mu SP-12, Technics 1200 turntables . . . you have to remember that this was before hip-hop became the giant money machine that it is today — thus much of the recording took place in whatever way was the least expensive, in both terms of gear and facilities. And, in a way, that was good: One was forced to get creative, rather than just plugging in a well-known, high-end piece of gear. At Calliope, mostly due to Chris's vision, we had things fairly early on, including Lexicon PCM-42 digital delays and, slightly later, the PCM-60 digital reverb. Mic-wise, we were using the AKG C414s and C451s for tons of different applications. Personally, I didn't become familiar with some of the "big-studio" staples like the Lexicon 224, EMT plates, Urei 1176, Neumann U87, and so on until our budgets allowed us to go to the "big" places. But on top of what little we had to work with, we pulled a lot of things out of my personal collection: AKG C24, C12a, Shoeps m221, Neve 2254e — not pieces normally associated with hip-hop, but I loved the way they sounded when put into that context.

EQ: Looking back, what do you feel that album left the hip-hop world in terms of recording?

BP: *The Low End Theory* was the first album, as far as I know, that had samples put together with such detail as to form fairly complex and sophisticated sounding tracks. The feel was great — it sounded like a bunch of very good players conversing — but because it was so skillfully reconstructed, and the beats were so deep, it actually went places those "real" players never could. Also, arguably, it was the first time, or at least one of the first notable examples, where so much attention was paid to sonics on a hip-hop record, and the world took notice. This record is a testament to the startling creativity of Q-Tip, Ali Shaheed, and Phife Dawg.

But as far as the engineering/mixing goes, why did it come out sounding the way that it did? Part of that was due to the felicity of having a Neve console at our disposal. It was partially due to that album being cut during a very aggressive time of discovery for me as an engineer. Part of it was due to Tip standing over my shoulder at the mixes saying, "more snare, more kick." But mostly it was due to that intangible confluence of people, the times. It was a record that really took on a life of its own.

Overall though, it comes down to the same elements that make a good record in any genre: compelling performances of great songs. Although it may be truer in theory than practice, if you have those elements, you can make a great record on an answering machine. EQ

MANLEY LABS TNT

Exclusive preview of one very different box

by Glenn Bucci

When Manley introduced the Slam! (a stereo limiter/two channel tube pre combo) a few years back, many raved about how amazing it sounded: It had tons of gain, yet offered a very smooth sound. Unfortunately, the price tag kept the piece out of many a salivating studio hound's hands, and many of us were left crying out to the gods, "why can't Manley just come with a single Slam! pre minus the compressor so we can actually afford one?" And while those poor chaps sat lusting out of their price range, those of us that had been so blessed also wanted some extra features out of the Slam! — namely, variable impedance control and a composite channel, British console-inspired pre for added flexibility.

So Hutch, Manley's chief designer, hid away in his laboratory, slaving over what would soon become Manley's newest outboard offering. Although *EQ* normally doesn't review prototypes, when something is a product is this close to production and of such interest to our readership, we couldn't resist. This two-channel, half-tube/half-solid state TNT mic pre marries the celebrated Slam! sound with the classic discrete character of the "No Tubes" design — at under half the price of the aforementioned product.

OVERVIEW

The TNT is a heavy-duty 1U rackmount, split into two very different channels — with the left side being the Slam! tube pre and the right being a brand new solid-state pre. A toggle switch does power on/off, while input/output overload levels are clearly represented by two sets of LEDS (green for normal, red for overload). The Tube Channel includes a phantom power toggle switch that enables a full +48V, phase button that reverses the polarity 180°, low cut filter switch (flat or 80Hz), three-position toggle that chooses the input impedance (600 Ohm, 2400k, and 10k), unbalanced instrument jack that disables the XLR when in use, gain knob that peaks at 70dB (perfect for ribbons or other mics that need a lot of gain), and a trim control that runs from 10 to +10.

The solid-state channel (Cool Channel) sports a similar layout. However, the filter options come in at 60Hz and 120Hz, as well as flat (options the tube channel unfortunately lacks). The impedance control on the solid-state side is a five-way rotary switch with settings from 300 Ohms to 2 Megohms. The Cool Channel also offers two exciting features new to Manley: an Iron knob that goes from –1 to +3 and a Color toggle switch that allows you to choose from Clean (normal state — very low noise/high accuracy), "70s," or "60s" (both of which are much more colorful). Cool channel, indeed.

Flipping the unit around, there are XLR ins and outs for each channel, as well as a 1/4" unbalanced output jack. On the Cool Channel, the 1/4" can be used with +4 or –10, while the tube pre has a +4 unbalanced output. And if you rip the unit open (we love to to do that!), you can see that the sealed relays were placed on the circuit board to allow for the shortest possible signal traces.

According to Manley Labs, the final version of the TNT will offer additional DIP switches for added color — taking the total harmonic distortion of 0.03% to about 0.3%. There will also be a secondary switch that will produce a similar amount of distortion, biasing an output transformer to give more low frequency distortion (or "warmth"). Furthermore, the ability to add a clipper after the first tube stage, so the user can overdrive the pre by turning up the input, will be offered and on the Cool Channel, the Iron knob will be updated for added character as well.

APPLYING THE TNT

Each channel/pre of the TNT has its very own distinct personality, with the Tube Channel sounding very warm and the Cool Channel being very clean, detailed, and musical. Thanks to the range of control offered by the TNT you, however, are not merely stuck with one sound or the other — a middle ground is readily available by just playing with the settings a bit. And this is good, as there is no such thing as a "silver bullet" pre: Different sources demand different treatments. And the TNT's ability to shape the sound it's receiving, from barely to dramatically, is exactly why it excels as being a great "go-to" pre.

The Cool Channel offers a very clean, almost "three-dimensional" type of sound that's both smooth and subtle. I found that, when working with a solid-state mic, the Cool Channel was my best bet — but that was in the context of working with a vocalist who had a great, controlled voice. It benefited from the "trueness" of the Cool Channel, as it needed a pre that was "open" and detailed with no added coloration to the signal, except for just a hint of smoothness. In these situations, I turned the impedance knob to its highest setting, which really brought out the high end.

But, when it was time to use a tube mic, I kept shying away from the Cool pre and exploring a bit of the added warmth present in the Tube Channel, as the effect is not overwhelming (unless you want it to be) — you can get a very clear, sharp sound out of the Tube Channel. I especially liked the Tube Channel, which reminded me a bit of the sound I get from the Langevin DVC; when paired with a Røde K2 or an Audio-Technica 4060, it allows me to preserve the character of the mic if I want, or alter it with strong color.

When miking an acoustic guitar, the Cool Channel gave a punchy, crisp top end that was full and up front — which made it a better choice for more balanced-sounding instruments (Martin guitars, for example). Conversely, when the artist was using a brighter-sounding acoustic guitar (a Taylor in this case), the Tube Channel seemed to be the better option.

I ended up using the Cool Channel's "60s" and "70s" option for that oft-sought vintage console sound, especially on the acoustic guitar, but also on some of the more naturally silky vocals; it affected the depth positively, opening up the sound just enough and clipping in a good way, adding a very gentle distortion. This feature alone made me feel as if I had doubled the amount of available sounds of the box, giving me a lot of breathing room. Also, after I had dialed in my settings, I found that utilizing the Iron knob is another feature that changes the tonality of the signal without reducing the depth and openness. If anything it acts not unlike a high frequency expander when turned counterclockwise — especially if you use the 1/4" output, as it adjusts the output transformer. Essentially it's sort of like getting a variety of transformer colors, and even what might be the opposite sound of a transformer. This knob allowed a change to the sound which was much more subtle compared to the "60s/70s" switch.

CONCLUSION

All in all, the TNT is one of the more flexible boxes out there. With a wide variety of control options, you can get numerous pre sounds out of the two channels and, most importantly, they all sound damn good. If you are looking for a tube pre to treat signals coming from brighter instruments, from sharp vocals to cutting guitar to brass or cymbals, the Tube Channel is a comfortable, warm-sounding pre. And if you need a very true sound for already smooth vocals, full drums, bass, or overdriven guitars, the Cool Channel just can't be beat. And the best part? You get all of these options in one unit, which is why the TNT comes with, and will continue to come with, high recommendations. EQ

Product Type: Two-channel/two mic preamp combination rackmount box.
Target Market: Studios wanting the Manley Slam! sound without the (overtly) painful price point, while also looking for solid-state pre capabilities.
Strengths: Solid construction, flexible performance, high-end sounds with a relatively affordable price tag.
Limitations: Tube Channel filter options not as expansive as Cool Channel filter options.
Price: $3,000 list
Contact: www.manleylabs.com

Reviews

CHANDLER LIMITED GERMANIUM PREAMP/DI

Getting into the thick of it

by Jay Matheson

Chandler's new Germanium is truly a completely different mic pre. Though boasting an effective front panel (1/4" unbalanced DI jack, rotary Germanium Drive input gain knob, Pad, Thick switch for low frequency rise, Feedback rotary knob, and a VU range switch with an accompanying meter), it's the guts that are really impressive.

Using classic transistors in all class A, transformer balanced circuits, the Germanium has a classic sound, but with a few forward-thinking features. Altering the Germanium Drive and Feedback controls produces results that drastically change the preamp's character — with Feedback in particular affecting aggression and "size." But the Thick option is perhaps the most addictive; it significantly smoothes out the lows, enriches the signal, and improves the sonic balance.

APPLYING THE GERMANIUM

I first used the Germanium on a hip-hop vocalist with a bright voice, and who usually needs a little help to get a bigger, meatier sound. His default signal path is an AKG 414 into a Daking 52270H followed by an Empirical Labs Distressor and, finally, a Digi 192 interface. As an experiment, I swapped out the 52270H for the Germanium; sure enough, engaging the Thick switch made the lows bigger and "creamier," while turning up the Feedback above 5 made the vocals more thick and aggressive. Overall, the vocal took up more space in the mix, making the vocalist sound "tougher." It even sounds like the track was cut to 2" tape, not my HD2 Accel rig.

The Germanium adds so much color that a vocalist I later cut with a "darker" voice needed some extra highs during the mix to keep his voice afloat. But, when I needed a cleaner sound later on, disengaging the Thick switch and being a bit more conservative with the Feedback control gave a "truer" sound — so no worries.

Next up was acoustic guitar ('87 Martin Shenandoah), miked with a Neumann U87. I tested the Germanium by printing tracks using several different settings, then comparing the results. On the first track, I set the input at 10, Feedback at 1.5, and kept the Pad and the Thick switches off. This produced a well-balanced sound, save the lows were not rich enough and the attack was a bit stiff. Turning on Thick added a significant low/low-mid boost — but with no noticeable low end peaks. The sound's overall character changed so dramatically that you could have sworn the mic, or even the guitar, had been swapped out.

I preferred the guitar with Thick on, so I engaged the Pad to test the unit at higher Feedback levels. Setting the input to 8 and Feedback to 9 produced a more aggressive sound, with the overall balance and mid-range detail besting the previous settings. Furthermore, the pick attack sounded smoother; generally, the guitar sounded as if it had been lifted from a great '70s LP.

Finally, I checked out a '71 Fender P-Bass sent direct through the Germanium's front panel. Usually I would go into a Chameleon 7602, which has great lows and mids, but as I needed that extra low thud I used the Germanium's Thick switch. The mids weren't as pronounced, but the extra low boost made the Germanium the correct choice; checking with an analyzer, I noticed the low end response was flat down to 20Hz. The Germanium produced the biggest low end that I've ever heard in a direct bass, while still sounding balanced and solid.

CONCLUSIONS

The Germanium dispels the myth that solid-state pres can't offer the same kind of thickness as tube preamps. I could get some serious girth out of the signals, and manipulate them in ways different from merely increasing the gain. Some of the tracks had a sound that hearkened back to the Stones or Pink Floyd — no tape emulation needed. This alone made the Germanium a permanent addition to my studio.

But, as much as I love this box, it's not perfect. A helicopter-like oscillation appeared in some instances when the feedback was set near maximum, and the Germanium seemed to increase background room noise when used on quiet sources. Neither of these issues seriously hindered me, however, and I would still recommend the Germanium to nearly anyone. EQ

Product Type: 1U rackmount preamp/DI Box
Target Market: Mid-high end studios wanting a pre/DI box that adds significant, yet smooth, low end.
Strengths: Thick option lives up to its name. Feedback pumps life into sounds. Front panel design well-suited for DI applications.
Limitations: No low cut filter. Brings up noise with quiet sources.
Price: $1,150 list
Contact: www.chandlerlimited.com

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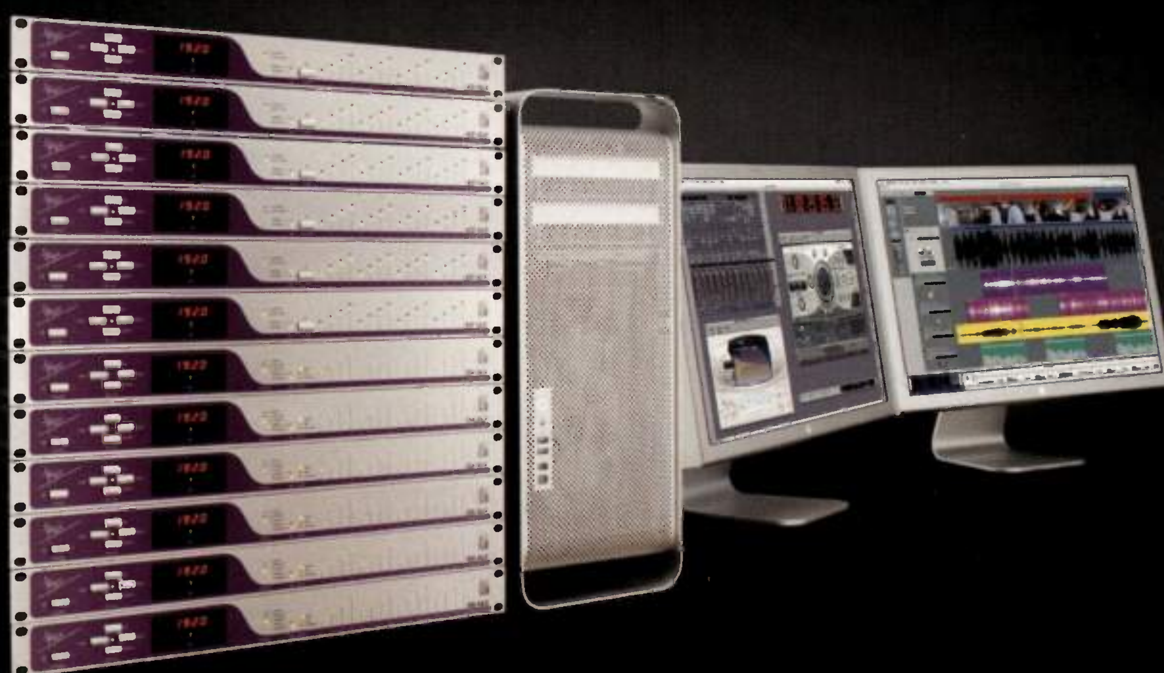

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PETERSON STROBOSOFT TUNER

The strobe wheel tuner goes virtual

by Craig Anderton

Reviewing a tuner seems about as appealing as watching C-SPAN: Vaguely educational — and soporific. But the Peterson Strobosoft had enough surprises and "hmmm, that's cool!" features to keep me more than interested.

BASICS

Peterson's hardware strobe tuners are the top-of-the-line in tuner-land. They've now stuffed that technology into StroboSoft, available in Deluxe (reviewed here) and Standard versions. These are 26MB downloads; however, the "StroboSoft Suite" boxed version (with Deluxe software, a 1/4" to 1/8" adapter for cheapo computer soundcards, a DVD on tuning and intonation, and printed manual) is available from dealers.

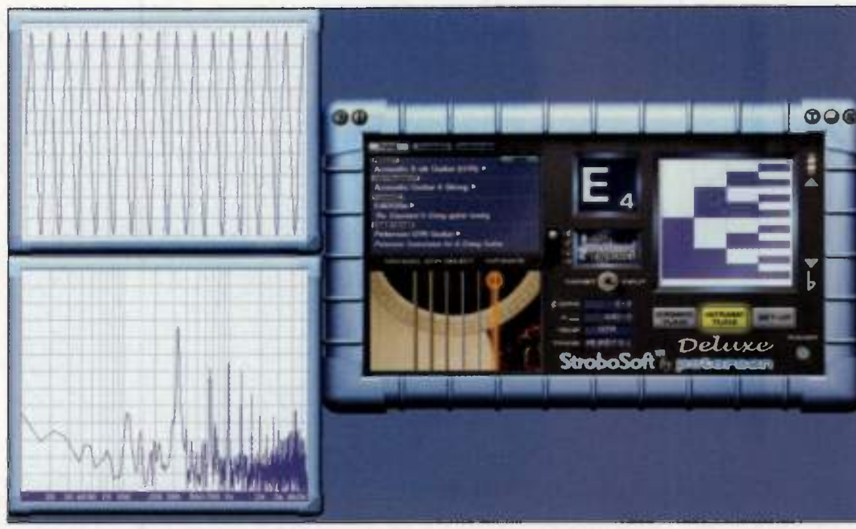
The software is cross-platform (Windows XP/2K, Mac OS X 10.4 including Mac Universal) and stand-alone. It supports Windows MME/ASIO drivers (although it seems optimized for MME), Mac CoreAudio, and talks to your soundcard input of choice. There's an issue with Pro Tools HD, as it doesn't allow running other apps using the same audio device; you have to use one or the other, or a separate soundcard. But that's a problem at Digi's end, not Peterson's.

The Standard version offers chromatic tuning, while the Deluxe version adds an outstanding instrument tuning mode. There are multiple presets for "sweetening" the tuning in different ways (including a preset for Buzz Feiten tuning guitars), as well as tunings for 7-string guitars, 5-string bass, dobro, violin, open tunings, alternate tunings, cello, and more — and you can create custom presets. Accuracy is exceptional, as it's within 0.1 cents.

APPLYING STROBOSOFT

I plugged my PRS guitar into the E-mu 1820m interface mic input, and got started. As it gave a clean signal I didn't need the noise filter. But if you're tuning an instrument through a mic, the noise filter cancels out consistent noise like hum.

Chromatic Tune mode is most like the traditional Peterson strobe tuner. It recognizes the input note automatically, although you can also select a note manually. You can choose different temperaments (including four types of meantone and six well-tempered tunings), set transposition for different keys, and of course, alter the base reference from 440Hz.



The oscilloscope screen is in the upper left, the spectrum analyzer in the lower left, and the Instrument mode window (with sweetened tuning) toward the right.

The Instrument tune screen offers the option to use various presets. I was curious about the "sweetened" tuning, which supposedly gives "sweeter" 4th and 5th intervals. The difference is subtle, but I indeed noticed a different, and indeed more pleasing, tonality on barre chords. There's also a sweetened tuning for bass that's optimized for playing along with stretch-tuned keyboards. These are powerful, creative features unique to StroboSoft. Furthermore, an intonation option recognizes both the fundamental and octave, which makes adjusting intonation incredibly simple.

EXTRAS

The Deluxe version has an oscilloscope display (fun eye candy), and a spectrum analyzer which shows what your input is seeing — this exposes whether, for example, there's hum or other garbage in your signal that could throw off the tuning (remember, don't tune your guitar too close to the computer itself, as noise can be induced into the pickups).

CONCLUSIONS

If you need to simply touch up your guitar's tuning, the tuners included with most software packages, or hardware units from companies like Korg and Boss, will get you "in the ballpark." But if you want to get into serious tuning, investigate sweetened tunings, and have presets available for all kinds of instruments and playing options, there's nothing with the StroboSoft's extreme degree of accuracy (when you're dealing with sweetened tuning offsets of a cent or two, 0.1 cent accuracy is essential), convenience, and depth. And if you're a guitarist, it's definitely worth the extra bucks for the Deluxe model. EQ

Product Type: Virtual strobe tuner.
Target Market: Musicians and studios requiring extremely accurate, computer-controlled tuning.
Strengths: Presets for different tunings. Useful intonation option for guitars. Standard chromatic or instrument tuning modes. Supports Buzz Feiten tuning system. Noise filter when tuning instruments using a mic.
Limitations: Doesn't work as a plug-in, only stand-alone.
Price: $79.99 Deluxe, $49.99 Standard, $39.99 Standard to Deluxe upgrade; StroboSoft Suite, $149.99.
Contact: www.strobosoft.com

MACKIE ONYX SATELLITE

Eat at home, or get it "to go"

by Craig Anderton

For some, the Onyx Satellite will be a solution in search of a problem. For others, it will be the perfect solution to their problems. But look closer: While disguised as a FireWire audio interface, it has something extra that lifts it out of the "me too" product category.

BASICS

The Onyx Satellite, a cross-platform FireWire interface, is a "desktop" interface with multiple patch points — but incorporates a bus-powered"pod" that can lift out and go with you for mobile recording. The pod contains Mackie's Onyx mic preamps that accommodate mic and instrument inputs, stereo "control room" outs, gain controls, primitive (four LED) metering, and two headphone outs.

The "dock" section offers more, but is at heart a 2-in/6-out interface. The two ins, which have analog insert jacks, also each have switches to choose among XLR mic, high-Z unbalanced instrument in, or two balanced TRS line ins. There are two stereo control room outs, as well as four line outs (3-6, which correspond to like-numbered ASIO outs when using a host sequencer). An output switch chooses between "1-2" mode (the Control Room output signal goes to your choice of the A or B outs, with level determined by the Control Room Level control) or "1-6," where the Level control acts as a master volume control for the four mono line outs and two stereo A/B outs. Assuming your host does surround and you can assign the desired channels to the desired ASIO buses, we're talking genuine surround mixing.

There's also a talkback section with built-in mic (and heavy compression, so it will pretty much pick up whatever you have to say), and a source switch to choose between monitoring the inputs and monitoring the DAW out.

APPLYING THE ONYX SATELLITE

As with any FireWire audio device, make sure your system is tweaked. With Windows, install the hotfix (http://support.microsoft.com/kb/885222/en-us) for FireWire devices running on Windows XP

SP2. For Mac fans who encounter the dreaded "FireWire whine," go to http://developer.apple.com/tools/download and download the CHUD tools. This adds a processor option to the System Preferences menu under "Hardware." Open it up, uncheck "Allow Nap," and banish your FireWire woes.

The Satellite's key feature is that it can be patched into your system when you're working at home, but you can take the pod interface for mobile recording and not have to re-patch when you return. The entire unit, pod included, is sturdy and feels like it could survive the road — as long as you didn't apply a lot of downward or sideways pressure to the knobs. (Mackie will be offering an optional carrying case for the pod.)

Despite the Satellite's convenience factor, there are limitations. The mic pres only produce +38V of phantom power; some mics may balk at that, although for most this won't be a problem. Also, there's no MIDI I/O. A more curious issue involves source monitoring: A switch can choose input or DAW, but no combination of the two. Huh? Fortunately, Mackie says a firmware update will be available by the time you read this that allows mixing in the DAW signal with the switch set to input. So, you can either listen to the DAW with "zero-latency" monitoring from the input source, or switch over to DAW and monitor your input through the host program.

This isn't as flexible as an associated mixer applet; as things stand, there's a control panel for setting the sample rate (up to 96kHz) and sample buffers (128 to 2,048 samples) — that's it. A software mixer would make the Satellite more versatile. (However, Mackie expects to release a high performance driver that can drop latency down to 64 samples or less.)

In terms of sound, though, the Satellite scores big. The instrument ins are clean and crisp, as are the mic pres.

CONCLUSIONS

If you don't need the dock/pod concept, there are better values for both mobile and desktop interfaces. But if you fit Mackie's target profile, nothing else is equivalent, and you'd have to buy two interfaces to get the Satellite's functionality. Bottom line: This will be a perfect fit for some, but not for others . . . in either case, though, it provides a welcome alternative. EQ

Product type: Mobile *and* desktop 2-in/6-out FireWire audio interface.
Target market: Smaller studios and recording musicians that do both mobile and desktop recording.
Strengths: Excellent mic pres. Innovative design. Control room outs. Accommodates surround mixing. Drivers are solid, even at low latency.
Limitations: No MIDI I/O. No applet to blend source and DAW out. Mic pres only put out +38V.
Price: $519.99 list.
Contact: www.mackie.com

BEYERDYNAMIC MC 840

Can the MC 840 put the "pro" in "prosumer"?

by Kris Force

The Beyerdynamic MC 840 — an incredibly compact large diaphragm condenser mic — weighs in with an impressive feature set, sophisticated design, and a price point far below mics of similar quality in the "professional project" realm. On paper, it's one of the most easily attainable (and versatile) high-end mic locker acquisitions unveiled in the past year; let's see how it holds up.

OVERVIEW

Sporting a gold vaporized double diaphragm condenser design, the MC 840 — complete with a sharp and sleek black finish — comes equipped with five switchable polar patterns (omnidirectional, cardioid, wide cardioid, hypercardioid, and figure-eight), two switchable two-stage rolloff filters (6dB/octave at 80Hz and 160Hz), and two stage pre-attenuation settings (–10dB and –20dB). At 0dB attenuation the MC 840 accommodates 127dB; with –10dB this increases to 137dB, and to 147dB at –20dB. Signal-to-noise ratio is low (1 PA, approximately, to 70dB).

Operating on a pressure gradient principle, the MC 840 is a transducer-type condenser whose frequency response is stated as 30Hz to 20kHz. However, the frequency response curve is not as transparent as implied: There is an audible boost from 150 to 500Hz that's most emphasized in the figure eight directional pattern, and even more so with closer proximities. The omnidirectional pattern is perhaps the best setting in terms of transparency, with only a –1dB dip from 3–4kHz.

APPLYING THE MC 840

I had numerous voice tracks for various projects waiting to be recorded, so the MC 840 arrived at a perfect time for testing; for a comparative standard, I tested it alongside the venerable Neumann TLM 170. With absolute flat EQ settings and matching impedance on the Mbox2's internal mic pres, I tested the same sound source, simultaneously, with both mics; the diaphragms were as close to sharing the same reference point as possible, and both were set to matching rolloff and attenuation.

My impressions of the MC 840 were:

- The MC 840 endured pops and physical handling more than adequately.
- During an off-axis coloration test, the diaphragm capsule inside the head grill had a reasonably smooth acoustic response for all directional characteristics over a wide acceptance angle.
- With reflections arriving at different angles, the diffusion field response is relatively even.
- Tonal coloration differences were most audible at an ambient proximity of 3–4 feet with the more directional polar patterns.
- The best ambient sound was achieved with the omnidirectional setting.
- In my personal project room, the 160Hz rolloff filter came in very handy for eliminating extraneous room tones.

Compared to the TLM 170, the MC 840 tracked consistently with 2–3dB more gain (with an accompanying drive presence) and an evident, yet predictable, boost from 150–500Hz. This very well may be attributed to the "new and improved low-noise preampl fier and impedance transformer without transducer" design many may notice as being carried over from the MC 840's predecessor, the MC 740. Regardless, there were no negative surprises; overall, the mic offers tremendous configuration versatility, which makes it easy to offset deficiencies relative to your recording source and environment.

CONCLUSION

While it may appear unfair to compare the MC 840 with a "holy grail" item like the TLM 170, the TLM 170 had the closest matching feature set and, while the MC 840 is significantly cheaper, it definitely held its own. The MC 840 does exactly what it has been hyped as doing, which is a bit of an anomaly in this world of vehement marketing departments claiming that every product they unleash is the next "must-have" acquisition for your studio.

All in all, the MC 840 is a fabulous pro-level mic at a prosumer price point that's versatile enough to be at home in many different scenarios, from the recording studio to the film set. Having worked extensively in several studio and production environments, the Beyerdynamic brand has always been perceived by my colleagues and me as professional-grade, utilitarian, sturdy, and trustworthy . . . and the MC 840 reinforced this belief. EQ

Product type: Studio condenser microphone.
Target market: Recording enthusiasts looking to add a pro-level large diaphragm condenser to their mic cabinet without breaking their wallet.
Strengths: Wide array of incredibly helpful features with a friendly price tag.
Limitations: None.
Price: $1,599 list
Contact: www.beyerdynamic-usa.com

M-AUDIO
ESSENCE OF CHINA

Contact: M-Audio, www.m-audio.com
Format: CD-ROM with 264 Acidized WAV loops (551MB total); 24-bit/44.1kHz
Price: $49.95

Essence of China: Sounds like a perfume, eh? And in a way, it is; this CD-ROM provides the atmosphere for adding a whiff of traditional Chinese music (not the club music they play in Shanghai!) to your productions.

There are only two folders, Chinese Flutes and Chinese Zithers (unfortunately, there are no Erhu samples, which would have contributed to a more comprehensive collection). But within what's included, there's a fair amount of variety. If there was a machine that could select one loop from each folder at random intervals and crossfade them together, you'd have a decent soundtrack for an evening out at the local Chinese restaurant.

Okay . . . but what if no one has called you to do the soundtrack for the sequel to *Crouching Tiger, Hidden Dragon*? As it turns out, the loops are acidized (and reasonably well, though you'll need to tweak them for serious stretching), so they're suitable for adding a dash of exotica into other projects. Bear in mind that these are indeed traditionally Chinese, so they automatically add that Chinese vibe to whatever you're doing; however, there is a definite beauty and grace to these sounds, so you (and your listeners) might find this a trip well worth taking.

In any event, as hip-hop and world productions increasingly add a taste of the exotic (and "new age" music always has), these samples will slide right in. And unlike the myriad sample CDs with tabla sounds and Arabic percussion, this one provides something unique and classy; check out the audio example at www.eqmag.com.
—*Craig Anderton*

POWERFX
UK GRIME

Contact: PowerFX, www.powerfx.com
Format: "Downloadable CD" with 435 Acidized WAV loops (532MB total), also available in ReFill and Apple Loops formats; 16-bit/44.1kHz
Price: $49

There are two certainties in life: New dance music strains will mutate and evolve, and sample providers will conjure up suitable samples to support them. Enter Grime, which is sort of like Electro meets Glitch, but raises the bar for creative soundcrafting.

The acidization for these files is quite good, with only a few missed transients — definitely above average. And the sounds are a huge amount of fun. Some of the standouts are FX Beats, which are actually great additions to any rhythmically-oriented track, and the Toprhythmz & Perkuzion folder, which doesn't have a lot of files — but the ones that are there are fine complements to the other loops. Grime World includes some world-oriented samples, and there are also folders for Bass, Synth, Strings, useful one-shots, and several other loops.

Now, I'll admit I'm a sucker for hardcore underground stuff that sounds best in a club where last night's cleaning crew couldn't really get rid of *all* the beer smell, but even so, this is a standout collection of samples. You can throw just about anything together, and the samples themselves are sufficiently inventive that you really can't go wrong (following the rule I learned from DJ Dave Holland: "No matter how bad you are that night, if your loops are good you can only fall so far").

Want proof? Go to www.eqmag.com, where an audio example shows what you can do with these samples and about 30 minutes of spare time — and *UK Grime* has about 415 more samples where these came from. —*Craig Anderton*

Power App Alley
by Craig Anderton

WAVES GTR

Use GTR as a vocal processing strip, and even add a synthesized harmony

OBJECTIVE: Use the "stomp box" effects in Waves' GTR guitar processor to do vocal processing.

BACKGROUND: Although GTR is marketed as a guitar amp/effects simulator, the stomp box effects are derived from Waves' standard plug-ins and are well-suited to other instruments. This example shows how to do vocal processing in Cubase SX 3.

steps

1. Insert the GTR Stomp 6 Stereo plug-in into one of your vocal channel's inserts.

2. Insert Compressor as the first effect in the "pedalboard."

3. Similarly, add the Pitcher, EQ, Doubler, Delay, and Reverb (in that order, going from left to right).

4. On the Pitcher, set Min Pitch to 3.00 (minor third harmony) and Max Pitch to 4.00 (major third harmony).

5. Create a MIDI track and assign its output to the Stomp 6 Stereo plug-in. Make sure this is the active track in the next step.

6. Right-click on the Pitcher Pitch control and select Learn, then move a MIDI controller (*e.g.*, mod wheel). At one extreme of the controller, the harmony will be a minor third, and at the other extreme, a major third. This allows dialing in the desired harmony interval in real time as you move the controller.

7. Edit the other FX settings as desired, then after all the parameters are set as desired, save the vocal strip preset for later recall.

tips

- If you want to process several vocals similarly but not identically, insert a Stomp 4 Stereo with Compressor, Pitcher, and EQ in each vocal channel to provide basic dynamics, harmony, and tonal control. Then, insert a Stomp 4 Stereo in an aux bus with Doubler, Delay, and Reverb. This allows different amounts of compression and EQ, as well as different harmonies, with each vocal track; but by feeding into the same aux bus, they'll all have the same ambient effects.

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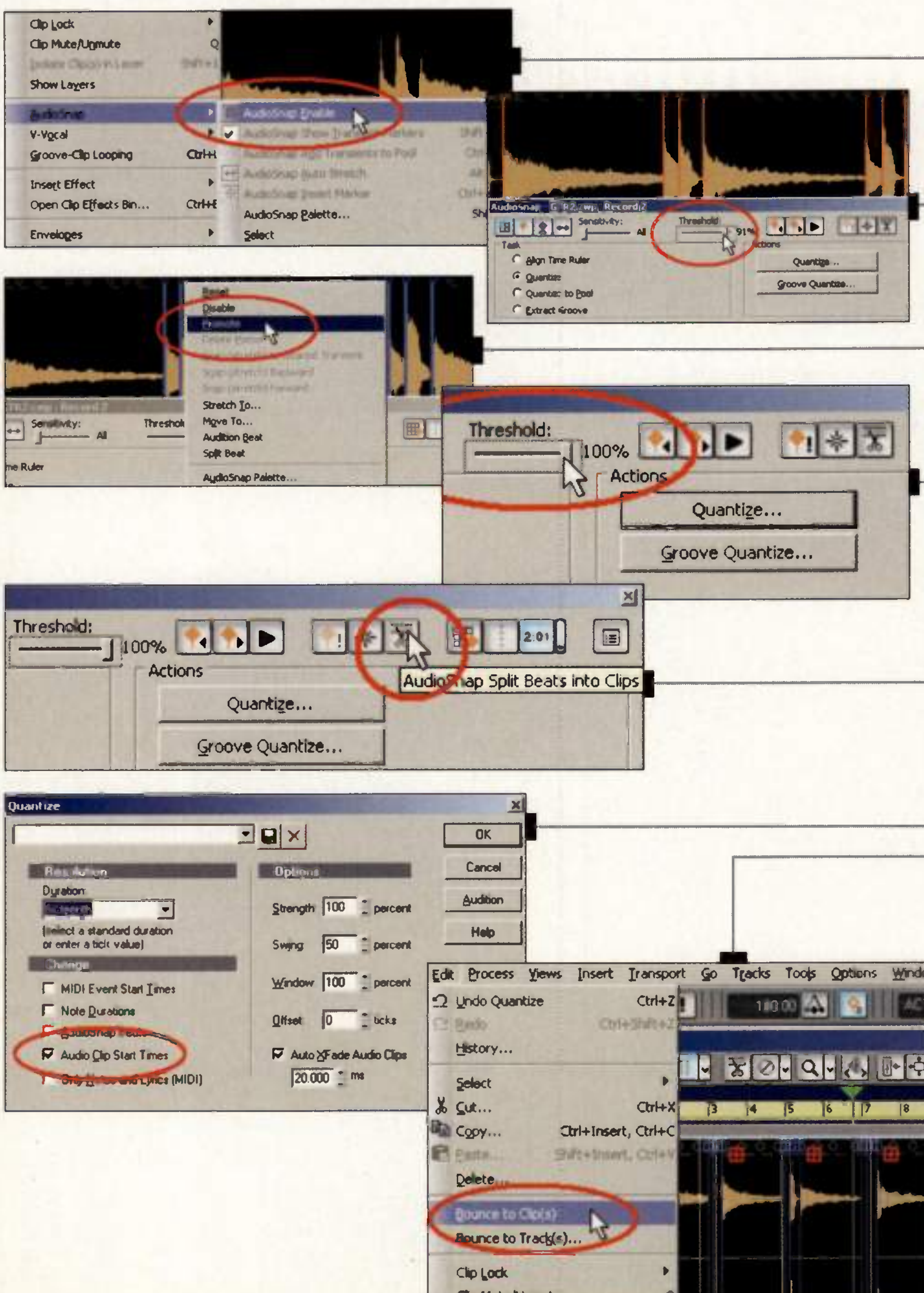
CAKEWALK SONAR 6

Quantize your rhythm guitar part

OBJECTIVE: Quantize the chords in a rhythm guitar part so that they follow the beat more tightly, or even fall exactly on the beat.

BACKGROUND: The AudioSnap feature can define the start of chords by marking transients, then split the clip at those transients. You can then quantize the clip start times.



steps

1. Right-click on the clip you want to quantize, then select AudioSnap Enable (or type F12).

2. Adjust the AudioSnap palette's Threshold control so transient markers (the thin orange lines) appear at the start of each chord.

3. Ctrl-click on each transient marker (it turns blue when selected) that marks the start of a chord, then right-click on any one of them and select Promote.

4. Set Threshold in the AudioSnap palette to 100%. This leaves only the markers you've "promoted," and insures that splitting will occur at no other transient.

5. Make sure the clip is selected, then click on the AudioSnap Split Beats Into Clips button.

6. Select the clips you want to quantize. Click on the Quantize button, then set the Resolution and Options as desired. Under Change, check Audio Clip Start Times, then click on OK.

7. Select all clips, and then go Edit > Bounce to Clip(s). This converts your quantized clip to a standard clip, and disables AudioSnap.

tips

After quantizing, some notes may move ahead in time, thus leaving a gap before the next note. In this case, prior to doing Step 7, you might want to add a fade to the first note so that it has a smooth decay.

Hi Five!

Introducing Auto-Tune 5

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Once again, Antares raises the bar. (Unless, of course, it's sharp. In which case we lower it.)

FOR NEARLY A DECADE, AUTO-TUNE has stood alone as the overwhelming tool of choice for professional pitch correction. Combining transparent processing and superb audio quality with a clear, easy-to-use interface designed to let you get the job done with a minimum of hassle, Auto-Tune is used daily in thousands of studios around the world to save editing time, ease the frustration of endless retakes, save that otherwise once-in-a-lifetime performance, or create unique special effects.

Now, with Auto-Tune 5, we have once again raised the bar. Featuring even better pitch detection, a slick new interface, and a host of productivity enhancements, Auto-Tune 5 is the best sounding, easiest-to-use pitch correction tool in the history of the (known) universe.

Key New Auto-Tune 5 Features Include:

- Improved pitch detection algorithm
- A Humanize function that makes it even easier to get realistic, natural-sounding results
- Realtime natural vibrato adjustment
- Seriously larger Pitch Edit Display
- Realtime pitch tracking display
- Selectable clock source
- Sync to host transport in Graphical Mode (host dependent)
- Multiple simultaneous Graphical Mode instances (host dependent)
- Enhanced Envelope Display functionality for easier navigation in large files
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- Compatibility with pen tablet input
- PowerPC and Intel Mac compatibility
- And lots more...

Auto-Tune 5 is available in TDM and Native formats for Mac OS X and Windows XP. Upgrades are available for registered Auto-Tune owners.* Check it out at your local Antares dealer or come to our website for more details and a fully functional 10 day trial version.

*If you purchase any version of Auto-Tune 4 after September 15, 2006, you will be entitled to a free upgrade to Auto-Tune 5. See our web site for details.

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In the Studio Trenches

by Phil O'Keefe

GROUPING, BUSING, AND PARALLEL PROCESSING

Let your tracks live in a parallel universe

One of the really cool features of most DAW software is a customizable virtual mixer, where you can create layouts and templates for recall at a later date. But in terms of customization, I'm also a huge fan of being able to add auxiliary channels to the mixer for functions such as effects sends and returns.

Usually with an affordable hardware mixer you are limited to a just few aux busses, so those tend to get used up pretty quickly for reverb, headphone monitoring sends, etc. But in software, adding an almost unlimited number of sends is usually just a few mouse clicks away. It certainly beats having to sell your hardware board and buy a new model just to get some more aux sends! Having lots of available aux sends opens up the possibility of using some of them for "bus compression" or parallel processing. By parallel processing I don't mean dual CPUs in your computer, but parallel audio paths for the direct signal and the processed signal, with each on a separate fader.

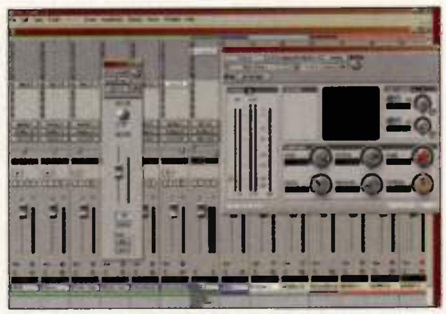

Fig. 1: Drum tracks 1-6 are all sending some of their signal to Bus 1-2. This goes to a Compressor/Limiter, whose output returns into the virtual mixer via track 7.

ENTERING THE PARALLEL UNIVERSE

Suppose you have a set of drum tracks, or several background vocal tracks, or multiple acoustic guitar tracks, and you'd like to compress all of them a bit — either to smooth them out, or add a bit of punch. You could insert an individual compressor inline on each track by using your software mixer's channel insert points, and sometimes that may indeed be your best option. It will certainly allow you to use different attack and delay times — or a completely different compressor — on each track if desired. But it takes more CPU power to run all those compressor plug-ins simultaneously, unless you call on some external hardware assistance (*e.g.*, TC Electronic PowerCore, FocusRite Liquid Mix).

Besides, sometimes you just want a bit of compression across all the drum tracks, or background vocal tracks, using the same type of compressor. This is where routing aux sends to a compressor inserted on an aux return channel can come in handy. The concept is similar to using an aux bus and aux return as a reverb send and return, instead of inserting a separate reverb plug-in for each track. Here's how to set it up.

1. Add a stereo aux return to your software mixer layout. In Pro Tools, do this by hitting Ctrl-Shift-N, and selecting "1, Stereo, Aux Return" in the ensuing dialog box.
2. Upon creating your new stereo aux return, assign the input of that aux return channel to an aux send. In Figure 1, the input for the aux return channel is set to aux ("bus") 1-2.
3. Add an aux send (again, using aux/bus 1-2) to each track you want to send to the processor. If you have "Send levels follow groups" enabled in your Pro Tools preferences (under the "automation" tab), you can then raise or lower all the aux sends on any grouped set of tracks by simply raising any one of the aux sends — and all of the aux sends assigned to the same bus will follow suit for every channel of the group. Of course, you can disable the group and adjust the send levels for any tracks individually as well.
4. Adjust each send's panning individually. With drums, I normally use the same panning for the aux sends as for the stereo mix.
5. After setting the levels and panning, tap the signals off of the original tracks, and bus them to your aux return channel; this is where any inserted effects or processor plug-ins will process the signal before sending it to the stereo mix bus via the aux return channel fader. This allows you to bring the compressed signal up on a separate fader, and blend it in with the original, unprocessed tracks.

This technique makes it easy to fine-tune the amount of compression you want, and because it's working in addition to the original source tracks, you get a more balanced and natural final result, with less "squash." Just raise the aux sends until you're hitting the compressor as hard as you want, then slowly raise the aux return fader until you hear the desired mix of unprocessed and processed signals.

Of course, you can insert other types of plug-ins either before or after the compressor on the aux return channel. Placing an EQ plug-in before the compressor will tend to change the way the compressor responds, while placing it immediately after the compressor will tend to shape the overall sound and timbre of the compressor's output. Neither approach is "right or wrong" and both can be useful, so experiment with both approaches and decide which one sounds best. 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.

by Michael Molenda

SONIC POOCH SCREWING

5 ways to make your guitars sound truly awful

The concept of kick-ass guitar tone is 100 percent subjective, and therefore vulnerable to all kinds of verbal cage matches, evangelistic ejaculations, and outright rebukes. As a result, any how-to article on crafting fabulous guitar tones in the studio is immediately suspect — even if the author is a big-time producer or engineer who has produced massive hits driven by brilliantly recorded guitars. Brilliant? *Who says those tones are brilliant?*

See what I mean? Even wildly successful artists are shackled to the whims of an impassioned and opinionated mob of tone zealots.

But, even in an arena of rampant subjectivity, you can tip the scales toward suckitude. There are certain elements of a guitar tone that, to many musicians' ears, heighten the perception of amateur — or simply bad — sounds. You can hear tone-challenged guitars on myriad DIY releases, as well as on numerous audio files uploaded to www.myspace.com and other online music-sharing communities. Sometimes, less-than-sterling guitar sounds even sneak into major-label releases. And, in these cases, no amount of debate or spin will likely change a listener's view that someone has screwed the pooch.

Now, you may be an iconoclast or a gambler — a sonic rebel who struts horrible and ill-conceived guitar tones into the public domain, and then dares more genteel recordists to classify the fruits of your genius as garbage. If so, you are one courageous buckeroo, and to help ensure that your guitar tones are derided as meerkat poop, here are some surefire tips for murdering the beatific sound of your favorite 6-string.

1: TWEAK IN A VACUUM

Spend tons of time soloing your guitar tracks, adjusting the EQ and other signal processing until each guitar sounds marvelous all by itself. Although those guitar tones must ultimately work with all the other sounds in your mix, don't ever reference the guitars to anything else. That's a sign of weakness, partner. Guitars rule — screw the drums, vocals, bass, keyboards, and so on. If the track sounds a bit thin, muddy, and/or bottom heavy when the guitar tracks are inserted into the main stereo mix, consider remixing — or drastically lowering the level of — all the instruments that are interfering with your spectacular guitar tones. And don't forget to solo any tracks that you're remixing, as well. You certainly don't want to confuse your timbral focus by considering all the individual tracks of your mix as one homogeneous spectrum of sound.

2: TAKE A SHOWER

Everyone knows that guitars sound best when hosed down with gallons of reverb. Find the biggest, baddest Hall patch, and pour it on. Make those reverb tails longer than the contrails from a 757 on a frigid winter morning. Be sure to obliterate any aspect of sonic clarity and punch. Bask in the giddy joy of the Concert Hall reverb on your solo as it smashes head-on into the Large Room reverb on your rhythm guitars. You paid for that digital reverb, so use it! Yum.

3: CRUSH 'EM LIKE BUGS

Compression is a stupendous tool for making guitars appear louder in a mix, and, as every guitarist wants his or her guitars to scream like banshees from hell, be sure to max out the parameters on your compressor of choice. Start with a ratio of 10:1 at a threshold of –20dB, and choose a fast attack and a slug-slow release. You should hear all kinds of pumping and breathing as the compressor destroys the dynamics of your playing, which, in turn, will make it easier for you to boost this garbled muck of guitar goodness to the tip-top of your recorder's output meters. You'll also dig the groovy sustain the compressor will add to your guitar sound — which will goose those enormous reverbs you're applying to the guitar tracks to even more magnificent levels of ambient bliss.

4: GO LOW RIDING

Fat guitar tones are godly, so grab those EQ controls, and boost anything under 100Hz by 6dB or more. Go for a truly morbidly obese tone that swallows everything in its wake. Smother that bass! Suffocate the kick drum! Asphyxiate the toms! Churn those bass frequencies into a vortex of angry mud!

5: CELEBRATE THE PAIN

Of course, it's not enough to have the absolute lowest guitar tone in the galaxy; you also must have that searing, rip-the-hair-from-your-eyebrows midrange that slices through a mix like a lightsaber through Luke Skywalker's wrist. Hit those EQ controls again, find the most obnoxious clanging frequency imaginable (start at around 1kHz to 3kHz), and crank those suckers up by 10dB. Yeeoow! Them's some brutal kerranging guitars. Perfect.

THE GLORY OF ONE'S OWN WORLD

See how easy it is to process your guitar tones until they'll make it near impossible for a listener to enjoy your song — or, perhaps, hear anything else besides the guitars? (And if someone else is mixing, don't forget to tell the engineer to turn up the guitar real loud — they love that!) This is what self-absorbed mixing is all about, and sissies need not apply. After all, if you are truly a confident creative being, then you shouldn't care if your ghastly guitar tones drive audiences away. Fear not. Someday, the public will understand you. 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.

by Lee Flier

HOW MANY TAKES DOES IT REALLY TAKE?

Here's how to improve performances during overdub sessions

One of the most radical changes in the flow of a session since the advent of near-unlimited track counts and non-linear editing has been the way overdubs are tracked. It seems a couple of common scenarios took root almost immediately, and didn't always work to the benefit of the end result.

The first oft-used technique I'll call "performer laziness" (although the performer may or may not be the decision-maker, and others may have more generous terms for it such as "economical" and "time-saving"). With the "lazy" technique, portions of the song that repeat (such as choruses) are often performed only once and then copied to the places where they repeat.

The second scenario is sort of the opposite problem; I'll call it "comp madness." This is where the performer does a seemingly endless number of takes, either of the entire song or specific sections. Each comp is on its own track and the mix engineer and producer then decide which phrases to use from each comped take, and edit them together to make the final track. Sometimes even a combination of the two is used, with say a vocalist doing multiple takes of the same chorus without knowing which one will ultimately be used on which repeat.

But didn't these techniques exist long before DAWs? Well. . . .

THE ANALOG WAY

Back in the analog tape days, both of these techniques were sometimes employed, but a lot of things conspired to limit their abuse. The "lazy" technique was extremely difficult if not impossible with tape, and was limited to being used on the entire mix rather than individual tracks. And "comp madness" could only go so far, as the number of tracks was always limited. Generally, if a singer or instrumental soloist flubbed part of a take one would simply have to punch in over the mistake, and comps were usually just a few takes of the entire song.

In retrospect, although it's comforting to be free from worrying about track limitations and it's convenient to be able to cut and paste a good phrase over a weaker one, there were also certain advantages to having those limitations. It may appear to be saving time and/or increasing creative possibilities to opt for the lazy performer or comp madness technique, but it does affect the performance and doesn't often prove to be such a time saver after all. Why? In the case of the "lazy performer," there are often no dynamic changes between the first chorus and the last, which makes the song seem dynamically flat from start to finish, even though the mix engineer may be adding in other instruments to make it appear something is happening. In the case of "comp madness," the performer doesn't really have the opportunity to put together a take that flows. Trying to sing or play a passage when you're unaware of the context of the arrangement is a real challenge; it's hard to know what kind of dynamics to apply to the performance — and even harder to give a song real emotional depth or a "storyline" that has a clear beginning, middle, and ending. The performer is essentially taken out of this decision process.

A BETTER WAY TO DAW

So, how to mitigate these problems? First, you have to want to! Think about it long and hard, and decide whether what you're doing is truly for artistic reasons, or reasons of convenience. Second, understand why these techniques were developed, and use them in a way that makes sense. For example:

- The "lazy performer" is a handy tool for composing, as it allows you to try different song structures quickly. It may even provide a backdrop that inspires you to come up with lyrics, or other musical ideas. But before you start doing any "keeper" tracks, you should have already committed to the song structure. That way, you can let all of the performers play and sing the song all the way through, in a context as close to the song's final structure as possible.
- The best thing about comping is that it lets you warm up without pressure; you can practice a part over and over, but if lightning strikes and there's a great take, the record button was on. So, use comping to focus on getting one or two complete takes that are strong, rather than thinking of it as a way to get a good phrase here and there, which are then stitched together laboriously.
- Try letting musicians play through a whole section when recording a comp, not just something like comping individual phrases, so that they can have a feel for the context of the performance and aren't "flying blind."
- Put together a good rough mix for the musicians and vocalists before the overdub session; if "everything and the kitchen sink" isn't in the mix throughout and the performers have some clue how the song is going to ebb and flow dynamically, it'll be much easier for them to turn in an emotional performance that's appropriate for the song.

Sure, working with DAWs has some real pitfalls, like dealing with updates and system incompatibility issues. But many a common pitfall of working with DAWs has much more to do with how the DAWS are used — and you can fix those problems without even having to buy any new gear! 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 and countries. She can be contacted via the band's website at www.what-the.com, and also moderates the "Backstage With the Band" forum at www.harmony-central.com.

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by Gus Lozada

Dare to Fix it in the Mix!

Timing is everything . . . and there's nothing like getting parts to groove together

Once we get our tracks into the digital domain, everything is perfect, right? Uh . . . no. Okay, maybe not, but once it's in the digital domain, everything is possible. Right? Well, almost! Although good recording starts at the source (see Phil O'Keefe's column in the Aug. '06 issue) by getting the best possible sound and performance, the second part of a good recording implies putting all those performances together in a way that swings and flows.

Fig. 1: Sonar's V-Vocal can do automatic pitch correction, add or subtract vibrato, and control dynamics of vocal (and other mono) tracks in real time — but it can also change timing and "groove."

WHEN BAD THINGS HAPPEN TO GOOD RECORDINGS

Situations like recording an occasional off-beat drum hit, or having tracks of different musicians recorded in separate locations (or loops from several collections) that prevent getting the same overall "feel" within all of your tracks, can sometimes be solved by checking into your DAW's "groove tools." Most DAWs have at least one built-in time-stretch, groove-correction routine. But what we're talking about here is *not* quantizing everything to some robotic grid, but to let the "groove" of one part influence other parts.

Pro Tools users have known about "Beat Detective" (BD) for a while. BD is a actually a collection of several separate functions, which will allow you to do functions like extract a recorded track's groove and apply it to other tracks — audio and MIDI — so they can all play in harmony, and with the same groove. Other tricks are also possible, like playing drums at different tempos and matching the feel of different loops. And, you can quantize an audio loop to any grid value you want. Even better, BD comes free in all Pro Tools versions 7.0 and above.

When Ableton Live appeared on the scene, one of its premises was that it could meld together various audio sources, regardless of their original pitch and tempo. Although a related article in this issue discusses using Live's "warp markers" to correct drum parts, you're certainly not limited to drums or correcting errors: You can mark beats in an audio file and move them to change the feel of the original performance. For example, perhaps you have a great loop, but it feels stiff and over-quantized. You can move the warp markers to have the snare lag a bit, or a hi-hat hit anticipate the beat. Also note that warping can be applied to a recently recorded track in real time, immediately after you stop recording.

WANT BETTER GROOVES? TRY SOME ACID

This technology of identifying a loop's tempo grid and saving it as meta-data into the file was first established by Sony's Acid, which created the concept of "Acidized" files. In many cases, you can just import a file into Acid, and it will accurately adjust the loop tempo to the project while preserving the file's original "groove." If not, you can tweak the "acidization markers" for individual slices to perfect the stretching process.

Of course, while Acid excels in managing and creating Acidized files (and the latest version of Acid turns the program into a full-featured DAW), other programs like Steinberg Cubase, Cakewalk Sonar, and MOTU Digital Performer can import Acidized files, as well as with other stretchable formats like REX files.

Although you cannot edit an Acidized file's slice markers in Cubase, you can use a "warping" process similar to Live's to correct slices a little bit, and then use "hit points" to generate warp markers in recorded tracks and edit them to make all of your tracks groove together. And don't forget that Cubase was the first program to accept REX files generated by Propellerhead's ReCycle program (ReCycle is a great tool on its own to capture and modify a loop's groove, but note that it's a separate application that does not integrate with DAW hosts).

Sonar has also gone deep into stretching options; it was the first DAW other than Acid to be able to import and edit the slices of Acidized files and in Version 5, they added Roland's Variphrase technology in their V-Vocal processor (Figure 1). This not only corrects tuning problems (similar to Antares' Auto-Tune), but also the timing of vocal tracks. Furthermore, Version 6 introduces AudioSnap, which is a "toolkit" of timing correction, groove-altering, and stretching options.

Fixing in the mix doesn't sound like such a bad idea after all, does it? In fact, when used sparingly to help music "flow" better, it can sound great. EQ

Gus Lozada is a contributing editor to several printed and Web media. He is currently tour- as the front man of WoM (www. com.mx), hosts clinics around Latin America about music production, and moderates "Nuestro Foro," Harmony Central's Spanish-language community. He should be rich by now.

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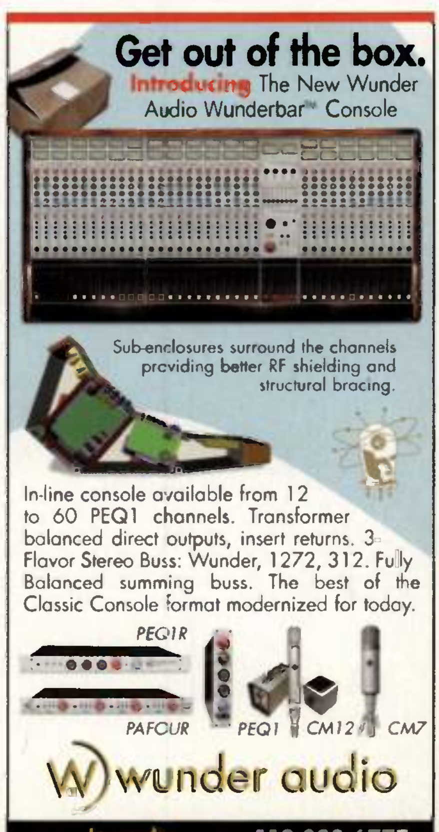

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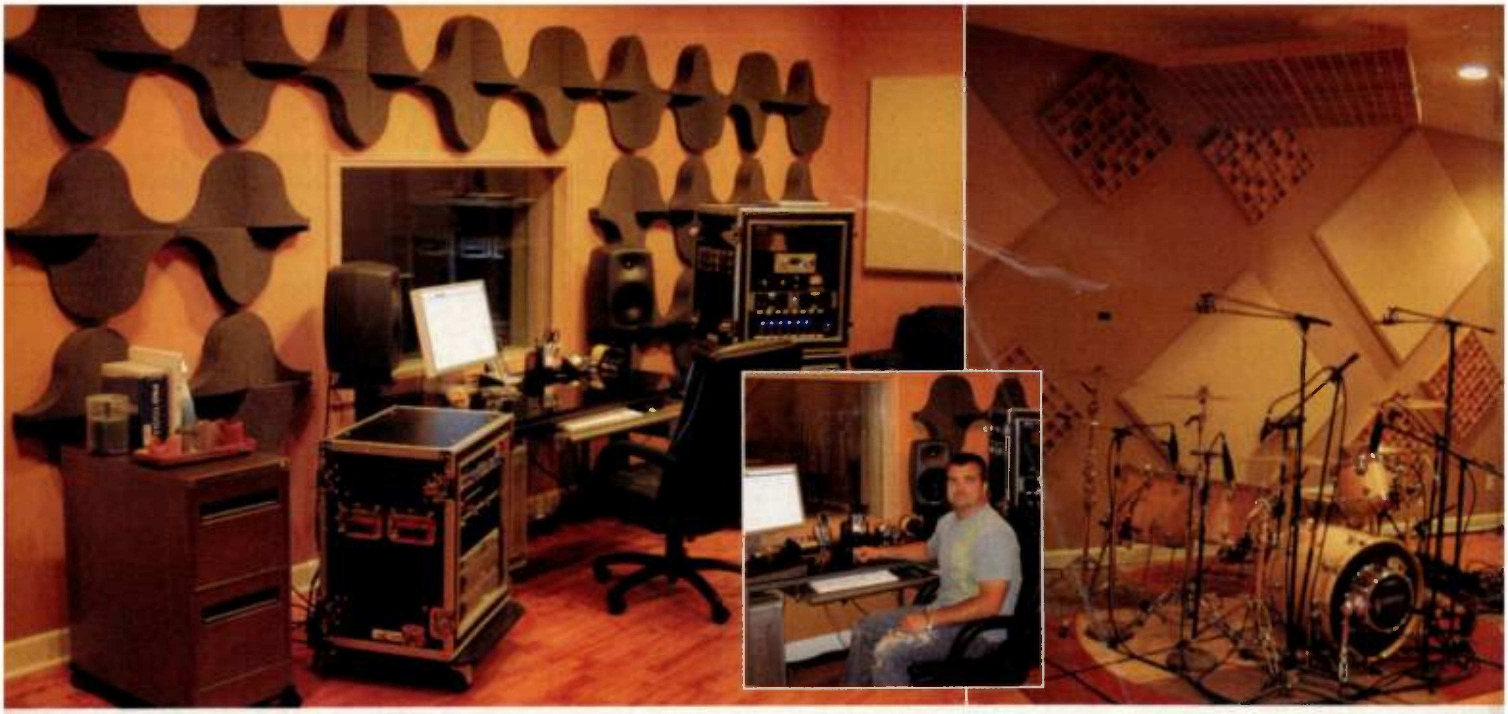

Room with a VU

by Matt Harper

STUDIO NAME: Rivergate Studios
LOCATION: Gautier, MS
KEY CREW: Chris Henderson
CONTACT: www.myspace.com/rivergatestudios
COMPUTER: Macintosh G5 with Apple 23" HD Cinema Display
SOFTWARE: Pro Tools HD7 with Massive Pack 4 plug-Ins
INTERFACE: PreSonus Central Station with CSR-1 remote
STORAGE: Glyph GT 103 with three 80GB hard drives
MONITOR MIXER: Hear Technologies Hear Back Hub with six mixing stations
MONITORS: Genelec 8050
DIRECT BOXES: Avalon U5
PRES: Focusrite ISA 428; PreSonus ADL 600, M80; Universal Audio 2610, 8110
MICS: AKG B-XL II/ST; Audio-Technica 2020, 4050; Beyerdynamic Opus 65, Opus 88, Opus 99; Neumann U87 U147; Royer R-112 Matched Pair; Sennheiser MD 421; Shure SM57; Studio Projects T-3; Yamaha Sub Kick
OUTBOARD: PreSonus ACP 88, Universal Audio 1176, LA-2A
BACKLINE (INSTRUMENTS): DW 5 pc. Collectors Series with Zildjian A Custom Cymbals; Fender 1957 Jazzmaster, 1962 Mustang, 1971 Stratocaster, 2005 Stratocaster, 2005 P-Bass; Gibson 1957 Les Paul Jr., 1968 Les Paul, 1971 SG, 1971 Les Paul Deluxe, Gary Rossington SG, Gary Rossington Signature; Martin HD-28; Paul Reed Smith Custom 22, Hollowbody II, Single Cut; Silvertone; Taylor 810CE
BACKLINE (AMPLIFICATION): Ampeg 8x10, SVT Classic; Bogner Uebershall; Diezal 4x12 (straight), VH-4; Fender 1965 Blackface Twin Reverb (hand-wired); Hughes and Kettner 4x12 (slant), Alex Lifeson Signature, Triamp Mark II; Marshall 1960 4x12 (slant), JCM 800, JCM 900; Mesa-Boogie 4x12 (slant), 4x12 (straight), Dual Rectifier, Single Rectifier; Mills Acoustics AfterBurner 412B; VOX AC30 Blue, Valvetronix 2x12 Combo.

NOTES: "I had built, roughly, a 40x60 metal building on my property a while back, and decided to use it as a studio," owner Chris Henderson tells us, taking a break from his busy schedule as lead guitarist for rock juggernauts 3 Doors Down. "I ended up building another 40x60 room inside the already constructed building, so that it already had sub-walls with the inner gap. And I built it that way all by myself. I was a contractor in my old life before the band, so I knew how to build, but I had never built a recording studio before. I made my share of mistakes, but the room ended up working really well. I did a couple of records there for Redding and Halfdown Thomas, and 3 Doors Down actually did all of our pre-production there. So in that sense it was functional. It did what I set out to do."

But then disaster struck, in the form of Hurricane Katrina sweeping through in the fall of 2005 — flooding the entire facility and effectively erasing all of Henderson's hard work. "When it happened, my heart sank," the guitarist says. "I was on tour at the time, and I wasn't able to get back down there for three weeks — so the water stayed pent up, leaving a ton of black mold over everything. I had to tear it down on the inside and start over, but I got a chance to fix all the little mistakes I made the first time around."

So from the ground up Henderson started anew, procuring a few pieces of his favorite gear and constructing a Pro Tools project studio that is a true recording guitarist's cornucopia. Shying away from the thought of running a large-scale console, Henderson works primarily "in the box," utilizing some choice vintage outboard gear and smart selections for mics, pres, and, most importantly, indisputably great instruments to achieve a sound comparable to the major studios he has recorded in during his stint with one modern rock's most celebrated bands.

But the most notable aspect of Rivergate Studios isn't its gear, or the personality behind the board, but the altruistic approach the facility takes to the area's Katrina-ravaged musicians/bands, many of which are left without the funds to record as they attempt to rebuild their lives. "All the recording studios that were there didn't survive and I'm the only one who has rebuilt. All the clubs that these bands can play at are gone and since you need to play to make money, these bands have no way of making money to record. My goal is to provide a place where Gulf Coast bands can record and not have to worry about paying a ton of money. If they come from a Katrina area, I charge nothing — not until these guys start getting places to play so they get back to making a living." EQ

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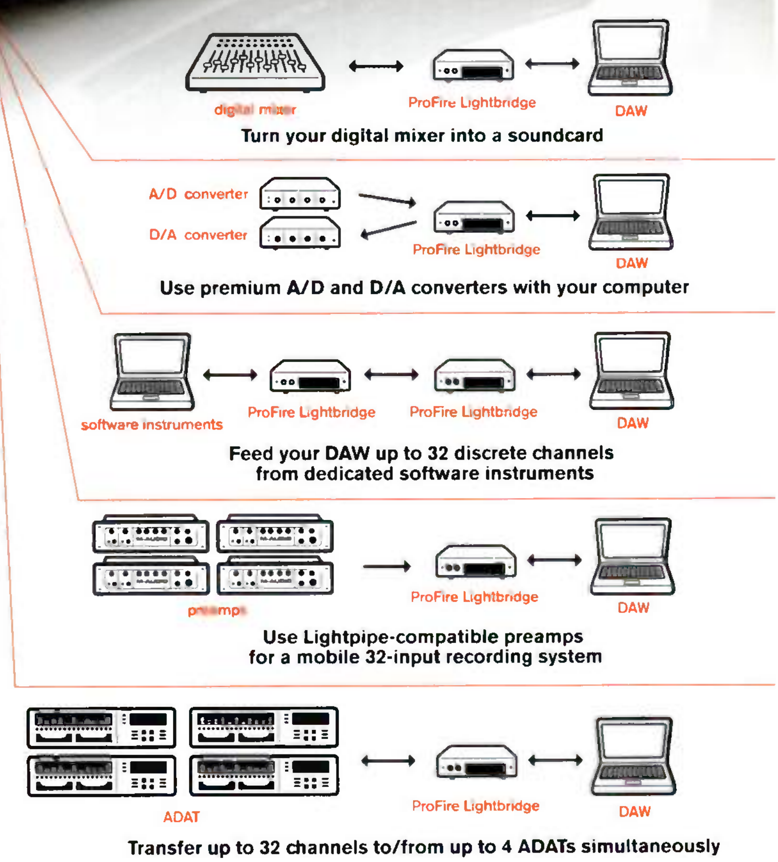

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