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Glen Ballard

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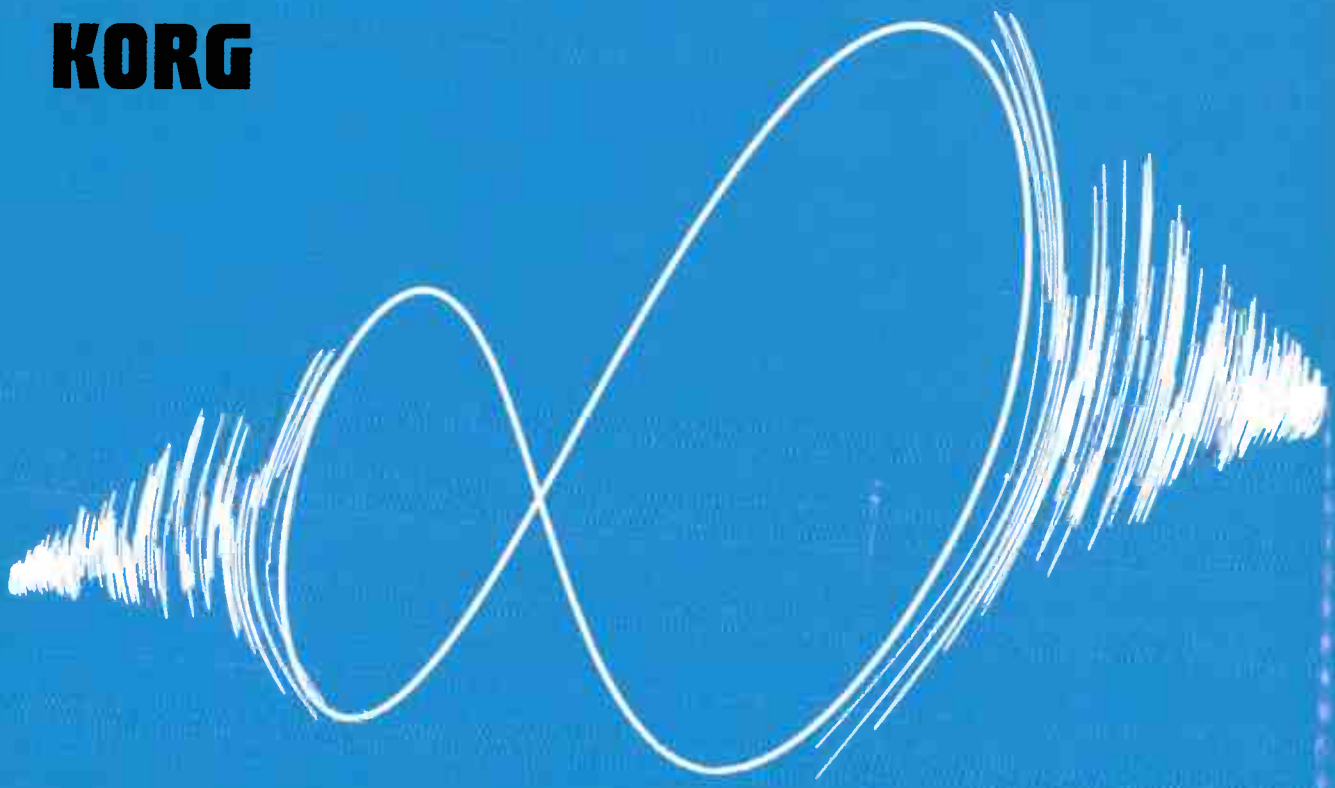

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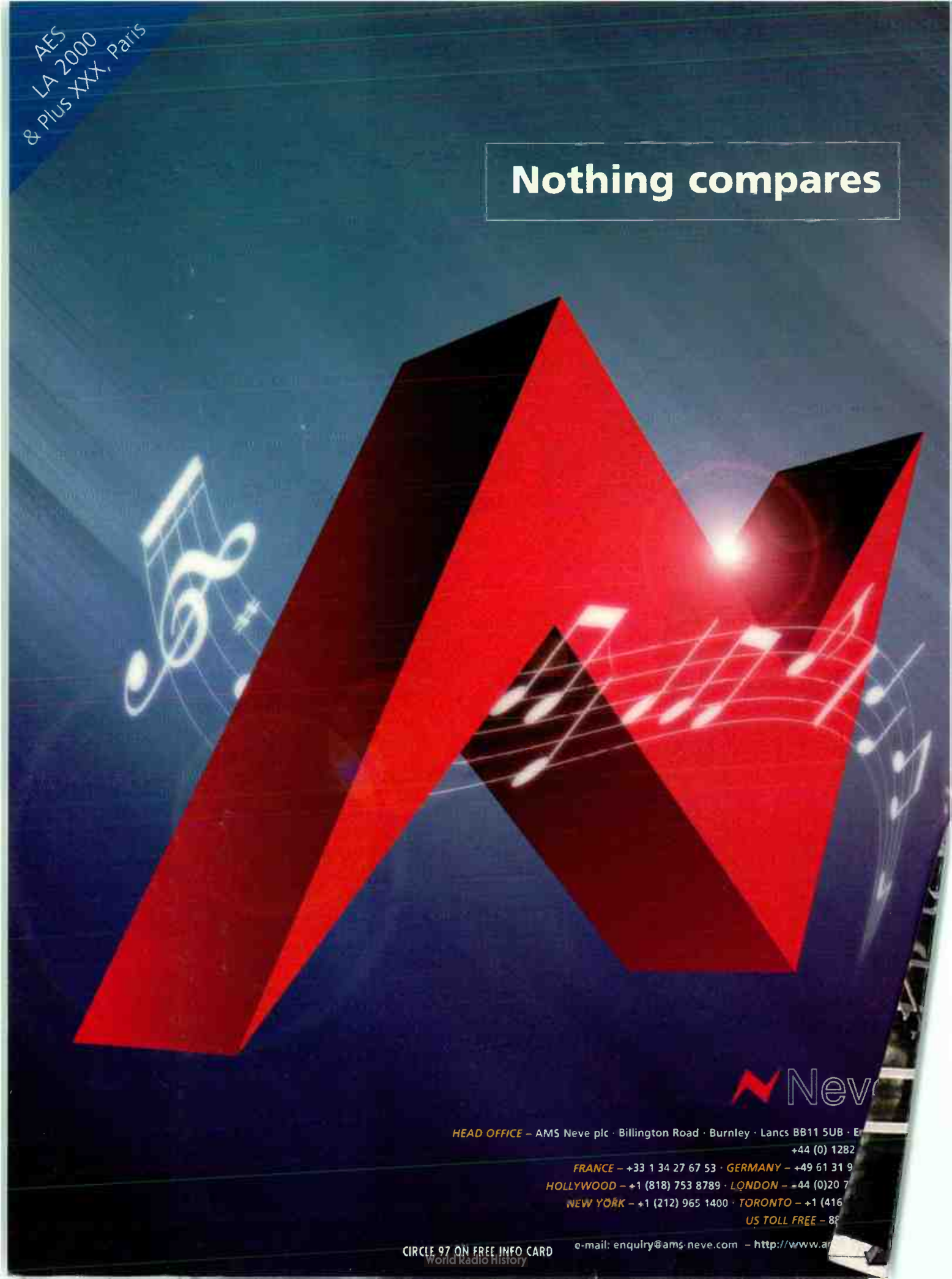

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Glen Ballard and his guitar.
Photo by Con Keyes.

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EDITORIAL

Lucy's False Teeth

I work long hours, so it's not unusual for me to find myself leaving my studio/office to walk home at 1 or 2 a.m., sometimes even later. Crossing Manhattan at that hour is always entertaining. To minimize the "entertainment value," I trundle along at a fairly blistering rate — head down, peripheral vision set on "high," focus entirely on putting one foot in front of the other.

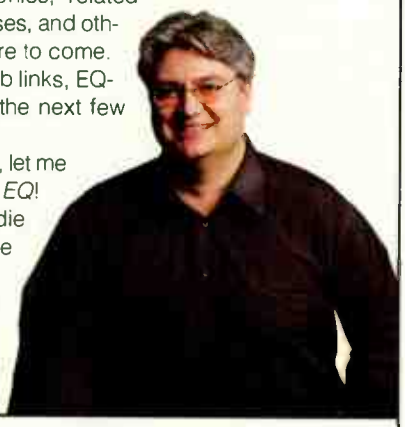
On one such sojourn, I found myself overtaking a nice-looking couple who were strolling at a relaxed pace and chatting quietly. Being in "Late-Night Forced March" mode, I blew past without really paying attention to what they were saying. About a block later, their conversation penetrated — at least the snippet I had walked through. "Did you hear about Eddie?" the fellow had asked his lady. "Lucy hit him with her false teeth...."

I stopped dead in my tracks — what could this possibly be about? I was positive I hadn't misheard. I turned around hoping the couple would catch up and pass me by. My intention was to shadow them like some sort of late-night gossip-stalker, in hope of hearing at least the conclusion of what had to be an amazing story.

Sadly, by then, the couple was gone. Even more sadly, I'll never know the true story behind Lucy's dental strike. What had Eddie done to deserve such treatment? Had Lucy thrown her dentures, or had she clenched them tightly in her hand and smacked him? Apparently the teeth had connected with Eddie; had the poor man survived? Giving up on getting answers to these and myriad other questions, I chalked the incident up as yet another in a growing string of New York "moments."

But Eddie, Lucy, and the teeth weren't to be forgotten quite that easily. I named an acoustic guitar piece in honor of the event ("Lucy's False Teeth"), but even that wasn't enough to put the ghost to rest. Clearly there was a lesson to be learned here. And I've figured it out: "If you go through life too focused or with blinders on, you'll miss a lot of interesting things along the way." What does that have to do with project studios? I have no idea, but it's probably something pretty darned profound or I wouldn't have thought it up.

No, the point of this true-life parable is that I don't want you to miss that you hold before you a brand-new, re-born EQ. The magazine has undergone a top-to-bottom graphic overhaul courtesy of the immensely talented Greg Gennaro under the guidance of art director Marshall Moseley and managing editor Tony Savona. I'm thrilled with the results, and I hope you'll find the new look cleaner, more exciting, and easier to peruse. But we've also added a number of new things, and *that*'s what I don't want you to blow past without noticing. Chief among these? In most articles you'll find a "Web Link" box. The information and URLs in those boxes is your key to a new, expanded *EQ* experience. Follow the Web links to audio examples, additional text, photos, graphics, related articles from past issues, e-mail addresses, and other Web resources. And there's still more to come. For example, to support all of those Web links, EQmag.com will also be recreated over the next few months.



On behalf of the entire magazine staff, let me be the first to welcome you to the new *EQ*! And please, if anyone knows how Eddie and Lucy are getting along, let me know....

—Mitch Gallagher
mgallagher@uemedia.com

EQ

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BPA INTERNATIONAL

United Entertainment Media

Glen Kolotkin shows off his gold record for Jimi Hendrix's *Electric Ladyland*.

WHODUNNIT

In your June 2000 issue, it was stated by Glen Kolotkin that he was the remix engineer for Jimi Hendrix's *Electric Ladyland* album. Since I had recorded most of Jimi's music at that point and had mixed *Electric Ladyland*, I find his claim curious to say the least.

I can attest that there is not one tape in the entire Hendrix Family Library that bears either CBS Studios or Mr. Kolotkin's name on it.

When *Electric Ladyland* was re-mastered in 1997 by myself and George Marino at Sterling Sound, we (John McDermott, curator of the Hendrix Library, and I) found the original 1/4-inch masters with my handwriting on them. These are the originals — they match every note on the CDs.

It *is* true that when the album was completed in 1968 the tapes were sent to Columbia Studios for *mastering*. The engineers here at the time made a complete hash of the sound and to add further insult to injury, printed the title on the acetate labels incorrectly as "Electric Land Lady"!

The album was subsequently remastered by Reprise in Los Angeles. If Mr. Kolotkin remembers Jimi coming to CBS Studios in an attempt to remix the album, it certainly didn't appear on any subsequent tapes that we are aware of. The album was mixed and recorded by Eddie Kramer and Gary Kellgren with help from Jimi Hendrix.

As far as the "multiple phasing" using three tape machines — this is a technique I had been experimenting with in London at Olympic Studios in 1967. When it came time to mix *Electric Ladyland*, both Gary and myself were into trying *many* different techniques *including* the accidental 3-D effect that Mr. Kolotkin refers to. In fact, it was standard procedure to have multiple machines in the control room for voice doubling and phasing.

In the December '99 issue of *EQ*, Craig Anderton writes that Tony Bongiovi was doing most of the engineering on *Electric Ladyland*. This is so blatantly untrue as to be laughable. While it is true that Mr. Bongiovi was an engineer at the Record Plant NYC, it was as an assistant initially. Mr. Bongiovi may have done the odd overdub or parts of a basic track, but to suggest that he did the lion's share of work on this album is entirely untrue.

As far as the flanging — as previously stated, the VFO system was a permanent fixture at the Record Plant and the direction and implementation came from either Gary Kellgren or I. The tone of the article strangely suggests Mr. Bongiovi did most of that, which he definitely did *not*! Please ensure that the correct story is told to your readers.

Eddie Kramer
Cold Spring, NY

[*Glen Kolotkin replies: It is true that Jimi walked into remix room 412 at Columbia Records' studio at 49 E.52nd St., New York. He told me that he had been working on the album for over a year and that he was not happy with the mix.*

In addition to remixing the album, Jimi brought in a female friend. On one of the days, I recorded Jimi playing his guitar and his friend on the harmonica, right in the remix room on a four-track half-inch machine! The tapes are most likely in the Iron Mountain vaults to this day. I remember the final night finishing up late. They had to keep the mastering man after his shift to do the mastering. Both Jimi and I watched the master being cut.

As you can imagine, with Jimi Hendrix in the studios the word spread fast and many people knew of his session. Don Puluse was there and confirms he remembers talking to Jimi. (Don is now the Dean of Technology at Berklee in Boston.) With over 23 Gold and Platinum records to my credit I don't have to make up any stories!]

[*Craig Anderton replies: Actually, I did not write that, "Tony Bongiovi was doing most of the engineering on* Electric Ladyland*." I was specifically referring to those times when Hendrix ran over into our time, and during those sessions, Tony was indeed doing most of the engineering. I think the article made it very clear that I wasn't a participant in the* Electric Ladyland *sessions, just a lucky observer. My comments weren't meant to diminish anyone's contributions; in fact, the article states that Gary Kellgren and Eddie Kramer are credited in the liner notes for engineering.*

As for the VFO, I mentioned this because I'd heard that Hendrix's flanging was produced by the engineer pressing his thumb on the tape reel. Tony said this wasn't true, and now Mr. Kramer has confirmed it was a VFO system. But remember, this mention was in the context of an article saying that we benefited from techniques originally developed for Hendrix projects. Tony never claimed to have invented the concept, but he was in fact the person who hooked it up for us, which is what I reported. If it needs to be clarified that Mr. Kramer and Mr. Kellgren came up with the system, that's an interesting historical point, and it's duly noted.]

"With over 23 Gold and Platinum records to my credit I don't have to make up any stories!"
—Glen Kolotkin

ERROR CORRECTED

In his review of the Yamaha CDR1000 (*EQ*, July 2000) Howard Massey writes (p. 116):

"[O]ne strong advantage of using CD-Rs and CD-RWs for both mixing and reference purposes is that, unlike DAT, they don't use error correction; that's why, even in the most stringent conditions, DATs are rarely bit-transparent."

Wrong. Both CD and DAT formats use heavy error correction, and it works: both DATs and CDs are bit-transparent under all but the most gross conditions, such as damaged media or misaligned players. I've proven it with subtraction tests.

It would be helpful if professional DAT and CD players had error meters that showed us how close we were coming to losing anything, and when we were operating in a lossy (error concealment) mode. If the manufacturers were up-front with this information, there would be a lot less mystery to digital media, and we could all have real confidence in the integrity of our sound.

Dan Dugan
via e-mail

[*Howard Massey replies: I stand humbly corrected on the first point. Both CD and DAT do indeed employ error correction; however, this is no absolute guarantee of bit transparency. I agree entirely with Dan's suggestion that professional players offer error meters — something that audio professionals have long been campaigning for.*]

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CD-R PROBLEMS

I use the cheapest CD-Rs I can find for giveaways and I just burned one that "ticks" every 3-5 seconds, but only on one CD player. I burned it at 1x thinking it would burn better (I usually burn at 4x). This is the first time I've had a problem like this. Is it because it's a cheap CD-R? What's actually causing the "ticks," error detection? And why on only one machine? Other CDs play fine on it.

—*alphajerk*

Try the same thing with a different CD-R. Does it click on the same player? CD-Rs aren't guaranteed to play on every player. All adjustments made to a player when it's manufactured are ±10%. A player that works is off in a good direction, while one that doesn't work is off in a bad direction.

The same thing happens with DAT machines. Sometimes a DAT won't play back in one machine, but will play everywhere else.

—*Roger Nichols*

How old is the CD burner? Newer CD burners adjust the laser intensity to the recording medium, which results in more consistent and accurate pit dimensions on the CD-R surface. For older burners, the advice I've received is to burn at 2x for masters — otherwise you get poorly formed pits on the CD-R, resulting in potential playback problems.

—*Quin*

It's a relatively new burner. I've burned over 2,000 CD-Rs on it at 4x with no problems; it was only when I burned at 1x that the problem developed.

—*alphajerk*

Then I think this would point to an "overburn" problem — the media can't handle the longer laser bursts and the pits are poorly formed, making the disc hard to read. There's a load of info on CD-R generation at **www.cloversystems.com**. These people make CD analyzers and are a great source for related info.

—*Quin*

PATCHBAY CONFUSION

A friend and I are combining our overgrown home studios into a non-commercial project studio. I'm struggling with the patchbay design. I want to keep things as portable as possible, because we want to be able to take gear out for "live to two-track" sessions. I'm leaning toward a TT patchbay w/ XLRs on the rear panel. Does anyone know of such a patchbay with TT front panels and XLR rear panels? Are there good reasons why no one uses this sort of patch bay? Good reasons why I shouldn't go this route?

—*sjp*

Have you thought about using ELCO connectors, attached to a snake, with an XLR fan out? You'd have to organize your racks based on functionality, as each snake would typically service only one rack, but you'd have an easier job with connectors, less cable, and would save a load of money in the process.

—*Quin*

I am not aware of a company manufacturing a TT patchbay with XLR connectors on the back. Of course, it's easy enough to wire XLR connectors to the back of a TT bay. Anything goes when it comes to wiring patchbays.

If you're open to other options, then I'd recommend rear-panel DLs (a ZIF connector made by ITT Cannon) and maybe one or two XLR panels on the back of the patchbay rack. This would allow you to maintain a certain number of plug-and-play XLR points on the bays and still have an efficient means (via DL) of connecting the other racks.

You could use DL-to-XLR breakout cables; whenever a significant amount of gear changed, simply change the accompanying harness. The patchbay wiring (which is typically the most fragile) wouldn't change, giving you a long life expectancy for the system.

Another important question is how many bays you'll need. If it's only one, then the all-XLR option might be the way to go. Two or more bays and I would start thinking about a DL/XLR combination.

—*David Frangioni*

MIC PRES...WHAT MAKES 'EM GOOD?

I'd like to know basically what makes an expensive mic pre better sounding than a cheaper one. You read the specs (and the reviews) and begin to wonder if there's really a sonic difference that equals the price difference.

—*ckirkp1021*

To get an idea of the cost difference, break down the chain inside the box.

Input (phantom) filter: One-pole (cheaper) or two-pole (better low frequency response).

Transformer or non-transformer: Transformer front-ends are far easier to design and implement, and given other parts savings, the less-expensive alternative.

First gain stage: Tube, bipolar, or JFET. I think transformers are less transparent than bipolars. FETs are noisier and a little less linear. Tubes guarantee reliability problems compared to solid-state, but have that "now" sound (for reasons I, for one, can't fathom), and always seem mushier than solid-state. It's quite difficult to design a stable, wide gain range, linear bipolar first stage, and probably the more expensive option.

Power supply: Wall-wart, switcher, or custom low-noise linear. Again, the latter is more expensive, but better.

Case and connectors: Steel chassis versus punched aluminum, gold versus. silver versus tinned something-or-other connectors. You get the idea.

Is there a sonic difference? I think so. The question is *always*, "Does the sonic difference outweigh the increased cost?" But I suppose the question that you'll first need to answer for yourself is, "Can I hear the difference?"

—*George Massenburg*

The differences between preamps are like the differences between camera lenses. At first, you may think that they all sound (or look) the same, but with experience your tastes and discrimination develops and you realize there really is a big difference.

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On snare, it may be how much of the room ambience is picked up. Or how well you hear the rattle of the individual snares at the end of a hit, or how clearly you can tell where on the snare head the stick landed. These are excruciatingly small details to some — inaudible differences to others. But once you've learned to detect and appreciate the differences, they're not subtle.

After years and years of comparing, I thought I had figured it out. Then we put 33 preamps together in the same room and listened blindly and didn't make distinctions by price, class, design, name, or tube-ness. Guess what? I learned a thing or two. The truth is "it's all in the ears." Just as beauty is in the eye of the beholder, sonic beauty is in the tools of the engineer.

—Lynn Fuston

BACKGROUND VOCALS

How can I get my background vocals to sound tight? Is it mainly the talent or is it the outboard gear?

—sngwrtr_prdcr

There are many ways to get backround vocals "happening," although it truly is the arrangement *and* the performance. Try double, triple, quadruple tracking; changing tape speed (if you're analog, or have a varispeed on your computer) between tracks (you know, like ABBA); getting great singers and balancing them on one mic. Find that hooky, ear-catching "thang." You do it by using your imagination, and then executing what's in your "mind's ear." Listening to a lot of music really helps...from doo wop to Motown to Stax to Chess to Babyface to Roy Rogers, and especially The Sons of the Pioneers. Listen to what the early folks did...everything now is based on that.

—Ed Cherney

Go for the talent. The arrangement is a big thing, too. A well-done vocal arrangement can have much more effect with just a few voices.

Technically, I like multiple singers in a room, each on their own mic. The performance seems much better. Have a keyboard in the tracking room for instant pitch checks and arrangement discussion. The arranger should be in there with the singers, because the mixed sound of singers in the control room monitors can be deceiving in terms of the pitch.

Work hard on the headphone mix. It shouldn't be necessary to have individual mixes because they're supposed to sing as an ensemble. And don't think you can fix it in the mix with AutoTune; you can end up chasing your tail because singers relate their pitch to what they hear — fix up one singer and the next one is out of whack.

—ozbass

NOISE FLOOR IN PC RECORDING

I'm a little confused about mic noise specs, preamp specs, and sound card specs. I am using a Midiman Delta 44, Cakewalk 9.0, CoolEdit 2000, and a Mackie CR1604. When looking at the metering on CoolEdit 2000, the noise floor is around –54 with an AKG 414B-ULS mic turned up to normal level, –73 with the mic turned down, and -84 with no cable plugged into the front of the Delta 44 breakout box. I am recording at 16-bit, so in theory I should go down to –96, right? The noise I am getting seems to be ambient noise. What levels should I expect? What's reasonable in a home/project environment?

—Ronnie

As a comparison, –54 is the noise level of two-inch 24-track analog at 30 ips with no noise reduction. If that's the best you can get, then it isn't bad — people made records that way until Dolby and digital showed up.

A normal project studio that's said to be "soundproof" really isn't, unless you have built a room inside of a room with a floating floor isolated from the surrounding foundation, which costs lots of money. Usually the soundproofing is reflection control for high frequencies, but not for transmitted low frequencies. You may not hear it, but it still shows up on the meters.

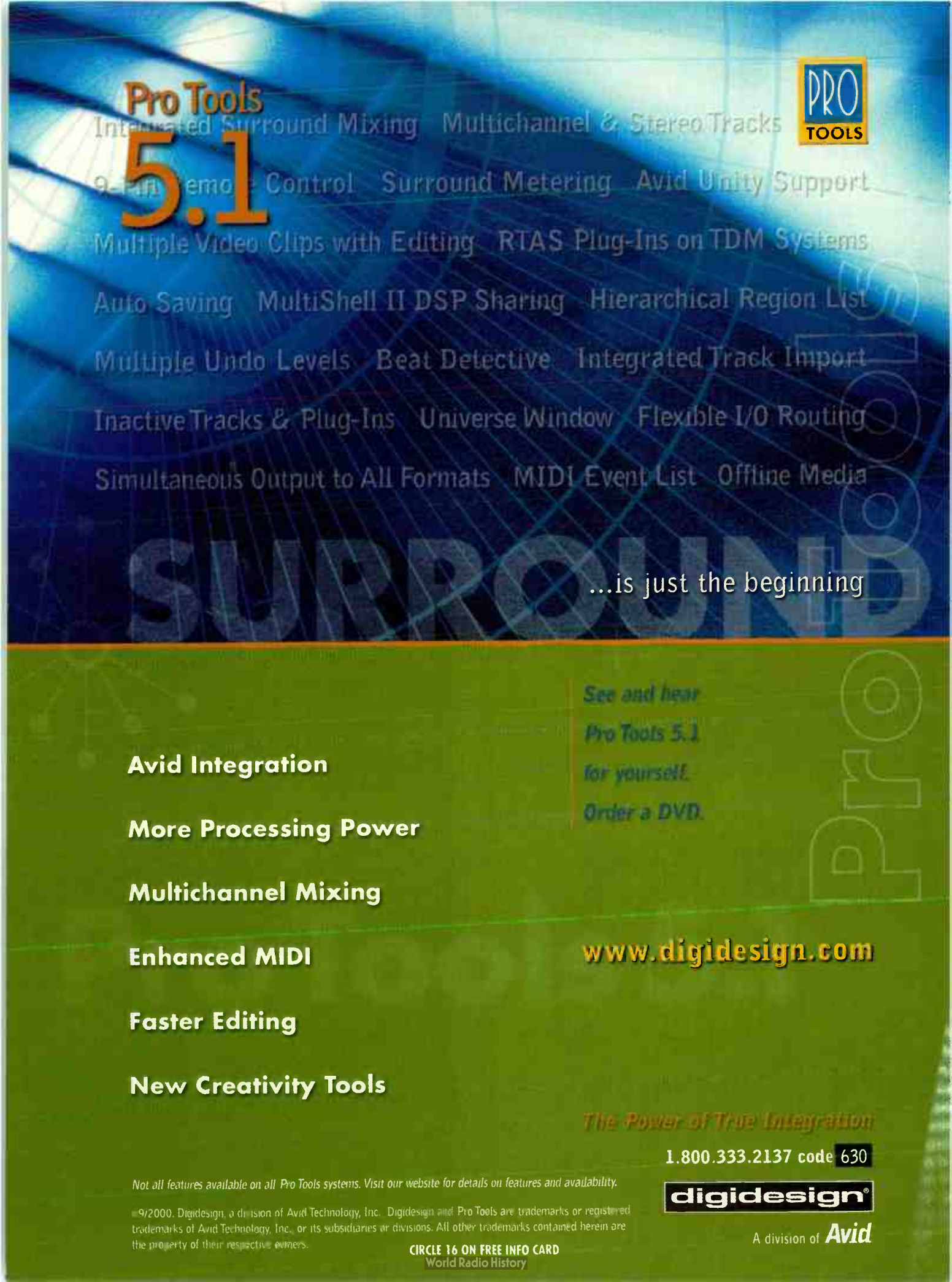
Just buy one of those cheap noise level meters at Radio Shack to measure the ambient noise level with the computers and everything turned off. That will be the best you are going to be able to do. There's more to buying a mic pre than the noise floor. The quality of the sound seems to me to be the deciding factor.

If you're going to be recording to hard disk, just mute each instrument between sections where it plays. Also, usually the instrument will be pretty loud and the mic gain will be turned down, thus lowering the noise floor. Lots of studios that cut hit records in the '70s weren't much quieter than today's home studio, so I wouldn't worry about it for now. You do the best you can, and crank out the hits.

—Roger Nichols

WEBLINK

Have a question you'd like answered? Visit Roger Nichols, George Massenburg, Ed Cherney, Al Kooper, and David Frangioni online at www.eqmag.com



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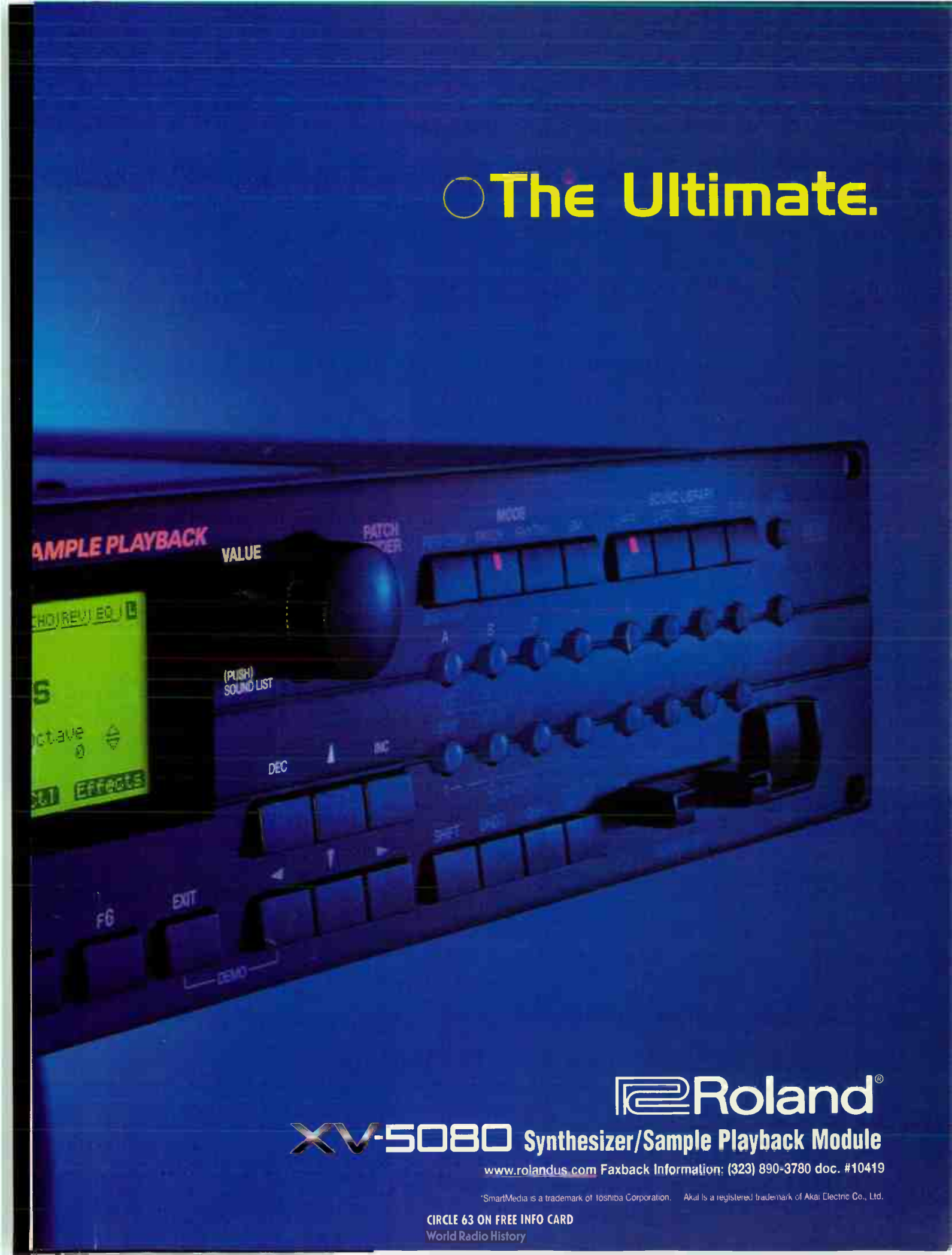

AES PREVIEW

The Product: SeaSound Soloist
The Basics: Hard disk recording interface
The Details: Everything you need to get audio into your computer — a PC- and Mac-compatible PCI card plus rackmount breakout box with stereo line inputs, high-quality mic and guitar preamps, dual 24-bit/96 kHz A/D converters, and a headphone amp — all in one comprehensive package. Additional features include six-segment VU meters, a MIDI activity light, input monitor, and computer monitor and control room level controls. **Contact:** SeaSound, 415-485-3900 or **www.seasound.com**. Circle *EQ* free lit. #111.

The Product: Countryman B6
The Basics: Lavalier microphone
The Details: Said to be the world's smallest lavalier mic, the B6 measures a mere one-tenth of an inch in diameter, making it ideal for broadcast and live performance applications. Tiny enough to be taped to the performer's head or body or worn discreetly on the collar, it comes with replaceable protective caps that not only prevent perspiration and makeup from entering the microphone, but can also alter the mic's frequency response to match the desired application. **Contact:** Countryman, 650-364-9988 or **www.countryman.com**. Circle *EQ* free lit. #112.

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AES PREVIEW

The Product: Sadie RADiA and ARTEMIS
The Basics: DAWs
The Details: RADiA is a PC-based entry-level editing platform that provides four inputs and outputs and up to 24 replay tracks at 48 kHz. ARTEMIS is supplied as a fully configured, rack-mountable turnkey system in a range of digital and analog input/output configurations, all with timecode support and four channels of RS422 interface, as well as onboard SCSI. 20-bit A/D conversion and support for 24-bit 96 kHz and 192 kHz audio editing is standard.
Contact: Sadie, 615-327-1140 or **www.sadie.com**. Circle *EQ* free lit. #113.

The Product: Dolby DP570
The Basics: Multichannel monitor
The Details: The DP570 Multichannel Audio Tool is an all-in-one device that enables producers to monitor and enhance the quality of their multichannel mixes. It allows the user to monitor multichannel audio, check downmixes, re-assign channels, and quickly create Dolby Digital metadata and hear its effect in real time. In conjunction with Dolby E and Dolby Digital codecs, the DP570 provides extensive control over what consumers hear at home.
Contact: Dolby Laboratories , 415-558-0200 or **www.dolby.com**. Circle *EQ* free lit. #114.

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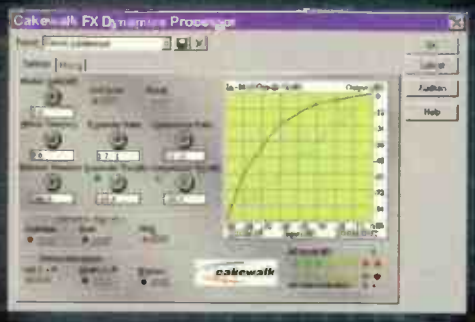

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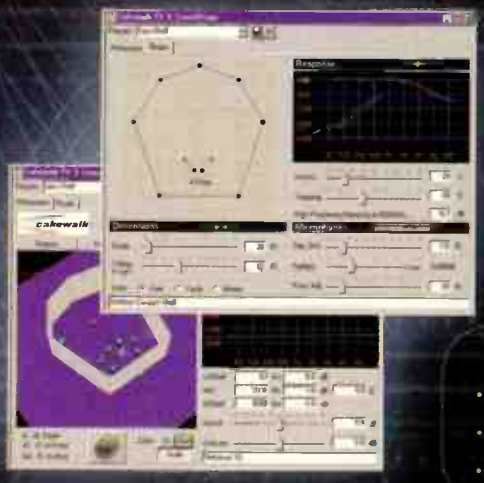

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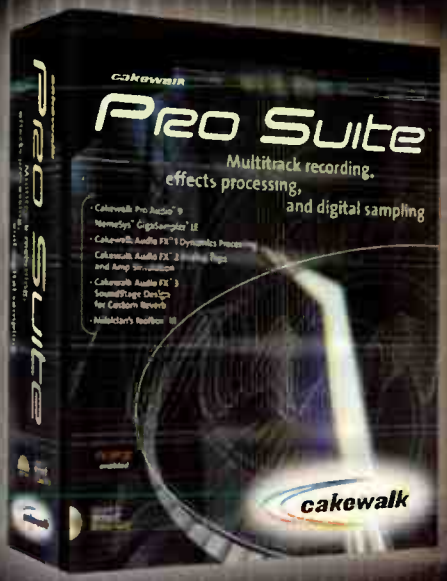

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CIRCLE 11 ON FREE INFO CARD

GT Electronics VIPRE

By Steve La Cerra

GT Electronics has introduced the Variable Impedance Microphone Preamplifier, or VIPRE. Based completely on Class-A vacuum-tube circuitry, this new preamp employs no transistors, integrated circuits, or electrolytic capacitors in its signal path. And as the name implies, VIPRE is one of the few mic preamps currently in production with variable input impedance. Further, GT Electronics' engineers have provided an adjustable rise-time function, further expanding VIPRE's sonic possibilities.

Microphone sound quality can grossly vary depending upon how the mic must "push" against the input of the mic pre to which it is connected. That resistance to "push" is the input impedance of the mic pre. Impedance mismatches between microphones and mic pre's can attenuate high frequencies or result in a loss of signal level. By providing the option to adjust input impedance, GT Electronics' engineers make it possible to achieve the maximum in performance from a given microphone. Owners of ribbon mics will appreciate this feature because ribbon mics tend to be particularly sensitive to input impedance.

At the input stage of the VIPRE is a multi-tap input transformer; the taps can be selected using a front-panel switch. Each tap provides a unique impedance allowing the VIPRE to be tuned to your microphone.

Impedance settings include 300, 600, 1,200, and 2,400 ohms. A transformerless input is also provided for a completely different audio character, and a front-panel instrument input, featuring a 20 dB pad, is included.

In addition to variable impedance, VIPRE also provides user control over "rise time" or slope. Rise time describes how fast the audio circuit responds to a transient. Faster rise times allow the attack of a transient to come through more clearly, maintaining the aggressive qualities of a sound; slowing down the rise time helps smooth out sibilant sounds, providing an increase in the "vintage factor." Rise time of the VIPRE may be adjusted in five steps from six volts per microsecond to 0.75 volts per microsecond — a value akin to the transient response of vintage audio gear.

Other front-panel controls for the VIPRE include coarse and fine controls for gain adjustment. These aren't pots, but stepped attenuators. The coarse attenuator provides 5 dB steps while the fine attenuator provides 1 dB steps. Besides ensuring accurate repeatability of gain settings, the attenuators allow GT Electronics to maintain a completely balanced audio path throughout the device. Maximum gain from the VIPRE is 75 dB. A VU meter shows the signal level from –20 dB to +4 dB, but may also be switched to an expanded view (–60 dB to +9 dB). There are three further settings that place 0 dB VU at +10 dB, +20 dB, and +30 dB, for more of a "peak response" in high-gain applications. Complementing the VU meter is a pair of lights, one indicating 3 dB below clipping, the other indicating clipping.

GT Electronics specifically developed a power supply with massive storage capabilities for the VIPRE, ensuring that the high-level stages of the circuitry get the power they need, while rectifier hum is filtered down to inaudible levels. This helps the VIPRE capture the full dynamic range of the source without running out of gas on loud transients or strong low-frequency signals. ■

What is it? A single-channel, vacuum tube-based microphone preamp with user-adjustable input impedance.

Who needs it? People who use a wide variety of microphones.

Why is it a big deal? It is one of few (if not the only) mic preamps currently in production that offers adjustable input impedance.

Special notes: By matching its input impedance to the microphone's output, VIPRE helps get the best possible sound out of your microphones.

Shipping: This fall.

Suggested retail price: $2,199

Contact: For more information contact GT Electronics, a division of Alesis Corporation, at 800-525-3747 or visit **www.gtelectronics.com**. *EQ* free lit. #101.

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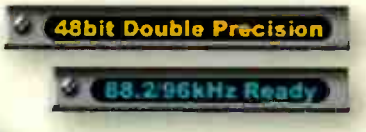

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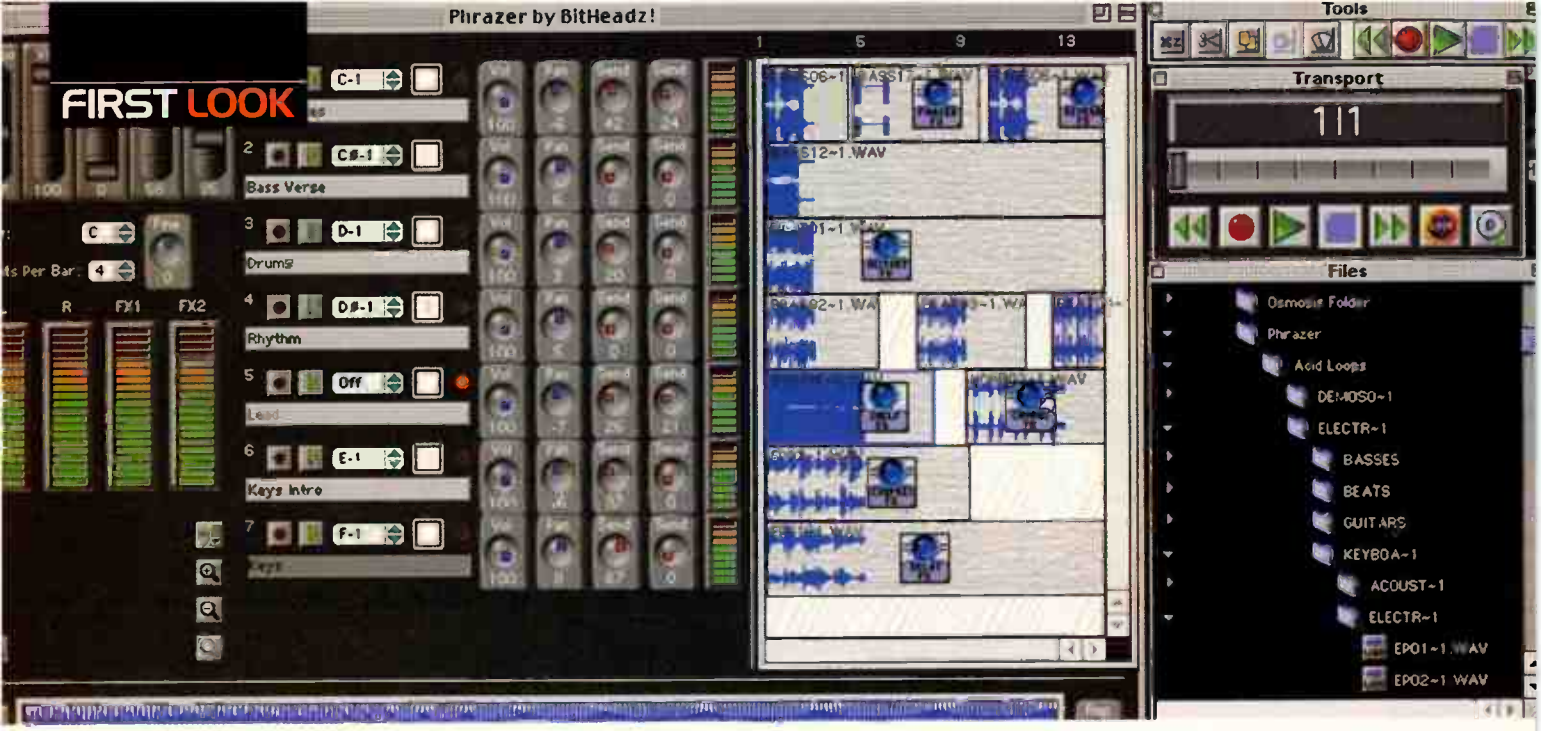


BitHeadz Phrazer

By Steve La Cerra

A recent introduction from BitHeadz is Phrazer, a loop-based composition and arrangement program. Optimized for Apple G4 computers, Phrazer supports ASIO, Direct I/O, DirectConnect, MAS 2.0, ReWire, and Sound Manager drivers. Phrazer is intended to bridge the gap between high-end multitrack audio/sequencing programs and consumer-level looping programs, providing a friendly, yet powerful way to mix audio loops and phrases on-the-fly.

Phrazer lets users import audio files in a variety of formats such as AIF, WAV, SDII, CD-audio, Acid, and UDS (BitHeadz' Unity DS-1 sample format). Files are collected into a Phrazer "session" where they can be manipulated in real-time for tempo changes and/or pitch changes. Tempo may be changed without affecting pitch, and vice-versa. A fine-tune function is available for tweaking the pitch. Several different loops or phrases may be assigned to the same track within the session, and you can specify the number of times a loop will repeat before Phrazer moves to the next loop in the same track. (This is a departure from most looping software, which requires a new track for each loop or phrase.) Each track features mixing controls capable of automated volume and pan changes, as well as DSP-based effects including chorus, flange, reverb, delay, parametric and shelving EQ, four-pole filter, compression, and distortion.

Phrazer's main window displays tracks horizontally, while an optional mix window can display tracks vertically, "mixing console" style. Two global effects are available for use on the L/R mix bus while an unlimited number of insert effects may be used per track. Different effects can be inserted on the same track at different points in the session (or song).

Sessions may be played either from the computer's hard drive — as in most digital audio editing programs — or directly from RAM. The switch between RAM and disk-based playback doesn't require restarting the computer.

Tracks may be triggered from any standard computer keyboard or via MIDI keyboard — enabling you to start and stop loops or phrases from your MIDI rig. Multiple tracks can be triggered simultaneously using different MIDI note numbers. Two modes are available for gating tracks: "instantaneous" mode, where a track will play for as long as the key is held down, and "toggle" mode, where a track will play continuously until the same key (MIDI or QWERTY) is pressed a second time. Separate user preferences are available for both MIDI and QWERTY keyboards.

Phrazer also includes a sample editor for tweaking audio files and zeroing in on loop points. Since support for external MIDI communication is provided, Phrazer can be used with a MIDI keyboard as a stand-alone phrase player, or in conjunction with a MIDI sequencer for recording performances. Phrazer will also be able to import Unity DS-1 samples, and supports the Unity version 2.0 "oscillator stretching" feature. Oscillator stretching allows a sample to be "split" or "sliced." Once a sample has been split, an oscillator speed control allows manipulation its tempo without changing its pitch. Samples that have been split inside Unity DS-1 will retain this same functionality when imported into Phrazer.

Once your loops and phrases are imported, edited, and mixed, Phrazer can render the stereo output of the session as a disk file for export to other digital audio applications. ■

What is it? A program that allows importing of samples, loops, and audio phrases, and facilitates their arrangement into new compositions.

Who needs it? Anyone interested in loop- and phrased-based composition.

Why is it a big deal? Phrazer bridges the gap between high-end digital audio/sequencing software, and consumer-level audio editing software.

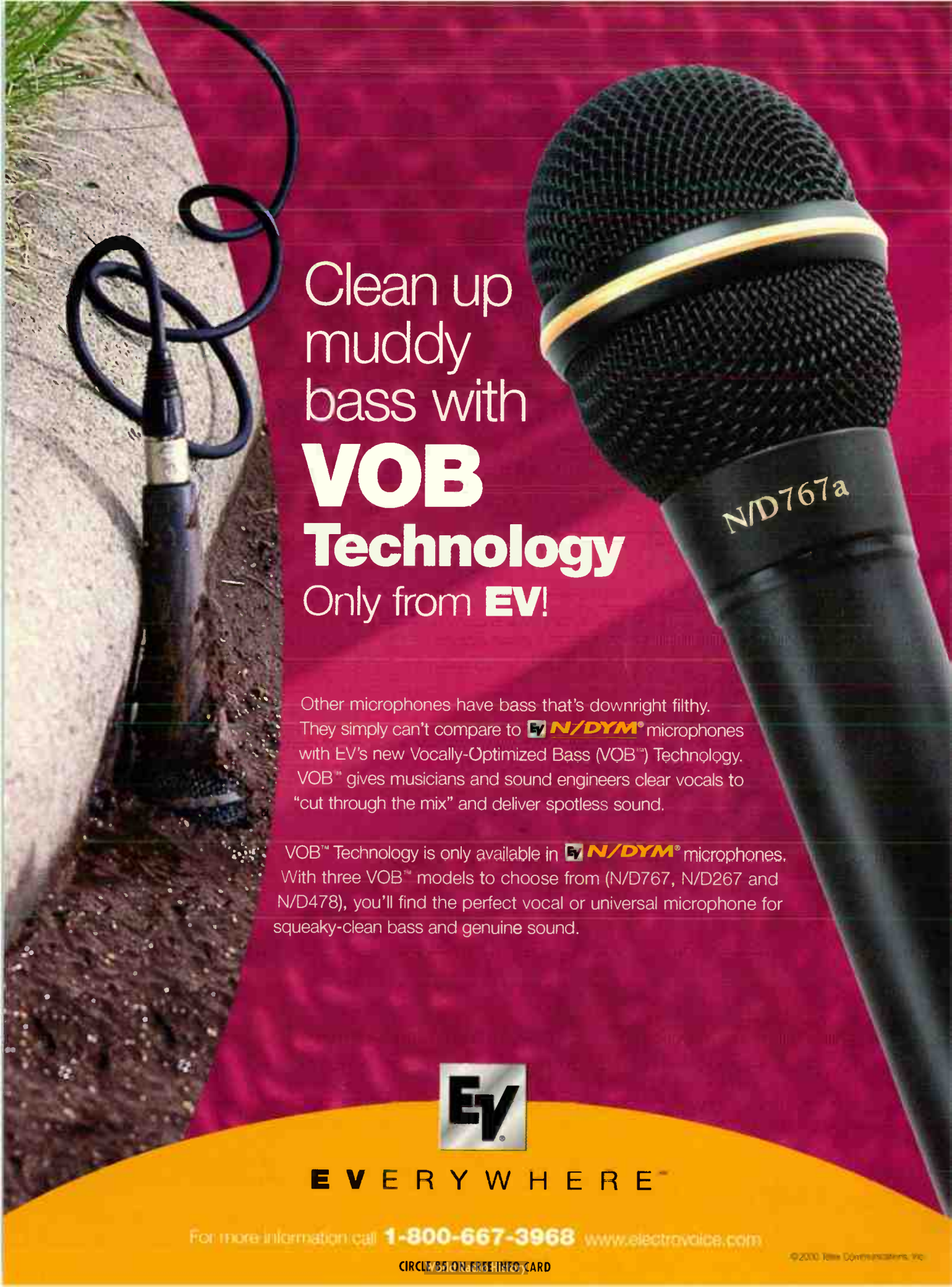
Special notes: Phrazer can import a wide variety of file formats and is capable of importing multiple loops or phrases to the same track.

Shipping: Last quarter of 2000.

Suggested retail price: $399

Minimum hardware requirements: Apple G3/266 MHz computer, 64 MB RAM, OS 8 or higher, and 350 MB of hard disk space.

Contact: For more information contact BitHeadz at 831-465-9898 or visit **www.BitHeadz.com**. *EQ* free lit. #102.



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CIRCLE 85 ON FREE INFO CARD

Paul Umbach

EQ poses six questions to an up-and-coming engineer

There are gigs, and then there are dream gigs. Being the "in-house" guy for the hottest record label on the planet firmly falls into the latter category, as 32-year old Long Island native Paul Umbach will gladly attest. Hooked on recording at the age of 15 (he'd already been playing guitar, drums, and piano for years), Umbach embarked on his career with a single-minded enthusiasm that paid off shortly after he decided to make the move to the Big Apple.

EQ: How did you land your current gig?

Paul Umbach: In December 1998 I phoned a friend named Steve Lunt; we had produced several bands together over the years and he was working in A&R at Jive Records at the time. I told him about my interest in getting engineering work in Manhattan, and he said, "I've got something better for you." They were looking for a programmer to run the writing room for the pop department, and Steve knew that my abilities — both musically and technically — would be a perfect match for the job. Timing truly is everything!

I'm the in-house musician, producer, programmer, engineer, and A&R person, and I work closely with Zomba Publishing as well, re-demoing the catalog as needed. I've worked with Britney Spears, Aaron Carter, Don Philip, Steps, Ellie Campbell, N'Sync, and the Backstreet Boys.

My schedule is fairly steady — I work Monday to Friday from 10-8 unless we're tracking or mixing a master. Having regular work hours has saved my marriage!

How did you get started in engineering?

I'm essentially self-taught. After college, I worked as a staff engineer at Legend Studios on Long Island for three years. I always tried to make my demos sound like records, tried to get the most out of every piece of gear. Once I started to use better equipment, my techniques were simplified and my choices expanded.

Where do you see yourself in five years time?

I want to continue to write, to produce, and to discover talent. However, I'd also like to be playing a bit more golf!

Who are your heroes in engineering and record production?

Kevin Gilbert, Jon Brion, George Martin, Steve Chapin, Nigel Godrich, David Foster, Walter Afanasieff, and Max Martin.

What are your favorite current recordings, and why?

The Shaming of the True by Kevin Gilbert. Completed after his death in 1996 by Nick D'Virgilio and Jon Cuniberti, this is an epic rock opera with artistic sensitivity. As with most of his work, Kevin conceived it, wrote it, engineered and produced it, and performed most of it. It's true brilliance!

Bachelor #2 by Aimee Mann. It's got great songs and great arrangements, awesome musicians, and she's got a great voice. Also, anything from Cheiron in Sweden: Max, Kristian, and Rami have made the most memorable pop tracks in the last few years. Next time you hear a Britney, Backstreet Boys, or N'Sync track that they've done, listen to how well all the layers of sound work together, and how dynamic the mixes are. They're great songwriters as well!

If you were stranded on a desert island and could only take five pieces of studio gear with you, what would they be?

My Mac with Logic Audio, my Mackie HR824 monitors, my Fender Telecaster, a Korg Triton, and an AKG C-12 mic.

WEBLINK

E-mail Paul Umbach at paulumbach@zomba.com

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THE BODY

PHOTO BY HOWARD SHERMAN

Carter Burwell (left) and designer John Storyk.

PHOTO BY DAVE KING

Body of Work

Composer Carter Burwell scores feature films in a NYC loft

STUDIO NAME: The Body
LOCATION: New York City
KEY CREW: Carter Burwell, Dean Parker
CREDITS: Film composer Carter Burwell has scored more than 50 motion pictures including *Blood Simple, Fargo, O Brother Where Art Thou?, Hamlet, Psycho III, Raising Arizona, Miller's Crossing, Waterland, Rob Roy,* and *Gods and Monsters.* Dean Parker is assistant to Carter Burwell.
MIXING CONSOLE: Euphonix System 5; 24-fader frame with 96 digital inputs, 72 digital outputs, 48 analog inputs, and 24 analog outputs
MONITORS: Genelec 1038A, Custom design by Ted Rothstein
AMPLIFIERS: Crown Microtech 1200 and Studio Reference II; Rane MA6S, XTA DP226 digital processor, Chevin Research A500
RECORDERS: TASCAM DA-98 [3] with IF-88AE digital audio interface [3], Panasonic SV3700, SV3800; Pioneer CTW54 cassette deck
OUTBOARD GEAR: Empirical Labs Distressor [2], dbx 166, 120XDS; Aphex Aural Exciter Type C, TC Electronic Finalizer 96K, White Electronics 4675 graphic EQ, Brainstorm Distripalizer, Digidesign 888, Z-Sys 64x64 Digital Router [2]
EFFECTS: Eventide H3000SE, Lexicon PCM70, PCM80, PCM90; TC Electronic TC2290, Synton Synthvox vocoder, Roland VP70 Voice Processor
MICROPHONES: AKG C414 [2], C406 lapel mics [3]; Sennheiser MD421 [2], Shure SM81, Beta 58, Beta 57, VP88 (stereo mic); Beyer TGX480, Neumann TLM170 [2], Audio-Technica AT822
MIC PREAMPS: dbx 760X [2]
SAMPLERS/SYNTHS: Yamaha TG77, Korg Wavestation, Roland JD990, S760 [6], JV1080 [2], D50, S770, SP700 sample player; Sequential Circuits Prophet VS
COMPUTERS: Apple PowerBook G3, PowerMac G4, MOTU MTP AV [3], MTP II [2]
SOFTWARE: Digidesign Pro Tools, Coda Finale, MOTU Digital Performer
SYNCHRONIZATION: MOTU DTP, Digidesign SSD (SMPTE Slave Driver)
VIDEO EQUIPMENT: Pioneer Plasma Video Display, JVC BR-5622DXU SVHS, Panasonic SVHS
POWER CONDITIONING EQUIPMENT: Furman: PL+ [2] and AR-Pro
STUDIO NOTES: Carter Burwell had been looking for a space in the Tribeca area of NYC "for a long time but I like to have a lot of light and views and that's not easy to find in this area. I've lived here for several years and tried for the first time to separate life and work, keeping my studio in Hell's Kitchen where it's been until now. Separating life and work doesn't work for me because I have no concept of where to draw the line. I'd rather have the option to work at any time, and also to walk out of the studio and be home. Due to the particular shape of this space, the only area that would accommodate the studio dimensions is where the bedrooms and bathrooms were located. Since the studio would have to replace these rooms, I knew we were talking about a complete renovation of the entire space. The only logical thing to do was to bite the bullet. It was a huge project. I had an architect for the residential part (Cathy Chia) and John Storyk (Walters-Storyk Design Group) was my first choice for designing the studio. Cathy developed designs for the residence, and we started working months before I spoke to John — I wanted to have a concept for the look and feel of the living space first. I kept the studio as a sort of 'black box' in the drawings because we didn't know what was going to go there. We left it alone until Cathy and I came up with ideas for the residence — it's a minimal

THE BODY

design with a few beautiful materials, taking advantage of all the light in the space and not cluttering it with a lot of architectural details. Keep it simple — which of course, is much more complicated than it seems!"

"Once we had those ideas solidified, I brought John in to design the studio. He came to my old studio to get a sense of the way I work. Then John and Cathy started working together, discussing how we'd divide up the responsibilities. John had ultimate responsibility for all the acoustical aspects of the space and how it would be utilized — the layout of the room and the location of the gear. Cathy had the responsibility of making sure that the look of it — the surfaces really — would match the rest of the residence. Ted Rothstein handled the audio wiring design."

"One of the things I loved about this space in the first place is that we have beautiful views of New York. Having lived in so many places in New York over the years, I ultimately resolved it necessary for me to have windows in the studio. I need to have light and to be able to look out the window to see things. I spend most of my life in there, and I'm not willing to spend all that time in a dark room. For months John and I went back and forth with drawings. He'd send me a plan where the windows would be three feet high. I'd send it back saying 'no they have to go to the ceiling.' Then he'd send another drawing where the windows were four feet high. I'd say 'no they have to go all the way to the ceiling' and he'd send back a drawing where they were five feet high! I thought that in the worst case — if the windows were doing something terrible to the sound — we could put curtains over them to pull during critical listening sessions. John said 'No. we're making it perfect.' and somehow he did it!"

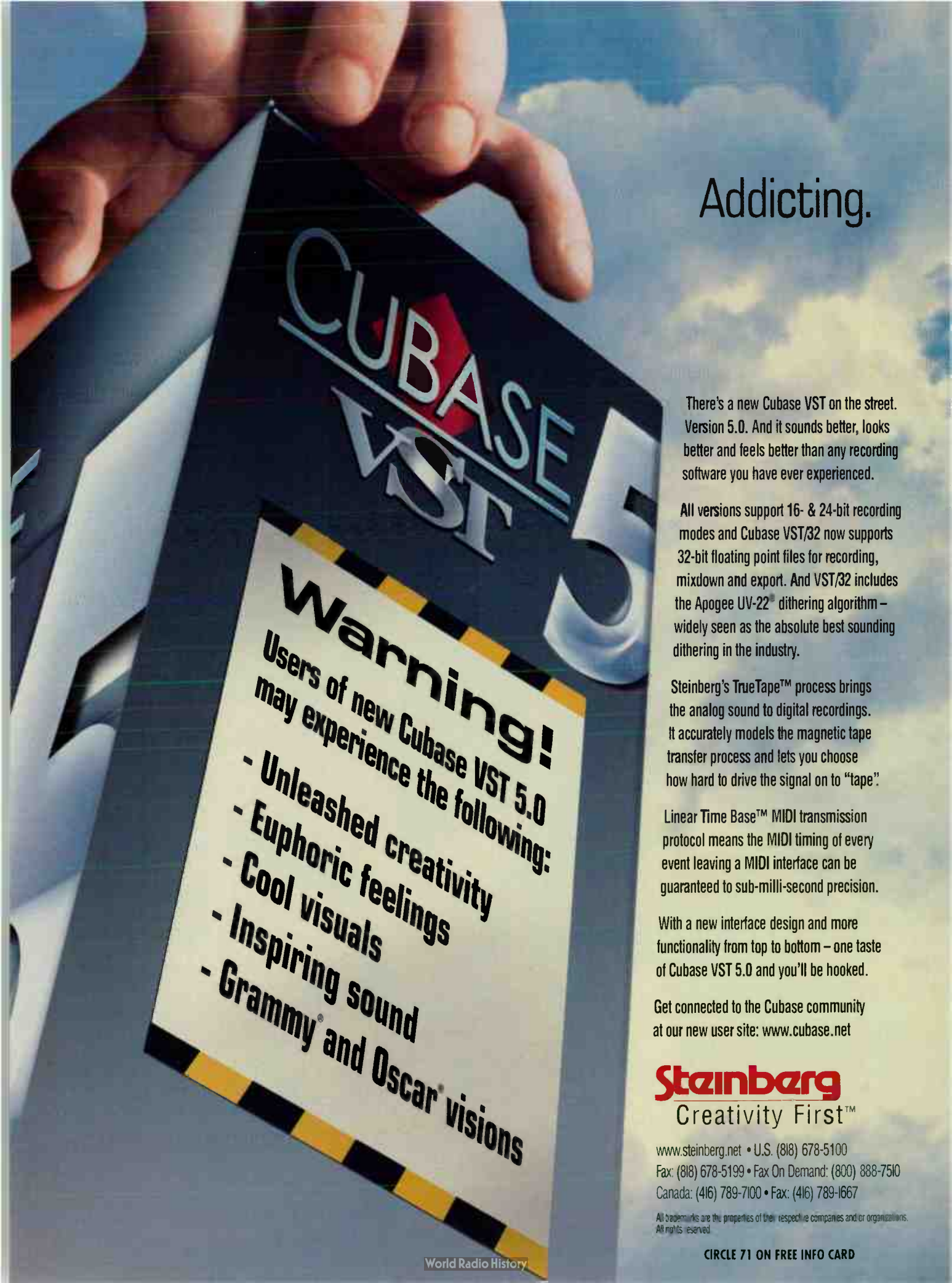
FAVORITE EQUIPMENT: "There's several pieces of new equipment in there which are fabulously enjoyable," continues Carter. "I know it sounds silly but the Pioneer Plasma display really looks wonderful. It's so bright that even having all these windows and daylight has no effect at all on the picture quality. My Apple G4 has a Cinema display, which is the largest LCD made for a computer. By now I've gotten used to it, but of course when I use any other computer I feel like the screen is too small (and the machine is too slow!). I use the computer for essentially everything in my life, so having such a fast machine affects me all the time. I run Digital Performer for composing, then transfer my files into Finale to prepare my scores. I use Pro Tools for digital editing and premastering."

JOHN STORYK ON DESIGNING "THE BODY"

"The basic design, entrance and location of the iso booth [in the studio] was fairly obvious. The room worked better as a wider room rather than a deeper room. The toughest issue really was getting all that equipment into the room. Carter has a lot of gear and he needs all of it. Once the gear was configured and the orientation of the room was established, center lines were created. Initially I considered placing the windows in a perfectly symmetrical fashion, but doing that would have meant sealing off one of the windows on the south side or the room's right side. We decided to slightly abandon the perfectly symmetrical approach so as not to seal off that window. The vocal booth kind of had to be where it is, so that window fell into place. In theory one would mirror the vocal booth window on the opposite side of the room, creating an inner window and then disregarding the outside windows. That would probably have been okay, but we wanted maximum natural light.

"In general, the overall symmetry of a room — particularly for a 5.1 room — is very important. But there are moments when you can abandon it. A more interesting issue is the big rack in the rear right, which is *not* duplicated on the rear left (or you wouldn't be able to get into the room). We had no choice but to put that rack there, and it did frighten everyone a bit. There was no other place to put the gear. I was worried about the possibility of reflection off that rack but it turned out that when we did the tests on the room you don't really see it much [sonically].

"Typically you enter a control room on the side, slightly to the rear of the room. I generally try to get *something* to happen exactly on the opposite side of the room that involves a door. Sometimes there's a vocal booth, sometimes there's another way in and out of the room (which is always nice). Maybe there's a machine room...if not, I'll create a reason for the door...maybe add a microphone closet. For me the real reason is so that there are two doors in perfectly symmetrical locations. There are some important issues regarding low-frequency absorption: if you're going to have that surface on one side it should be replicated on the other side, so that the low-frequency absorption is consistent on each side of the room. There was no way to do it in Carter's room because we were dead-ended into the corner of this small room. That scared me a little bit at first, but the impulse responses of the room look very good. Even with 5.1 the right-rear speaker didn't pick up a reflection from that rack. In fact the response from the rear speakers is almost identical to that of the front speakers."



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ROOM WITH A VU

COWBOY TECHNICAL SERVICES RECORDING RIG

PHOTO BY C.S. GRAY

Cowboys in Brooklyn

A practice space becomes a studio

STUDIO NAME: Cowboy Technical Services Recording Rig
LOCATION: Williamsburg/Brooklyn, NY
KEY CREW: Eric "Roscoe" Ambel, Dan Pifer, Tim Hatfield
CREDITS: Eric "Roscoe" Ambel has played guitar with Joan Jett and the Blackhearts, Del-Lords, Roscoe's Gang, and Steve Earle and the Dukes. As a producer he's worked with Bottle Rockets, Nils Lofgren, Blue Mountain, Go To Blazes, and Cheri Knight. Dan Pifer is assistant engineer, plays bass with Mary McBride's Pancake Party Boys, and was formerly with God Street Wine. Tim Hatfield has worked with Steve Jordan, Waddy Wachtel, Keith Richards, and Godfrey Diamond.
CONSOLE: Neotek Elan 32-input, 24-bus with patchbay
MONITORS: Hafler TRM 8, TRM 12 subwoofer; JBL EON 10, Furman HDS-6 cue system, HR-6 cue boxes [6]; Audio-Technica ATH D40fs [8], Fostex, and AKG headphones
RECORDERS: Otari MX-80 two-inch analog 24-track, DTR-8S DAT recorder; Studer B-67 1/4-inch analog two-track, Roland VS1680, Teac CD-RW 700 CD writer, W415 cassette deck
OUTBOARD GEAR: Neve 1067 mic preamp/EQ [4], Demeter VTCL-2 stereo tube compressor, Avalon 737SP mic preamp/compressor/EQ [2], Manley Variable MU Stereo Compressor, Dual Electro-Optical Leveling Amplifier [2]; UREI LA-4 opto compressor [2], dbx 900 rack with 903 compressors [3] and 902 de-esser [1], Altec 436c tube compressor, Universal Audio 1176 limiters [2]
EFFECTS: Lexicon PCM41, PCM60; Eventide H3000, Instant Flanger; Behringer Dual Intelligate, Composer; Roland 501 Space Echo, TC Electronic Finalizer Express, Alesis Quadraverb, Korg Dual Delay
MICS: Manley Reference Cardioid, Audio-Technica AT4060, AT4050 [2], AT4033, AT4041 [2], AT3525 [2], AT3528 [2], ATM25, ATM23 [4], AT4051, AT4054, AT4055 [2]; Earthworks SR-77 [2], Beyerdynamic M500, M88; Shure Beta 87
MIC PREAMPS: Earthworks, Demeter VTMP-2a, Telefunken V-72 [4]
POWER CONDITIONING EQUIPMENT: Furman IT1220 Balanced Isolation Transformer
WIRING: Mogami, Canare, Monster Cable
INSTRUMENTS: Hammond A-100 organ, Story & Clark spinet piano, 1969 Premier drum kit, 1970's Ludwig metal snare drum, Pearl piccolo snare
GUITAR AMPLIFIERS: Supro amps [4], Fender amps [8], cabinets [4]; Marshall amps [2], cabinets [2]; Dean Markley cabinet, Silvertone, Sound City amp, Demeter SSC-1 Silent Speaker Cabinet
STUDIO NOTES: According to Eric "Roscoe" Ambel, "The Cowboy Technical Services Recording Rig is currently parked at a practice spot in Williamsburg, Brooklyn. It's a two-inch, analog recording system set up 'no control room' style. Various stations such as guitar, keys, bass, drums, and vocals are set up and ready to go. The basement room (23x27 feet) opens out into a common area with a bar, couches, and a ping-pong table."

"Construction of the room was done by Rich Ross. Originally we moved our equipment in here to store it while we were working on the wiring. But then I just fired it up and did some recording and mixing with people I know. It's a very comfortable room. Knowing the instruments, mics, and preamps makes it really easy to focus on the music in there. I've had quite a bit of experience with the 'no control room' thing in the last few years. Hypercardioid mics, small amps, and the Demeter Silent Speaker Chamber (for a tube bass amp sound with no bass leakage either way) make it work."

FAVORITE GEAR: "The first thing I got was the Manley Dual Electro-Optical compressor, and it all took off from there. I like to mix through the Manley. Having the Roland VS1680 and a snake that brings it up on the patch bay gives me the opportunity to use tracks that start at home or mobile. The snake is also great for getting guitar pedals in and out of the board."

RCA "LBJ"

PHOTOS BY EDWARD COLVER

RCA "LBJ"

The mic that fooled an American President

MICROPHONE NAME: RCA LBJ Microphone
FROM THE COLLECTION OF: Stephen Sank, Champlain Valley Speaker Company, Albuquerque, NM
YEAR OF MANUFACTURE: circa mid-1960s
TYPE OF MIC: Dynamic
FREQUENCY RESPONSE: 80 Hz to 12,000 Hz
DIRECTIONAL CHARACTERISTICS: Non-directional
EFFECTIVE OUTPUT LEVEL: -67 dBm, referenced to a sound pressure of 10 dynes per square centimeter (@ 1 kHz)
HUM PICKUP LEVEL: -112 dBm, referred to a hum field of 1×10^{3}
OUTPUT IMPEDANCE: 30, 150, or 250 Ohms
EXTERIOR FINISH: "TV gray"
MIC NOTES: Here's a bit of MicroPhile trivia: what microphone looks like an RCA BK-1A, but has two XLR plugs at the end of its cable? The answer is a mic that very few people ever knew about: the RCA MI-11025. Also known as the "LBJ Microphone," this specially designed microphone actually contains two complete RCA BK-6B (MI-11017A) miniature dynamic lavalier mics in a housing that appears very similar to the company's BK-1A microphone. The LBJ Microphone was designed by Jon R. Sank for President Lyndon B. Johnson's 1964 re-election campaign.

The story goes that Johnson had a very strong dislike of the clutter resulting from multiple mics on his podium. His dislike was in conflict with the technical need for a backup microphone during public addresses, just in case one mic failed. So without his knowledge, Johnson's staff commissioned RCA to design and build a dual mic that looked completely like the BK-1A that LBJ was accustomed to using — all the way down to the cable.

To accomplish this feat, a pair of BK-6B mics were set into a very precisely-machined piece of nylon-type plastic, which firmly held the mics and provided screw threads for the top windscreen cover of the outer housing (approximate dimensions of the BK-6B are 2.6 inches long x 1 inch diameter). This top screen and the main body casing were in fact specially designed for this mic, as the BK-1A parts weren't quite suitable. Even the cable was specially designed as a pair of balanced mic cables in a single round jacket — completing the facade of a single mic (perhaps this was the first application of "multi-pair" cable).

Please note that the specifications listed above pertain to the original RCA BK-6B lavalier microphone.

Historical data furnished through the courtesy of Stephen Sank (stephen_sank@rjknet.net).

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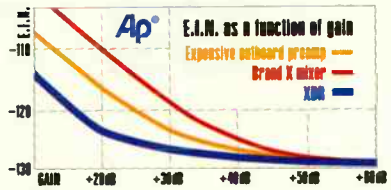

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"Great product. XDR's are a great innovation and the value's unbeatable." J.K. Boise ID

"I hooked up my brand spankin' new AKG 3000 and all I can say is 'cool'." T.D., Waukesha WI

"I've been though numerous small consoles that were noisy. This one isn't, so it's a keeper!" J.F., Boca Raton FL

"Gorgeous preamps. Nice job, guys." J.C., Provo UT

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"Excellent frequency response... excellent mic preamp. Worthy of our Neumann mics." C.M., Atascadero CA

"Killer mic preamps!" R.A., New York NY

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Spinal Tap is
"Back From The Dead"

Spinal Tap

Suspects: Derek Smalls and David St. Hubbins (Nigel Tufnel was not present due to a severely sprained pinkie.)

Background: *Newsweek* calls *This Is Spinal Tap* (Rob Reiner's first film) the "funniest movie ever made about rock and roll." Coinciding with the re-release of the 1984 film in theaters and a new DVD version released in September, the band has recorded a new single, "Back From The Dead." Suspects were questioned during sessions at Henson Recording and Mastering Studios (formerly A&M).

Notes: Producer Jeffery C.J. Vanston confided, "I received a lot of criticism for taking this gig and disturbing the band's slumber — kinda like when those explorers wanted to raise the Titanic. Well, I thought it was time for Spinal Tap to resurface and take on the iceberg again! My aim from the beginning was to resuscitate their dated sound while also paying homage to whatever is on the charts today."

MR. BONZAI: Have either of you ever had a spinal tap?

Derek Smalls: Never had a spinal tap myself, but "memorable and painful" seemed to sum up what we wanted to be as a band.

David St. Hubbins: There's a sexual maneuver called a "spinal tap" and I procured one of those in Singapore. It burned.

Will your kids continue the Tap legacy?

Derek: I don't have any children, so it would be hard for them.

David: My only known offspring played for a while in a punk band, gave it up for a bad idea...he sells T-shirts now in Venice, California. And he's doing very well. So don't hold your breath.

What about the Internet? Can you protect your huge success from the big rip-off?

David: Not a chance. That's why we're surrendering most of our creative rights in a blatant attempt to become better known.

Derek: On Tapster.com, we're offering free downloads of "Back From The Dead." We're ripping ourselves off, and that's really the artist's best escape — that's the answer to Metallica: Rip yourself off. Then you've got nobody to complain about.

How does it feel playing this dark metal music with one foot in the grave?

David: I can think of no more appropriate place to keep one's foot.

Derek: What do you mean by that?

Well, uh, you're not exactly kids anymore...

Derek: But that doesn't mean you have one foot in the grave. To me, you've got a foot in the grave if they've got you hooked up to tubes and they put a catheter up your willie. None of us are in that situation, so we still feel that if you don't look at the calendar, and you don't look at the clock, and you don't look in the mirror, then it feels the same.

What are your main recording tools?

David: MIDI, ASCII, finger sandwiches, and Napoleon brandy.

Derek: Pro Tools — at least a couple of the Pro Tools, a good long cord, and you need a mic, and a nice long power cord as well. I don't mean a "power chord," like what we play, but a splitter, AC. That's all I need.

Do you have any special effects processors?

Derek: To me, all the special effects really belong to the land of guitars. I want the bass to be clean, and loud, and strong, and dominant. The more effects you put it through, the more you get in the way of that. Bass is basically like breath and the guitars are like speech. So, you want special effects on the speech, but on the breath you just want it clean and pure, and straight.

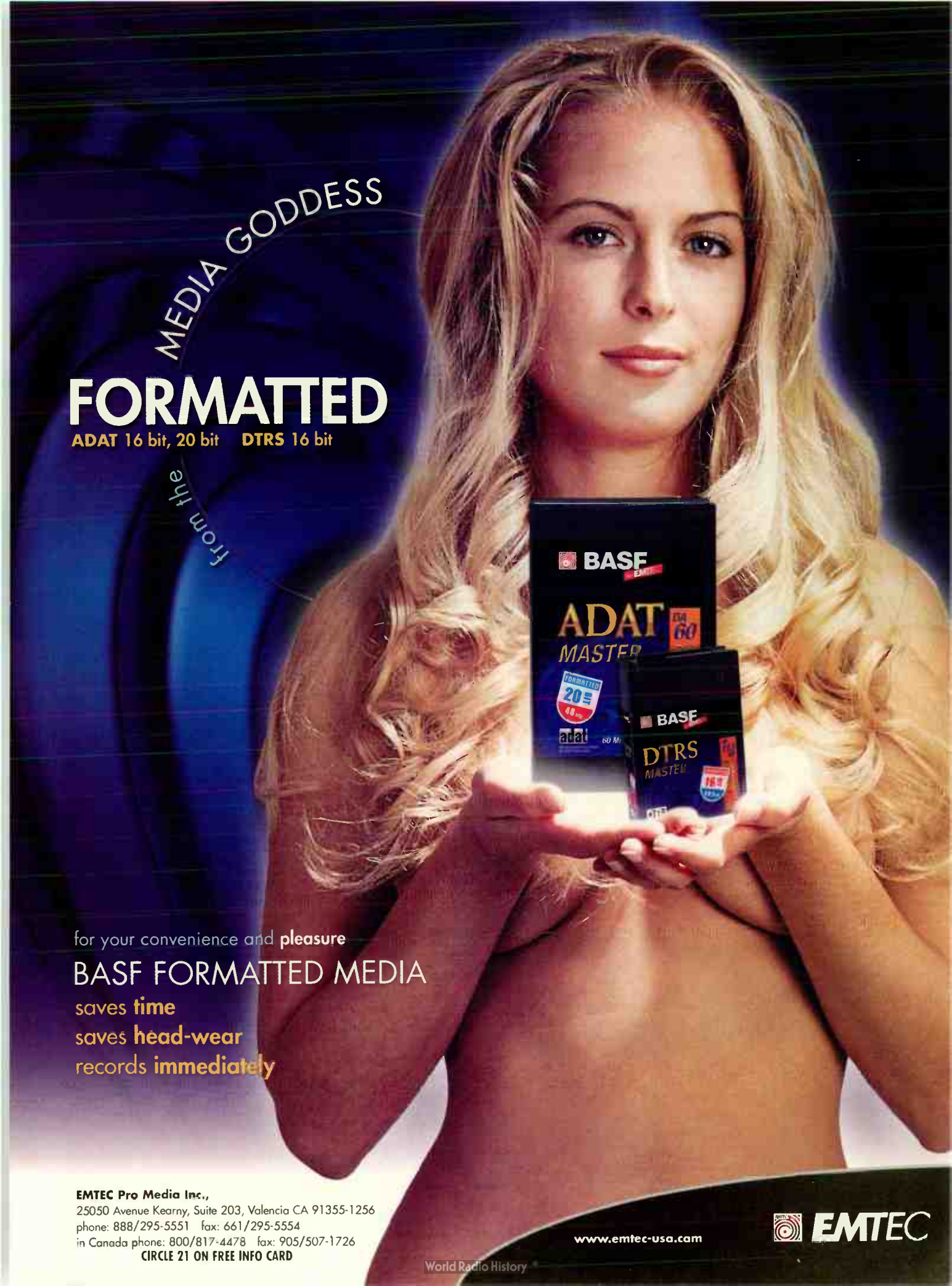
David: I use the Kaori X30, an experimental compressor that actually increases your guitar's atomic weight.

Could you compare the recording process today with that of yesteryear?

Derek: Much quicker, much more focused, we're able to get much louder without being louder — I don't know how they do that. Much better take out food and much better parking, really, to be truthful about it. Always had trouble finding a place for the Lamborghini in the old days.

David: More snacks available in the studio

SPINAL TAP PHOTO BY MR. BONZAI



these days. And the engineers are younger. And crankier.

Did you ever lose anything?

David: No, it's all here.

Derek: My virginity, my way to the stage — in that hatchet job of a movie. Of course, we have found our way to the stage a lot, but you don't see that, do you? Some money, thanks to Mr. Ian Faith — lost that. Car keys on about 2,000 occasions.

What's wrong with the Music Industry?

David: Two things: the Music and the Industry.

Derek: It's an Industry — you just said it. Shouldn't be an Industry. Should be the Music Colossus, or the Music Cooperative. You can't have a "Music Industry" — what do you call that? It's a moron.

What music would you like played at your funerals?

David: Something of mine, so my estate will become more valuable. My mum didn't raise no fools...

Derek: "Back From The Dead."

If you could go back in time before the birth of recording, what would you like to hear?

David: Al Jolson begging for work.

Derek: I'd like to hear what it sounded like in Beethoven's head after he went deaf.

What did you learn from Ozzy Osbourne?

David: The location of the secret restroom at the Rainbow.

Derek: Don't bite the heads off chickens — they'll come home to roost.

Who were your musical heroes when you were getting started?

Derek: Bix Beiderbecke, Stanley Holloway — English entertainer, great musical artist, and Sir Thomas Beechum, just for his

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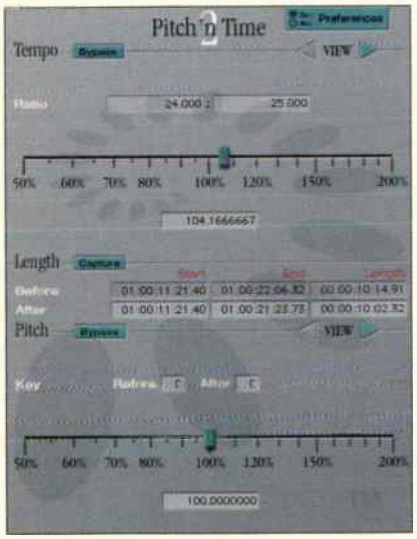
Pitch 'n Time, originally released in December 1999 by Serato Audio Research, has already been described as the best pitch and tempo control tool on the market today. It has received rave reviews from a number of pro audio magazines and was recently nominated for the 2000 Mix Technical Excellence & Creativity award.



With the release of version 2.0, we at Imaginary Gadgets have built on that tradition of excellence to deliver a product which incorporates exactly what you have been wishing for.

Introduced in this release is an impressive array of new features including multi-channel mode, waveform overviews, tempo mapping, time morphing and pitch mapping which together make Pitch 'n Time 2.0 an essential tool for your collection.

New in Version 2.0

Multi-Channel Mode allows you to process up to 48 tracks together while maintaining their original phase coherency.

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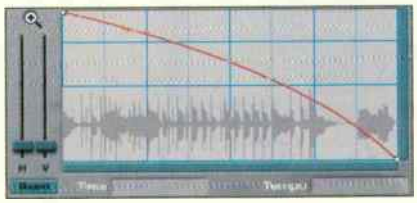
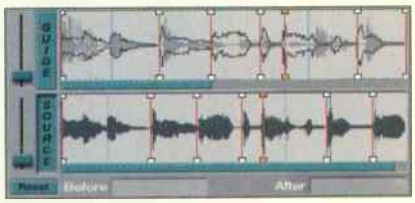


Tempo Mapping gives you the power to create variable tempo ramps and changes over the length of your sample.

Pitch Mapping allows you to create amazing pitch-bends and record-scratching effects by drawing your pitch change right on top of the waveform.



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sense of domination.

David: George Formby and the guy who sang "Sukiyaki."

Who do you respect and admire today?

David: Those artists who never give up no matter how many times people suggest they do.

Derek: Björk, just for having an unpronounceable name and making everybody try very hard to pronounce it. I think that's just a great act of manipulation. And I like some of the boy bands. I think we in Spinal Tap have gone about as far in looking dangerous and threatening as you possibly can — to see people looking as safe as you possible can, is an interesting other way to go.

What was your most frightening experience in a recording studio?

Derek: I guess it would be on the original "Sun Never Sweats" session, when I came in one day and they said, "Oh, we've wiped all the bass parts off. Sorry." And then I realized they had just muted them. I was new in

▶ continued on page 152

JEFFERY C.J. VANSTON ON PRODUCING TAP

The "Back From The Dead" sessions with Spinal Tap were nostalgic for all involved (except the second engineers, who had never heard of the band). We started the project at a secret rehearsal hall on the west side of L.A. (it's the only building on the West Coast where Tap can rehearse at "show" volume).

The recording sessions took place at Henson Recording and Mastering Studios (formerly A&M), studio D. Drummer Skippy Scuffleton (AKA Gregg Bissonette) bravely joined in on drums, safely tucked away in his own booth away from the band. Engineer Bill Drescher had the unenviable job of harnessing all the relentless sound. We had to use [Sennheiser] 421's for everything — the only mikes able to take the sheer pounding.

Tap has always recorded strictly on analog tape (at +11 of course), but we were very conscious of going for a modern sound this time. We recorded directly into Pro Tools first, to get that digital "sound," and then transferred to analog — you see? There's that clever "twist" that Tap is famous for. Next, we transferred the vocals back to Pro Tools for "plug-in day." We really went wild. It's amazing how the time flies when you're able to do fun things with plug-ins like manipulate pre-reflection reflections, tune tom-toms, remove natural room sound, things like that. We tried about 34 different plug-ins on David's lead vocal, but stopped the minute we heard the beta version of Outa-tune 1.0143. We all looked at each other in awe — he sounded just like he did 20 years ago!

Derek was interesting. In some ways, he was the most helpful member of the band, organizing food orders and keeping an eye on the air conditioning levels. When he sang background, I saw a beast come alive as he unharnessed his emotional "Yeah!" and "No!" shouts in the chorus, one time thrusting his fist right into the mike while attempting to rock-pose on the last note. That's what the studio gets for trusting their 251 to the loudest band in the world.

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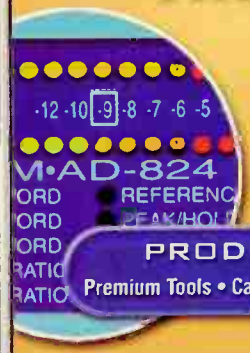

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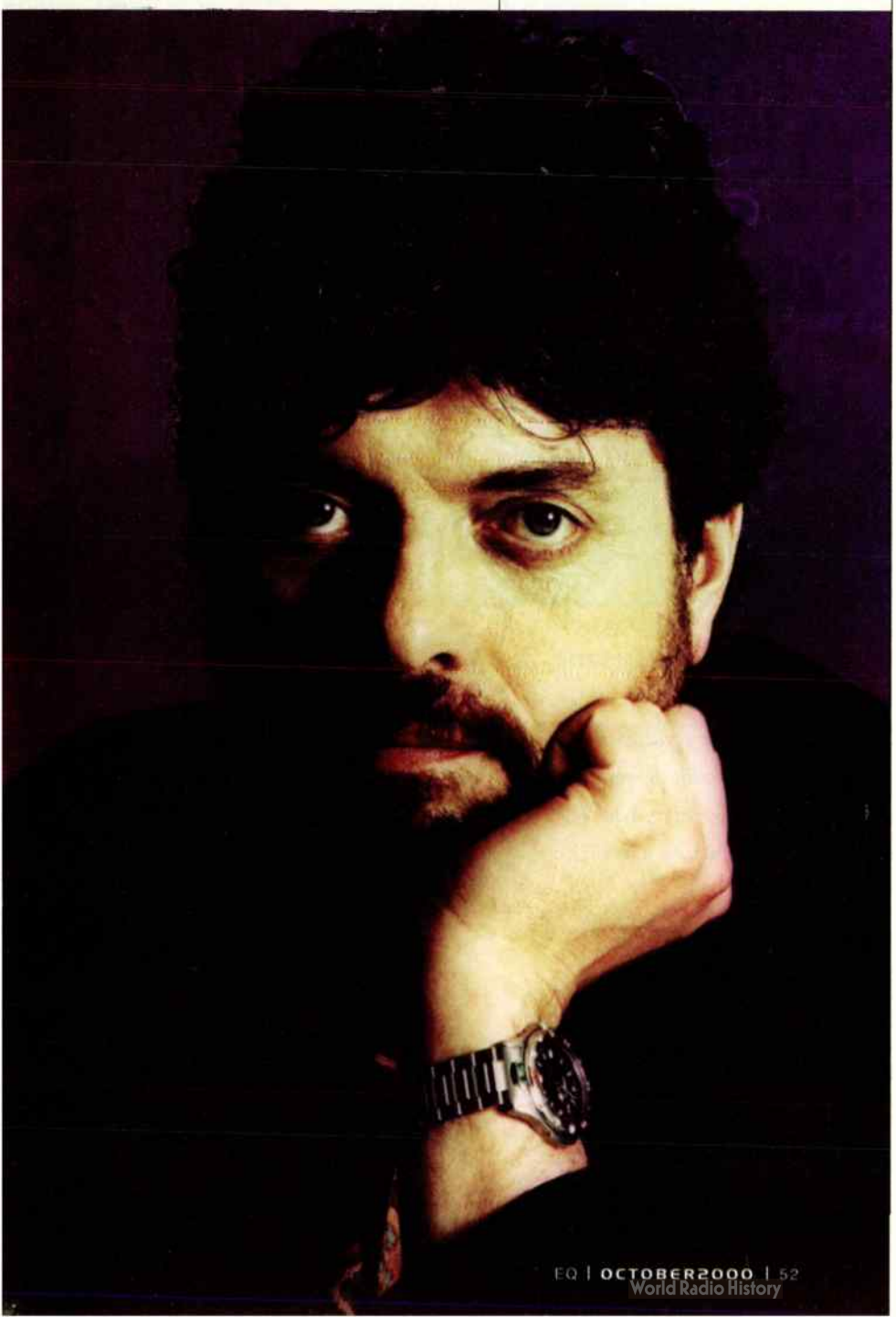
Dark Side Of The Moon, synchronized tape delays, and mixing without drums or bass

Alan Parsons

By Howard Massey

There are hit records, and then there are hit records. Pink Floyd's 1973 Dark Side Of The Moon *was unquestionably the high-water mark in the band's career, both in terms of creative and commercial achievement. Occupying a staggering 14 years on the* Billboard *charts (an all-time record) and ultimately selling over 25 million copies, the success* Dark Side *enjoyed has been unparalleled in the history of recorded music. Though he has enjoyed a rich career in the years since, Alan Parsons will forever be known as the man behind the board for this incredible accomplishment.*

Brought up through the ranks at London's famed Abbey Road studios, Parsons assisted for the likes of Geoff Emerick, Ken Scott, and Glyn Johns in the late '60s and early '70s — learning at the feet of the masters, as it were. By the mid-'70s, he made the transition to producer, overseeing hit records by Al Stewart, Pilot, Steve Harley, and John Miles, as well as striking out on his own with the Alan Parsons Project, a group of musicians assembled to render Parsons' own musical vision.



EQ: Was more care taken with *Dark Side Of The Moon* than other recordings you've been involved with?

Alan Parsons: Not really. At the beginning, when we were getting drum sounds for the first track, it became quite unpleasant, with Nick [Mason] saying, "No, it's not there yet; I haven't heard my drums sounding like drums yet." We would agonize through that, and then finally we would say, "Yes, we think it's okay now; let's do the track." Then, after that first period, they would just leave it to me; there would be no question of the drums not sounding good, because they must have just started trusting me. Somehow everybody relaxed and got into it and started working as a team.

One of the hallmarks of that album was synchronized echoes, but those were the days before digital delays. How were you able to achieve synchronization so successfully?

Good question. It was quite extravagant to put a varispeed on a [tape] delay machine, but if it was considered important — if a delay needed to be in time — we would do it. But a lot of the guitar solo echoes were actually generated by Dave [Gilmour]'s own gear, on a Binson Echorec.

Were any of the basic tracks

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recorded to a click track?

They recorded to a [standard] metronome ticking away in another room. I daresay that electronic metronomes existed, but we didn't have one. But on "Money," for instance, the tape loop became the click track.

It's probably the first instance of a loop being used as the rhythm track.

Probably. We started off trying to edit that loop sonically and found that there was just no way of getting it in time; the only way to do it was by physically measuring the length of one of the sounds and then duplicating that exact length — otherwise it just wouldn't work. It took quite a bit of doing, and it was made more difficult by the fact that it was a four-track loop on one-inch tape, because we wanted it to work in quadraphonic as well as in stereo.

Was *Dark Side Of The Moon* recorded in 16-track or 24?

16-track.

Were there any reductions?

There were. In fact, it was nearly all second-generation — non-Dolby on the first generation, Dolby on the second generation. And if they had their way, the first generation would have been Dolby as well; I was actually the one who was anti-Dolby. They kind of insisted, and it probably was sensible to Dolby on the second generation, but we got rid of so much noise with Kepex noise gates, which were the new thing at the time. Bass and drums were reduced down to two tracks a lot of the time. I'm still to this day praying that I get the opportunity to remix the album one day, so that we can sync up [and work off] the original multitrack tapes.

How much time was spent mixing *Dark Side Of The Moon*?

Between two and three weeks. We usually got a track a day done. Sometimes a track would go to two days, but it was generally pretty quick.

Do you have one particular mic that you use for vocals most of the time?

No. [*Chuckles.*] Since the Audio-Technica 4033 came out, I favor that. But I'm happy with any good mic. [Neumann U] 47s, I've had a lot of luck with, but they're unreliable; FET 47s are usually fine. [Neumann U] 87s are rather boring as a choice; I've tended to avoid 87s as a vocal mic, just because it's what everybody else was using, though I've often got good results with them.

The vocals on your records have good presence, but they're never even close to sibilant.

That's probably because I favor adding top during recording and de-essing afterwards. I have almost never recorded a vocal flat. I like to hear the sound at the back of the throat, and if it's at the expense of increased esses, then I'll take them off later. I boost 10k shelving, as a general rule. But I don't like hard top on vocals; I always like the sort of breathy, airy top, so sometimes I might start at 12, even 14k, and do more of it. With some voices, you can literally go as much as 6 [dB] at 14 [kHz], and it won't be at the expense of esses.

Do you ever roll out mids to create the smile curve?

Occasionally, but I'm more likely to do that on strings and electric guitars, where you're likely to get that hard, clangy top end. Especially if you're close-miking strings, you tend to get undesirable 2, 3, 4k nastiness. You just put a bit of airy top on and take a little 1k, 2k out — that usually sorts it out.

"You can succeed through breaking the rules and doing nothing else"

You always use reverbs and delays in a very creative way. How are they set up?

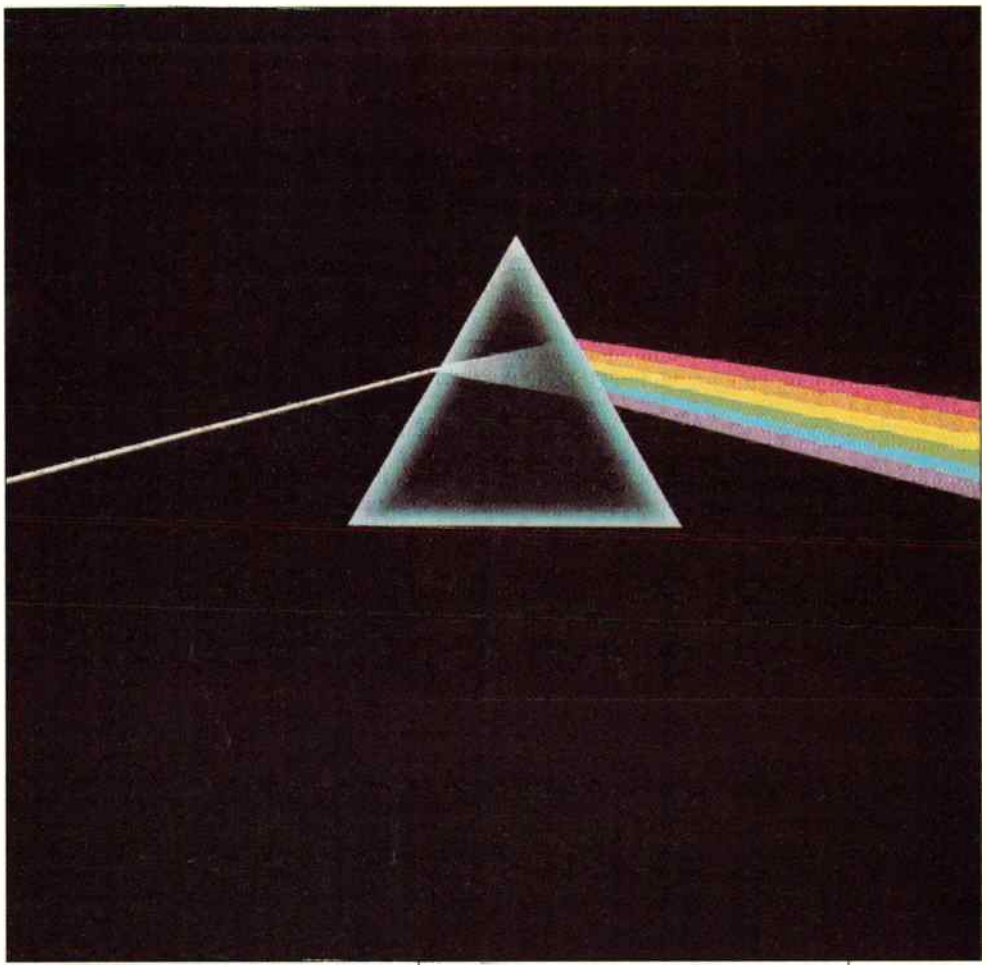
There's always delay on the send, and the send is nearly always mono. I think the predelay is the most important component of any reverb — it adds character — and it's the first button I reach for when I put on a reverb.

Do you typically EQ the sends or the returns?

I'd much rather EQ the sends than the returns. I'll roll lots of bottom out; occasionally I'll roll off top end as well. I'm quite happy for a very middly signal to be going into the reverb.

Do you use tape machines for the delays?

I don't possess any analog tape machines any more! [*Laughs.*] For the last several albums, I've been working with digital machines. And digital delay is fine; you can achieve most of the artifacts of tape if you want, just by rolling off a little bit of the top end on the send, it'll come



back a little duller with each repeat. You can overdrive it, you can distort it.

Are you big on compression?

I hate compression with a vengeance. I avoid it. I'm a great believer in the dynamic range being preserved. I'll limit rather than compress a vocal in order to get it at the best dynamic range on tape, but I'll still be riding it [manually] on the mix. I never use compression on drums; I just hate that sucking sound.

Unless it was an effect, I would never want to hear a limiter or a compressor working. That said, there are classic examples of great compressed sounds on vocals; "Lady Madonna" is one of them. It's a great vocal sound, but it is a special effect.

When you record bass, do you take a DI signal or amp signal or both?

Always DI; I can't be bothered with an amplifier. They're noisy and they crackle and they hiss and they often don't represent the sound of what the bass player wants because they're turned up too loud. At least with the DI, you're hearing the instrument. You can emulate the sound of just about any amplifier with enough processing inside the control room. I haven't miked a bass amp for years.

Do you have a favorite DI box for bass?

Not particularly. I always use passive DIs, not active ones. I remember doing an evaluation of DI boxes, and there were only subtle differences between the different models.

When you record electric guitar, do you use the standard miking technique of a 57 up against the grill?

[*Laughs.*] No, never. Every engineer I've ever come across — especially live engineers — has always had the mic touching the cloth, and the first thing I do is move it away literally a foot. Let's hear what the *amplifier* sounds like, not what the cabinet sounds like! I've always thought that most people mic guitar amps too closely. They supposedly make it up with an ambient mic, but I much prefer to find a mic position that works and process that, rather than mix in too much ambience.

What mic do you favor for electric guitars?

I like the Neumann 86 for guitars. I always use condenser mics on a guitar amp, never dynamics — they're too telephoney. That's an unfair expression for a very good dynamic mic. but by comparison to a condenser, you're not going to get the bottom end.

So you use a small-diaphragm condenser mic about a foot away from the amp?

Yeah. I might have it even further away if it's a very loud 4x12 cabinet — as much as four feet away. I think people can be over-worried about the separation aspects of a guitar on a live tracking date. It's much more likely that the guitar is going to go elsewhere rather than anything else getting into the guitar mic. You can be four feet away from a guitar with a full band playing, and you're not going to hear very much of what else is going on; it's going to be the loudest thing in the room.

Are you a leakage fan or is that something you fight?

I don't worry about it. I use it if it's there; if it's a problem, I'll try and get some volumes adjusted. The worst case scenario is live drums and piano — that's always the nightmare — but the way around it is to move the piano as close as possible to the drums, which is contrary to what you'd think. I move it closer, because it's the time delay that's the problem, not the actual separation.

"I hate compression with a vengeance."

Do you use room mics a lot?

I think that's a fashion thing. Room mics became very fashionable when the Phil Collins gated snare thing became

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trendy. They never really excited me hugely; I prefer the versatility of making those decisions after the event rather than with the event.

Do you always multi-mic drum kits?

Nearly always, but I try and steer away from using two mics on the snare. I prefer a good over-the-top snare sound, with perhaps a little more EQ than I'd like to get the brightness I need, because the hi-hat tends to creep in if you add too much top. I usually mic the toms over; I very rarely have any luck under-miking them, unless the bottom skin is off.

I always use a [Neumann] KM 84 on the snare; I could never get a sound I was happy with using any mic other than a KM 84. I'll use anything on the hi-hat, really; I'm not fussed about that. And I use dynamics on the toms — they're the only things I ever use dynamics for, apart from the kick drum, where I usually use a D112 — that's the punchiest kick drum mic I've found.

Also, whenever I can, I put ribbon mics overhead, because I like to pick up a little bit of the room, and they're figure-of-eight. That was something Geoff Emerick taught me. I have a pair of Telefunken 4038s, and if I've got them with me, I'll use them for overheads.

How will you EQ the drum kit?

On the snare drum, lots and lots of 10k shelving, to make it fizz — a close condenser mic on a snare always sounds kind of puffy. If I'm using the ribbons overhead, I'll put a bit of top end on those, because they're dull-sounding mics; if I'm using condensers overhead, I'll use them flat. Hi-hat, because you're using so little of it, I would tend to make it flat as well. Toms, it depends what you want out of them, really. If you want a boomy tom, then you put a little bottom end in and take a bit of midrange out; if you want a hard, clicky tom, either leave them alone or add some mid.

Do you use the overheads for full-range signal, or do you filter out the low end?

I don't think I've ever found the need to filter them much. There's something about the general air that you create with the overheads; if I was filtering off the bottom too much, I'd feel that that wasn't being achieved.

So you're really creating the drum sound from the overheads, using the individual spot mics to fill in.

I'd say that. The single most important things in rock 'n' roll — let's face it — are the kick and the snare. That's really what you need, and everything else is on top of that. I've always found it slightly difficult to get the toms to be in the same room as the kick and snare; that's something you have to sometimes fight a bit to get — to make it sound like it's coming from the same source. And that's where the overheads can help — the air around them can help make the drums sound like they're all in one room.

Probably the hardest thing for the project studio recordist is to get the low end right — to get it tight without being flabby or woofy. What kind of tips can you pass along for achieving this?

The key is understatement; don't overdo the bass. A lot of records are so kick drum-driven; it becomes so much of a feature. You look at the meter and the kick's right up to zero and all the rest of the music is down there.

Are you saying it's more to do with level than EQ or compression?

It's mostly level. There are a lot of records I dislike where

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Everything you wanted to know about dBu and dBV ...but were afraid to ask

dB Demystified

by David Stewart

Most people who've been in audio even a short time are acquainted with our friend the dB. Short for "decibel" (1/10 of a Bel), the dB was conceived nearly a century ago by the scientists at Bell Labs, who defined early standards for measuring signal and sound levels that are still in use today. However, modern equipment is so different from what was used back then that many previously important concepts now appear irrelevant. And along the way, the dB has been twisted and turned to measure different things under different circumstances, leaving us with a legacy of terms and concepts confusing to even the most savvy among us. Those with an ambition in audio engineering better know this stuff, but can those with a simple desire to record their music get by without it? For the most part, the answer is yes — but there are some potentially serious twists and turns along the way that you need to be aware of.

The most notable of those is the dreaded –10 dBV versus +4 dBu issue, which you've probably already encountered. It's likely that many people would naively connect the two kinds of equipment together except that they usually use different kinds of connectors (providing the first clue that they aren't compatible). Of course they *can* be connected together using adapters — and it turns out that the different connector types have more to do with balanced versus unbalanced signals than the level differences — but the results of making such a connection are usually poor. Levels don't match. Meter readings are all wrong, and you hear noise or distortion. So what's going on?

The problem can be summarized this way: +4 dBu and –10 dBV each represent a reference voltage defined as the nominal operating level of audio equipment. However, dBV and dBu *each use a different reference voltage,* which means the difference between –10 dBV and +4 dBu isn't 14 dB (a common but erroneous assumption).

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Don't worry, we're going to make this as painless as possible! Both the dBV and dBu are decibel references to a specific voltage *without regard for impedance.* This means they only specify the voltage present and not the power transferred in a connection. The "zero reference" level for the dBu (in other words, 0 dBu) is equal to 0.775 volts, while 0 dBV is equal to 1 volt.

The formula for working with voltages is 20log (V1/V2). By plugging the zero reference voltages into the formula, and applying our long-dormant math skills we can determine the voltage specified by the +4 dBu and –10 dBV nominal operating levels:

20log (X/0.775) = +4 dBu

Solving this equation, we find that X, the voltage specified by +4 dBu,

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WHO CARES?

Does this whole dBu and dBV thing really matter all that much? Can't you just plug in your gear and live with the levels being off some, perhaps making up for it with trim or fader adjustments? The answer depends on what you're doing. In critical situations where you want the lowest noise floor and maximum dynamic range, yes, it really is important to have the levels properly matched. In other applications, you can sometimes "fudge" it, especially when using equipment that has input and/or output level controls — your ears should always be the final judge.

equals 1.23 volts.

20log (Y/1) = –10 dBV

Solving this equation, we find that Y, the voltage specified by –10 dBu, equals 0.316 volts.

Now it starts to become apparent where the problem lies. When analog audio equipment shows zero on its input or output metering, you know (assuming it's calibrated properly) that the level is at its nominal or "zero reference" level. For example, if you feed a typical hi-fi (–10 dBV) cassette deck a 0.316 volt signal, its VU or LED meter should be right at zero (just before going into the red) when it's input is set at "unity gain." Similarly, if you put a 1.23-volt signal into a professional multitrack machine's +4 dBu inputs, its meter should go to zero (again, assuming unity gain).

But if you connect the output of the –10 cassette deck to the input of the +4 multitrack, what meters at zero on the cassette is way down on the multitrack. While this specific scenario may not be that common, the need to connect –10 gear to +4 gear is commonplace. For example, a typical audio mixer may have a maximum output of +26 dBu, which is 15.5 volts. Plug that level into a piece of –10 dBV equipment and the meters will be off the scale and unit will likely be in full distortion. You'll get a correspondingly weaker signal by plugging a –10 device's output into a +4 input on the mixer.

In the land of the dBV, the above mentioned 15.5 volt signal equals 23.8 dBV. "Huh?" you're probably saying. "That's 2.2 dB *less*, so it should be a *lower* voltage level." But that's not true; remember the reference voltage difference between dBu and dBV: 15.5 volts is *both* 23.8 dBV *and* 26 dBu. There's always a 2.2 dB difference in dBu versus dBV with *any* given voltage. You can prove this by plugging their reference voltages into our formula:

20log (1/0.775) = 2.2 dB

This one fact has caused the most confusion for people who try to understand the difference between dBu and dBV. If we plug in the actual voltages we use for +4 dBu (1.23) and –10 dBV (0.316) we'll get the *real* difference between the two:

20log (1.23/0.316) = 11.8044 dB

Voila! There's approximately 11.8 dB difference between the zero reference or nominal signal level of equipment rated at +4 dBu versus –10 dBV. (By the way, did anyone notice that 11.8 + 2.2 = 14?)

Metering

What does this look like on your meters? A –10 dBV device outputting a signal at its "zero" meter level is outputting –10 dBV, which we have shown to be 0.316 volts. This is its zero reference and that level is equal to –7.8 dBu, but shows up on the meters of a +4 dBu device as –11.8. (See the accompanying chart to make more sense of this.) In order to get the +4 dBu device up to its "zero" level on its meter, you must drive the –10 dBV device until its meters register +11.8 dB, at which point it is outputting 1.78 dBV, or 1.23 volts (which equals +4 dBu). Most –10 dBV devices can deliver this much voltage, but it puts them awfully close to clipping. At this point, you're about out of headroom on the output device (at nearly its maximum level), while you're just beginning to drive the +4 dBu device. When the –10 dBV device peaks or clips, there's still tons of headroom (probably at least 11.8 dB) available in the +4 dBu device

Of course the tables get turned the opposite way if you're driving a +4 dBu device into a –10 dBV device, where a reasonably high output signal from the source device will usually cause overload in the target device. Perhaps you can turn the level down at one end or the other, but this may compromise your signal-to-noise ratio. For these and other reasons it's generally not advisable to interface the two different kinds of equipment together without some sort of gain stage in between that properly changes from one level to the other. There are many such devices available in various price ranges. Most of these also take care of balancing the unbalanced signals and vice-versa.

Pro Versus Semi-Pro

Why is –10 dBV equipment sometimes considered "consumer" or "semi-pro"? The –10 dBV standard arose because manufacturers were looking for less expensive ways to build equipment. The components required to drive 0.775 volts (with headroom to spare) into the 600 ohm loads of the day were relatively expensive. As equipment with higher input impedances came along, designers realized they could use lower voltages and save money. The –10 dBV standard took off in the consumer hi-fi industry, while the pro audio and broadcast industries were still building and using gear referenced to dBu and dBm (a standard based on power transfer rather than just voltage). Once –10 dBV gear began to cross over into professional use, the trouble began.

It's important to understand that there

ONLY TWO STANDARDS? YOU MUST BE KIDDING

You might think that all equipment is rated at either –10 dBV or +4 dBu, but that's not the case. After all, why have just two standards when you can confuse the issue with more? 0 dBu, +8 dBu, 0 dBm, 0 dBV, and 0 dB (whatever that means) are a few of the operating levels you may find on equipment. And don't forget about "dBv" (little "v"), which is the old way of expressing dBu. There are valid reasons for some of these other level standards, so when you encounter equipment using them, you just have to deal with the situation.

TABLE I

Meter Reading	dBV	Volts	dBu	Meter Reading
+16	+6	2	+8.2	+4.2
+11.8	+1.8	1.23	+4	0
+10	0	1	+2.2	-1.8
+7.8	-2.2	0.775	0	-4
+4	-6	0.5	-3.8	-7.8
+1.8	-8.2	0.338	-6	-10
0	-10	0.316	-7.8	-11.8
-2	-12	0.250	-9.8	-13.8
-2.2	-12.2	0.245	-10	-14
-10	-20	0.1	-17.8	-21.8
-30	-40	0.01	-37.8	-41.8
-50	-60	0.001	57.8	-61.8

A summary of the different meter and dBu/dBV values for various signal voltages. Numbers highlighted in blue represent dBV and dBu reference voltages, while numbers highlighted in red represent dBV and dBu zero meter readings.

really isn't much of an inherent disadvantage to a 0.316 volt (–10 dBV) nominal level compared to a 1.23 volt (+4 dBu) signal in today's equipment. However, since the +4 dBu standard is usually applied to high quality "pro" gear, while the –10 dBV standard is often used with unbalanced connections — and sometimes on gear of questionable quality — the stigma that –10 dBV isn't as "good" as +4 dBu has lingered in our industry. It isn't always true, and in setups where you're given a choice between the two, it's sometimes preferable to use the –10 dBV unbalanced connections rather than routing your signal through another gain stage to balance it and raise the level up to +4 dBu. It all depends on the gear and your specific circumstances.

Plug It In

Just a little bit of knowledge is all that it takes to understand how dBu and dBV relate to one another. Having put on your propeller beanie and thought your way through this article, you now have the background required to figure out what those two reference levels actually mean in terms of interfacing your gear in the real world. Never fear a decibel again!

This article is based on a version previously published on the Sweetwater Web site (www.sweetwater.com). Special thanks to Don Albanese for his assistance with this article.

David Stewart is president and founder of DSL Studios in Louisville, KY, and one of the bigwigs at Sweetwater in Fort Wayne, Indiana.

Back To Basics

by David Torrey

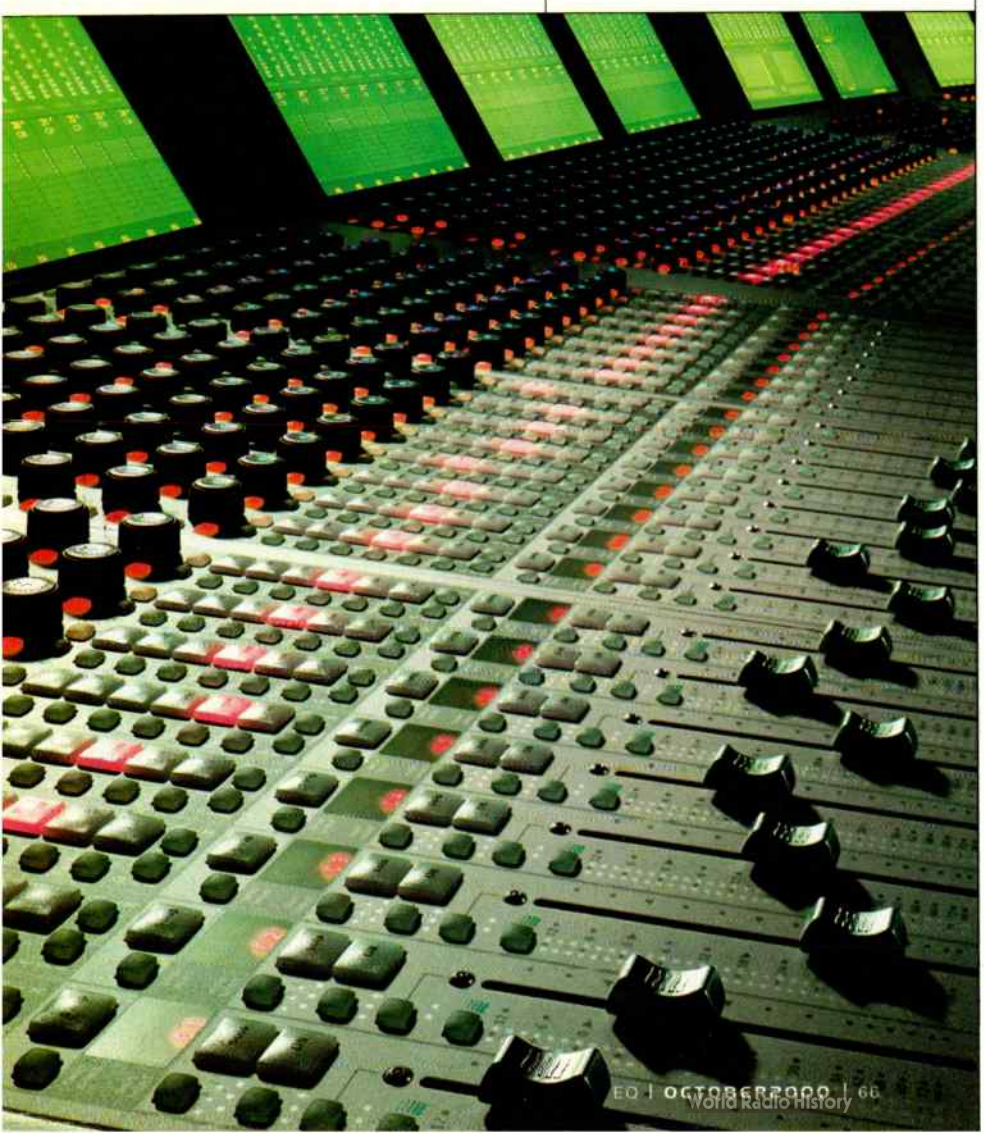
There are a lot of things that go into creating a great-sounding mix; as well as things that should be avoided to prevent hassles later in the production process. As a mastering engineer, I see the same problems over and over — problems that could be avoided without too much difficulty. Here's how to create problem-free mixes that are ready for mastering.

Placing The Tracks

Positioning a mix's tracks involves both determining how instruments are positioned from left to right in the stereo field, and creating a sense of depth "behind" the speakers. The two common positioning tools are the pan pot, which positions a sound by controlling its relative volume in each channel, and reverb, which creates a sense of space. Short delays at the beginning of a reverb signal, called early reflections, create "room sound" and a feeling of size, while the longer reverb tail can simulate a large space and add depth.

Panning Tips: Keep instruments that convey power and solidity in the center of the stereo field, unless you want a retro ping-pong stereo feel. Kick, snare, bass, lead vocal, and most instrumental solos all benefit from being positioned in the center. Tracks such as doubled rhythm guitars, keyboards, and background vocals can often be panned out to create a sense of space. Toms and drum overheads are usually placed at about halfway out so that they sound natural, but still give the mix some width.

Monaural Compatibility: One hallmark of pro mixes is that they sound great in mono. This is important for songs that will be played on FM radio. In fringe listening areas, most stereo receivers automatically revert to monaural reproduction to reduce noise. You can't control when this will happen, but you can make sure that you don't aggravate the situation. Frequently hit the mono button on your console when you record and mix. This combines the left and right channels into one signal that's sent to both speakers. When you switch to mono, the stereo image will collapse to the center, but both the tones of the instruments and their relative volumes should stay the same. If the sound or level of an instrument changes drastically, there's a phase problem that needs to be addressed.



Mix Balance

After the instruments are positioned, you need to set the correct volume and tone for each. Volume is fairly easy to set. We've all heard well-balanced mixes on CD and radio. If you can clearly hear what an instrument is playing, then the volume setting is close.

The tone part is harder. For discussion, tone can be broken down into two components, spectral content — bass, mids, and highs — and attack, the leading edge of a note. On a wide-range instrument, the fundamental tone conveys power, the lower harmonics (midrange) convey sense of fullness or presence, and the higher harmonics, attack and definition. A few examples:

Midrange: Guitar and keyboard leads need to be forward in the mix. To make this happen, consider using the midrange EQ control. The 1 to 2 kHz frequency range is critical to the body of the sound. Players doing metal styles tend to cut this range out to create an aggressive sound, but if you pull it too far back, the presence will disappear. If your EQ

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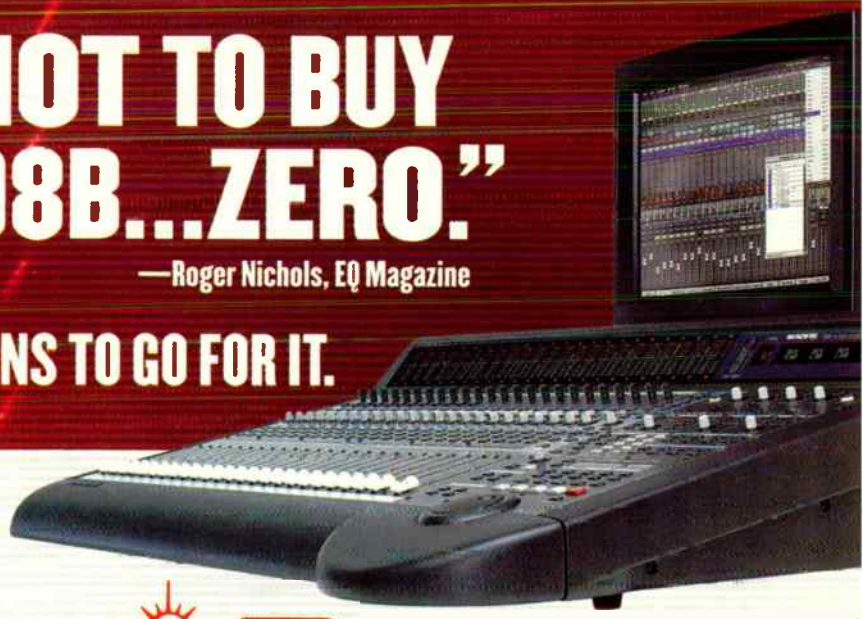

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includes a bandwidth control, start with a one-octave setting.

Enhancing Attack: Many engineers assume that treble EQ is the best way to produce attack. While it's common to chase clarity this way, you can end up with the treble cranked on every channel, and things will start to sound tinny. Try using compression, which can control the attack and smooth out volume variations at the same time. Adjusting the compressor's attack control to a slow setting will let the sound's leading edge pass through, while the body of the sound is reduced in level. This gives increased definition. A faster attack setting will clamp the leading edge, so the body of the tone will dominate for a fatter sound. Set the release control so that the compression cycle is finished by the time the next note is played. With the right setting, you'll have a clear tone without the added grit that sometimes comes with treble boost.

Tighter Low End: Compression can do a lot to give the bottom end more weight. The attack of a kick drum or bass guitar will get your attention, while the body of the sound moves the air. It's worth spending whatever time you need to get solid lows. You can build anything on that foundation. If you fail, the track will never have power.

Sub-Bass: Watch out for frequencies below 40 Hz. They eat up headroom in the mix and are too low to be heard on a many systems. An exception might be five-string basses or hip-hop drops that are designed to rattle the windows in the car next to you. Even in those cases, much of the power comes from the second harmonic, around 80 to 100 Hz. Many engineers roll off everything below 40 Hz to make the mix sound cleaner.

Snare Level: Most "pop" tunes benefit from a strong backbeat. You can mix snares hot because they're staccato, but they'll sound weak without enough low/mid body. Try stepping on the initial peak of the sound with fast-attack compression, and allow the body to come through as the gain control circuit releases.

Vocals: Everyone loves "air" and vocal clarity, so 10 kHz gets boosted. Consonants become more understandable, and you hear throat sounds. If you love this, be sure to add a de-esser to your equipment rack. De-essers can control just the excess sibilant sounds that lie in the 8 to 13 kHz range, while leaving the detail and body of the sound untouched.

Add some low-ratio compression to help the consistency of the performance, and you're 90% of the way to solid vocals.

Your Other Monitors

Every mastering engineer has faced mixing disasters that can be traced back to good people using bad monitors. It's tough to make good mix decisions if you can't trust what you hear. To make matters worse, most speakers aren't placed in the optimal location in the listening room, and the room itself contributes even more problems. In short, your speakers may be lying to you, especially in the bass frequencies.

What's the solution? You could spend next year's mortgage payments on speakers and room treatment...or maybe not. Instead, let's talk headphones. You already have several pairs in the studio for overdubbing, right? Forget those — we need reference quality. I'll stick my neck out and recommend three fine examples: Sennheiser HD600, Grado RS2, or my favorite, Stax Lambda Nova with tube amp. These carry street prices of around $330, $450, and $1,500 respectively. Ouch! Not cheap — but worth every nickel.

Listen to five minutes of well-mixed commercial music on any of them. Then listen to one of your mixes. The differences will be easier to spot with headphones. This is because you've eliminated the room from the equation, and your ears are hearing only direct sound. Plus all of these units are pretty accurate down into the deep bass. Think of them as nearfield monitors on steroids. Headphones can't take the place of monitors; you'll still need speakers to check panning, since phones exaggerate the stereo image. And phones won't hit you in the chest with massive bass tones the way that speakers can. But they can be a great second reference and a good magnifying glass for questions of balance and tone. You will hear stuff that was lost through your speakers. Consider headphones one more tool for achieving the Holy Grail — mixes that translate well across many speaker systems.

Mixing Technique

It's easy to get overwhelmed when you have an album to mix. Try slicing the work into manageable chunks:

1. Work on a short section. Focus your attention on a section of the tune no more than sixty seconds long. Include a piece of the main hook, part of one verse, and some of the bridge if possible. If you have a digital machine, set it up to play back the section as a loop. Now work on the tones. Auditory memory is short. There are times when you might want to loop ten seconds of music to concentrate on the sound of one instrument.

2. Start with the basic tracks.

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Bass, drums, one rhythm instrument, and lead vocal. Go for clarity and impact first. Work until your short section sounds killer, then document the EQ, level, and compression settings. You'll be able to use most of what you learned so far on the remainder of the tune.

3. Add parts one at a time. Resist the temptation to dial in all the awesome groove tracks and cool licks that the guitar and keyboard players added. That's the way of the Dark Side. If you give in, you'll use up your resources too soon, the mix will become dense and lose its focus. There will be less tension and release, which is what involves people emotionally in the music.

4. When the mix gets away from you, remember the silence. (Ah, deep wisdom, Grasshoppa.) All kidding aside, you want to hear the instruments clearly, so become sensitive to that point where too many things are going on at once. When you can't hear any silence behind the notes, cut away instruments until the energy of the mix returns. In extreme cases, return to the settings you saved before your ears and judgement became toasted. Take a break and try again.

5. Maintain Sane Levels: Don't record super-hot signals onto a digital mixdown deck. With DAT and CD-R, any level above –6 dB means you have a 16-bit recording. There's no advantage to pushing it harder. If the clipping indicator comes on, you've already lost music. Try for maximum peak levels of –3 to –1, and let the mastering engineer bring the volume up further, if necessary.

If you use a Finalizer-style maximizing processor during mixdown, try applying just a few dB of low-ratio program compression, to tighten the sound up a bit. Avoid using the Crush-O-Matic preset; if you change your mind later, you'll have to remix the album to undo it.

Recap

If your mixes have a good sense of space, clear instrumental and vocal tones, and energy that builds to a strong finish, the mastering engineer can spend more time building on the strong points and less time fixing problems. The final sound will work on radio, home and car systems, and everyone will be happy. Have at it!

David Torrey started building audio gear when he was nine years old (BC!) and hasn't stopped yet. He designs the toys at DRT Mastering (www.drtmastering.com), which specializes in fat analog sounds. Email him at davidt@drtmastering.com

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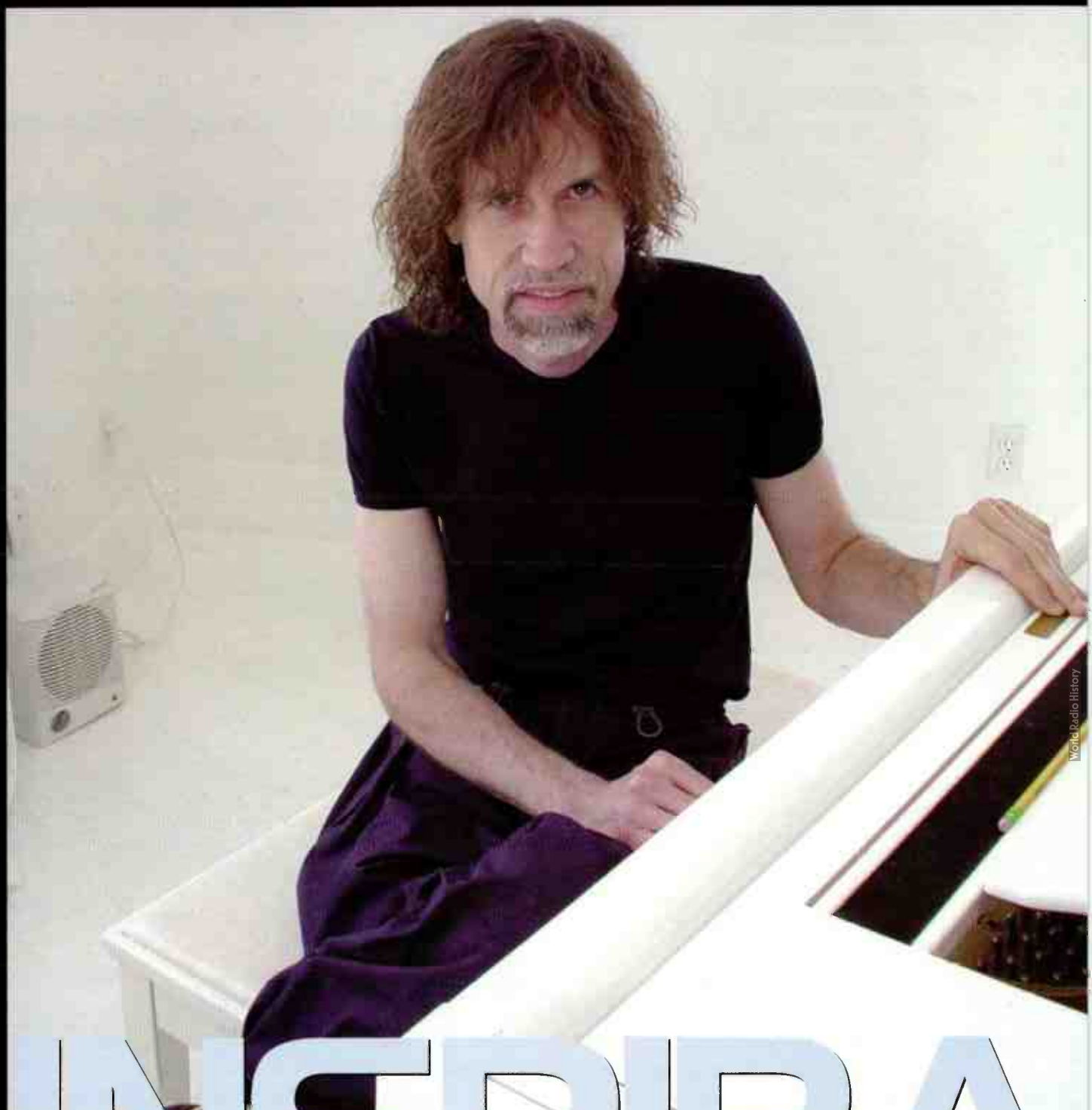

INSPIRA COLLABO

TION RATION

A conversation with Glen Ballard

BY HOWARD MASSEY • PHOTOS BY DAVID GOGGIN

In 1995, a young Canadian singer burst upon the scene, releasing an album that forever changed the landscape of the music business. Her name was Alanis Morissette, and the album was *Jagged Little Pill*. Not only did it yield three hit singles and dominate the charts for close to a year, it ended up selling over 30 million copies worldwide, setting an all-time record for a debut recording.

And did we mention that it was recorded on ADATs in a home studio?

Glen Ballard was Morissette's producer on that fateful album and on her follow-up *Supposed Former Infatuation Junkie*. In the past year, he's reprised that success with No Doubt's critically acclaimed album, *Return Of Saturn*.

A protégé of producer/arranger Quincy Jones, Ballard already had a long and distinguished track record on the day that Morissette first appeared at the door of his home studio with a sheaf of lyrics and an idea about making a new kind of record. He was the man behind the board for the Wilson Philips hit "Hold On," as well as a string of successes for a diverse crop of artists, ranging from Teddy Pendergrass to Curtis Stigers to Barbra Streisand to Paula Abdul. An accomplished instrumentalist, arranger, and songwriter, he had already written hits for the likes of Michael Jackson, Aretha Franklin, Al Jarreau, George Strait, and Earth, Wind & Fire.

Today he heads up his own record label — Java Records — and is currently working with Lisa Marie Presley and fledgling singer/songwriter Celeste Prince; there are even rumors afoot that he may soon be collaborating with an illustrious ex-Beatle. But no matter where his career takes him, he will forever be known as the man who, in true harmony with an artist of rare vision and talent, guided a record from a home studio to the top of the charts, proving once and for all that, yes, it can be done.

EQ: How did you begin the process of recording *Return Of Saturn*?

Glen Ballard: We spent some valuable time in pre-production going through every song and deciding whether or not it belonged on the album. If it did, then we took the next step of really making the arrangements as strong and as tight as they could be.

It really is an old-fashioned production, in the most positive sense.

Yeah, it is. They really wanted to create the record within their unit, and that's understandable; they have such a distinctive sound and they've played together for so long that it's a great advantage. So we did do it the old-fashioned way, in terms of really focusing on the arrangements and the songs. The hardest part was selecting the 15 or 16 songs that we felt deserved to be on the album, because a band can be really emotionally attached to them; they become like their children. The way I

describe it is, "There's only so many seats on this bus for this trip." [*Laughs.*]

I gather there were some major emotional tensions during the making of the album, almost like what Fleetwood Mac went through 20 years ago. Were No Doubt able to channel their interpersonal problems in a positive, creative way, as Fleetwood Mac did?

I think so. It's no secret that [lead singer] Gwen [Stefani] and [bassist] Tony [Kanal] had a romantic history in addition to their creative history. That's always a component for anybody who's writing songs; you tend to go to the most emotional place if you're being honest. Gwen is so without guile and without any sort of emotional artifice — I think she's incapable of lying in her lyrics! [*Laughs.*]

Out of the long history that they've had together, there have to be lingering deep feelings even if they're not being romantic at this moment in time. So that was certainly there for them to draw on from a writing standpoint; when there's something that emotional, it continues to be abiding. I think all the people in the band love each other in the general sense of the word because of all the struggles they've been through. They drew on that, and they really resisted the idea that someone like me would come in and help them write "hit songs." They really felt that they had to conjure up all the material from within their tight circle, and I respected that. I always considered it my primary mission to be there to help them make the record *they* needed to make. Certainly it is an honest and real reflection of where they are at this moment. Gwen didn't want to write "Just A Girl" sideways, because she said, "Hey, I'm not just a girl anymore; I'm nearly 30 years old and I want to reflect where I am right now."

In general, have you found that interpersonal tensions can serve as a creative spark or make for a better record?

I don't necessarily subscribe to that theory. If there is something like that going on, it is important that you try to channel it in a positive way, but I've seen it go the other way just as many times, where people aren't

Glen Ballard and his assistant, Rachel Cleverley

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collaborating. Music is one of the most collaborative media; I'm coming from a background where musicians have to play together and really listen to each other for it to really work. I like to create an atmosphere where people feel safe and creative; I think there's sufficient turmoil in anyone's personal life to draw on. [*Laughs.*] But when it comes time to play the game, if I may use a sports metaphor, it's important for the team be together, so I try to create an atmosphere of harmony, literally and figuratively. I believe that when people have a little more of a comfort zone, they perform better, though I suppose that in the writing process, those tensions can be channeled in the right direction.

But I don't want you to get the impression that No Doubt were an unhappy group of people; they actually had a great time. I don't presume to know much of the details of Gwen and Tony's relationship, nor is it my business, but to the degree that it impacted on what we were doing, I didn't find that it was negative.

Has Lisa Marie Presley inherited her father's musical sensibilities?

Absolutely. She's not a carbon copy of Elvis — she's distinctive in her own right — but to the extent that a musical gene is there, she's got it. Her record has emerged as something that is a real testament to her talent and her distinctive place in the universe.

> *"If it doesn't feel right, it's probably never going to be right"*

Is she writing as well as singing?

Yes, she wrote the entire album, either with me or with Cliff Magnus, or with Billy Corgan. She's emerged as a very gifted lyricist and as someone who's completely uncompromising in her approach. If there's any influence of Alanis Morissette, it's that she inspired a lot of people — especially women — to speak their mind and to not shrink from any topic and to not shrink from talking honestly. Lisa's got a lot to say, she's got a great sense of humor, and a unique take on life in general. It's a fierce record, and I love it.

What sort of style is it?

It's got elements of pop, blues, and rock, but it's a distinctive sound. It's certainly not shy and retiring; it has an element of aggressiveness to it, it has an element of cool, funky, slightly Southern in its flavor, which is not surprising. [*Laughs.*]

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How much of *Jagged Little Pill* was actually recorded in a home studio?

It was 75 percent done in my home studio, which certainly is a professional situation, but not a commercial studio. Everything was created in that environment, just with the two of us. When we overdubbed drums and other musicians, that was kind of postproduction, almost.

All of it was recorded on black-face 16-bit ADATs, and it was never bounced to any other format — it was mixed off the ADATs. We even used the ADAT's onboard A/D converters. There were no upgrades, no extras, no nothing — it was just off the rack. And I have to tell you, I think those original ADATs sounded amazing; in some ways, I miss the sound. I can't tell you why, but it just jumps out at me.

Was it mixed in a professional studio?

We did some overdubs on one song at Westlake, but the entire album was mixed in my studio on a Euphonix console.

Did you also edit directly on the ADATs?

Yeah, the only edits we did were from ADAT to ADAT. It was usually just arrangement, maybe adding another four bars of guitar, or whatever. We would just do offsets; we got really good at that!

So there was only the one analog-to-digital conversion — through the ADAT's onboard converters — and then it stayed digital all the way through.

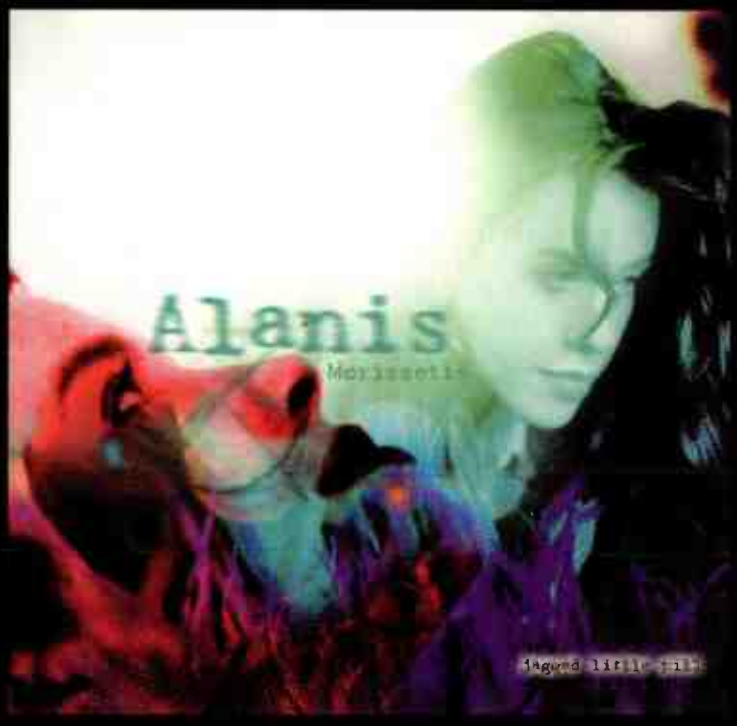
Yeah. I think one of the big positive factors was having a great mic for Alanis to sing through — a vintage C12 — going into a Demeter preamp, into an LA-2A, and then straight to tape. So we had a pretty nice analog stage for a lot of the stuff before we got it to ADAT.

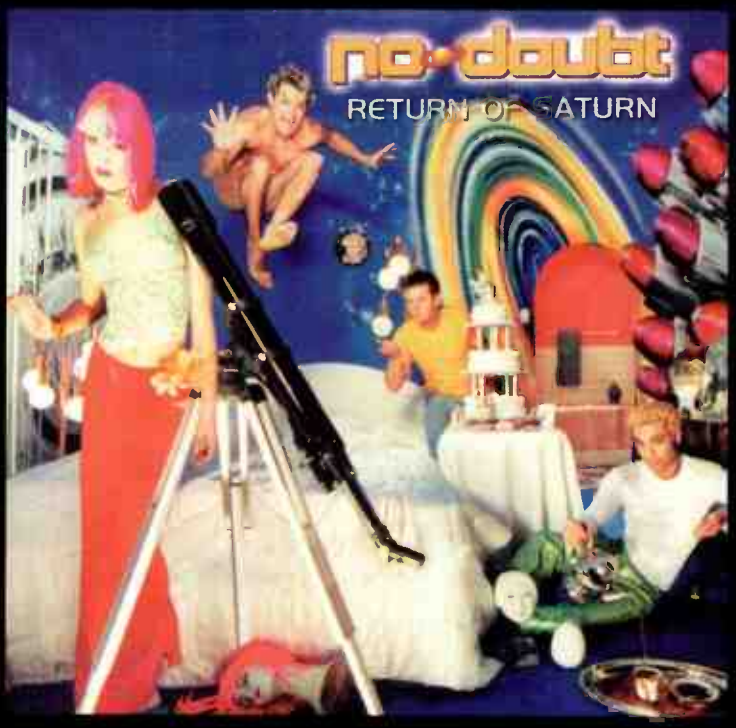
There was no equalization on the vocal on the way in?

None. Occasionally, with some of the guitar stuff, there might be a little bit of board EQ. I also have a Pultec equalizer, which I'd sometimes use lightly on the bass. I also have a Massenburg stereo EQ, which we'd occasionally put across the mix [bus].

Was the vocal chain you used with Alanis developed after a lot of experimentation with different mics and preamps?

Well, I have a pretty decent selection of tube mics — I've got a C12, an M 49, a U 47, and a U 67. The first time I heard this particular C12, I thought it had incredible top end and yet it was really warm in the lower mids, so it's got this wonderful combination of brilliant highs and still a real warm, tube thing. I always loved that mic, although it's not right for every singer. But the first time Alanis sang with me, I just put it up





because I thought this would be the right mic for her, and I've never changed.

So you didn't audition other mics on her?

No, I didn't. I just loved the way she sounded on it so much — it was right. At one point, the capsule fried, so I used the U 67 on her, and it was much inferior. So I got the C12 fixed really quickly! [*Laughs.*] It's a delicate baby — it's older than I am!

How much compression do you apply to Alanis's vocal?

It was just under medium compression, and I was riding the level to tape with my left hand at the same time. Her voice has an incredible dynamic range, and I usually have one or two chances to get it on tape — she's a one-take singer — so it was a heightened experience for me! [*Laughs.*] And after cutting a track all day, engineering everything, it could get pretty interesting.

On "You Oughta Know," I definitely scorched the vocal — the preamp was too high, and it was distorting — but there it is! [*Laughs.*] It was literally a one-take vocal, and it was so stunning, that was it. I knew I had a couple of scorch marks on that one, but nobody seemed to mind. Certainly she had the courage to say, "I love this — let's not take all the life out of it by redoing and sanitizing," and I certainly enjoyed that kind of attitude of, "Let's put it on tape with energy and leave it there."

Was *Supposed Former Infatuation Junkie* recorded on ADATs also?

It was done on 20-bit ADATs. Same vocal chain, same mixer — the Euphonix at my studio. We mixed to DAT and to half-inch analog, and, like *Jagged Little Pill*, I think about two-thirds of the final mixes came off the analog half-inch, and a couple of the mixes, we just liked the way they sounded on DAT. So it was pretty much the same setup — the only difference was that it was 20-bit ADAT.

It's a denser recording, though. Did you use more tracks?

It's denser, yeah. Again, we started with everything just the way we did it before — the two of us in my studio. Then we took it into Royaltone, which is a wonderful, big commercial studio, and the band played along with the basic tracks that we had. So there's a lot more of her band involved on that

INSPIRATION & COLLABORATION

EVERY GREAT PRODUCER NEEDS A GREAT ASSISTANT

Landing a gig with a multi-platinum mega-hit producer is the dream of many a young assistant engineer. For the past three years, Rachel Cleverley has been living that dream, working with Glen Ballard on projects with No Doubt, LIT, Terence Trent D'Arby, and Brian Setzer.

In many ways, location played a big role in Cleverley's landing the job. Growing up literally around the corner from legendary engineer Bill Bottrell and going to high school with Bottrell's children, she soon found herself working in Toad Hall, his personal studio. Bottrell taught her everything from lining up tape machines to mic positioning and doing punch-ins, even giving her an opportunity to do an overdub with Sheryl Crow. ("I went in, turned on the studio, and punched in Sheryl's pedal steel," she recalls. "I was nervous but it went just fine.")

So all the technical skills were in place when fate placed Cleverley and Glen Ballard at the same party. Learning that he was looking for an assistant, she quickly landed the gig, and hasn't looked back since.

"Rachel brings so many skills to the table, she already understood the process of making records, and she already understood that the process is an art and not a science." Ballard tells us. "Although we do apply as much science to it as we can, it's a process that can be relatively open-ended and is demanding. She seemed to have a great personality, plus she's intelligent and had serious knowledge of a studio, so obviously she was the right person."

"There's a lot of multitasking going on here that she not only doesn't shrink from, she embraces. She certainly makes it possible for me to do all the different things that I do in a given day and she's usually about three steps ahead of me in terms of helping to create environments for artists to thrive in. Certainly part of that is on a technical level — making sure that everything's working and the multitude of things that go into that — but it's much more than that. There's this other element that has to be there that Rachel has a really great understanding of: Providing comfort and support to the artists that come in, so that they do feel like they are in a place that they can be creative and do their best." —*Lisa Roy*

record than on the first record; it's really more of an ensemble.

Were the overdubs also recorded on ADATs?

Yeah, everything. We just kept it in that format. But we probably didn't go beyond 32 tracks on any song. On *Jagged Little Pill*, I don't think we went beyond 20, ever, and some songs had just nine tracks. *Infatuation Junkie* is denser; it has more tracks, more words, more backgrounds, more everything. [*Laughs.*] And because it's denser, it takes up more room sonically.

What's the story behind the guitar hum on "That I Would Be Good"?

[*Laughs.*] That's a 60-cycle hum with some reverb on it. I played the song on my Telecaster, and it's got this single-coil pickup, so it's a noisy guitar unless you're sitting just right. But because I was recording it myself, I couldn't really do anything about it — I just hit record, jumped over, grabbed the guitar, and started playing

> *"What we sometimes do in the studio is spend 90 percent of our time on stuff that's worth about 10 percent."*

on it; I was sitting on the floor with Alanis, and she was actually singing it while I was playing it. I listened back to it and thought, gosh, that hum has got to go. Later, she went home, and before she came in the next day, I replayed the guitar and cleaned it up. When she came back in, she freaked out, saying, "What happened to that guitar?" I said, "I didn't erase the original, I promise!" So she made me put it back.

The hum almost sounds like it's changing pitch along with the chord changes.

It's an aural illusion, a harmonic accident. I know I'm blowing my street credibility here, but it was a complete accident! [*Laughs.*] Not only that, but I was trying to get rid of it! I was saying, "We can't live with this," but she always fights for the authenticity of the moment, and she's always right. I guess that giving yourself the opportunity is what's important — being willing to go with it.

I don't think the listener cares about perfection. They just want to feel the emotion of the communication. That comes in many forms, but what we sometimes do in the studio is we spend 90 percent of our time on stuff that's worth about 10 percent — the absolute detail of everything that, at the end of the day, probably doesn't matter. It doesn't mean that we don't obsess over it and try to make it perfect — or at least as good as it can be — but sometimes it's almost like it's either there or it's not. It's got to be there as a basic thrust, or a lot of the other stuff doesn't matter. It's just got to feel right, and if it doesn't feel right, it's probably never going to be right.

And the real art of the producer is knowing when it is right, having the courage of your convictions.

With the technology that's available to us now, it's very easy to postpone any kind of decision-making, because you can literally record as many takes as you want and keep everything. So you can postpone what you should really be deciding on right then and there. Also, you can clean up everything: you can tune vocals, you can time-adjust — you can really make it all perfect, on paper. But the real challenge is to know when not to use all of that stuff, to find out the essence of what it is you're trying to accomplish. You can

make it perfect — that's easy, the tools are there — but that's not what we're doing here. We're trying to get the magic, and the magic is none of those things. Those tools can help, but more than ever, you've got to feel it — it's almost like you have to not think about it and really be emotional with your approach, because that's still at the heart of it.

That sounds like the kind of thing you would have learned from Quincy Jones.

Absolutely. Anything that's worth knowing about making records and about dealing with artists, I would give him full credit for. Starting with his deep love of music and how deeply he *feels* music, even as he's deeply educated in all forms of music. He probably has the deepest knowledge of music of anyone that I know, whether it's the origin of African rhythms and talking drums, whether it's orchestral, or jazz, or pop — he's fluent in all the musical areas. To him, it's all about the feeling of it — creating an atmosphere where people feel safe and creative and warm and want to give it up out of joy and love. So certainly he's been the biggest influence on me in that regard, and I love him like a father.

What's your favorite studio toy these days?

I've been developing a deepening relationship with the Pro Tools platform over the past year. It's certainly become kind of indispensable for me. I like the plug-ins, but I prefer to not go too far in that direction, and I still like to be able to bring up each channel individually and not combine stuff within Pro Tools, especially when it's time to mix. I experiment with all the plug-ins, and they're all fun, but I try not to go overboard with it. I like the aspect of the 24 bits and the noise floor being so low; I like the way it sounds now better than ever. But most processing I'm doing outside of Pro Tools; I use it mainly as a recorder and as an editor.

What advice do you have for the reader who wants to be the next Glen Ballard?

I think the best advice is to get hands-on with everything you can. Learn Pro Tools, because you can accomplish so much in that one box now — you don't really need anything else. At the same time, learn as much music as you can to go with it, because one without the other isn't enough. I started out as a musician and I learned engineering and programming just because it was an easier and more efficient way for me to express myself as a musician and as an arranger. But if you have both of those things, you're in good shape.

And learn to be sensitive to artists; if you're working with other artists, give them an extra measure of patience and encouragement. It's kind of like tending your garden — it can be very satisfying.

You know, I'm always studying. It seems like I'm getting software updates every two weeks for everything I've got, and I'm still studying music; I'm still trying to make up for my days as a music student, when I wasn't such a great student. So it's a constant learning process, and if you learn one thing every day, at the end of the year you'll be a lot smarter, I promise.

Portions of this interview were excerpted from Howard Massey's new book *Behind The Glass*, now available from Miller-Freeman Books.

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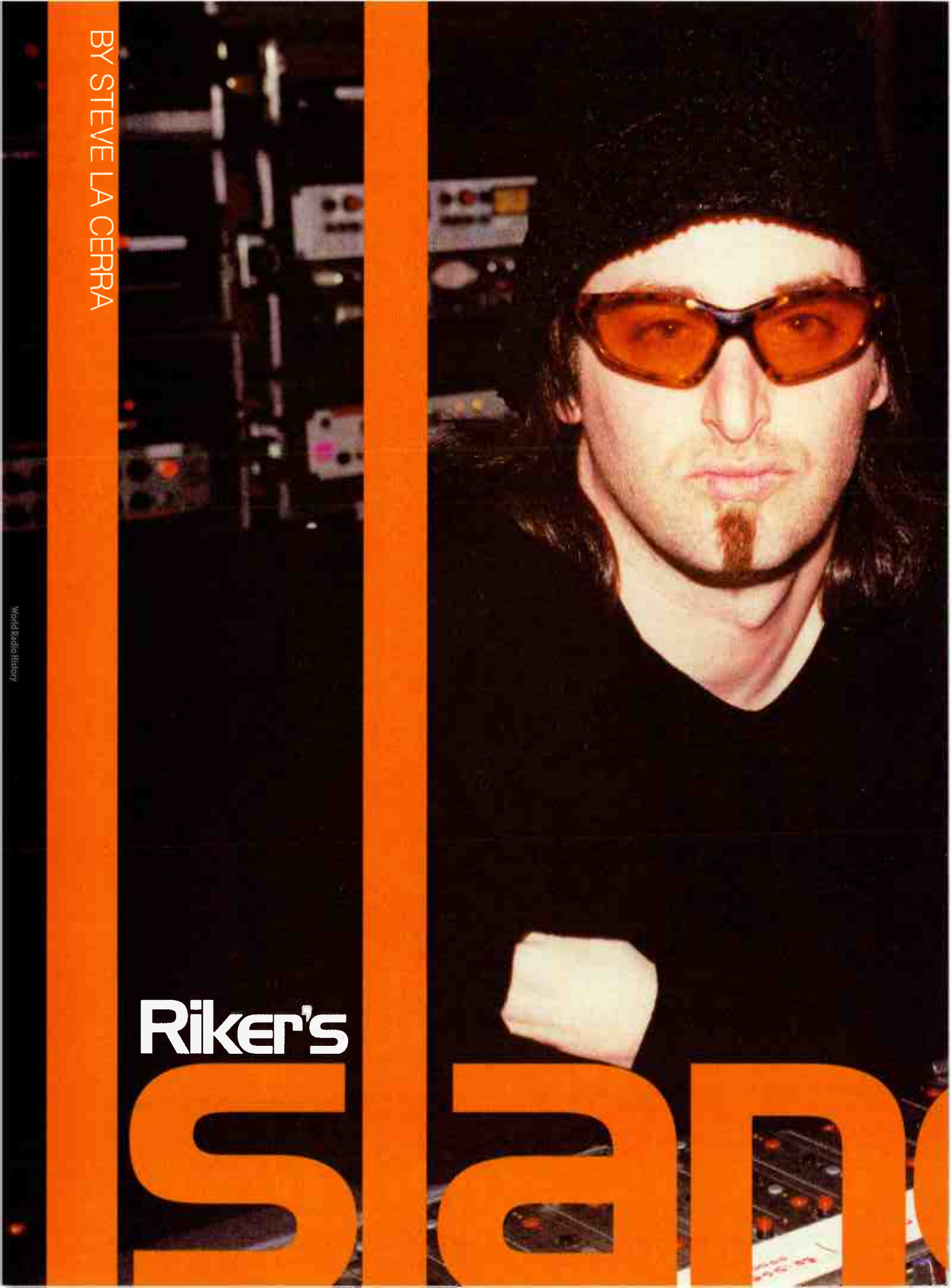
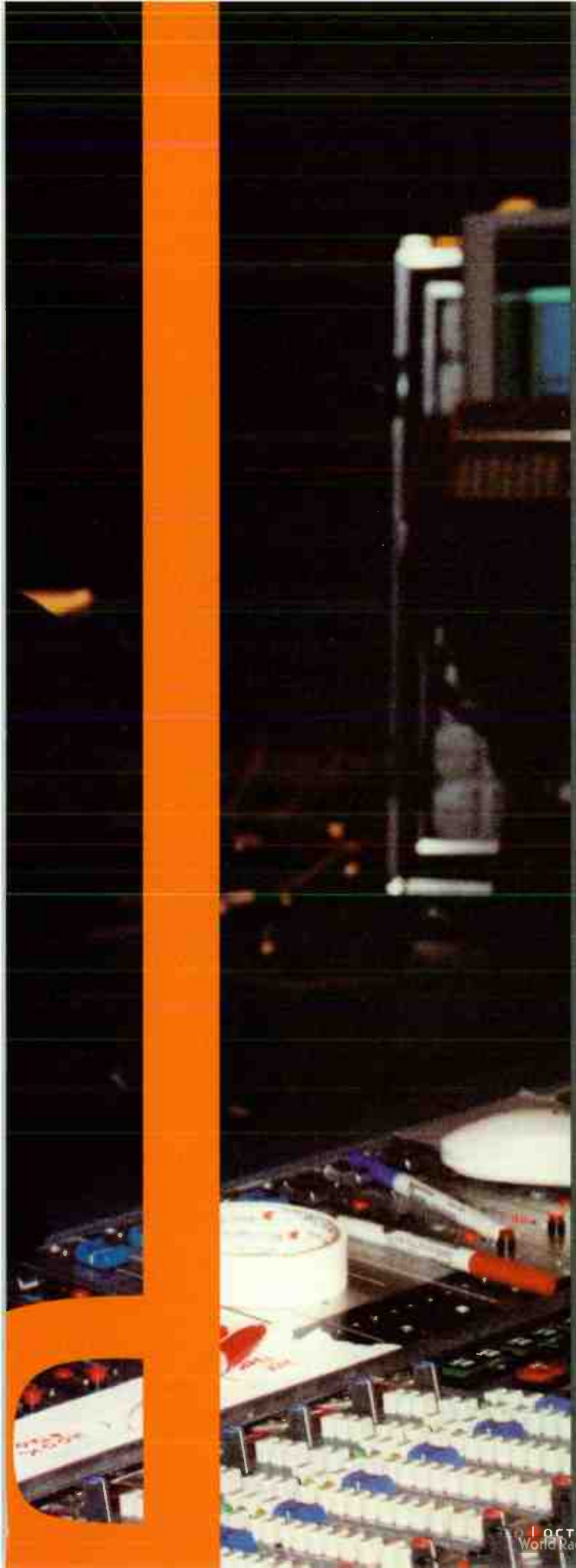


Riker's slap



Master mixer Warren Riker on bottom end, comp'd tracks and...country hip-hop

Any guy who's only 29 and has already won two Grammy Awards has got to be doing something right. Having mixed tracks such as Korn's "Freak On A Leash" (remix), Santana's "Do You Like The Way," Lauryn Hill's "The Miseducation Of..." and "Sweetest Thing," as well as tracks for Michael Jackson, Fugees, Aerosmith, Everlast, and Sublime, Riker's credits are — shall we say — extremely diverse. When *EQ* caught up with Warren, he was mixing a Virgin Records project from Jason Downes, a new country hip-hop artist soon to be released. That's right — country hip-hop. While the concept sounds strange, the music sounded way cool.

EQ: How in the world did you come upon doing a country hip-hop record?

Warren Riker: I have no idea! [*Laughs.*] But the funny sh*t is that two weeks ago I'm sitting in my dentist's chair and he says to me, "I got this great idea: country hip-hop. *Country hip-hop*." My dentist is always trying to come up with a weird new concept. I'm in the chair with his tools in my mouth goin' "grrrrryeahhhh right, yyyyyyyeah gggright." Two weeks later I get this call: "We have a country hip-hop record for you to mix." So my dentist gets to hear it first. Anyway, the artist is Jason Downes with Milk as the producer. He's got some really cool tracks on there. It's very strange to listen to because it's so different. I really don't know what to expect.

When you receive a project to mix, are you provided with roughs? Or do you just want to hear it fresh?

Most of the time I'd rather not hear the roughs. A week before we mix, I'll listen to the songs once or twice and get a feeling for the record. Then I won't listen to it again. The artist can listen all they want for their own heads and I'll jump out [of the control room]. But at that point it's a completely different animal. There's nothing worse than someone putting up a rusty cassette and saying, "The bass sounds like this on the cassette." And I want to kill them! If someone is calling me up and trusting me to mix their record they're usually willing to give me the shot to deliver what I hear for the song. At the end of the night when everybody is listening I'm willing to make everybody happy. After all, it's their record, not mine. I'm not the type of engineer/producer who says, "This is my mix and you can go screw yourself..." I'm here for the artist. If the artist asks me to do something then I'll do whatever it takes to make them happy.

Are you one of those engineers who prints a boatload of mixes?

No! I don't do recalls, and I don't print 50 mixes. I print one mix and an alternate. I usually make between six and eight passes of stems, with tones that I'll be able to lock up. If for any reason we need to come back and change something, we drop it into Pro Tools, line up the stems and push everything to 0. There's our mix again and from there we can change anything.

How do you organize the stems?

Riker's Island

I usually record drums and bass together. Listening to the song and learning the arrangement, I find instruments that are playing against each other and then group instruments accordingly. I'll split the stems so that there isn't a ton of sh*t playing together at once. Maybe keys and synths, guitars and pads... Then just the lead vocal, just the background vocals, and just the ad lib. It might take an extra half-hour to do at the mix session, but it's going to save the label $5,000 instead of coming back into studio to do a recall that never works. Guys who are printing a hundred mixes with vocal up, hi-hat down, guitars over to the left, drums down, bass up, and this left and that right — it doesn't make any sense. It's bullsh*t and gets you into trouble. You *know* if you give them that many they're going to say, "Well, can't you take this part of mix 43 and put it into mix 57?" No way. At least I know I'm out of the gate safe when I leave here. And the assistants love me because they don't have to do any recalls. So I get anything I want! [*Laughs.*]

How much time do you spend per song?

In around six hours I'm usually at the point of sweetening and tweaking. A usual mix date is a day: eight to 12 hours. On an album, I'll start a song today, mix all night, leave it up, maybe listen to it at home (or not), come back the next day with fresh ears, listen, print it down. Then I'll put up another one, leave it up that night. That's the ideal way for me to mix. And when I say I'm mixing for 12 hours, I'm not sitting in front of the console for 12 hours. It's more like hang out, work for an hour, walk around the studio for a bit, mix for another hour. When you're focusing like that for an hour straight it's a draining feeling because you're zoned for so long.

Sonically I like to build everything as we go instead of trying to fix it in the mix. I usually record effects as I track the instrument. Sometimes I'll set aside a stereo pair of tracks for something odd where I'm not sure if I like the effect. Nine out of ten times the mix is good in a few hours. Then somebody breaks out with, "Well, you know we have to do a little vocal overdub, and lock up this and redo that…." That's the most frustrating thing to me. You're trying to mix and the next thing you know, "We gotta fly these vocals in, we gotta overdub that." That kills me. If I'm in the project from the start and we book a studio to mix, there's nothing else to do but mix.

When mixing vocal tracks that you haven't recorded, have you had experiences where they're not comp'd?

Sometimes. That's an ass-kicker right there, because comping can take hours. If I have the luxury of going to another song, I'll have the band comp the vocal while I start mixing the other track. I don't mind if I have three different vocals and there's a line here or there that needs to be fixed. But when I'm given three multitrack tapes to mix from, one multitrack is filled with just vocals, and nothing is comp'd, then it's like, "Yo dude, this is no longer a one-day mix. This has become a two-day project." We had so many background vocal tracks for "Killing Me Softly" [Fugees, *The Score*] but I only had one 24-track machine. All along the way we were bouncing harmonies. We might have gone six generations down when the comp was done but it was cool, and it was on one tape.

CAREER DIVERSITY

When asked how he manages to keep from being pigeonholed into working on one type of music, Warren reveals, "In the beginning, I was worried because all I was doing was hip-hop. I had to take a few chances and not take everything that was thrown my way to protect myself from getting stuck. I really made a conscious effort to do different kinds of projects, even if they weren't the next big thing. There were projects I turned down, so I'd be able to make other things happen. In the past year I have been focusing on getting different-sounding projects so that it's not like 'Oh well, you mixed this big hip-hop record, so here's another hip-hop record for you.' You're brain kind of gets tired doing the same sh*t and you end up going through the motions. That's the last thing you want to do.

"Lately I have been mixing as well as producing projects from tracking through to mixing. It seems either I'm involved from the start on a production level, or someone hands me a record to mix. Producing from the start is more fulfilling. But mixing is something that's really fun."

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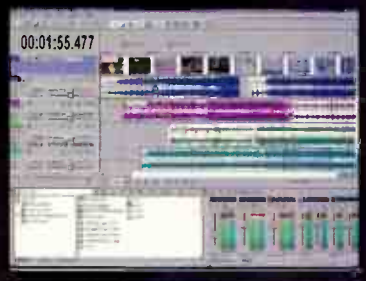

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Riker's Island

The decisions were made. When you do that many vocal tracks, experience shows that the vocalist is going to make the same mistake on each track every take.

If you're not giving them direction, it's going to stay the same. What I have been doing lately with Pro Tools is give them five tracks back to back. Run, run, run. On the first one we make sure the phrasing is cool and they have the right concept of where the song has to go. Then just give me five tracks and see you later, because you're not going to get them any better than that right now. I'll put together a good one. Anything in there that you don't like, we'll go in and fix later. But to sit doing take after take after take, and do comps and then more takes — it kills the song. It pulls every bit of life out of it.

How much of your mixing is done on mains versus nearfields?

If I'm working at a room I'm comfortable with, I use the mains a lot. Like here [Chung King in NYC], I'm comfortable with their Augspurger's. I know what the room sounds like so I'll use the mains a lot. I mix on all kinds of different speakers. My favorites are these little... [*Makes a sour face.*]

...Radio Shack speakers? I have a pair of those, too — $15.99 each!

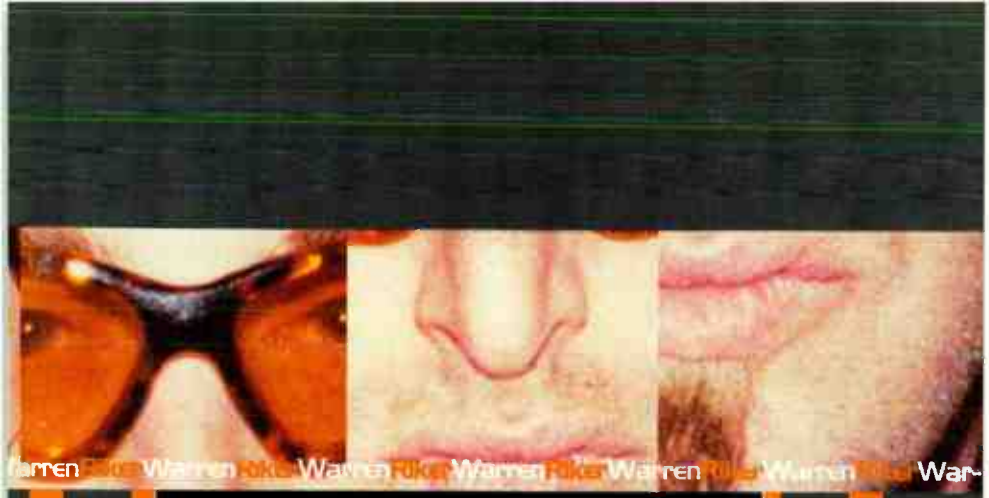
I do everything on those! From start to finish I'm constantly on the Shack's. I have 30 pairs around here. Everybody is stealing them from me! Every time I'm done with a record, I'll turn everything up and blow

WARREN RIKER'S TIPS FOR BETTER BASS

All the power in the mix is coming from the low end, explains Warren Riker, "because it takes up so much space. Really controlling it has so much to do with the relationship of all the bass instruments, and whether they're completely in tune with everything else." Here are Warren's tips for improving the bottom end of your mixes:

- Listen carefully to the bottom end. "Most of the time in the beginning of the mix, all I'm working on is the bottom. I'll roll everything off the top of the tracks, sit and listen to the bottom."
- Try listening on a pair of small monitors. "Those little Radio Shack mama-jammies help me make sure the low end is going to translate to other systems."
- Try delaying certain sounds. "This can help the bottom not only 'feel' big but also to 'hear' big. Use a slight delay of a few milliseconds to make one bass sound line up with another. Moving a track slightly or changing its EQ will somehow — through the phase shifting that results from EQ'ing — allow me to get things to sound bigger. It's not just dumping a crap-load of bottom end on a kick drum and a bass."
- Experiment with compressors. "If there's a lot of powerful, transient bass cranking away I'll use compressors sidechained to each other. For example, if I have a kick drum and bass happening at the same time, but not really together, I might feed the bass into a compressor and sidechain it. The kick track will feed the sidechain of the bass compressor, so, when that kick is hitting, that bass will duck just a little bit. It depends upon the flow of the track. It might be the other way around."
- Set up a couple of bass channels with different processing. "That's the balls right there. Use those channels in addition to the originals (if you're delaying a signal, do it at the source and then drop it back into the mix)."
- Use a crossover to split a bass sound. "Maybe just feed the sub output of the crossover into a compressor or a dbx subharmonic box."

"Just like anything else, there's nothing that works on every mix every day. This thing might sound as big as God one day. Then the next day you process the bass the same way and it's not working. You might not need anything. It might be bigger by leaving it alone."



MIXING TO DAT VERSUS HALF-INCH

Warren typically mixes to "half-inch analog tape or DAT with Apogee converters. Most of the time, the DAT is fed straight from the console outputs, but once in a while I take the output of the half-inch machine and go to a second DAT. At any of mastering sessions I've ever attended it's either the half-inch or the DAT mix — never the DAT from the half-inch. I vary from using half-inch at 15 ips, 15 ips with SR, or 30 ips.

"Whether we use the DAT or the half-inch mix for the record depends on what the song is like sonically, the condition of the half-inch machine, the hours on the heads, and whether it was tweaked out before the session. One day an analog machine may sound great, and the next day the same machine might not work. Or the Apogee DAT may sound better for whatever reason. I can't say exactly why. When it's done you sit back and listen for which mix sounds better. It's never really been like 'all the DAT's sounds better' or 'all the half-inch mixes sound better.'"

them to bits! They keep on changing them, though. They started with the Minimus 7's, then recently it was these Optima's and they're like little heavy-duty-shielded Shack's. Then they ported them and the port bugged me out. I complained to the guy at the store, "Dude, you can't do this to me. This is my *mixing* speaker!" Then they cut the port out. (They listened to me — *ha*!) I also use ProAc Studio 100's, NS-10's, Auratone's, UREI'S, the Studer mono speaker on the tape deck.

What kind of monitor levels do you mix at?

Either really low or really loud. If I'm making critical balances with, say, vocals, I'll probably listen at around 85–90 dB. For blends I like to listen at a real low level. To get the impact and feel, I'll crank it up for maybe half a song. To see what air movement is going on. Most of the time I'm down on the little joints for the whole time.

The relationship of the bottom end to

Riker's Island

the rest of the instruments is easier to judge at a lower level.

Because once you open the door and the bass gets out [from the mains], it's everywhere. You don't know where it's coming from. It feels good but you really can't tell. If people come in to listen, that's the hype button right there. Turn it up, get everybody fired up, and it clears the room out. Then I go right back down to the bottom [the smaller monitors] — unless I know the monitors up top *really* well. Most of them sound like sh*t whether you know them or not. It's knowing what they're doing. I don't use them for anything other than listening loud. [For Warren's tips on mixing bottom end, see sidebar.]

When you're doing a live session, how involved do you get in the sounds of the instruments?

As much as I can without annoying the musicians. That's your gig right there: to make all those sounds work together. Most of the time, we know we're cool with each other before we go into the studio. So no one has any problems if I say, "Hey look, I know you love this setup for your live show, but right now it ain't happening. Let's start from scratch and go for something better. If what I'm going for isn't as good as this, I'll be glad to put it back. But let's at least give it a shot." You have to be a bit of a diplomat. It's not something you're doing out of your own ego. You're just trying to make a good record and it's not for any other reason than that. Just like anyone else mixing a record, when their sh*t comes on the radio they want to know it's theirs.

That's tough to do.

Of course it's tough. It revolves around the music that you're mixing. If you're mixing something that sounds like everything else, you really have to fight through that to bring something else out and make it different. That's what I work at, trying to bring out something that makes a record sound different.

What else is coming up?

For the next few weeks I'll be mixing Jason Downes. After this, I'm supposed to be mixing the new Jive Jones record for Columbia and possibly the Long Beach Dub All Stars. Hopefully we'll get working on Lauren Hill's next CD, but right now she's writing.

WEBLINK

Warren Riker may be reached at rkrsisind@aol.com.

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> *H.D. Wells, Bromley, England, 1899**

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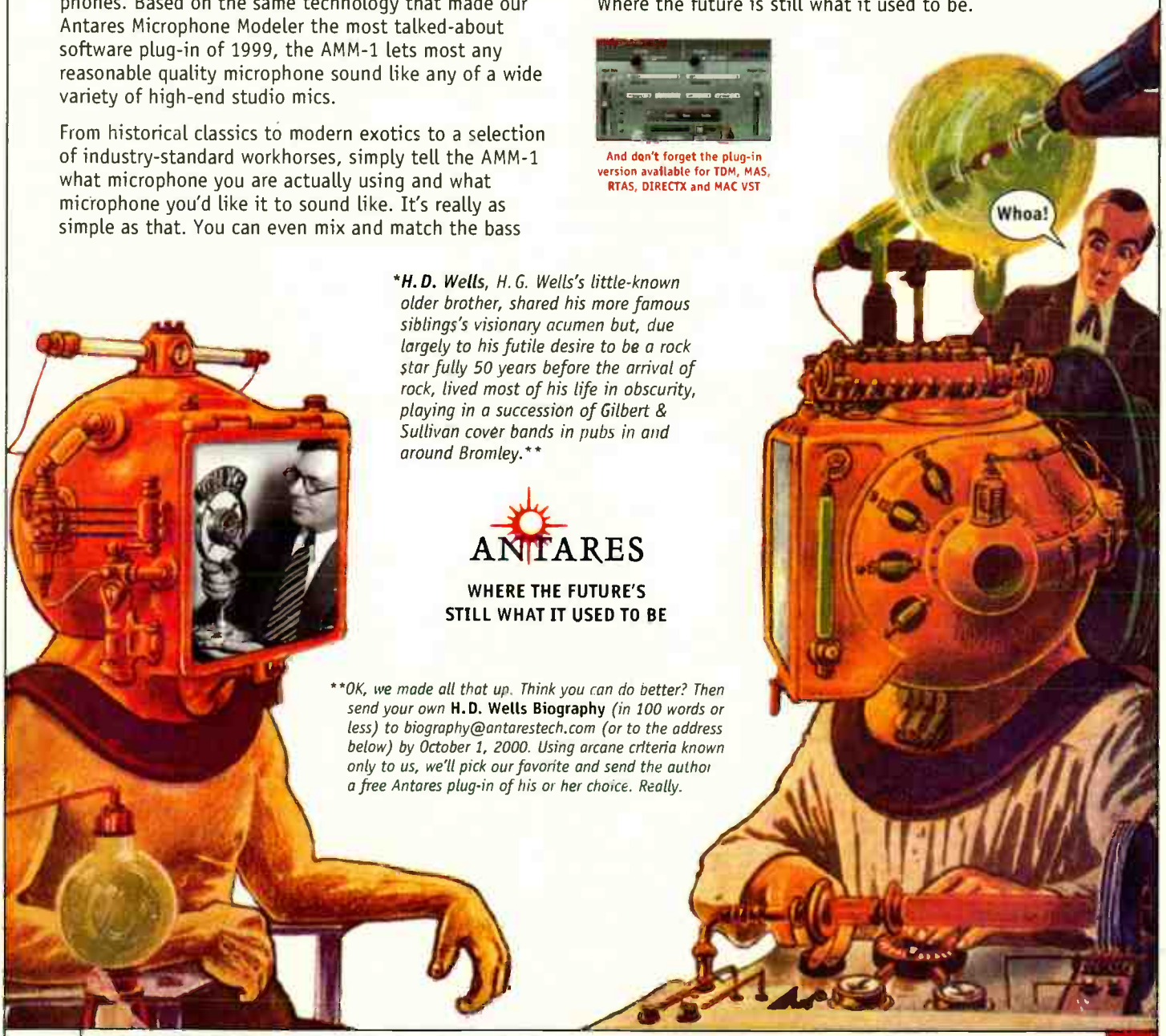

H. D. Wells, *H. G. Wells's little-known older brother, shared his more famous siblings's visionary acumen but, due largely to his futile desire to be a rock star fully 50 years before the arrival of rock, lived most of his life in obscurity, playing in a succession of Gilbert & Sullivan cover bands in pubs in and around Bromley.***

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WHERE THE FUTURE'S STILL WHAT IT USED TO BE

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02

CIRCLE 04 ON FREE INFO CARD

Home Is Where The Art Is

SUZANNE CIANI ON HER NEW ALBUM, HER NORTHERN CALIFORNIA PROJECT STUDIO, AND WORKING AS AN INDEPENDENT ARTIST

By Jonathan Miller

Dubbed "America's First Lady of New Age Music," Suzanne Ciani has worked in some of the world's classiest facilities, yet chose to record her latest, Grammy-nominated studio offering a little closer to home — well, *in* her home, actually.

"Every time I record an album, I think it is my last. I've thought about this since my first, *Seven Waves*, when I was sure I had summed up everything that I would ever have to say. But I am always wrong, proving to myself again and again the mutable nature of certainty. A good stopping point might have been nine, like Beethoven, or a numerical milestone like 10. Yet here I am with album #11 — surprise!"

So read the liner notes from *Turning,* composer Suzanne Ciani's latest release. "...the flowering of my living here in northern California, after 19 years in New York City."

In A New York Minute

Suzanne's road to becoming the "First Lady of New Age Music" has been long and winding, innovative, exciting, and ultimately, successful. As a child she fell in love with the melodic classical masters Bach, Beethoven, and Mozart, teaching herself to play piano and read music. Fast-forward to the late-'60s; as a music undergraduate at Wellesley College, Suzanne took a field trip to the Massachusetts Institute of Technology (MIT), a trip that would have a profound impact on her musical future. There she was introduced to a professor attempting to coax a computer into re-creating the sound of a violin — something we take for granted today, but unheard of back then.

Gaining a Master of Arts in Music Composition at the University of California at Berkley in 1970, Suzanne studied at Stanford with Max Matthews, the granddaddy of computer music, and John Chowning, whose frequency modulation techniques were later used by Yamaha and their family of FM instruments.

But it was Don Buchla, one of the founding fathers of analog synthesis, who would set Suzanne on the road to musical glory by employing her on the assembly line in his Oakland loft. Thus began a love affair with her own Buchla Series 200 modular system that would see Suzanne transiting to New York City, where she tutored Philip Glass and Patrick Moraz, among others, in the art of the synthesizer, before establishing herself as an in-demand session player.

As president of Ciani/Musica, Inc. from 1977 to 1987, Suzanne pioneered electronic sound design for television, producing numerous award-winning commercials for General Electric, Pepsi, General Motors, AT&T, and Coca-Cola, whose infamous "pop and pour" sound effect was created entirely on her beloved Buchla!

Music-to-picture projects followed, with scores including *The Incredible Shrinking Woman* (1980) and the 1986 Moscow Peace Prize award-winning feature film documentary *Mother Teresa*. Between 1991 and 1993 Suzanne provided the score for ABC's popular daytime drama *One Life To Live*.

Ciani's earlier-mentioned debut album was initially released in Japan in 1982, but it was not until 1986 that she really broke through as a recording artist. Her second album, *The Velocity Of Love,* helped define the New Age genre on the radio; its title track proving to be a big hit. Eight more albums and several Grammy nominations followed.

PHOTO © SUZANNE CIANI

Home Is Where The Art Is

Which bring us to where Suzanne is now. Returning to those liner notes: "*Turning,* the album, has grown out of my life...and it's grown from the inspiration of the natural beauty of where I live and designed my studio. I've recorded all over the world, but this time I knew I wanted to be home, in my own very personal environment."

That "very personal environment" is a picturesque oceanside former hunting cabin in Bolinas, California, one of Marin County's beach towns. Signs pointing to it are always being removed by the reclusive locals (largely well-to-do former San Franciscan hippies), but, once you find it, they're surprisingly friendly — just like Suzanne Ciani.

Oceans Apart

Though New York's frenzied lifestyle and the serenity of northern California are oceans apart, one can't help but wonder if fate played a supporting role in bringing Suzanne to Bolinas. "The mystery is why I ended up in northern California, and not L.A.," poses Suzanne. "That really was a 'life' choice. In my last year in NYC, I discovered I had breast cancer and I knew a change of environment was due — a move from the 'super-fast-track.' I went from the center of Manhattan to the end of a dirt road and never looked back."

Yet, even prior to moving, Suzanne had considered it artistically necessary to record outside the New York City confines of her Ciani/Musica Studios from her fourth album onwards. "*History Of My Heart* was recorded while I still lived in NYC. I went on a writing and recording sabbatical to northern California to seek inspiration and change gears from the NYC machinery. I was in the habit of travelling to record my albums as my New York studio had a rather frantic ambiance with all the production going on. *Pianissimo* was recorded in Los Angeles, and the new material for *The Private Music Of Suzanne Ciani* was also written and recorded in L.A. *Dream Suite* was entirely written in Bolinas and recorded in Moscow and Capri, Italy. *Pianissimo II* was recorded at Skywalker Ranch — an incredible 'local' resource. *Suzanne Ciani And The Wave: Live!* was recorded at the Hearst Theater in San Francisco, and, finally, *Turning* was written and recorded in my Bolinas studio — the very first time I actually made a final master 'at home.'"

"Just prior to recording *Turning,* I acquired Mark Of The Unicorn's 2408 hardware and software, and it is truly remarkable," says the composer. "By working in Digital Performer, I can integrate acoustic tracks with synth tracks, record 24-bit, and, using Waves software, do a lot of the processing right within the computer. The best part of the situation is the impact it has had on my performances at the piano. The recording process is very relaxed and self-controlled, and I believe I get a heightened intimacy of sound and feeling."

PHOTO BY JOSEPH GRECO

SUZANNE CIANI AND HER BAND PERFORM LIVE

OCEAN AVENUE STUDIO GEAR

SYNTHESIZERS/ SAMPLERS/MODULES
- Alesis D4
- Alesis QS8
- E-mu Proteus 2
- Korg M1R
- Kurzweil K2000
- Roland D-550
- Roland JD-800
- Roland S-770
- Roland M-OC1

KEYBOARDS/CONTROLLERS
- Kurzweil K2500
- Yamaha Disklavier C3

COMPUTER HARDWARE/ SOFTWARE
- Adaptec Toast
- Apple Macintosh G3 (333 MHz, 128 MB RAM, 9 GB HD)
- Apple Macintosh 8500
- Digidesign SoundDesigner II
- Digidesign Audiomedia card
- Passport Encore
- MOTU Digital Performer 2.5
- MOTU MIDI Timepiece II
- MOTU Video Timepiece
- MOTU 2408

PROCESSING
- Lexicon 224X
- Lexicon 300L
- Waves Native PowerPack

PERIPHERAL HARDWARE
- APC uninterruptible power supply
- Furman power conditioners

MIXING
- Yamaha O1V (with ADAT interface)

RECORDING
- Alesis ADAT XT20
- TASCAM DA-45HR (24-bit DAT)
- Yamaha CRW8424 (CD burner)

SPEAKERS
- Genelec 1031P

MICROPHONES
- Neumann U 87
- Schoeps CMC6-MK4 (matched pair)

VIDEO
- Panasonic AG-7350 SVHS (VCR)

Home Is Where The Art Is

Intimate it is, yet Suzanne isn't averse to accepting a little outside assistance on occasion. "Peter Kelsey, my engineer from L.A., came up during the pre-production stages and set up the Shoeps mics for me at the piano. He created various memory slots for me in the Yamaha 01V mixer for recording piano and overdubbing various instruments, complete with bussing and EQ, and all I had to do was push a few buttons to accomplish all the recording. Then he came back up for the mix."

California Dreaming

Ocean Avenue Studio, Suzanne's undeniably beautiful recording retreat, is in a corner of her living room, beneath a large window overlooking Stinson Beach. "Environment is extremely important to me and my work," she says. "My studio is situated on a cliff, with the sea below, a view of San Francisco across, and some glorious hills to the left where the moon rises. From my piano I can watch the migrations of clouds, birds and butterflies, and the ever-changing patterns of light and movement on the water."

More inspirational surroundings you'd be hard pressed to find; little wonder, then, that the lilting tones of *Turning* came so easily to this 'composer-in-residence.' "Another impetus for me to record here was that all of the songs had been written in this environment, were in some way attached to it, and I wanted to capture them authentically."

"The acoustics in this wooden cabin are superb, though this defies all logic," Suzanne continues. "All those years I spent in enclosed New York spaces, worried about 'leakage.' Except for the patter of rain on the roof, nothing seems to interfere with the signal here, except the occasional construction project. It is such a gift that the technology of digital recording has advanced to the degree it has to allow me to work in this spectacular spot."

Quantum Leap

That technological leap wasn't apparent from the outset, however. "For me, it was a big leap of possibility to record here, and I didn't see the possibility right away," concedes Suzanne. "I had produced a track for Asia earlier, which I recorded and mixed here using an Alesis ADAT, and the recording sounded very good. I had kept a Neumann U87 from the NYC days. So, I began to wonder if perhaps I *could* record an album here. I looked into Pro Tools, but it did not feel right to me. I guess, after the Synclavier years, I had trouble adjusting not only to the notion of a 'home' computer-based system, but one whose producers, again, seemed to have a monopolistic view of themselves."

Instead, Suzanne sought her own down-to-earth recording route. "I found that the 2408 was just about to come out and I thought, 'interesting coincidence of events; perhaps I should check it out.' It was a dream, seemingly made-to-order for my project. Although I would only have eight outputs, as the music was of 'chamber' proportions, it was exactly adequate."

PHOTO BY JOSEPH GRECO

CIANI IN HER PROJECT STUDIO

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"I love writing at the Yamaha Disklavier, where I can capture my ideas spontaneously," she says. "Then the structuring and arranging of a piece I do in Digital Performer, including printing out the musician's parts. It's a long way from the old New York days when we would have to leave the music under the copyist's door by midnight to see it appear at the studio the following day!

"One particularly useful gift of the technology was using the Disklavier for playback while auditioning various mic placements for the piano. It provided a perfect 'control' for the experiment: the performance was always exactly the same, so we could easily compare the recorded sound of different placements, which we documented with a Polaroid camera."

Not that Suzanne's recording past has entirely been forsaken. "I still use my beloved Lexicons, the 300L and PCM80. The Kurzweil K2500 has found a place of honor in my studio. It's a wonderful controller, and a very intelligent machine. The Genelec speakers are important, as they provide a standard and reference for my engineer."

Given that *Turning* included acoustic instruments such as violin, oboe, cello, English horn, flute, bass, and guitar — not forgetting Suzanne's piano — an obvious question is whether Ocean Avenue Studio was adapted to meet these demands. Suzanne responds, "I would say that rather than having adapted the studio, I adapted to *it*. As it is small, and as I only have one great mic — other than what's on the piano — I recorded the players one at a time. We have performed live together frequently, so getting an ensemble feel was not a problem. And, of course, missing tracks could always be 'roughed in' by the synths. It was effective to work 'one-to-one' with each musician, to be able to concentrate so specifically."

"I LOVE WRITING AT THE YAMAHA DISKLAVIER, WHERE I CAN CAPTURE MY IDEAS SPONTANEOUSLY. THEN THE STRUCTURING AND ARRANGING OF A PIECE I DO IN DIGITAL PERFORMER, INCLUDING PRINTING OUT THE MUSICIAN'S PARTS."

Virtual Versus Reality

Ciani isn't entirely swayed by sweeping predictions of the physical recording studio disappearing into the virtual world. "My experience is that we are in a bit of a hybrid situation," she responds. "I have processing and mixing options in Digital Performer, the Yamaha 01V, even in the synths themselves — so many options, making sense of it, finding your path, can be a challenge. I am a fan of 'virtual' mixing consoles. I love the graphic interfaces that allow you to do microscopic surgery, and the amount of automation available. I am not a fan of the emulation of acoustic instruments for the most part. I think that when synthesis became this literal sort of thing, the boat was missed. I am a fan of digital processing in all forms, of creating 'virtual spaces.' Even in the

PHOTO BY JOSEPH GRECO

Buchla days, I always used a quad setup and voltage-controlled reverb and Doppler shift to sculpt the spatial landscape."

"If the question has to do with being in favor of the democratic availability of so much studio sophistication, I must say I am thrilled. My only caution, having been through so many evolutions of technological possibilities, is that the architecture of such systems remains flexible enough to allow creative choices and interactions on many levels."

Riding The Independent Wave

"For me, the most important artistic control is that of the intellectual property itself. To maintain legal ownership of one's work is fundamental to an artist's career. The last bastions of major label control are the physical distribution of product and the ability to get airplay — yea for the Internet, which is showing artists a new possibility, and giving them hope and a vision of standing for their own needs! A lot of speaking-out is being done now to expose the abuses of the music industry. It is no secret that artists have not had the necessary leverage to take care of themselves vis a vis the majors. Often there is the notion that an artist is independent due to the lack of major label contract offers. This is an inappropriate value judgement. The artists of the future will choose to control and own their own creations."

Suzanne must have had something along those lines in mind when co-founding Seventh Wave Productions, an independent multimedia company with her husband, entertainment attorney Joe Anderson, in 1994. Their Winter/Fall 1996 catalog states, "We are an artist-friendly label with the mission of releasing high-quality instrumental music and supporting the work of our artists through support at retail, distribution and fair dealing." At the time of that mission statement, there was a small roster of artists on the label and its then newly formed Atmospheric Records imprint. Now, several years down the independent business freeway, Seventh Wave has been "downsized" into effectively becoming an outlet for Suzanne's product.

Suzanne explains, "We had a great start and built a strong artist roster. The music business is always filled with vagaries and obstacles, and this time it was the collapse of the independent distribution system that hurt us. My female intuition tells me that this was no accident — that the majors were behind unplugging the system that was strong enough to bother them. Basically, Alliance bought up all these vibrant independent companies for good money, then kind of flushed them down the toilet and they all but disappeared. Our distribution never fully recovered. Still, I feel very strongly that being independent is the most worthwhile approach."

Caught In The Net

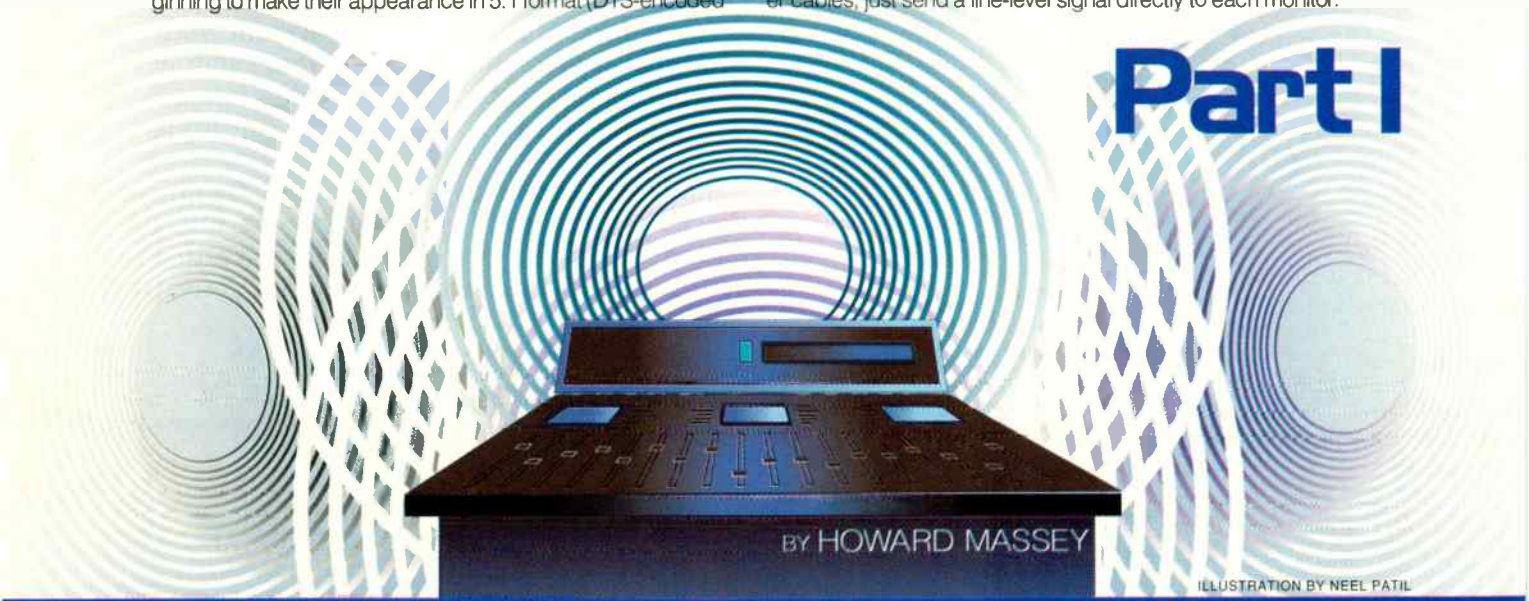
Suzanne readily complies when asked to expand upon the implications of the Internet for the independent recording artist. "The possibility of direct artist distribution through the Internet is very important. Even if the reality at this time is less than convincing in terms of financial rewards, the possibility itself, that there is a viable and growing alternative, has opened up criticism and exposure of music industry abuses. It has not exactly spawned a dialogue yet, but that is where we need to go.

"There will always be those artists who crave the endorsement of a major label and are willing to give up their 'rights.' And there will always be consumers who will buy physical CDs. But these are not the exclusive options anymore — and that is a fruitful situation. I share the view that music distribution should be modeled more on book publishing, where the author retains

▶ *continued on page 150*

Surround Sound in the Project Studio

Part I



BY HOWARD MASSEY

ILLUSTRATION BY NEEL PATIL

How to bring your project studio into the surround sound era

You've no doubt been hearing terms like "5.1," "repurposing," "downmixing," "bass management," and the like being bandied about over the past year or two. As is evidenced by the theme of this year's AES Convention ("Surrounded By Sound"), the audio industry has been re-energized with the coming of surround sound, and with that renewed enthusiasm comes a slew of new technologies and buzzwords. This three-part series will provide a primer in the basics of surround sound and, more importantly, will tell you what you need to do to get your project studio surround-ready.

Why this rapidly growing interest in surround? The main reason is that digital technology is finally at the point where it allows audio recordings to be accurately stored and reproduced in multiple channels (something that analog technology didn't allow, which is one reason Quad didn't take off in the '70s). But the conspiracy theorists in the industry will tell you that the real main reason is money — lots of it. Dozens of the world's largest manufacturers have poured billions of dollars into the development of DVD, banking on big returns in sales of players, speakers, and media. Be that as it may, the good news is that surround sound — where audio is coming from multiple speakers strategically positioned around you — makes for a listening experience that's much closer to reality than the decidedly artificial stereo format we've become accustomed to over the past fifty years.

Since its launch in late 1997, DVD-Video has proven to be the fastest-growing consumer format ever (even faster than VHS or CD). But it's the imminent arrival of DVD-Audio (with its much higher quality multi-channel audio) that's caused studios around the world to begin investing in surround sound gear. Major producers and engineers including Elliot Scheiner, George Massenburg, Dave Tickle, and Bob Margouleff are spending a good deal of their time doing surround sound remixes of classic albums and TV music specials, and new productions are beginning to make their appearance in 5.1 format (DTS-encoded CD and/or DVD-Video) as well as in traditional stereo format.

The Components

Let's start with the basic requirements. To get started doing surround sound mixing, you need the following:

1. A mixing board or audio interface with six buses in addition to the stereo bus. This could be an analog eight-bus console, just about any of the new digital boards, or a multi-output computer DAW. In a pinch, you could use *any* mixer that has at least six aux sends, but that's an awkward and somewhat inelegant solution.

2. A place to store eight tracks of audio. This could be a dedicated eight-track MDM (such as a TASCAM DA-88 or Alesis ADAT), but it could also be any six-track-or-more hard disk recorder, or any analog multitrack, for that matter. Bear in mind that these six tracks have to be in *addition* to the multiple tracks you'll be mixing, though they don't have to necessarily reside in a separate machine. For example, if you have two ADATs (which provide a total of sixteen tracks), you can use ten of those tracks for recording the song and the other six for storing the surround mix. On the other hand, it's good to have a dedicated eight-track recorder at your disposal for this purpose, since you can put a synchronized stereo mix on the other two tracks, allowing you to quickly switch between the two mixes for comparison purposes.

3. Five full-range monitors and at least one subwoofer, plus appropriate power amplifiers to drive them. If possible, the five full-range speakers should be the same make and model. But if you have two great stereo speakers that you want to integrate into the system, by all means do so, and then buy another three speakers that are as much like them (in terms of frequency range and response) as possible. Self-powered monitors simplify setup by eliminating the need to run speaker cables; just send a line-level signal directly to each monitor.

The truth is out there.

Twentieth-Century Fox, Newman Scoring Stage, Los Angeles, CA

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CIRCLE 101 ON FREE INFO CARD

Surround Sound in the Project Studio

Note that I said at *least* one subwoofer. Most industry experts recommend the use of two wherever possible, and producer and surround sound evangelist Bob Margouleff (see sidebar) agrees. "If you can afford two subwoofers, that's a better way to go. It gives you more dispersive energy so you don't have to work them as hard; therefore, you get better transient response."

4. A healthy dose of creativity. As we'll see in future installments, surround sound mixing is a whole lot more than just deciding which guitar track to put in front of you and which to put behind you. There are numerous pitfalls to avoid but there's also a whole new universe of sonic possibilities waiting to be explored.

One thing you *don't* have to necessarily buy is an encoding device, though it can be helpful to monitor through an encoder just to see how the conversion will affect your audio. (We'll talk more about encoding in a future installment.) Ultimately, encoding will probably be the province of the mastering studio; all you have to do for now in your project studio is get your six-channel mix together (in addition to — not instead of — your stereo mix).

That's it for this month, in upcoming installments, we'll talk about encoding, equipment interconnections, speaker placement and calibration, surround mixing tips and pitfalls, and more. Stay tuned to this space!

WEBLINK

Web-surfers can find extensive information on surround sound at www.dolby.com, www.dts online.com, www.tmhlabs.com, www.surroundassociates.com, and www.RobertMargouleff.com.

MARGOULEFF ON SURROUND

Bob Margouleff is a producer best known for his work with Stevie Wonder, as well as artists such as Devo and Oingo Boingo. These days, he's firmly entrenched in the front lines of surround sound, producing 5.1 remixes of Boyz II Men, Marvin Gaye, Tangerine Dream, and numerous classical recordings.

"The recording industry is now a hundred years old," Margouleff observes, "but what happened a hundred years ago is that we gave up one of our most important sensibilities: what I call 'vector' — that is, where the sound is coming from."

"Up until now, we have been attempting to create a three-dimensional image inside a two-dimensional space. That's why we invented panning, reverb, and echo — to create the illusion of depth in space. I think of stereo as being an objective experience — we create as much illusion of depth and space as we can, but in the end it is a two-dimensional medium, like a picture. Surround, on the other hand, is subjective audio. It allows the listener and the performer to occupy the same space."

"Don't forget," he adds, "that a hundred years ago, Berlioz wrote the *Symphonie Fantastique*, which was scored for the horns in the rear of the hall. Wagner wrote works for orchestras so large that musicians had to play under the stage, in the foyer, out in the audience. Church and sacred music has always been three-dimensional, and that's the reason it's so magical and mystical: The pipe organ was in the back of the church and the choir was in the front, with everything designed to fill the space surrounding them. That is a very, very powerful force that we threw away for a hundred years. Now we're finally at a place in the technology where we can deliver it."

Margouleff's project studio doubles as his living room. Nestled among the overstuffed sofas and wall hangings is a Yamaha 02R digital mixing console, a pair of TASCAM DA-98s, and a modest assortment of outboard processors. Strategically positioned around the console are five Genelec 1032A self-powered monitors and unobtrusively hugging the far wall are two Miller & Kreisel subwoofers.

"My studio," he comments, "converged in my living room, and I think that many project studios will ultimately do the same, sharing the same platform with the home theater."

Margouleff concludes, "Where I want to be going, and where I think a lot of musicians are going to want to go, is to write for this medium, to create new material based on the fact that we are no longer living in a two-dimensional space. We are going to return to the third sensibility — the cat's out of the bag, the horse is out of the barn, the genie's out of the bottle. We *are* going to step forward into this technology, and very soon."

Introducing
WaveCenter/PCI

OK, you see what's happening: digital mixers are looking pretty cool. After all, they've got incredible sonics, built-in effects, and the automation capabilities you could only dream about before. But if you hook that puppy up to the NoiseRacket analog soundcard that came with your computer, you're right back in ****ville. (Rhymes with "Snapville.")

Here's a better idea: keep it digital with our new **WaveCenter/PCI** card. It has all the connections you need to integrate your digital mixer into your computer-based studio. Transfer up to 10 channels of digital audio simultaneously using ADAT lightpipe and S/PDIF, with 24-bit resolution on all channels. Use one of the built-in MIDI ports for mixer automation, and the other to connect your synthesizers.

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BY MITCH GALLAGHER

Roland VP-9000 VARIPHRASE PROCESSOR

A NEW APPROACH TO SAMPLING AND PITCH, FORMANT, AND GROOVE MANIPULATION

For the past few years, many musicians, composers, and engineers have been wondering when Roland would release a new sampler. The company's last offering, the S-760, while still a great machine, is swiftly falling behind in features and especially in terms of polyphony and RAM capacity. So when Roland introduced the VP-9000 at the Winter NAMM show this year, a lot of musicians and engineers were anxious for a closer look. Was this the long-awaited update to the company's S-760 sampler? In a word, no. [Roland suggests their XV-5080 as a solution for playback of S-760 libraries.] Instead, Roland has given us a new technology and a new take on sampling and sound manipulation. The VP-9000 especially excels at phrase and loop manipulation, but there's much more to it than that.

Roland VP-9000

VARIPHRASE PROCESSOR

▶ROLAND VP-9000

MANUFACTURER: Roland Corporation, 5100 S. Eastern Ave., Los Angeles, CA 90040. Tel: 323-890-3700. Web: **www.rolandus.com**.

SUMMARY: A unique device that will change your concept of sampling and sound manipulation.

STRENGTHS: Real-time sample manipulation. Formant processing. Groove processing. Unique re-pitching ability. Processing available while sampling. Three independent effects processors. COSM effects algorithms.

WEAKNESSES: Limited polyphony. No crossfade looping. Keymap only splits 12 ways. No external word clock input. No keypad for number entry. Can't extract samples directly from an audio CD.

PRICE: $3,295 **EQ FREE LIT. #:** 122

Overview

The VP-9000 uses traditional Roland structure and terminology: Common, Parts, Performances, etc. If you're familiar with this from earlier products (back to the venerable D50) you'll have no trouble, otherwise you'll have to get acclimated. I had no problem diving right in and doing some fairly deep editing, even without cracking the manual. Speaking of the manual, two are provided, a Reference book, which offers detailed coverage of various features and parameters, and the User Guide, which is a tutorial-style step-by-step guide to accomplishing common tasks.

Here's the rundown: The basic sound source in the VP-9000 is the sample. Samples are assigned to phrases, which is where the sample is assigned to a key, etc. Phrases make up Parts, which contain global parameters for the phrases. Six Parts, each of which can be on a separate MIDI channel, make up a Performance. A limitation you'll bump into fairly quickly is the VP-9000's six-voice polyphony. After becoming accustomed to boxes providing up to 256 voices, six seems pretty meager, especially since stereo samples use up two voices of polyphony. But you truly can do a *lot* with those six voices.

There are three independent effects processors in the VP-9000, one a multi-effects, the other two dedicated to chorus and reverb, respectively. Each Part has its own effect send level, and each Part, as well as each effects processor, can be routed to one of three sets of stereo outputs. The effects all sound good. Included are COSM modeled effects such as various Boss stompboxes and guitar amp simulators, bit rate conversion, resonant filters, and more.

The VP-9000 comes stock with 8 MB of RAM, but can be expanded to a total of 136 MB with the addition of four 32 MB SIMMs. Installation of RAM is a breeze; under a "hatch"

ROLAND VP-9000 SPECS

Analog I/O	Mono 1/4-inch balanced input (front panel @ –50 to –10 dBm) Stereo 1/4-inch balanced inputs (rear panel) @ –20, 0, or +4 dBm) 6 1/4-inch balanced outputs (balanced +10 dBm, unbalanced +4 dBm)
Digital I/O	S/PDIF optical and coaxial
MIDI I/O	In, out, thru
Other I/O	Headphone out, 25- and 50-pin SCSI connectors, 3-pin IEC power connector
Internal disk drive	250 MB Zip
External backup	Optional CD-R, CD-RW
Maximum RAM capacity	136 MB (8 MB stock)
Maximum sample time	8 MB: 25 seconds/stereo, 50 seconds/mono 136 MB: 425 seconds/stereo, 850 seconds/mono
Maximum samples in RAM	1,024
Sampling frequency	Sampling/playback: 32, 44.1, 48 kHz Importing: 8, 11, 15, 16, 22.05, 24, 30, 32, 44.1, 48 kHz Internal: 44.1 kHz
Data resolution	16-bit linear
Internal processing	24-bit
A/D and D/A resolution	20-bit
Frequency response	20 Hz-20 kHz (–3 dB)
Polyphony	6 voices (stereo samples use 2 voices)
Multitimbral capability	6 part
Effects processors	3 (multi-effects, chorus, reverb)
Effects types	40 multi-effects, 8 chorus types, 9 reverb types
Supported sample file formats	AIFF, WAV, Roland S-700 series, Akai S-1000
Sample input processors	Compressor, limiter, noise suppressor

PHRAZER

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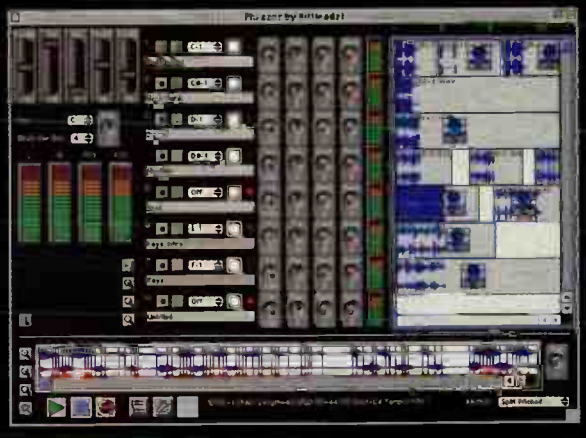

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Roland VP-9000

VARIPHRASE PROCESSOR

• **Solo** — for dry (no effects) monophonic vocal and instrumental lines
• **Backing** — for rhythmic, percussive material
• **Ensemble** — for synth pads, choirs, and similar sounds with complex harmonic structures and/or effects

You also need to establish the sample "mode," which has three aspects: First, whether the sample will play monophonically or polyphonically. Second, how the sample will react when a note is playing and another note is triggered. The sample can re-trigger (play from the beginning with each key strike), Time Sync (playback for the new note starts at the playback point of the current note), Legato (playback takes over from the previous note, but at a new pitch), and Step (note plays back until it reaches the next Event, and stops — see below for more on Events). Third, there are three trigger settings, gate (sample plays for as long as a key is held), trigger (pressing a key either stops or starts playback), and drum (sample plays back in its entirety each time a key is pressed).

In addition to basic sample editing, you can get deep into programming the response of a sample; looping, velocity response and curve, level, key follow, LFOs, portamento, and more are all provided.

The VP-9000 can automatically divide a sample into individual "Events" based on the transients in the sound; each of those Events can then be assigned to a separate key. This lets you break up a drum loop into its component hits and assign them to notes so you can play them from a keyboard. If necessary, you can go in and manually add or delete Events. This lets you split a sample that the VP-9000 might not otherwise be able to deal with as well as correct the VP-9000 if it mistakenly identifies (or misses) a particular Event you want to include.

For all this power, the VP-9000 isn't really set up to serve as a conventional sampler for playing, say, piano sounds. A keymap is limited to twelve one-note zones/splits on consecutive notes on the keyboard, and there's no provision for velocity switching, etc. You could set all six Parts to the same MIDI channel, set up keymaps for each that cover different ranges, and make an "instrument." I actually did this, but it's not worth the bother. The bottom line: The VP-9000 won't replace your conventional sampler for playing instrument multisamples from a keyboard or sequencer.

Variphrase

The technology behind the VP-9000 is something Roland refers to as "Variphrase." Essentially Variphrase allows you to independently process the pitch, tempo, formant, and groove aspects of sampled material in real time. It's possible to do similar pitch and tempo manipulation in software applications such as Sonic Foundry's Acid, but also having access to formant and groove manipulation and real-time control moves the VP-9000 to an entirely different level.

Three knobs at the upper right of the front panel, collectively referred to as the Variphrase Controls, are the heart of controlling the process. Their labels describe what they control: Pitch, Tempo, and Formant/Groove. This emphasizes that there's a lot you can do with the VP-9000 in real time. Grab these three knobs, and the parameters they control will follow, whether you're changing the tempo, pitch, or modifying the groove or formant. You can also toggle the effects processors on and off from dedicated front panel buttons.

> *"In order for the VP-9000 to work its magic, samples need to be 'trained' or encoded with their original pitch and tempo."*

Groove can do some pretty impressive things. You can make a swung rhythm sound straight or make a straight rhythm swing, which is amazing to hear. But I found it even more fun to push phrases past where they were happy, which results in odd distortions and resonances.

The tempo control is effective through a fairly wide range. As with similar time compress/expand processes, once you get outside the effective range, you'll start to hear artifacting — but as with the Groove

V-PRODUCER

At the AES show (September 22–25 in Los Angeles), Roland will announce new V-Producer software for use with the VP-9000. V-Producer will function as a computer-based sequencing and encoding tool for VP-9000 samples and parameters. This includes on-screen arrangement of audio phrases, six-channel mixing and effects setup, plus more precise graphic editing of pitch, time, and formant. With the accompanying V-Trainer software, more sophisticated wave training (encoding) can be performed on WAV and AIFF format files. V-Trainer can also batch-encode multiple samples.

V-Producer can sync to other software using MIDI clock or MTC, and complete V-Producer arrangements, including phrase arrangements can be saved as Standard MIDI Files. Up to six VP-9000's can be controlled by V-Producer for a complete Vari-Phrase-based audio workstation.

The V-Producer software will initially be available for PC with a Mac version slated for 2001. Current system requirements include a 233 MHz Pentium MMX processor (Pentium II or greater recommended) running Windows98, 64 MB RAM or greater, a minimum of 20 MB available hard disk space, and a Zip drive (when using V-Trainer). Price had not yet been announced at press time; V-Producer will be available through authorized Roland dealers.

processing, this can be pretty cool.

Formant lets you play with the sound's resonant characteristics – you can convert a male voice into a female-sounding voice, and vice-versa. But you don't have to go that far. For me, the Formant control was more effective for subtly adjusting the resonance of a vocal phrase as it was pitch shifted, particularly when trying to create stacked harmonies from a single melodic sample. The pitch shift algorithm in the VP-9000 is very good over a range of about an octave. Beyond that, you'll hear it. But when you combine the pitch shift with the formant control you can sometimes stretch this and keep things sounding natural.

It's hard to describe exactly what you can do with some of these features. At the most *basic*, you can load in samples with differ-

▶ *continued on page 114*

In REVIEW

Amek Channel in a Box Mic/Line Preamp/EQ/Compressor

Rupert Neve's latest preamp/EQ/compressor design is a pure winner

Upon receiving the Amek Channel in a Box (CIB), I eagerly mounted it in my rack and plugged it in. The first thing I noticed were two small, glowing icons — the Amek logo in blue against a white background, and the *Pure Path* logo in white against a blue background. With these two backlit logos, in combination with a plethora of red, yellow, and green LEDs and two LED-style meters, the CIB certainly looks cool enough! Overall, it's got a decidedly vintage look but with a bit of a modern twist, featuring white lettering on alternating backgrounds of dark gray and dark blue. The lettering was a little small for my eyes at first, but after awhile it didn't really bother me. The meters are easy to see and the main gain knobs are larger and have a yellow face as opposed to the white-faced knobs used for everything else. The buttons are backlit and color-coded so it's easy to see what's operational at a glance. Overall, I felt the unit was laid out in a fashion conducive to getting work done.

Mic/Line Pre

The CIB's mic preamp sounds fabulous! It has a very detailed character that's warm, silky, smooth and very musical. It doesn't exhibit noticeable coloration and seems to work well on a wide array of microphones. I tested it with a Soundelux U99 large diaphragm tube mic, a Royer R-121 ribbon mic, a Rode Classic tube mic, a Beyer Dynamic MC834 condenser mic, an Audio Technica AT4041 condenser mic, a Shure SM57 dynamic mic, and an AKB D112 dynamic mic. I used it on vocals, acoustic guitar, electric guitar, kick drum, and a variety of miscellaneous percussion instruments. No matter what I threw at it, it performed flawlessly. It was significantly better than Mackie VLZ mic preamps in both noise and overall sonic character; in general, it seemed fuller, cleaner, and more natural. I also compared it against an Oram MWS and, while they were similar in sound quality, I picked the CIB in most blind A-B tests. When used on guitars and kick drum,

By Rob McGaughey · soundsauna@earthlink.net

it exhibited fullness in the low end without any degree of tubbiness. In my opinion, this is one of the best mic preamps on the market.

The mic pre section of the CIB features two gain controls. One is stepped in 6 dB increments, while the other has only a center (0 dB) detent and offers the ability to fine-tune the gain. I really like this two-knob approach to trimming a mic pre; it's both easy and precise. There's also two backlit buttons for selecting +48V phantom power and for polarity reverse. An LED-style meter is available to monitor the input level; it's switchable between the mic and line inputs. Both inputs also have an overload LED that illuminates 4 dB before clipping.

The line preamp has its own set of inputs and works simultaneously with the mic preamp. It features a single trim control, a polarity reverse button, and shares the aforementioned LED-style input meter with the mic preamp. I ran a keyboard through the CIB into Pro Tools and compared it with running the same keyboard through a Mackie MS1402VLZ as well as running direct into the Digidesign 888 I/O input. The CIB, with no EQ or compression, exhibited a fuller sound, particularly in the low end. Another interesting feature is that the mic preamp is capable of handling +22 dB of level when set at unity gain, so you can use the CIB for two simultaneous line level inputs.

> *"The CIB's mic preamp sounds fabulous! It has a very detailed character that's warm, silky, smooth and very musical."*

Equalization

The EQ section of the CIB is split into three parts. Each of these shares a few common features. They're all capable of 18 dB of gain per filter, and each can be independently assigned to either the mic, line, or side-chain path. The first section, Filters, comprises high- and low-pass filters. The high-pass filter is continuously variable from 22–300 Hz, while the low-pass is variable from 2.5–25 kHz. Both filters employ an 18 dB/octave slope. The high-pass filter worked great for removing rumble on a vocal track.

The next section appears at first glance to be a pair of simple frequency-adjustable shelving filters. Actually, it's a lot more than that. The low-frequency component operates over a range of 30–300 Hz, with the ability to select one of four different filter shapes. In the standard shelf mode, the response curve remains flat up to the chosen frequency, where it rolls off at 12 dB/octave. Or, by depressing the Peak switch, you can select a more "bell-shaped" curve with a steep Q. This allows you to hone in on a certain aspect of the signal. A third option is "Glow." This feature makes the shape of the filter much less steep-sided, providing or removing a certain warmth. The fourth option is to select Peak and Glow simultaneously for a filter shape that falls somewhere between the two individual shapes.

The high-frequency shelving filter takes a similar approach, operating across a frequency range of 2–20 kHz. It also offers four different shapes: a standard shelf, peak (as above), Sheen (similar to Glow), and a combination of Peak and Sheen. The LF and HF shelving filters can be assigned in tandem to the mic, line, or the sidechain section of the compressor.

The third EQ section is a pair of traditional fully parametric equalizers. The Low Mid Frequency (LMF) equalizer is sweepable from 20 Hz to 1 kHz, while the High Mid Frequency (HMF) equalizer is sweepable from 500 Hz to 25 kHz. A Q control, which defines the bandwidth over which the EQ is applied, is adjustable independently for each band.

The EQ section of the CIB is easy to use but has amazing flexibility. Once you get a feel for how each of the filters sounds, you can quickly dial in the sound you want. I used it on vocals, guitars, keyboards, percussion, and kick drum. Without fail, the CIB EQ performed flawlessly — I particularly liked the Glow and Sheen filters. On acoustic guitar, the Sheen filter was outstanding, while the Glow filter added some nice girth to an electric guitar. Overall, I felt the equalizer was one of the best I've worked with.

Compressor

At this point you probably won't be surprised when I say the compressor section of the CIB is outstanding, too. It offers a threshold adjustable from –40 dB to +20 dB (according to manual, the front panel is labeled to +23 dB, and in fact the control turns past that point), an attack control adjustable from 0.3 to 300 ms, a release control adjustable from 0.1 to 10 seconds, a ratio adjustable from 1:1 to 40:1, and an output gain that ranges from –6 dB to +18 dB (again, per the manual; the front panel is labeled from –10 to +20). The compressor can be assigned to either the mic or line path. A Key switch routes the sidechain input to the opposite input of whichever one the compressor is currently assigned to. The sidechain path has an overload LED that illuminates 4 dB before clipping.

The compressor section provides an

▶ *continued on page 114*

▶AMEK CHANNEL IN A BOX

MANUFACTURER: Amek, Langley House, Third Avenue, Trafford Park Manchester M17 1FG, United Kingdom. Tel: +44 (0) 161-868-2400. Web: **www.amek.com** or **www.purepath.co.uk**.

SUMMARY: A rackmount "channel strip" from the renowned designer Rupert Neve that's ideal for tracking and mixing applications.

STRENGTHS: Pristine audio path. Mic preamp is one of the best. Glow, Sheen, and Peak features. Flexible and full-featured compressor.

WEAKNESSES: No front panel power switch. No master EQ bypass switch to turn off all sections simultaneously.

PRICE: $3,250

EQ FREE LIT. #: 118

AMEK CHANNEL-IN-A-BOX

► continued from page 113

LED-style gain reduction meter so you can easily see what the compressor is doing. There's also a Link switch that allows you to apply identical gain reduction across multiple CIBs, based on whichever path has the greatest signal. The most interesting button in the compressor section is labeled "&MM." This changes the compression curve to a softer knee that, according to Rupert Neve, "musically emulates the best of my past designs."

In practice, the compressor, like everything else on the CIB, was awesome. It could be used to provide anything from subtle compression to heavy limiting. The "&MM" feature was really useful; it allowed me to dynamically control audio signals in a more musical fashion. The sidechain and its integration with the equalizer, the true dual-path design, and the Key switch all serve to make the compressor a highly versatile tool.

Outputs

There's an independent balanced output for both the Mic and Line paths with a rotary knob to set the level, as well as a button that allows you to switch the input meter from pre-fader to post-fader for a true view of the output level. If you're into faders instead of knobs, an XLR connector lets you connect a remote fader that supplants the rotary knob on the output section for both the Mic and Line paths.

Conclusions

The CIB is the real thing — not something trying to emulate the real thing. It's also more versatile than it appears at first glance, since there's two completely separate audio paths (mic and line), with flexibility in assigning the EQ and compressor. $3,250 may seem steep, but the Amek Channel-In-A-Box will be worth it for those who demand the best. Rupert Neve has done it again — this is one truly awesome piece of audio gear.

Rob McGaughey owns and operates Sound Sauna Studios in Pittsboro, Indiana.

ROLAND VP-9000

► continued from page 111

ent original pitches and tempos, and the VP-9000 will make them all play back in perfect sync, for both pitch and tempo. You can grab the front panel controls and smoothly change the pitch, tempo, and formant/groove of a Part (or any combination of these for any combination of the six parts) and they'll all stay tightly locked together. But you can go far beyond these kinds of things; by pushing the various parameters, you can create entirely new sounds as well as twist, wreck, and mangle old clichés into fresh sounds — and keep in mind, *you can do all of this in real time.*

Loading up six sample phrases at different tempos and pitches, and having them all line up perfectly in time and pitch is a trip — and it becomes even more so once you start cranking on the Variphrase knobs as the sounds play back. Even if you've worked with Acid, there's something cool about playing the sounds from a keyboard and tweaking them from a front-panel knob.

When in solo/legato mode, you can do some amazing things, such as changing the pitch of notes within a sampled melodic phrase. You can even make a melodic phrase play back at one pitch, or take a phrase that was sung in a monotone and give it a melody. How well this works depends on a variety of factors. In some cases you'll hear the pitch skip to the new note, but by either programming or playing the MIDI control notes carefully, you can get astonishing results. While you can "perform" this kind thing from a MIDI keyboard, it really comes together under the control of a MIDI sequencer. I sampled phrases into the VP-9000, then set up MIDI tracks in Digital Performer to re-pitch their melodies. I was quite happy with the results, since I could really get in and tweak the MIDI control data in the sequencer.

Conclusions

The VP-9000 is one amazing box. If you're looking for new, fast, real-time ways to work with loops and phrases, you want to manipulate vocals, or if you want to get into creating new sounds, this box is for you. It's not cheap, and you only get six-voice polyphony, but overall the VP-9000 packs as much creative power into two rackspaces as anything I've worked with. I love it. ■

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By Howard Massey · hmassey@uemedia.com

Apogee AP8AD and AP8DA I/O Cards for Yamaha Digital Products

Upgrade the converters on your Yamaha 03D, 02R, PM1D, or D24.

After you've been working with digital audio long enough, you realize that its sonic qualities are largely a function of the converters being used, along with, to a lesser extent, the degree of jitter in the signal. That's why products made by manufacturers such as Apogee — widely regarded as one of the premier names in the converter field — are so prized.

With the release of their AP8DA and AP8AD expansion cards, Apogee allows you to add eight channels of Apogee-quality conversion directly to a Yamaha 02R or PM1D mixer, or to the Yamaha D24 MO recorder. Designed to fit into either a standard or short ("Mini-Y") YGDAI slot (each card takes a single slot), both products were also initially slated to work with the even lower-cost Yamaha 01V mixer, but currently don't do so. Although the manual doesn't reflect this, Apogee says the cards are now compatible with the Yamaha 03D mixer. One caveat: if you're installing a card in a "Mini-Y" device such as the PM1D or D24, you need to physically break off a couple of stabilizer strips. This is a simple operation, but once it's done, it can't be undone, so the card can't be reused in a full-length YGDAI device such as an 02R.

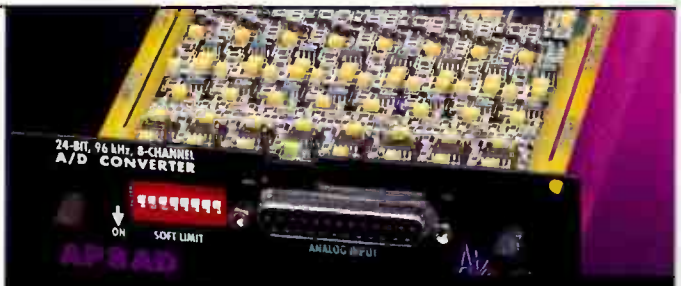
I had no problems installing the two cards in my 02R, which, after power-up, recognized them as "analog" cards. However, the 02R can't distinguish between the input (AP8AD) and output (AP8DA) functions, so an output assignment screen appears for the AP8AD, even though it serves no purpose.

The two cards can coexist comfortably inside the same host product, giving you high quality signal input as well as output (in fact, up to four Apogee cards can be installed in an 02R). But according to the manual I received, if an AP8AD is installed in slot 3 or 4 of an 02R (the upper slots), you can't put an AP8DA in the slot immediately below it. Apogee tells me that this problem has since been resolved; four cards should now work in the 02R regardless of slot positions.

Both cards utilize Apogee's renowned Ultra-Low Jitter Clock, and this undoubtedly has a lot to do with why they sound so good. But the main reason is the quality of the converters themselves, which are Apogee's proprietary 24-bit design, as opposed to the 20-bit resolution used by the 02R's stock converters. Both cards operate happily at either 44.1 or 48 kHz sampling rates and there are jumpers that allow them to operate at 88.2 and 96k, even though some Yamaha products (such as the 02R) don't support them yet. If and when Yamaha upgrades their products to allow for these higher rates, the Apogee cards will be ready to go with the flow. The brief but informative owners manual states that an upgrade will be made available at that time that will allow automatic sample rate selection, negating the need to physically remove the card(s) and change the jumpers.

Speaking of jumpers, both cards provide a set that changes the level setting for each channel independently. You've got your choice of three different levels: –10 dBV or either of two +4 dBu settings: 0 VU = 14 dBFS (18 dBu max) or 0 VU = 20 dBFS (24 dBu max). This allows the cards to interface with a wide variety of equipment.

Getting signal into or out of the cards requires a DB25 breakout cable (the same type and pinout used with, for example, a TASCAM DA-88), since that's the sole connector provided. This kind of cable can set you back 100 bucks or more,



▶APOGEE AP8AD & AP8DA

MANUFACTURER: Apogee Electronics, 3145 Donald Douglas Loop South, Santa Monica, CA 90405. Tel: 310-915-1000. Web: **www.apogeedigital.com**.

SUMMARY: Adds Apogee's high-end converters to your Yamaha 02R, 03D or PM1D mixer, or to the D24 recorder. A great way to transform analog signal to digital or vice-versa without having to go through external converters.

STRENGTHS: Great sound. Easy to install. Less expensive than eight channels of outboard Apogee converters. Can be set to any of three operating levels on a per-channel basis. Cards support 88.2/96 kHz sampling rates. AP8AD includes Soft Limit.

WEAKNESSES: Once installed in a "Mini-Y" device (i.e., PM1D or D24), card can't be used again in a full-size YGDAI device (i.e., 02R). DB25 breakout cable required but not included. AP8DA outputs pop when changing clock source or sample rate. For external sync, AP8AD requires clock signals that meet a certain standard of quality.

PRICE: AP8DA: $ 1,195; AP8AD: $ 1,495

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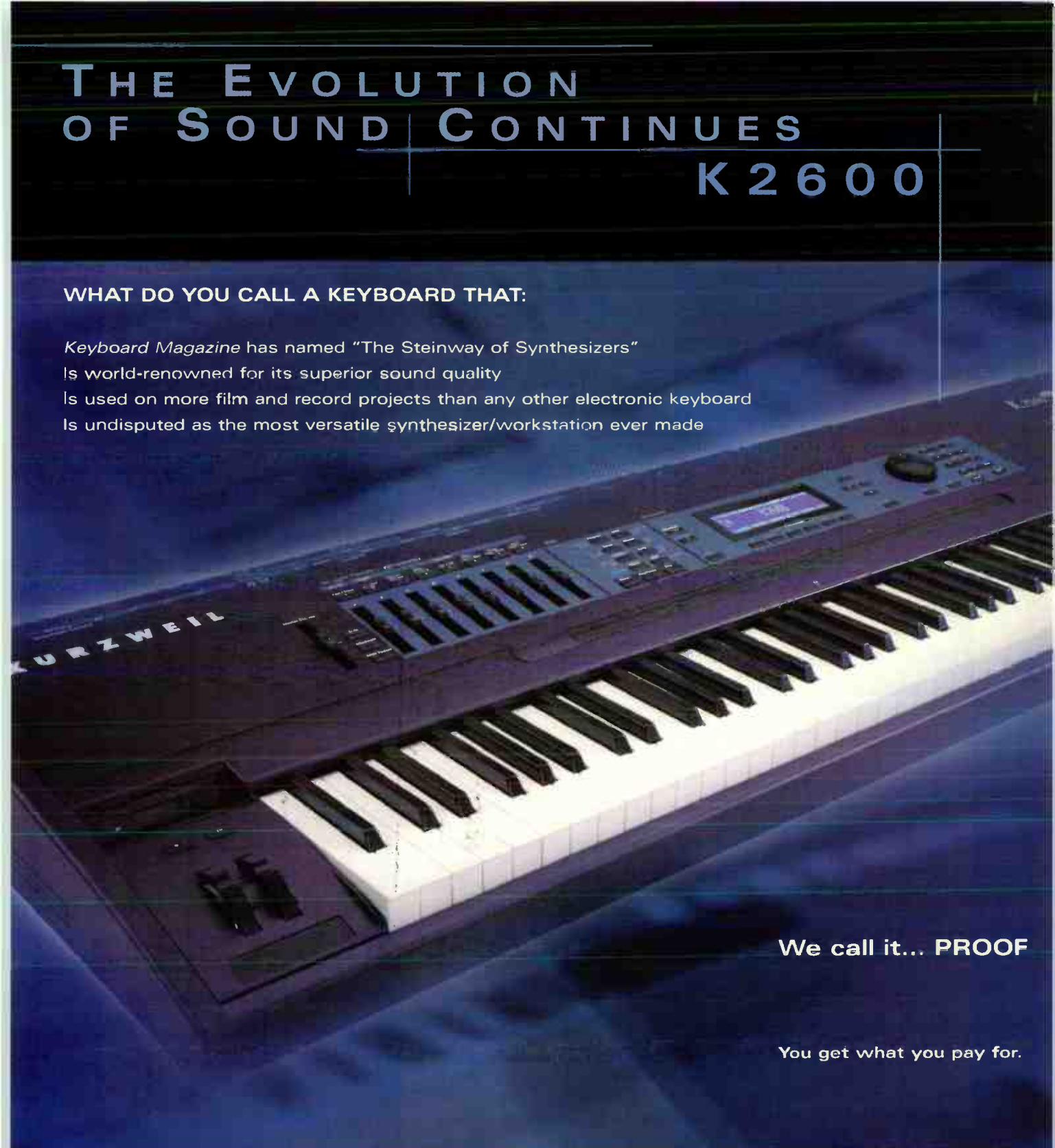

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unless you're willing to solder one up yourself. It's probably too much to expect that such a cable would be included with the card (it's not), but be aware that you're going to need to budget for one when you purchase one of these cards.

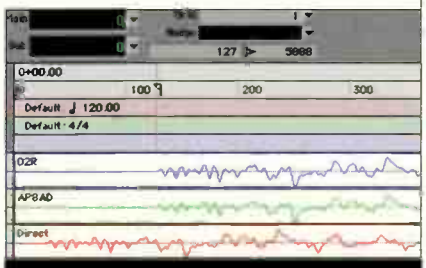

FIGURE 1. A snare drum hit was simultaneously sent to Pro Tools, an 02R input, and an AP8AD input. The latter two signals were brought into Pro Tools; all three signals were routed in the analog domain. The latencies of both the 02R and AP8AD are very low — 127 samples at 44.1 kHz, or just under 3 msec. More importantly, their latency is identical, so there are no time delays, phase cancellations, or smearing when using them together.

Having covered their common points, let's take a closer look at each card individually.

AP8DA

After installing both the AP8DA and Yamaha's CD8-AD card in my 02R, I used a balanced line-level switcher to alternately route the signal from each to a pair of Genelec 1029A monitors and a Genelec 1092A subwoofer. With the AP8DA's jumpers set to 0 VU = 24 dBu and the Yamaha card at its +4 level setting, my SPL meter confirmed that both cards were perfectly level-matched when outputting pink noise, so it was time to begin listening to some music!

The difference between the analog output of the two cards is clearly audible, though subtle. Audio played through the AP8DA had improved localization and better definition, particularly of low-level signals. The same signal played through the Yamaha card seemed somewhat two-dimensional, as if the sound were coming from a flat plane directly in front of the listener, whereas the Apogee card appeared to have a broader spread with greater front-back depth. This effect was even more apparent when listening to 5.1 surround sound material; each sonic component was placed more identifiably within the soundfield, and there was a greater degree of separation between the instruments. If I had to use one word to summarize the difference between the two, it would be "focus" — the sound of the AP8DA is clearer and more delineated than the Yamaha card.

The only negative is an audible popping when powering up with the AP8DA installed, changing sample rate, or going from internal clock to external word clock (or vice-versa). Apogee tells us that this is a result of using their low jitter clock for sync — the upside is an improvement in audio performance, the downside is that it takes a tenth of a second to lock, during which time this noise is emitted. Most studios follow the wise policy of turning on power amps/powered speakers last, but I'd also recommend powering down your monitors or turning their levels down (if they're connected post-monitor level control) before changing clocks or sample rates.

AP8AD

As before, my first step was to confirm that the levels between the 24-bit Apogee AP8AD card and the Yamaha's own inputs were matched. According to Apogee, the levels should be the same when the AP8AD jumpers are removed altogether (so that 0 VU = 24 dBu) and when an 02R input channel trim is set to minimum and its 20 dB pad is engaged. However, my SPL meter showed that the Yamaha inputs were 0.7 dB hotter than the Apogee's, so the appropriate channel faders were adjusted accordingly before beginning another round of extensive listening. (Apogee says this difference is due to a discrepancy in Yamaha's published specs.)

Interestingly, I felt that the sonic distinctions between the two A/D cards were more subtle than the differences between the D/A cards, though *EQ* editor Mitch Gallagher (who joined me for these listening tests) found the reverse to be true. But we both agreed that, especially on source material with high-frequency transients such as cymbals, shakers, and steel-string acoustic guitars, the sound of the Apogee card was more open, more airy. In short, there's the extra definition and greater dynamic range you'd expect when working with 24-bit signals instead of 20-bit (not to mention Apogee's low jitter clock). On the other hand, the stock 02R inputs imparted a slightly rounder tone that complemented other kinds of material — for example, heavy electric guitars or synth pads. These timbral variations provide a good reason for adding an AP8AD to your device's stock inputs; doing so gives you two different input options, each with a subtly different sonic signature.

Another feature offered by the AP8AD is Apogee's proprietary Soft Limit technology. This uses a fast attack and a logarithmic curve to tame overly hot input peaks. (See Figure 2.) Soft Limit can be turned on or off for individual channels using DIP switches thoughtfully located on the rear of the card — a smart design that precludes having to remove the card to access them. There's no adjustable threshold, so limiting doesn't actually kick in until you input sufficiently hot levels.

With the 02R's internal clock as master, the AP8AD is perfectly happy, but when syncing to other gear, the AP8AD requires external clock signals that meet a certain standard of quality. If you use a less stable external word clock, the AP8AD may generate some digital hash when changing sample rates. (You can mute the noise by pulling down the associated input faders or hitting the channel mute buttons.) The AP8AD grumbled when locked to word clock from the BRC in my rig, but when I tried it with another BRC, the problem lessened considerably; when I used an Apogee PSX-100 as clock source, all hash and noise disappeared entirely.

The latency of the AP8AD is remarkably low — some 127 samples at 44.1 kHz, or less than 3 msec — and is identical to that of the 02R's onboard A/D converters, so you won't have any time smearing or phase cancellations if you're bringing signal into the board using both types of A/Ds. (See Figure 1.)

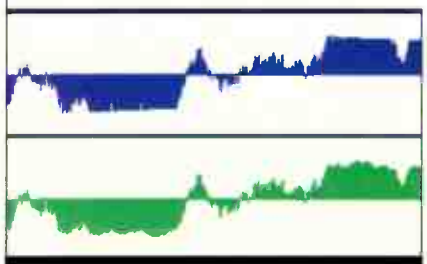

FIGURE 2. Compare the hard clipping and squared-off waves in the upper, non-limited signal with the rounder, less-distorted waves in the lower signal, which has been Soft Limited by the AP8AD. Also note that lower-level signal components are unaffected and therefore identical.

The Wrap-Up

The bottom line is that both the AP8DA and AP8AD provide audible improvement in the performance of the host equipment. During tracking, at mixdown, or if you're doing a lot of mastering work or critical listening, either or both of these cards can be an important add-on to your Yamaha console or MO recorder.

Howard Massey's latest book — *Behind The Glass* (Miller-Freeman Books) — is a collection of interviews with record producers.

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CIRCLE 08 ON FREE INFO CARD

By Steve La Cerra· slacerra@uemedia.com

Drawmer 1969 Mercenary Edition Vacuum Tube Mic Preamp/Compressor

A new take on the venerable Drawmer 1960 mic preamp/ compressor

Drawmer's 1969 Mercenary Edition Vacuum Tube Compressor is a two-channel unit with integral microphone preamps, as well as an instrument-level aux input with passive tone control. A modified version of the company's successful 1960, the 1969 Mercenary Edition was largely developed at the request of pro audio maven Fletcher of Mercenary Audio.

The circuitry of the 1969 is a hybrid design combining Burr Brown IC's with tube stages in the VCAs, and in the mic and instrument-level preamplifiers. Housed in a two-space rack chassis, the unit includes rear-panel jacks for XLR mic in, line in and out, 1/4-inch TS unbalanced sidechain input, and TRS insert jacks (ring = send, tip = return). Unlike just about every piece of outboard gear we've seen, the 1969 provides separate insert paths for –10 dBV and +4 dBu devices. A front-panel 1/4-inch TS jack for instrument input completes the unit's I/O facilities.

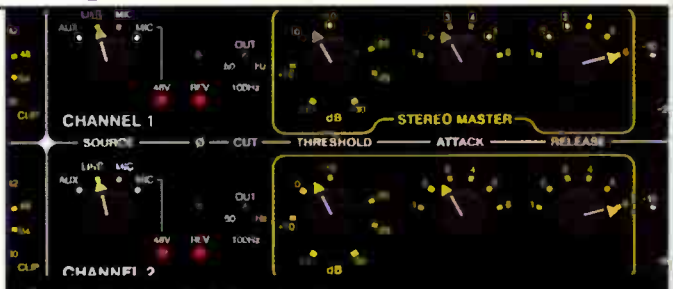
Front-panel controls for each channel include mic gain, source select (aux, line, mic, and mic with 48-volt phantom), polarity reverse, low-frequency rolloff (at 50 or 100 Hz), threshold, attack, release, and output gain. A VU meter is provided for each channel and may be switched to indicate either gain reduction or output level (very useful). In addition to the "normal" output of the compressor, a switch allows bypass or sidechain monitoring for each channel. At the far right side of the front panel are a power switch and a three-position stereo link switch, each with a trés-chic blue LED. Controls for the single aux input include gain, bass, treble, input sensitivity, EQ bypass, and normal/bright.

We found the concise manual to be useful, particularly the "quick set up" section. Simply patching a signal into one of the channels and setting the front-panel controls as per the quick guide got us off and running with excellent sonic results.

In Use

We used the 1969 Mercenary Edition in a variety of sessions for just about every audio source we could get our hands on. To get a feel for the microphone preamp's character we initially bypassed the compressor so that we were hearing only the preamp. Preamp gain is adjustable in 6 dB steps from 0 to 66 dB. It'd be nice to have finer resolution for situations where you're using the preamp with the compressor bypassed, because we sometimes found one setting too low and the next too high at our tape machine meter. When the comp is "in-circuit" you can use the continuous output level control to fine-tune signal level to any setting.

A clip light accompanies the mic pre gain control, but nowhere in the manual does Drawmer specify the signal level at which this LED turns on (though the manual does say it "illuminates when excessive mic gain has been applied and there is a danger of clipping"). It's also unclear if it indicates clipping at the input stage or after the mic amp. This irked us a bit because — judging from the signal level at our tape deck — the LED turned on way too early. When miking kick or snare drums we weren't sure if "clip" was lighting because the mic's output was too hot, or if we were in fact applying too much gain (if the 1969 had a pad it'd be easy enough to determine this). We decided to ignore the LED and trust our ears — which didn't indicate any distortion when the clip LED blinked on transients.



▶DRAWMER 1969

MANUFACTURER: Drawmer, dist. in the U.S. by Transamerica Audio Group, 4760 West Dewey, Ste 133, Las Vegas, NV 89118. Tel: 702-365-5155. Web: **www.transaudiogroup.com**.

SUMMARY: Two-channel vacuum tube compressor with built-in mic preamps.

STRENGTHS: Excellent sound quality. Capable of wide tonal variation. Separate insert loops at both –10 dBV and +4 dBu.

WEAKNESSES: Mic preamp clipping indication is vague. No pad on mic input.

PRICE: $2,950

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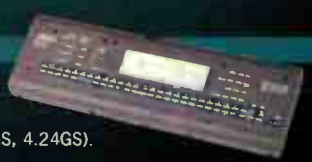

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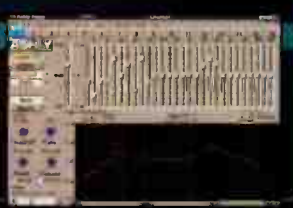

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After A/B'ing the 1969's pre against the other preamps in our studio, it became clear that its top end — from about 4 kHz and up — is extremely transparent and open, revealing as much detail in the source as a mic is capable of capturing. This characteristic was particularly nice on steel-string acoustic guitar, where the 1969 accurately translated the subtleties of attack on the strings as well as the resonance of the guitar body, and the ambient environment surrounding the instrument. A Shure SM57 on snare sounded great, with a quick transient response and meaty low-mids, while a CAD E-350 on a kick drum produced excellent low-frequency extension.

Since the unit compresses via "soft knee" principle, compression ratio is dependent mostly upon the setting of the threshold control. As you move the threshold control lower, the compression ratio gradually and very smoothly increases. To decrease the compression ratio, you raise the threshold. The attack time setting affects the compression ratio as well, though more subtly than the threshold. For a given threshold setting, shortening the attack time will increase compression ratio; lengthening the attack time will reduce the ratio. Once you grasp the concept, operation is a breeze.

Attack time may be switched between 2, 8, 15, 30, and 50 millisecond steps. Release times vary between three fixed (100 ms, 500 ms, and 1 second) and three program-dependent times (200 ms to 2 seconds, 500 ms to 5 seconds, and 1 to 10 seconds). Someone at Drawmer has definitely been doing their homework because we never felt like we needed a step between the attack or release settings (and of course the fixed steps allow precise repeatability).

An incredibly diverse tonal variety can be achieved by playing with the attack and threshold controls. For drum overheads we set attack to "2," and the release to "2" or "3." By adjusting the threshold from 0 down to around –10, compression is transparent; just enough to smooth out peaks. Bringing the threshold down below –10 and making the attack faster creates an emphasis on tom and cymbal decays. Drop the threshold low enough and you'll get that classic pumping-overhead-room tone. Ditto for vocals, which can be compressed "obviously" by using fast attack times and low thresholds or — by slowing the attack down to "3" or "4" — compressed transparently. A fast attack on kick drum resulted in a softening of the slap of the beater hitting the head, which in this case helped the kick sit nicely in the mix (a similar result may be obtained on bass). Using the mic pre and the comp together made for a great "direct-to-tape" path and the personality of the mic pre came through the compression process unscathed.

For stereo program material the 1969 offers a link switch with two modes where threshold, attack, and release settings from channel one serve as the stereo master. Due to the unit's discrete two-channel design, output level controls must be carefully matched. But once this is done, the channels track together without any shifting of the stereo image. In addition to link "on" there's a "big" link mode that makes the compressor less sensitive to bass frequencies. If engaged, the 1969 avoids ducking the entire program when strong bass frequencies occur (the type of FM-radio suck-down that occurs whenever a kick drum drops in on a ballad). Useful when strapping the unit across a stereo mix bus, this feature works well to restore the 1969's transparency, somehow leaving the bottom end untouched while compressing the rest of the frequency range (and yes — it somehow does make a mix sound bigger).

Drawmer definitely paid careful attention to detail in the design of the 1969. For example, the polarity reverse applies to both the line and mic inputs — as opposed to many preamps where polarity reverse only affects the mic signal. Ditto for the low-cut filters, which reduce problems such as mic stand rumble without obliterating the bottom register. The ability to insert at either +4 or –10 is a definite strength in matching external gear to the 1969 without losing headroom or creating distortion.

Conclusions

When you look at the "sticker price" of Drawmer's 1969 Mercenary Edition, you might gulp a bit at first. However the fact of the matter is that you get a lot for the money. It's a stereo unit that can be used as dual-mono, the mic pre's sound great, it's got a *huge* tonal palette, and the controls and switches feel like they're made to last. Taking that into consideration, the asking price isn't out of line compared to the competition — some of which may be giving you a single channel for only a bit less than the cost of the 1969. Definitely recommended. ■

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By Craig Anderton· anderton@musicplayers.com

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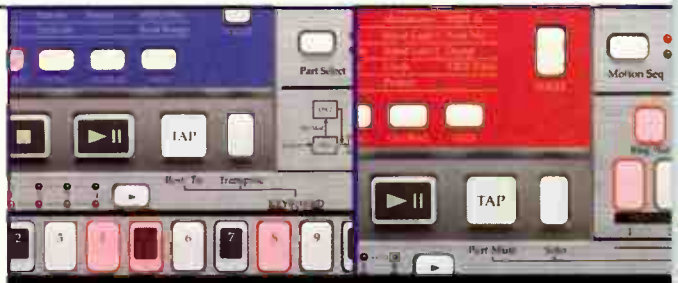
The three units are designed for stand-alone or system use; we'll cover them from both standpoints.

We Are Family

The machines have much in common, starting with small size — 11.75 x 8.5 x 2 (inches) — and light weight (about 2.75 lbs.). They're perfect for groovers on the go, as you can fit all three units — along with headphones, a small mixer, and other accessories — into a bag that will fit in an airplane's overhead compartment. Each uses a 9V/500 mA wall wart transformer. Although I prefer universal power supplies that can handle 100-250V, wall warts are the next best option, as you can purchase adapters to accommodate different voltages.

All three specialize in real-time tweaking. The controls don't feel as substantial as, for example, those in the Electrix series of groove-oriented processors, but given the number of functions packed onto a relatively small front panel, they're surprisingly playable. The knobs wobble a bit, but the buttons have a rubbery texture that absorbs hard hits very well. Strategic buttons light when selected, which really helps on a darkened stage. Overall, I think these babies will hold up fine with reasonably careful treatment.

There are two minor ergonomic missteps. All displays use LEDs, which wash out under daylight — something to consider if your all-night outdoor rave stretches into the morning. For studio and club use this is a non-issue. Also, the lettering in each unit's Common section (see sidebar, ES-1 Front Panel Modules) is difficult to read. It's bad enough dealing with smoke machines and rotating mirror balls without having to deal with tiny, low-contrast type.



KORG ELECTRIBES

MANUFACTURER: Korg, 316 S. Service Rd., Melville, NY 11747. Tel: 516-333-9100. Web: **www.korg.com**

SUMMARY: This smart, sassy "groove production system" blends real-time control with hip sound engines for rhythm, "analog" synth, and sampled sounds.

STRENGTHS: Compact. Excellent real-time tweaking options, which can also be recorded. Great effects with the ES-1. Pattern changes on the fly. High fun factor. MIDI sync. Good documentation. Vast numbers of knobs and switches promote ergonomic operation.

WEAKNESSES: Can't expand ES-1 internal sample memory. Wobbly knobs don't feel solid. Have to stop unit in order to change songs. EA-1 synth doesn't respond to MIDI volume or modulation.

PRICE: ES-1, $599. EA-1, $499. ER-1, $499.

EQ FREE UT. #: 117

Musically, the Electribes operate on the drum machine model: create *parts* of particular sounds (including processing), trigger parts rhythmically to create up to 64-step *patterns*, then string up to 256 patterns together into one of 16 *songs*. The ES-1 holds 128 patterns; the others, 256.

We'll look at ES-1 features in detail, as it's the newest in the line. Along the way we'll note some differences between this and the EA-1 and ER-1, and finally, what it's like to use them together as a system.

Sampling

First, note that this is a phrase sampler; it's not intended to replace keyboard samplers for piano sounds, etc. Rather, Korg designed it as a very capable phrase and rhythm production sampler for creating grooves and new rhythms from sampled elements. Internal storage handles up to 150 sounds (100 mono, 50 stereo), although given the maximum mono sampling time of 95 seconds, you'll probably run out of memory before you run out of samples.

The ES-1 ships with 53 samples; if you sample over these and erase them, you can't get them back. However, you can save samples (whether yours or the factory's) to SmartMedia cards, which can also store patterns and songs. If your computer can save to SmartMedia cards, the ES-1 can load WAV or AIFF files. Cool.

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-Tom Jung, Pro Audio Review

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-Gary Barlough and Richard Zwiebel, Mix

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-Tim Cooper, Studio Sound

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-George Petersen, Mix

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THE ER-1 AND EA-1

The ER-1 Rhythm Synthesizer resembles the ES-1 in many ways (motion sequences, architecture, programming, etc.), but digitally modeled analog-type drum sounds replace the sample-based engine. There are four "preset" drum sounds (claps, crash, hat open, hat closed) and four percussion synth sounds. Each synth sound has programmable pitch (with choice of two waveforms), modulation depth, modulation speed, modulation waveform, decay, level, pan, and low boost. The range of sounds these controls can produce is striking.

There's also a fabulous ring modulator that cross-modulates percussion synth 1 and 2, or percussion synth 4 and the audio input. Depending on your audio source, this can give some truly bizarre effects, especially with melodic inputs such as voice or guitar.

The EA-1 Analog Modeling Synthesizer is also a close relative of the ES-1. A two-voice synth does the sound generation. Each voice incorporates two multi-waveform oscillators with cross-modulation options (ring mod and sync), portamento, and resonant filter with attack or decay envelope (with a separate control for envelope generator intensity). You can use the 16 step keys as a keyboard (no pitch bend, though) that can shift to various octave ranges.

The audio input replaces oscillator 1, so the cool cross-modulation effects made possible by using osc 1 in conjunction with osc 2 can apply to an audio input. The headphone out is mono only — this isn't really a stereo device. Think of the EA-1 as a Roland TB-303 on steroids, and you'll have a pretty good idea of what it's about.

determines how fine the slices will be. There can be up to 64 slices, each having a sixteenth-note duration.

One advantage of the slice approach is that you can speed up or slow down the sliced sequence of parts without changing pitch, as tempo changes simply trigger each slice closer together (faster) or further apart (slower). But the ES-1 takes the concept further, as you can turn different slices on and off, thus changing the sample's character, or use a motion sequence to process a slice using the part processing controls (pan, level, pitch, reverse, etc.). This ability to deconstruct samples, then reconstruct them in novel ways, is hot.

However, slicing requires trial-and-error. Sometimes you need to change tempo a few BPM to capture attacks properly, and some samples (specifically those that are "groove" quantized or use swing), may not slice the way you want regardless of the tempo and sensitivity settings. Fortunately, the slices are placed across the step keys, so you can hit individual keys to hear what each slice sounds like. This will indicate how well your chosen settings work. For truly difficult samples, sometimes the best workaround is "quantizing" the sample manually in a computer-based digital audio editor, loading the sample into the ES-1, and then applying slice to the "splice-friendly" sample.

Bottom line: splice doesn't always work the way you expect, yet paradoxically, sometimes the results it produces are better anyway.

A System Approach

So what happens when you put all three Electribes together? Is it a true "groove production system?"

The answer: Combining the three modules together makes for an incredibly powerful rhythm section. You can hook them all together with MIDI, typically slaving two of the modules off the third. Starting and stopping the master will cause the others to start and stop, and patterns can also change in sync if desired. But they don't have to, which is sometimes better as you can keep a groove going on one machine while changing patterns on the others. If your thing is coming up with remixes, grooves, loops, and DJ tracks, the Electribes score a bullseye.

However, the Electribes aren't "lead" instruments in the conventional sense, so in most cases you'll want someone on turntables, synth, vocals, guitar, etc. to flesh out the sound. The EA-1 does recognize pitch bend and when connected to a MIDI keyboard, can make some convincing lead synth sounds. But it doesn't respond to modulation or aftertouch, so don't look for too much expressiveness.

For live performance, the Electribes score for meaty sound and the ability to get people moving. However, don't necessarily expect to do a continuous, four-hour set with them. You can't change the song or MIDI clock status (internal/external) while a unit is running, so doing seamless song-to-song segues isn't an option. There are workarounds, but clearly the units simply weren't designed for this.

However, what works brilliantly is to select among the various patterns in real time while all three are chugging along. Given the number of patterns and real-time control options, you could keep going for days just mixing and matching patterns from the three boxes. And it's a satisfying musical experience from a performer's standpoint, because all that tweaking is so involving.

Wrapping Up

We've skipped over some of the more

▶ *continued on page 152*

I/O

All three Electribes have MIDI in, out, and thru, AC adapter coaxial connector, and headphone out. MIDI is for sync and tone module applications; the EA-1 and ER-1 also use it for bulk dump/load.

The ES-1 and ER-1 have left/mono and right 1/4-inch outs. Thankfully, plugging in headphones doesn't defeat these. The EA-1 outs are similar, except the two outs each carry a "part" (synth voice), and plugging into only Part 1 gives a mix of the two.

The ER-1 has two audio ins that supplement the onboard tone generators; hitting the associated buttons lets you "chop" audio rhythmically (*e.g.*, cut a radio station or CD in and out of the mix), and one audio in can also provide an input to a ring modulator. The EA-1 has a similar single audio input that routes incoming signal through the unit's synth engine for filtering, modulation, etc.

Being a sampler, the ES-1 includes a 1/4-inch jack for stereo/mono sampling, a level control, and mic/line level switch. It also has a slot for a SmartMedia storage card.

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A talented musician, successful composer and recognized producer, Lenny White is considered one of music's real innovators. He's played with such greats as Miles Davis, Chick Corea and Stanley Clark, and has also collaborated on some of the most recognized and influential music of the past three decades, including the new CD *On the One*, with Sammy Peralta.

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Sammy Peralta loves music. That pure and simple fact comes through strikingly clear as he sits at his keyboard tinkering with half-written tunes. Sammy's background includes work with talents including Tito Puente and Willie Bermudez. "I have to be careful because I can get so lost in the music, I sometimes forget I have a family that would like a little of my attention too".

One last point: Sammy Peralta's new CD ***On the One*** featuring Lenny White was mixed entirely with LSR monitors.

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By Rob McGaughey · hmassey@ucmedia.com

Soundelux U99 Multi-Pattern Tube Mic

Soundelux's large-diaphragm tube mic proves to be a strong contender

The U99 is a good-looking side-address style microphone made of nickel, chrome, and brass alloy. I did find that the body of the microphone could be scratched when inserting and removing it from the spider-style shock mount if you're not careful. The capsule of the U99 is a dual-membrane, dual-backplate design, with a one-inch diaphragm diameter. The microphone contains only twelve electronic components, including an EF86 tube. The included external power supply, the P99, provides power to the microphone, so no phantom power is required. The P99 also contains a front-panel rotary control for polar pattern selection that's continuously sweepable from omnidirectional to cardioid to bidirectional. It maintains a fairly consistent frequency response through all settings. Also included is a 20-foot six-pin XLR cable that connects the power supply to the microphone. A spider-style shock mount is optional. According to Soundelux, all of the components in the U99, including the diaphragm, tube, and all electronics, are of the finest quality and specially selected. In the case of the diaphragm (which Blue of Latvia manufactures for Soundelux), it's hand-made and hand-tuned to meet tight tolerances.

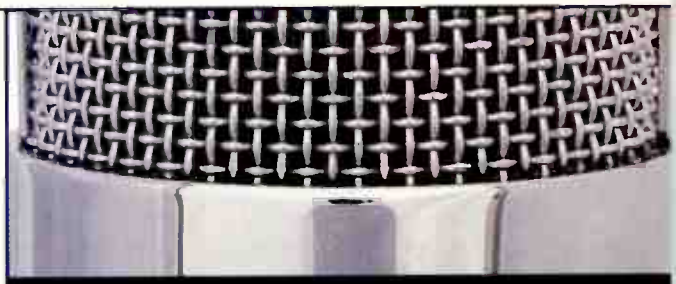
The U99 comes in a large, well-padded, heavy-duty ATA-style case. The case houses the microphone, the P99 power supply, the shock mount, the six-pin XLR cable, and a power cable.

In Use

According to the documentation that came with the microphone, the U99 needs to warm up for 60 minutes for optimal performance. This is typical of many tube microphones I've worked with. Also stated was a minimum warm-up time of five minutes. I compared the sound of the microphone after warm-up periods of five, 20, 40, and 60 minutes. I heard noticeable differences at five, 20, and 40 minutes, but the sound didn't change much, if any, after 40 minutes. Regardless, allow adequate warm-up time for the mic before putting it to work.

Unfortunately, the aforementioned documentation consisted of just a marketing sheet and a single page of notes on how to set up and use the microphone. It would be nice to get more details, such as frequency response diagrams for at least the main polar patterns, as well as some application notes. Even the frequency response graph on the included cut sheet could offer more detail.

My first test was to use the U99 on a male voice. I selected a cardioid polar pattern, grabbed a pop filter, and recorded some tracks. I was impressed by the smooth warm midrange accented with a slight boost in the 11–12 kHz range. The result was a sound that had a nice presence, excellent clarity, and a natural airiness. The U99 also did a fabulous job of preserving the dynamic range of the voice. For the sake of comparison, I recorded the same track with a Rode Classic tube microphone, a Beyerdynamic MC834, and an Audio Technica AT4050cm5. In a blind test I picked the U99 as my favorite. I felt the U99 and the Rode Classic were the top two choices; the U99 offered a smoother top end and better overall clarity. I repeated the test with the Rode and U99 in bidirectional mode. I still picked the U99 (although it



▶SOUNDELUX U99

MANUFACTURER: Soundelux, 1147 North Vine Street, Hollywood, CA 90038. Tel: 323-464-9601. Web: **www.soundelux.com**.

SUMMARY: The U99 is a superb large diaphragm tube microphone for tracking vocals yet versatile enough to be useful for a variety of other applications.

STRENGTHS: Warm, natural sound with outstanding clarity. Excellent vocal microphone. Wide dynamic range. Low noise floor. Comes in a large, well-padded, heavy-duty, ATA-style case that holds the microphone and necessary accessories including a shock mount.

WEAKNESSES: Minimal documentation. No built-in low-cut filters or attenuation. Body of the microphone scratches easily when inserting or removing from shock mount.

PRICE: $2,500. Shock mount, $250.

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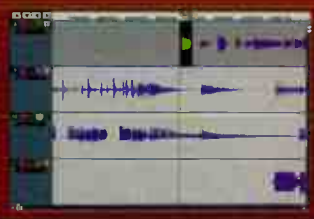

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SOUNDELUX MICROPHONES

The original Soundelux microphone, the U95, was designed to emulate a Neumann U67 microphone. The U95 was forced out of production due to the inability to obtain 6072 tubes that met the company's specifications. The U99 replaces the original U95 microphone and utilizes the same body but has a different tube, a larger transformer, and a new diaphragm. Soundelux claims that the U99 offers superior dynamic range, noise specs, and frequency response. The company also offers the U95S microphone ($3,500), which is designed to emulate a Neumann U47 microphone, and they are planning to release a new microphone, the Elux 251 ($5,000), which should be shipping by the time you read this. The Elux 251 is a clone of the venerable Telefunken 251 microphone.

wasn't as convincing as in cardioid mode), but I could see where in certain circumstances the bigger low end of the Rode Classic would be more desirable.

I also had the opportunity to try the U99 on a female vocalist as well as a second male vocalist. The U99 really worked well in both situations. It always exhibited a nice warm sound with a smooth top end and excellent clarity. I did several A/B tests and the U99 was the microphone of choice most of the time; however, the Rode Classic was selected a few times and was a close second overall. I did observe that the U99 was noticeably quieter than the Rode Classic in all tests. As a vocal microphone, I feel the U99 is a great choice.

Next, I tried the U99 on a Guild JF-35 acoustic guitar. I experimented with several microphone placements, recording both finger and heavy strumming playing styles. The U99 did a good job of capturing the sound of the guitar when placed about two feet away and aimed directly at the guitar's neck/body joint. While the U99 did work well on acoustic guitar, it wouldn't necessarily always be my first choice, except possibly in combination with another microphone.

I also took the advice of the included cut sheet and tried the U99 on a variety of drum and percussion instruments. I recorded various sizes of frame drums, hand drums, tubular chimes, shakers, maracas, a guiro, and a tambourine. I also used it as an overhead mic, placing it about six feet in the air above a drum set. In all of these percussion tests I was impressed with the results. I found the U99 to have good transient response, excellent clarity, and even when pushed hard, it maintained a smooth top end. Most importantly, the tracks recorded with the U99 seemed to have more dynamic range and sounded more alive when compared against tracks recorded with an Audio Technica AT4041, a Beyerdynamic MC834, and a Shure SM57.

Conclusions

With microphones, there's really only one question: How does it sound? It doesn't matter if it is a dynamic or condenser, if it has a tube, or how pretty it looks — it all comes down to the sound. I must admit that I was impressed with the U99. It does an outstanding job of balancing a warm, natural midrange with a smooth, present top end while maintaining plenty of clarity. I found that the U99 tended to give a smooth and consistent sound regardless of SPL and polar pattern. In short, it sounded great and should find a welcome home in many studios.

Rob McGaughey owns and operates Sound Sauna Studios located in Pittsboro, Indiana.

SOUNDELUX U99 SPECS

Patterns	Sweepable Omnidirectional to Cardioid to Bidirectional
Frequency Range	20 Hz–20 kHz (> –6 dB @ 20 kHz)
Sensitivity	21 mV/Pa
Equivalent Noise	26 dB unweighted, 16 dB "A" weighted
Maximum SPL	130 dB for 0.5% THD @ 1 kHz
Impedance	150 ohms
Capsule Size	1-inch
Tube Type	EF86

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CIRCLE 28 ON FREE INFO CARD

By Mitch Gallagher · mgallagher@uemedia.com

MediaForm CD-3701 Stand-Alone Automated CD Duplicator

Small-scale CD duplication just got easier

Most of us have needed to make multiple copies of CDs or CD-ROMs — whether for archiving, small-scale distribution, or to provide reference CDs for clients. Doing this on a computer-based CD burner is a pain. Not only do you have to load the software and data, but you have to shuffle blank CDs in and out of the burner — besides the fact that your computer is tied up while the copying takes place. An Alesis MasterLink provides one audio CD solution, but you still have to shuffle blanks in and out of the machine. MediaForm has a number of solutions for this dilemma in their line of automated CD-R duplicators; one of which, the CD-3701, is the subject of this review.

The CD-3701 comprises an internal brain/hard drive, SmartDrive2 CD writer, and a mechanical arm that automatically loads master and blank CDs into the writer. Setup is easy: Just de-box the unit, remove a foam shipping restraint, install the blank CD feeder bin (one screw), plug the machine into an AC outlet, and you're ready to go.

The CD-3701 is stand-alone; no computer is required. The master to be copied is loaded into the CD drive, and an image of it is created on the hard drive. Duplicates are then burned from the hard drive-based image; up to six images can live on the hard drive at once. Putting the CD-3701 into action is as simple as loading a stack of blank CDs into the feeder bin, putting the CD you want to copy on top of the stack, and pushing "Copy," then "Start/Yes."

In Relay Mode, each time the CD-3701 encounters a master CD in the feeder bin, it makes a new disc image and uses that to burn subsequent copies. For example, I wanted to burn copies of a Pro Tools backup that spanned six CD-Rs. Relay Mode made it easy. I just loaded the six discs with blanks sandwiched between them, pushed start, and went back to my usual loafing. Ain't technology grand?

Discs can be verified byte-for-byte against the hard drive disc images, and the SmartDrive2 has a "Full Scan" mode that can check the master disc before copying begins. For better quality copies, the SmartDrive2 utilizes ROPC (Running Optimum Power Control) to dynamically adjust laser output so it's optimized throughout the write cycle.

The CD-3701 also provides extended features such as password-protection for preventing unauthorized use. Antipiracy features include electronic watermarking, Smartstamp, which writes a user-defined code onto data CDs, and SmartRID (Recorder Information Data), which does the same thing for audio CDs. However, the manual warns that SmartRID codes may be mistaken for errors by mass-replication houses.

During my time with it, the CD-3701 performed flawlessly. I tried it with commercial audio CDs and Mac/PC CD-ROMs, MasterLink audio CDs and CD24 discs, DTS-encoded surround CDs, and computer-burner-generated audio and data CDs, and was rewarded with perfect copies each time.

For the casual user, a CD duplicator may be an extravagance. But for those making numerous CD copies, the fully automatic CD-3701 provides a fast and easy solution. ■

▶ MEDIAFORM CD-3701

MANUFACTURER: MediaForm, 400 Eagleview Blvd, Exton, PA 19341. Tel: 610-458-9200. Web: www.mediaform.com.

SUMMARY: Stand-alone CD copying and small-scale duplication doesn't get much easier than this.

STRENGTHS: Stand-alone. Relay mode. Multiple CD images can live on hard drive simultaneously. Easy to use. Broad disc format support.

WEAKNESSES: Pricey for casual use.

PRICE: $4,199

EQ FREE LIT. #: 120

CD-3701 SPECS

Minimum CD throughput	8/hour (650 MB discs, includes load/unload time)
Feeder bin capacity	75 blanks
Supported media	CD-R, CD-RW
CD write speed	12x/8x, with 4x/CD-RW capability
Supported disc formats	Audio, CD-ROM XA Mode 2, CD-ROM, Photo-CD, Enhanced Mixed Mode, CD-I, Video CD, CD-Extra
Dimensions	19.13W x 16.25H x 15.75D (inches)

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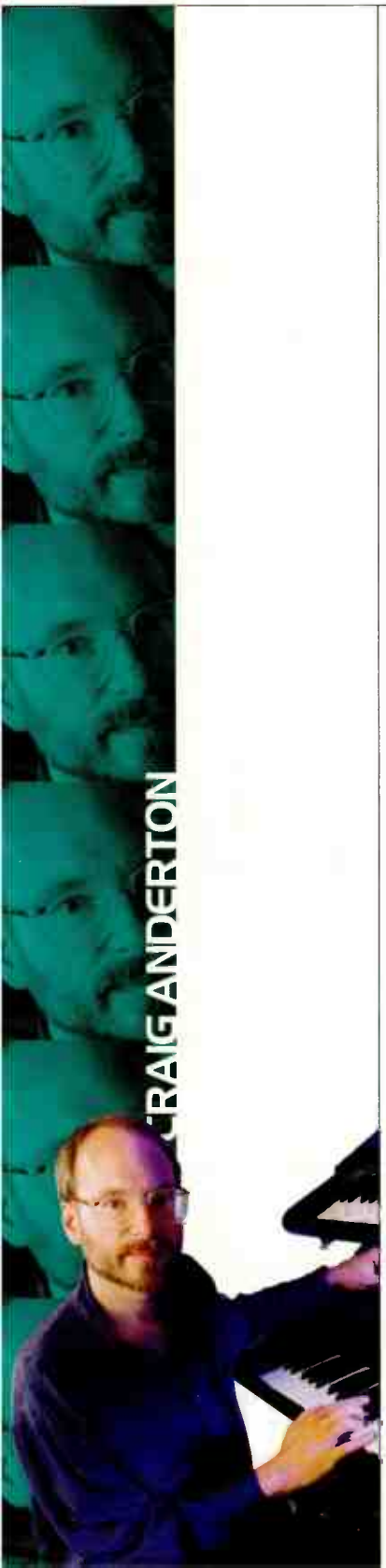
Creating Loops In Acid



Sonic Foundry's Acid is a loop-oriented, multi-tracking program that allows effortless time and pitch compression/expansion. Throw together loops of different tempos and keys, and Acid will make them all conform to your choice of tempo and key.

However, little has been said about Acid and the loop creation process. For me, this is one of Acid's strongest features, because I generally use Acid when creating audio-for-video — it's so easy to stretch or compress the tempo to make the music fit a particular scene or edit. But with videos, you'll often get requests for music that you'll never find in a loop library. (my favorite: "we need sort of funky, low-down rock and roll, but with a strong military character"), which is one reason to roll your own. Besides, it can be faster to create a loop than to audition sample CDs.

Creating The Loop Via Sequencer

I generally do drums and other "metronomic" tracks in a MIDI sequencer, and record the tone generator's output into a digital audio editor for trimming and processing. Because most loops are so short, having a sophisticated graphic editing interface isn't necessary; if a part doesn't work, I just play it over. Little tweaks, such as raising the level of a note, are often easier to do within the audio editor.

However, even if the displays for various hardware sequencers and software programs indicate the same BPM, the actual tempos may differ slightly. There are three ways to deal with this:

- Ignore any difference, trim to the downbeats, and let Acid do the needed time stretching.
- With short loops, trim the end to the theoretically correct length if the difference is only a few milliseconds, and don't worry that the loop is short or long.
- Use time stretching to change the loop length to the precise length. With small differences, the time stretching won't mess with the sound too drastically. Careful, though; some time stretchers will produce a file that is slightly shorter or longer than what you've specified in order to improve the fidelity. After processing, measure the length to make sure the program did what you wanted it to do.

Keep a calculator handy so you can convert tempo to time (60,000/BPM = ms per beat) for precise loop trimming. Try to avoid using the beginning of a sequence, as that's where tempo inconsistencies seem most common.

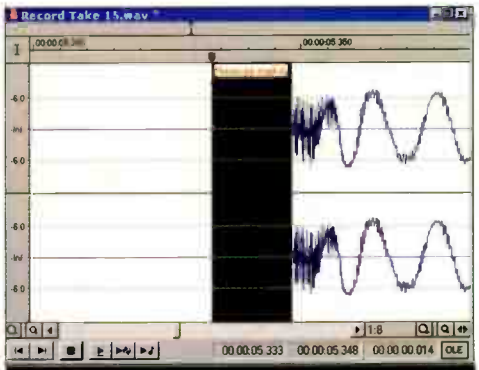


FIGURE 1: System latency can add a slight offset between the true beginning of the measure and the measure marker.

FIGURE 2: The bass track is about to be quantized.

Creating The Loop In Acid

You can also create loops using Acid's audio recording abilities. Set up backing loops you can play along with (you'll perform better if you play to a track), and set the total loop length and indicator for something like 16 measures. Turn on loop mode, and now Acid will play those 16 measures over and over.

Keep playing the part until you get it right. After 16 measures, Acid will loop playback, but the recording track will continue — what you play during the loop's second pass goes in measures 17-32, the third pass ends up in measures 33-48, and so on.

When you're done, open an audio editor and trim out the section you want to use as a loop. Acid is optimized to work with Sonic Foundry's Sound Forge, which places markers at each measure (see the article on creating guitar loops in the Aug. '00 issue).

Due to system latencies (at least that's my theory), the markers may not fall exactly at the beginning of the measure. Check this as follows:

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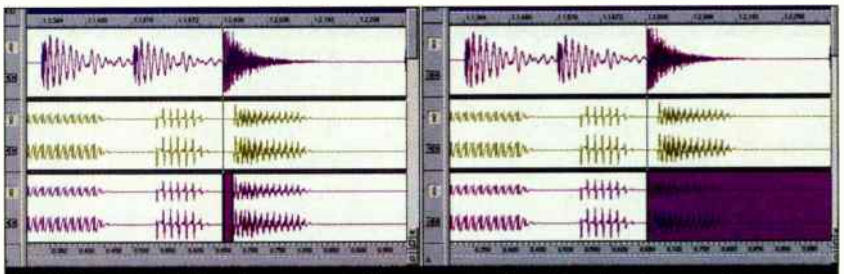

FIGURE 3: By defining the amount of lag, we can delete it and make room to snap the audio into place. FIGURE 4: The note attack now lines up with the sixteenth-note grid.

1. Loop a drum part in Acid that has a hit on the downbeat (*e.g.*, kick drum).
2. Solo this track.
3. Route the track back through your mixer or soundcard so that it's re-recorded into Acid.
4. Record a few measures.
5. Open up the file in Sound Forge.
6. Note the difference between the measure marker and the track's downbeat (Fig. 1), which in my case is 14 ms. Now you know to offset by that amount when trimming to measure markers.

If you're not using Sound Forge and are recording mono samples, you can create "markers" by setting up a click track loop in Acid (*e.g.*, clave on beat 1), and sending that into one of Acid's inputs. Send the part you're recording into the other input. When you open up the file in an editor, the click waveform will indicate the start of each measure, and you can trim to that reference.

Quantization Time

Okay, you've recorded your killer loop and trimmed it to the right length. But what if a note's start time is a little off? Here's your chance to use Acid's quantization feature. It's a bit time-consuming, but if you're patient you'll end up with a tight rhythm.

To start, load the loop that needs quantizing into Acid. The track type (as determined in Properties) should be a loop, not a one-shot or hard disk track. Zoom in close, set the Snap To function to sixteenth-notes (or whatever you want to quantize to), and get ready for surgery.

Refer to Fig. 2. The top track shows kick and snare. The next lower track is the unquantized bass track. The lowest track is a copy of the unquantized bass track, which we're going to quantize.

In this figure, the cursor is sitting at measure 1, beat 2 (the exact beginning of the kick waveform), but the bass note lags behind. We're going to cut the space between the cursor and the beginning of the note in track 3, then move the note forward to line up with the cursor.

Here's how to trim up notes that lag the beat:

1. Select the track where you want to quantize the note.
2. Place the cursor on the nearest sixteenth-note grid prior to the note's attack.
3. Type "S" to split the track at the cursor (this has just happened to the bottom track in Fig. 2).
4. Hit F8 to turn off the snap function.
5. Place the cursor at the precise beginning of the late note.
6. Type "S" again.
7. Click on the little sliver (indicated in magenta) between the grid position and the beginning of the note (Fig. 3).
8. Hit Delete to get rid of the excess space.
9. Type F8 to turn the grid back on.
10. Snap the note attack leftward to the grid, and *voilà*! Quantized bass note (Fig. 4).

If the note is ahead of the beat, simply click the cursor with quantize off at the head of the note, type "S" to split the part, turn grid back on, and slide the note to the right (later) until it snaps into place.

Of course, changing the note start in this manner also changes the notes "downstream." In some cases, you'll need to cut both the note's beginning and end so that other notes aren't affected by moving it. If moving the segment creates an undesired overlap with another segment, trim a little bit off the end to create a space.

The final steps are to set loop markers around the newly quantized measures, and from the Edit menu, choose Mix to New Track. The new track will meld all the quantized elements into one continuous track.

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Want to hear it for yourself? Go to www.eqmag.com. You'll find three audio examples: unquantized and quantized basses split in stereo, quantized bass with drums, and unquantized bass with drums.

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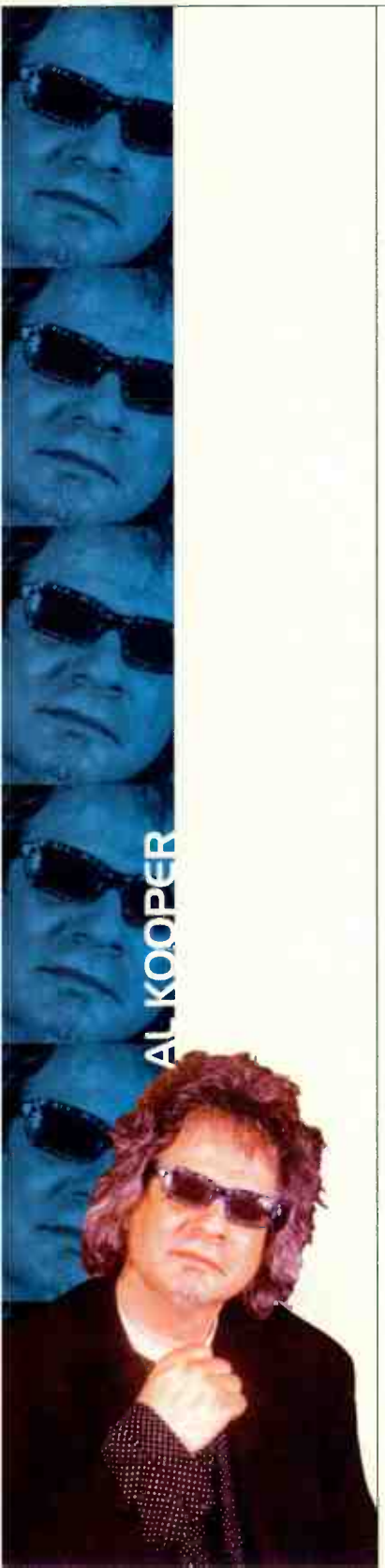

When the Alesis MasterLink was released, I really had no concept of what it did. I skimmed through a brochure, but it never really sank in. First I had to go out and buy a CD burner and record dozens of archival music CDs. *Now* I appreciate the MasterLink.

Okay. Say you like to create mix CDs; that is, CDs that have your favorite tracks on them — to play in the car or in the bedroom. With a typical CD burner, you try to go digital-to-digital to get the best sound. I was patched up through my DAT machine. I would record what I wanted on the DAT machine and then go digital-to-digital to the burner. It sounded great, but some tracks were louder than other tracks, some more bottom-heavy, etc. In that setup I just discussed, one is powerless to make corrections unless one is working with a full-blown computer in tandem with the burner. I don't know about you, but I like to keep my music in the living room and my computer in the office.

Along came the Alesis MasterLink. This box has:

- A hard drive that holds about 3.5 hours of music at 44.1 kHz/16-bit resolution.
- A CD burner.
- A DSP section that will raise and lower levels, compress, limit, EQ, copy DSP settings from one track to another, and peel grapes. Well...maybe not grapes *now*, but they're working on an update.

In addition, you can fade tracks in and out, trim intros and endings, join songs together, cut songs into segments, change the order of a playlist, and generally speaking, put all the ingredients of a musical salad together along with those peeled grapes.

I unpacked the box, read the manual, and slipped an audio CD in the burner so I could transfer a favorite tune onto the hard drive. I pushed the appropriate buttons and the machine went into overdrive. In one minute I had transferred a 4-1/2-minute tune. Pick out 15 songs in advance and you can get them onto the hard drive in less than 20 minutes. For a guy who routinely took over three hours to make 90-minute mix tapes, this was nirvana! If you want to transfer an entire CD onto the hard drive, just push the buttons, and it will be done in 20 minutes. Then it's tweaking time.

My modus operandi is to trim the heads and tails of all the tracks first. Then I go in and work on the sounds: adding EQ, matching volumes, etc. Then I check the segues to see if they're artful. I may switch the order around or delete a song. When that's complete, I play the soon-to-be CD as background music and go about my biz. If it passes *that* test, then I push the burn button. Burning the CD takes about 40 minutes. Still — twice as fast as real-time.

When, you may ask, does the other shoe ever drop?

Well, it did for me, but the Alesis tech department fixed everything *before* they knew I was a grouchy pro audio kolumnist. I found them to be courteous and sympathetic to my plight. Most of the fixes were achieved with the version 2.02 software. You can download this update or order it from the Alesis Web site at no charge.

> ***If you want great mix CDs made in the living room (where you've always made them) without hooking up a ton of crap to a computer, [MasterLink] is a great item for you.***

I actually went out and bought this product sight-unseen when I realized its possibilities. It's cheap, too, considering that it's a hard disk recorder, a CD burner, and a DSP processor for about $1,500 street price.

If you're like me, and you want great mix CDs made in the living room (where you've *always* made them) without hooking up a ton of crap to a computer, this is a great item for you.

Oh yeah — those wacky folks at Alesis also say in their rather straight-forward, not-as-large-as-The-Bible manual that this is an excellent machine for *mastering at home*. Imagine that!

Anyone want to buy my old CD burner?

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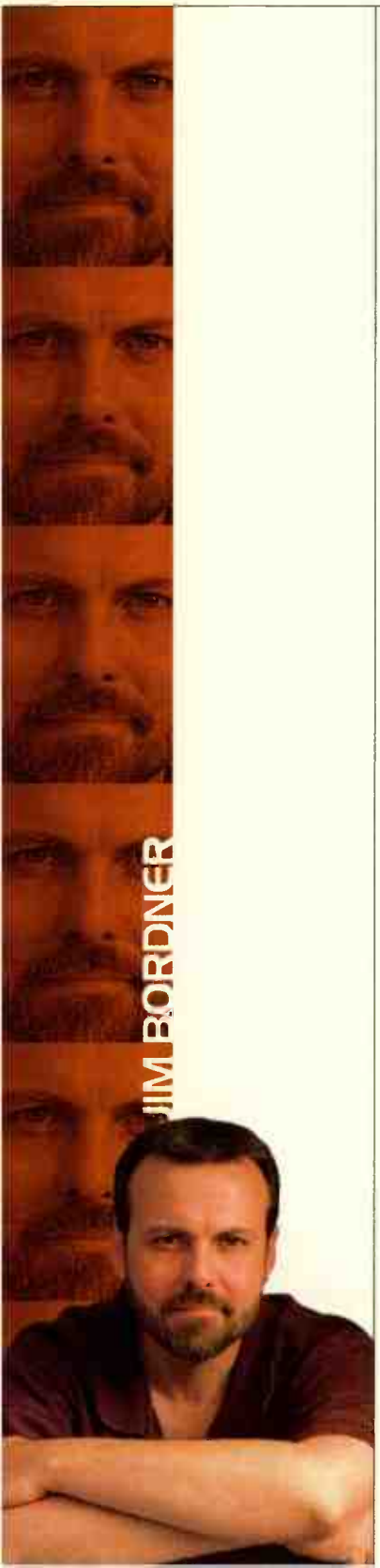
Putting tax deductions to work for your studio

Deductive Reasoning: The Home Studio



Benjamin Franklin (the owner of Franklin Audio, Inc., not the one you're thinking of) said it best, "Nothing is certain except: A. clients will always want changes two hours before the spot goes on the air, and B. taxes." Or something like that. I don't have my copy of Bartlett's handy, but you can look it up.

Of these two well-known stones in the Warrior's shoe, the more vexing is taxes. We hate the filing of receipts, the documenting of mileage, the paperwork, and especially writing the check for the estimated quarterly tax.

But the self-employed do have some respite. The truth is, we get a better shake on deductions than ever before. If we use the tax code to our full advantage (without trying to get more than we really deserve), our tax burden is really no more than any other working stiff, despite the Self-Employment Tax.

This month, let's talk about one specific type of deduction that's particularly friendly to the Guest Room Warrior: the home office deduction.

If you run a home-based studio, you get some mighty sweet advantages. The overhead is extremely low, so you stay more profitable. You can time-shift like a maniac, taking Monday off to mow the lawn or go to the beach, and then working late Wednesday to accommodate a client with a tough production schedule. You don't have to commute, saving yourself hundreds of hours a year or more: It's like getting 3 weeks vacation if you don't have to drive to work (although it does open up some deduction questions on your vehicle and mileage deductions, which we'll talk about in an upcoming column). And then there's the home office deduction, one of your best bets for reducing your tax bill.

There are lots of rules for qualifying for the home office deduction, but they really boil down to two major ideas. Your office/studio space needs to be used only for business, and it needs to be your *primary* place of business. In other words, it can't be a corner of your family room, and it can't be a place where you do a little paperwork in between jobs where you work at other places. For a gigging musician, whose work is done in clubs and studios away from home, the home office deduction is a little suspect.

You may have heard that the IRS has been cracking the whip on home-office deductors, and it's true, they are. But not on legitimate home offices. They're going after the professionals who have a regular office downtown but keep a computer in their study to "catch up on paperwork," or the consultant types who do all their work off-site and use their home office two hours a week to do billing. The home-based studio is usually in the clear. Don't believe the common misconception about not being able to write off the deduction unless you have a separate entrance for the business.

There are two kinds of expenses you can write off: direct and indirect expenses. Direct expenses are things you do exclusively for the business and the space it inhabits. If you remodel your studio, repaint it, put in new carpet, or hire a maid to clean it once a week, those are direct expenses, and you can write them off at 100% of their value (although most building improvements need to be amortized over time).

You may have heard that the IRS has been cracking the whip on home-office deductors, and it's true, they are. But not on legitimate home offices.

Indirect expenses are things you do to the building in general: a new roof, a fresh driveway, interest on your mortgage. Indirect expenses can be written off, too, based on the percentage of your home that's used for business.

Let's say you have a 2,600 square foot house, and your studio is in two rooms: a 12 x 14 control room and a small 6 x 8 walk-in closet you've converted to a vocal booth. Your studio adds up to 216 square feet, around 8% of your home's total square footage. So you can write off 8% of your indirect expenses, subject to the usual depreciation tables.

Some indirect expenses are written off 100% in the year of filing. You can take that same 8% (or whatever yours works out to be) of your utility bills, your lawn care service, anything you spend on maintaining your house. Because these kinds of expenses are intangible and have no ongoing value, there's nothing to depreciate. Major improvements such as a new furnace or a new deck will

The Guest Room Warrior's Top Ten Excuses During an IRS Audit

- 1. "You can't disallow those kegs... my clients demand cold draft beer!"
- 2. "We had to burn the receipts last winter just to keep warm."
- 3. "That pool table is part of my CEO compensation package."
- 4. "I thought if they didn't send me a 1099, I didn't need to report it."
- 5. "Well, I feed them and they sleep here... doesn't that make guitar players dependents?"
- 6. "Hey, if a Rolex isn't advertising, what is?"
- 7. "Taxes, schmaxes... Rockin' is my business!"
- 8. "You call this an income?"
- 9. "Sure it's a professional service... I can't work until I've had a good massage."
- 10. "Uh... I forgot."

need to be depreciated. A good rule of thumb is: if it changes the value of your home, it's got a depreciation schedule.

But remember (and here's where thing get complicated): If you choose to take a portion of your home care and improvement expenses as a business deduction, you may be subject to some recapture taxes when you sell the house. We did this couple of years ago, and the recapture only amounted to a couple hundred bucks, so to my way of thinking, the deduction was more than worth it. Now, if you live in a $500,000 house and are writing off the GNP of Portugal in maintenance and improvement costs, you may find the recapture taxes a bit more salty.

Those are the basics. As always, your mileage may vary, so the best advice is to talk to your accountant or a tax planner about your particular situation, especially if you're thinking of moving in the near future. In my first studio, I had only 130 square feet to call my own, and very little invested in the way of capital improvements. When we moved, we found a house with a clean, dry, unfinished basement, and spent about six grand building the new studio down there (I did most of the framing and finishing work myself). The new studio doesn't intrude on the living space, and we can claim almost 600 square feet of a 2,600 square foot house. As a result, I now resent spending money on home improvements just about 23% less than I used to.

We'll talk about some other deductions that can dramatically affect your tax bill in an upcoming column. Until then, remember that the Way of the Guest Room Warrior is to be honest and forthright, so that while others may jump in fear at the footsteps of the Audit Man, you remain calm and centered. See you next month.

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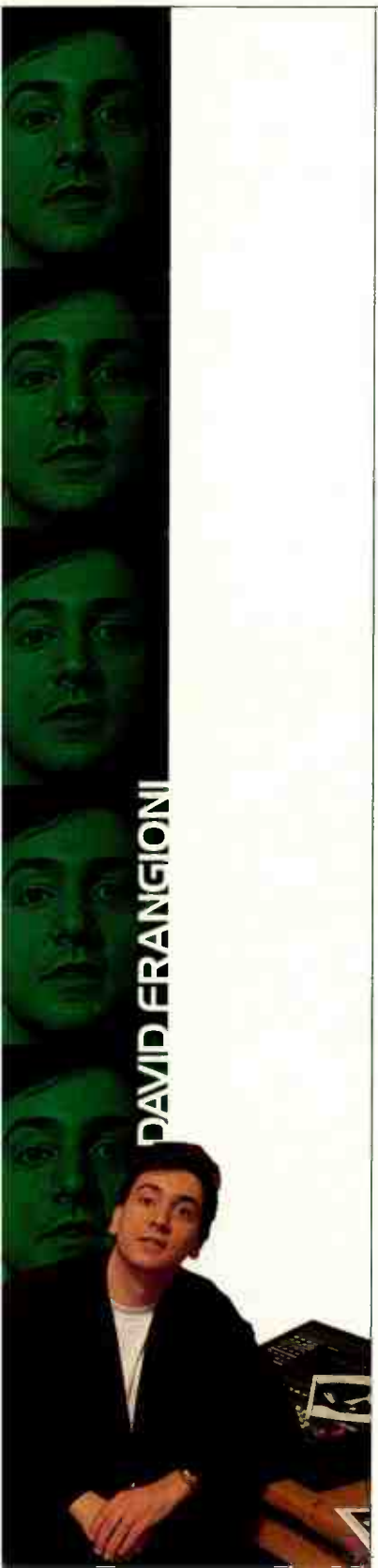
Have a question or comment for Jim Bordner? You can reach him at jim@gravitymusic.com.

DFMUSICPLAYER@AOL.COM

What to consider when choosing a backup format for your DAW

Moving Forward, How Do We Back Up?



The issue of backing up data is a long, fiercely debated topic. Some people see no benefit in spending lots of money on equipment that seems to do nothing more than a CD-R will do. Let's face it, to many DAW users, a DDS, AIT, or DLT drive costs a lot of money for something that can't even be mounted on the desktop! The reality is that investing (note the word "invest") in a dedicated backup solution is money well spent. You'll also need software to manage the drive, since most dedicated backup drives don't have proprietary drivers or software. And no, they typically don't mount on the desktop. Instead, they use a sophisticated (but often times simple to use) software front-end to track all of the data as it moves to and from the drive. This software will also keep a log on your computer of all of the backups performed, when they were done, and what's on each backup tape (or optical disc). For the Mac, the software of choice is almost always either Retrospect or Mezzo. On the PC, there are a lot more options, including Seagate Backup Exec, Microsoft Win2k Backup, and Retrospect for Windows, to name a few. Then comes the actual drive. Again, the choices are many, with the most popular (based on a poll that I recently took on my Web Forum) being CD-R, DDS3, and AIT a distant third. There's also DLT and DVD-RAM.

Now the question becomes three-fold: Which format or drive? Which software? Will it be compatible with all of my associates and clients? The answer to the first two dictates the answer to the third question (or vice-versa). The reality is that the software is fairly cheap (usually under $500), so one solution could become the standard and most people would be able to afford it. The choice of drive is a bit more complicated. CD-R is so popular because it's cheap and easy to get. It works, but having anything fewer than 2 GB of storage for backing up data can be pretty limiting. DDS3 or DDS4 are both fast, reliable methods for backing up data; however, the cost of the drive is between one and two thousand dollars, depending on what model you get. DDS can hold 12 GB of data uncompressed, while DDS4 can store 20 GB uncompressed. When you read the specs of a backup device, be sure to focus on the uncompressed capacity, as that's what matters most. Data compression can be a dangerous thing and often isn't used with precious audio files. Purchase a drive based on its uncompressed capacity, and if you choose to compress the data when backing up, then you'll have twice the space to do so.

If you back up with Retrospect to a CD-R, then chances are that you could go to just about any studio and get the data back with little or no problem (one way or another, just about everyone has a CD-ROM drive). If you used any other means of backup, you run the risk of not finding a lot of places that can read your data. You could always bring your drive and software with you just for the session. However, what do you do when you're in New York and your sax player is in L.A.? If your sax player has AIT and you've got DDS, then you're out of luck. This is precisely the problem that I have been hearing about from a lot of my clients. What do they do?

The general consensus is that the leading manufacturer (or two) of DAWs is going to have to endorse and/or design a package that unifies the whole issue of backup. This would influence and encourage enough users to adopt the same software/hardware package, since it would become that platform's *de facto* standard. There's no such thing as absolute in today's ever-growing DAW market, but the choices could be a lot more focused when it comes to backing up. In 1989, my Fairlight Series III (which I still have and love) only provided one way to back up data (using its system software), and that was to use the internal 60 MB tape drive; there really was no other choice. Now, in the year 2000, I see Fairlights for sale and every one of them has a 60-meg backup drive with huge libraries of sounds — all on the same 60-meg tapes, and totally compatible with just about every Fairlight Series III on earth. I believe that this concept represents what professionals in today's DAW age are asking for from manufacturers on all platforms.

When you read the specs of a backup device, be sure to focus on the uncompressed capacity, as that's what matters most.

Now, Back On The Bus

Last column, I talked about PCI bandwidth

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Still more on sharing your music with the world...or not

Hilarity, Hubris, and Heuristics

The constantly morphing universe of music online ignores the change of seasons. How did it get to be Fall? How the hell can we keep track? Read below for a sampling of the idiocy and inspiration presently defining Internet audio.

Digging around for some populist opinions on the moral nature of file sharing applications, we recently chanced across a chat area in a popular rock musician's Web site. Fans were asked their opinions and we offer the informal poll results below:

Q: Are File Sharing Applications Good or Bad?

They're great! — 39%
They're ok. — 45%
They suck! — 10%
Never heard of them. — 4%

That's apparently taken from 847 votes over a three-day period. During the same three-day period, the site's fan bulletin board displays over 200 posts. Of the 24 total posts (all favorable) mentioning Napster, 12 said Napster led them to sales and only one said they downloaded but didn't buy. Read on for some music lovers' comments about file sharing applications:

"At the beginning of the MP3 craze I searched for every tune I could find. The more I listened to the files the more I sought out the CDs because I wanted the best sound quality."

"I personally use it to listen to songs before I spend money on a CD I'm not certain about."

"I know all the MP3's I've gotten off this site I already have on CD, so I already paid my money!"

"I have downloaded several songs and am ready to buy the very albums those songs are taken from, the end result being that none of those artists have lost a dime from me — they've only gained."

"I would *never* download a note, because I've *always* believed that [the artist] deserves every penny."

"I think they [file sharing apps] are excellent tools to discover new artists and 'test' particular albums before buying. What if the car dealer wouldn't let you test drive the car before buying; would you still be inclined to purchase it?"

"File sharing programs have actually increased my music consumption."

Just Because Music Fans Like It...

Continuing with our theme, a court injunction against Napster that would have required shutting down the service has been appealed and reversed. The company is frantically working on a new strategy for its defense against a lawsuit by the RIAA. Amidst Napster's trial woes, the hot file sharing apps are now Gnutella and Freenet. Why? As distributed networks, both Gnutella and Freenet have no central organization to sue. Users are completely anonymous. Adding insult to injury, a renegade group of programmers tweaked AOL's Instant Messenger (AIM) service to enable it to route searches through Gnutella (known as "AIMster"). AOL isn't happy about being potentially liable for copyright infringement, but they're basically helpless.

There's power in threes (and we don't just mean celebrity deaths) as a trio of well-known musicians recently came out of the closet in favor of Napster and file sharing in general. The ubiquitous Courtney Love, Prince (one of the kings of the DIY ethic), and The The's Matt Johnson have all fired withering broadsides at the major label distribution infrastructure. Says Matt Johnson: "After much deliberation I have decided to offer, track by track, week by week, free downloads of my latest album." (**www.thethe.com**) The back story about Matt Johnson is that the album he's offering for free is a major label release.

And the promo beat goes on. Accepting the award for most ridiculous marketing idea this year is IUMA (a division of Emusic). The music upload site is awarding $5,000 to the first ten couples who name their baby "Iuma." As of this writing, one child has already been named. Your bemused FezGuys are considering paying $5,000 to couples to not inflict this on their children. Regardless of how we feel about it, IUMA has garnered an avalanche of press. IUMA has also been busy disbursing money to artists for Web page views and recently expanded to pay for audio streams and downloads as well. It may be a check for only eleven cents, but at least they're making an effort. All music upload Web sites should consider sharing at least some money with every one of their artists. After all, without the music there would be no site.

Speaking of Emusic, they've switched to providing subscription services: $19.99 for one month of access to their 125,000 songs — discounted to $9.99/month with a year subscription. Though Emusic will need to triple their customer base to make a profit using the subscription model, they offer it anyway, praying the service will seem attractive to consumers as well as a nice, legal alternative to Napster. Like so many other Internet music sites, Emusic (languishing in a sunken stock price) is desperately trying something, anything, to light a fire under their business.

And speaking of business models, MP3.com

(remember them?) has been busy cutting deals with the majors to re-establish their own subscription service. They dole out millions of dollars in damages and license fees and scrape for newsworthy items to keep people interested. Parsing the newsletter is simple enough. "Fresh look and feel!" Like the landlord, they've repainted, but the walls are still made of paper. "Partnering to open physical stores to download and burn your own CDs!" Hasn't this ancient idea been killed once and for all? "Send suggestions!" The last gasping cry from the drowning man...

At the MP3 Summit last June, MP3.com announced that they're going to become an infrastructure company or, in their own parlance, a "music service provider." "We're a utility company, but we deliver music instead of water or electricity," goes the party line. Whatever, kids. Recently the Web site also released an "open" set of tools to encourage other companies to create products and services that link directly into MP3.com's bloated databases. With no solution in sight for effectively and efficiently sifting through all their music, it sounds like MP3.com is dancing as fast as it can.

An Upload Site Review

Los Angeles-based major label Dreamworks Records has launched an online site **www.dreamworksdigital.com**. The Web site appears to exist basically to streamline the label's own process of fishing for new talent. To its credit, the site is very straightforward. A wash of bright orange color and the numerals "1," "2," and "3" greet us on first meeting. We chose "1" because our FezMom raised us right. We are shown the "Artist Profile" and asked to provide a lot of information. "Genre" is required field. There are nine to choose from. That's right, nine. Songs, according to Dreamworks Digital, fall into one (and only one) of nine categories.

We click through with our blatantly falsified profile into area #2: the Submission Agreement. It's all pretty standard, dense legalese but one section catches our eye. Point #5: "...DreamWorks Digital has access to and/or may create...materials and ideas that may be similar or identical to (your uploaded song)...you will not be entitled to any compensation because of the use by DreamWorks Digital of any such...identical material..." Kind of chilling, isn't it? It seems that one might upload a song to Dreamworks Digital and, a year later, hear your song on the radio recorded by someone else.

We submit to the sinister submission agreement and enter the download area. Options appear for uploading a song and a graphic file. At this point, the decision is made to not place a song on Dreamworks Digital. Not only is the Web site a faceless vortex, sucking music and giving nothing back (no musical material is available on the site), the site's disclaimers are enough to dampen the enthusiasm of even the most rabid accordion player. The "Rules & Advice" window intones that exile from the site will ensue should anyone upload more than one song over a three-month period. The "Don't Call Us, We'll Call You" edict is mentioned, as is an unintentionally hilarious recommendation (posing as thoughtful advice) for the person uploading a song to not "wait for the limo driver to ring your bell with a recording contract."

Ease of use — very simple
Design — thankfully, also very simple
Tech support — N/A
Expected user experience — shouting at the ocean
Overall — useful to the label, nothing groundbreaking,

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The FezGuys invite one and all to post questions and comments on the Threaded Discussion Area of www.fezguys.com.

SUZANNE CIANI

▶ *continued from page 99*

copyright ownership of the work. It is not right that in order to get distribution one should have to hand over one's entire future. I'm optimistic that viable independent Internet distribution will not be long in coming."

Back To The Future

Looking to the future, Suzanne sees herself increasingly returning to music-to-picture scoring — this time on her own terms. "I no longer want to be the 'gun-for-hire' I was in my New York City days," she states. "Part of my escape to nature involves downsizing, and I am committed to doing just as much as I can do without creating a staff. I truly enjoy working this way now, and for the appropriate film project I am more than enthusiastic."

At the time of our interview Suzanne was applying the finishing touches to one such project. "The film is called *Mother Teresa: The Legacy*," announces its composer, with more than a hint of pride. "It is produced and directed by Ann and Jan Petrie, who also did an earlier Moscow Peace Prize-winning feature-length documentary, which I scored about 10 years ago. This film is meant to represent Mother Teresa's work and teachings after her death. I was thrilled to be part of this important project and scored the picture using piano, violin, and cello, all recorded in my studio to picture. I was able to achieve a very intimate sound recording in my own environment, sometimes improvising variations of the theme on the piano at two in the morning. I truly believe that the nature of the recording environment had a direct and beneficial impact on the spirit of the music. The film will be released at the end of this year."

"I still want to tour and record and write my piano books," Suzanne maintains, "but, also, I love scoring. I'm good at it. I've worked to picture for the last 25 years, albeit mostly TV commercials. When I scored *The Incredible Shrinking Woman*, they told me I was the youngest woman to score a major film. Now women have become quite established in this area. I will not have the 'pioneering frontier' aspect that I have been used to in my career development, but I can make my artistic mark in films."

Onwards And Upwards

So, what can we expect next from Suzanne Ciani and Seventh Wave?

"Seventh Wave has begun to release other artists whose work I particularly enjoy," concludes Suzanne. "However, it is a new structure, where the artist gets essentially all the income from distribution, with a small override to Seventh Wave, and is responsible for all marketing costs, the marketing decisions being made by the artist with our guidance. This is yet again a new experiment in an alternative approach to a record label. Let's see how it goes."

Jonathan Miller is a British freelance writer living in...well, England. He specializes in the "ancient art" of the hi-tech music interview. Jonathan can be reached at jonathan.miller-musicmedia@virgin.net

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BONZAI BEAT

▶ *continued from page 48*

the studio in those days.

David: I once thought a journalist was the devil. We had to kill him. Don't tell anyone, okay?

Who do you think you were in past lives?

David: No one as famous as myself, more's the pity....

Derek: Me. I have a very strong sense of having been "me" in ancient Egypt.

What old saying do you hate the most?

David: "You know the old saying!"

Derek: "A stitch in time saves nine." Nine bloody what?

What animal do you identify with?

Derek: The panther.

David: The sea otter.

Who is the most amazing engineer you've worked with?

David: The tall one at Greenhouse West. Magic fingers!

Derek: This geezer over in London named Red Bottoms. He would just push everything — you'd look at the meters and they would all be in the red. That's how he got his nickname, Red. I'd ask, "Isn't that going to distort?" He would look at you with this gleam in his eye, and say, "Yeah?"

What makes a great record?

David: Three things: the hook, the hook, the hook. And the words.

Derek: Variety, purity, body, and flavor.

Have you ever witnessed a miracle?

David: I met a thirty-six-year old soccer mom who didn't drive a van.

Derek: No, although I have seen both loaves and fishes — but not together.

Any advice for getting a good start in the music business?

David: Be patient. Remember they're not going to like you automatically just because *you* do.

Derek: Ignore all advice.

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For your free download of Spinal Tap's new single "Back From The Dead" point your browser to www.tapster.com.

KORG ELECTRIBES

▶ *continued from page 128*

traditional features: step time and real time pattern recording, accents à la TR-808, etc. But the main deal with the Electribes is their almost off-the-scale fun factor. You can have a great time just tweaking the delay knobs...and then you realize there are many, many more options. There's something totally addictive about the real-time nature of these boxes. With the Electribe series, Korg has definitely nailed the groove experience.

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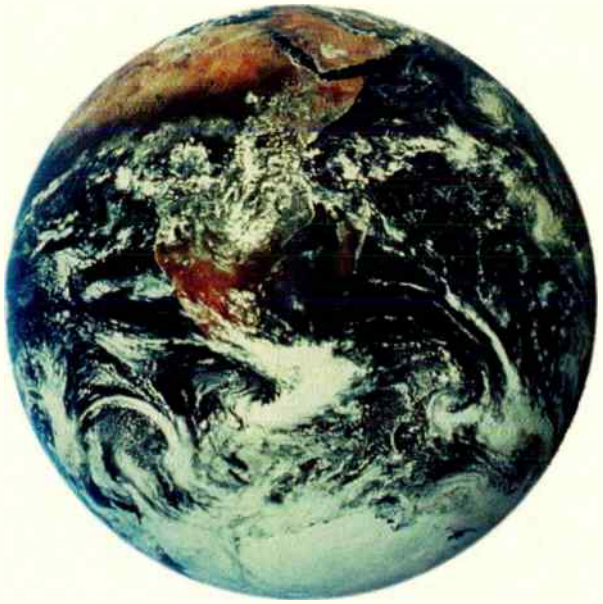
To see video demos and hear audio examples of the Electribes in action point your browser to www.musicplayer.com.



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ACROSS THE BOARD

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technology forefront. I was in a high-end audio store the other day and there were a bunch of guys comparing the new Sony DSD (Direct Stream Digital) with a 96k 24-bit CD player. They were arguing over which was doing the best job of reproducing what was on the CD. Of course they weren't comparing the same recording mixed in both formats. They were comparing two completely different performances by two completely different artists. How could you make a determination about which was the better vehicle for the final audio delivery system?

The only way to find out which is better is to record a live-to-stereo session that was encoded into both systems, DSD and 96k 24-bit. The only way to *truly* compare the two systems would be to record in both formats, and play them back while the live band played along. Then do an A/B/C comparison to see which one sounds the most like the original, not which sounds the "best."

Now that I have beaten this horse to death, how do I know that the horse is really dead and not just sleeping? I need a sleeping horse and a dead horse for comparison. Or do I need the original horse before he was dead. Or what if he died while sleeping? Talking about this is making me hoarse. How many bits in a horse's mouth? I remember when cowboy movies were two bits on Saturday....

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Have a question or comment for Roger Nichols? Visit him online at the *EQ* Boards, www.eqmag.com.

STUDIO TECH

▶ continued from page 146

issues when using DAWs on a Mac. Since then, I've had lots of valuable feedback from users seeking more information. Let me clarify another point relating to this topic: The SCSI card that you use has an important impact on the amount of data that is being carried by the PCI bus. However, for the most part, the hard drive that's connected to it does not. Sure, the hard drive is of critical importance to the performance of the system — in some instances, the drive can under-perform to the point where it actually brings the system down. There are many factors when choosing a drive (which I will cover at another time); however, PCI bandwidth isn't one of the necessary considerations when choosing a drive. While the hard drive needs to respond to the requests for data flow (to and from the card), it isn't occupying bandwidth that the SCSI card hasn't already used. You should actually be more concerned with the monitor that you choose to mate with your video card, since it can directly affect PCI bandwidth.

David Frangioni is an internationally known audio/video consultant, engineer, and producer, having worked with Chick Corea, Kiss, Aerosmith, MTV, Bee Gees, Desmond Child, and many more.

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Have a question or comment for David Frangioni? Visit him online at the *EQ* Boards, www.eqmag.com.

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- Headphone output with independent gain control knob
- 2 channel S/PDIF coaxial digital I/O
- 8 channel ADAT optical I/O can also be used as 2 channel optical S/PDIF

Pro Tools LE

- Supports 24 tracks of 16 or 24-bit audio and 128 sequenced MIDI tracks
- Sample-accurate simultaneous editing of audio & MIDI
- Real-time digital mixing capabilities include recall of all mixing parameters, support for edit and mix groups and complete automation of all volume, panning, mutes and plug-ins.
- Route and mix outboard gear in realtime
- MP3 and RealAudio G2 file support (Mac)
- Two plug-in platforms offer multiple options for effects processing— Real-Time AudioSuite (RTAS) is a host-based architecture that allows an effect to change and be dynamically automated in realtime as the audio plays back. —AudioSuite is a file-based format, that renders a new file with the processed sound.
- Bundled RTAS plug-ins include, 1 and 4-band EQ, Dynamics II- compressor, limiter, gate and expander/gate; Mod Delay - short, slap, medium, and long delays with modulation capabilities for chorus or flange effects and dither. AudioSuite plug-ins include Time Compression/Expansion, Pitch Shift, Normalize, Reverse.

MIDI Functions

- MIDI functions include graphic controller editing, piano roll display, up to 128 MIDI tracks and editing options like quantization, transpose, split notes, change velocity and change duration.
- MIDI data can be edited on the fly

MOTU AUDIO Hard Disk Recording Systems

The MOTU Audio System is a PCI based hard recording solution for the Mac and PC platforms. At the heart of the system is the PCI-324 PCI card that can connect up to three audio interfaces and allows up to 72 channels of simultaneous I/O. Audio interfaces are available with a wide range of I/O configurations including multiple analog I/O with the latest 24-bit A/D/A converters and/or multi channel digital I/O such as ADAT optical and TDIF I/O as well as standard S/PDIF and AES/EBU I/O. Each interface can be purchased separately or with a PCI-324 card allowing you to build a system to suit your needs. Includes drivers for all of today's hottest audio software and AudioDesk, multitrack recording and editing software for the Mac.

THEY ALL FEATURE–

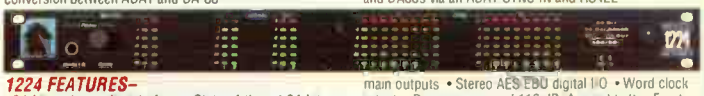
- Mac OS and Windows compatible • Includes software drivers for compatibility with all of today's popular audio software plus AudioDesk, MOTU's sample-accurate audio workstation software for Mac OS • Host computer determines the number of tracks that the software can record and play simultaneously, as well as the amount of real-time effects processing it can support • Front panels display metering for all inputs and outputs
- **AudioDesk Audio Workstation Software** for Mac OS features 24-bit recording, multi-channel waveform editing, automated virtual mixing, graphic editing of ramp automation, real-time effects plug-ins with 32-bit floating point processing, crossfades, support for third-party audio plug-ins (in the MOTU Audio System and Adobe Premiere formats), background processing of file-based operations, sample-accurate editing and placement of audio, and more

1296 24-bit/96kHz Interface Features–

- 24-bit, enhanced multi-bit 128x oversampling 96kHz converters • A-weighted signal-to-noise ratio of 117 dB • 12 Balanced XLR inputs and outputs can support two simultaneous 5.1 mixes • AES/EBU I/O with sample rate conversion both in and out • Compatible with existing PCI-324 cards (requires new PCI-324 driver)• Connect up to 3 1296 interfaces to one PCI-324 card for a total of 36 inputs and outputs or mix and match the 1296 interface with up to three of the other MOTU audio interfaces

2408 mkII FEATURES

- 7 banks of 8 channel I/O, 1 bank of analog, 3 banks of ADAT optical, 3 banks of Tascam TDIF, plus stereo S/PDIF. • Custom VLSI chip for amazing I/O capabilities • • Format conversion between ADAT and DA-88
- 8x 24-bit 1/4" balanced analog I/Os • 24-bit internal data bus for full 24-bit recording via digital inputs • Standard S/PDIF I/O for digital plus an additional S/PDIF I/O for the main mix • Sample-accurate synchronization with ADATs and DA88s via an ADAT SYNC IN and RS422



1224 FEATURES–

- 24-bit analog audio interface • State-of-the-art 24-bit A/D/A • Simultaneously record and play back 8 channels of balanced (TRS) +4 dB audio • 24-bit balanced +4 XLR main outputs • Stereo AES EBU digital I/O • Word clock in/out • Dynamic range of 116 dB (A-weighted) • Front panel displays six-segment metering for all inputs and outputs • Headphone jack with volume knob

CD RECORDING/MASTERING

ALESIS Masterlink ML-9600 High-Resolution Master Disk Recorder

The Alesis MasterLink ML-9600 is a 2-track 24-bit recorder that combines hard disk recording, CD burning, digital signal processing, and mastering functions to create compact discs in the standard "Red Book" 16-bit/44.1kHz format, or high resolution CDs that utilize Alesis' revolutionary CD24 AIFF-compatible technology. MasterLink is capable of recording and playing up to 24-bit/96kHz resolution CDs using the inexpensive, readily available CD-R media. The amazing sonic quality, powerful built-in tools and CD24 technology offers a uniquely versatile and affordable solution for everyone from large commercial audio facilities to project studios and recording musicians.

FEATURES–

- 24-bit 128x oversampling analog to digital and digital to analog converters
- Supports 44.1, 48, 88.2, 96 kHz sample rates and word lengths of 16-, 20- and 24-bit
- 20Hz-20kHz frequency response at 44.1/48 kHz sample rates
- 20Hz-40kHz frequency response at 88.2/96 kHz sample rates
- 113dB signal-to-noise ratio (A-weighted)
- Matsushita ATAPI CD-RDM drive allows up to 4x CD burning using standard CD-R discs.
- Built-in sample rate conversion & noise shaping to change sample rates & bit resolution as needed
- Reads and Writes 16-bit 44.1kHz Red Book Audio CDs
- Alesis' exclusive CD24 is a high-resolution mastering format that reads/writes files up to 24-bit 96kHz in the ISO 9660 disc format. AIFF caompatible file format that can be read by MacOS, Windows and Unix computer platforms.
- Built-in 3.2GB IDE hard drive
- Hard disk max recording times 95 min. @ 24-bit/96kHz 310 min. @ 16-bit/44.1kHz
- Create and store up to 16 playlists containing as many as 99 tracks

Analog Inputs and Outputs

- Balanced XLR connectors (+4dBu input and +19dBu max. output)
- Unbalanced phono (RCA) connectors (-10dBV input and +5dBV max. output)
- 1/4 inch TRS headphone output with level control

Digital Inputs and Outputs

- AES/EBU balanced XLR inputs and outputs
- S/PDIF unbalanced phono (RCA) inputs and outputs

Editing

- Gain control
- Cropping allows adjusting start and end points.
- Join and Split features allow combining and separating song sections.

DSP Finishing Tools

- Equalization, Compression, Normalizing and Peak Limiting

Includes

- Infra red remote control and rackmount brackets

marantz CDR-631 Professional CD Recorder

The CDR631 offer all the features and functions of the CDR630, its popular predecessor, but adds many features and functions that were previously unavailable. Its full complement of digital and analog connections lets you record your own CDs from audio sources such as CDs, LPs, cassettes, DAT, or even a computer.

Features–

- Pro and consumer CD-R and CD-RW compatible
- Track titles can be saved and edited in CD-TEXT format that can be read on CD-TEXT compatible CD players
- Memory buffer that prevents the beginning of tracks from getting cut off
- Menu selectable SCMS copy protection
- Digital and analog record level and balance control
- XLR-Balanced and RCA unbalanced analog inputs
- AES/EBU (XLR), Coaxial, and Optical digital inputs
- Unbalanced (RCA) analog and Coaxial digital outputs including Coaxial loop-out for unprocessed connection to other digital equipment
- IR remote control included

MICROBOARDS StartREC Digital Audio Editing/ CD Duplication System

The Microboards StartREC is the first digital audio editing system combined with a multidrive CD recordable duplication system for professionals. Audio is recorded to the internal 6.2 GB IDE hard drive using analog or digital inputs. Sample rate conversion is automatic. Tracks can be edited and sequenced using the StartREC's user friendly interface and up to 4 CDs can be recorded simultaneously. StartREC is the ideal solution for studio recording, mastering, post production or any pro audio environment requiring digital audio editing and short run CD-R duplication.

Features–

- 2X, 4X, or 8X recording speeds
- 6.2GB IDE hard drive
- Editing functions include move, divide, combine or delete audio tracks, add or drop any index or sub index, and create track fade in or fade out
- Coaxial SPDIF or AES/EBU digital input plus optical S/PDIF I/O
- XLR balanced and RCA Line inputs and outputs
- Automatic sample rate conversion from 32 and 48kHz
- Automatic CD Format Detection feature and user friendly interface provide one touch button operation
- Front panel trim pot and LCD display provide accurate input signal and time lapse metering
- SCMS (Serial Copy Management System) is supported regardless of the source disc copy protection status
- **StartREC Models Include:** ST2000 (2) 8x writers ST3000 (3) 8x writers and ST4000 (4) 8x writers

DIGITAL MULTI-TRACK RECORDERS

TASCAM MX-2424 24-Bit 24-Track Hard Disk Recorder

Co-designed by TASCAM and TimeLine Inc., the MX-2424 is an affordable 24-bit, 24-track hard disk recorder that also has the editing power of a digital audio workstation. A 9GB internal hard drive comes standard as well as a SCSI Wide port that supports external LVD (Low Voltage Drives) hard drives from up to 40 feet away. An optional analog and several digital I/O cards are available so the MX-2424 can be configured to suit your work environment. SMPTE synchronization, Word Clock, MIDI Time Code and MIDI Machine Control are all built in for seamless integration into any studio.

- Records 24 tracks of 24-bit audio at 44.1 or 48 kHz, or 12 tracks at 88.2 or 96 kHz. Up to 24 tracks can be recorded simultaneously using any combination of digital and analog I/D.
- Supplied 9GB internal drive allows 45 minutes of audio across all 24 tracks
- Wide SCSI port on the back panel allows you to add multiple drives. A front 5-1/2" bay available for installing an additional drive, or an approved DVD-RAM drive for back-up.
- ViewNet MX, a Java-based software suite for Mac and PC offers DAW style editing of audio regions, dedicated system set-up screens that make set-up quicker and easier and track load screens that make virtual track management a snap. Connects to a computer via a standard Ethernet line.
- Can record to Mac (SDII) or PC (.WAV) formatted drives, allowing later export to the computer. The Open TL format allows compatible software to recognize virtual tracks without have to load, reposition and trim each digital file.

Transport Controls-

- Jog/scrub wheel
- MIDI In, Out, and Thru ports are built-in for MIDI Machine Control.

Editing-

- Built-in editing capabilities include cut, copy, paste, split and ripple or overwrite
- 100 levels of undo
- Supports destructive loop recording and non-destructive loop recording which continuously records new takes without erasing the previous version.

Build-In Synchronization-

- TBUS protocol can sample accurately lock 32 machines together for 384 tracks at 96kHz, or 768 tracks at 48kHz.
- Can generate or chase SMPTE timecode or MIDI Time Code.
- Word Clock In, Out, and Thru ports

I/O Options-

- Optional analog and digital cards all provide 24 channels of I/O. There is one slot for analog and one for digital.
- IF-TD24- T/DIF module
- IF-AD24- ADAT Lightpipe module
- IF-AE24- AES/EBU module
- IF- AN24- A-D, D-A I/D module with DB-25 connectors

Software Updates-

- System updates are made available through a front panel Smart Card slot or via computer directly from the TASCAM web site.

DA-78HR Modular Digital Multitrack

The DA-78HR is the first true 24-bit tape-based 8-track modular digital multitrack recorder. Based on the DTRS (Digital Tape Recording System) it provides up to 108 minutes of pristine 24-bit or 16-bit digital audio on a single 120 Hi-8 video tape. Designed for project and commercial recording studios as well as video post and field production, the DA-78HR offers a host of standard features including built-in SMPTE Time Code Reader/Generator, MIDI Time Code synchronization and a digital mixer with pan and level controls. A coaxial S/PDIF digital I/O allows pre-mixed digital bouncing within a single unit, or externally to another recorder or even a DAT or CD recorder. Up to 16 DTRS machines can be synchronized together for simultaneous, sample accurate control of 128 tracks of digital audio.

Features-

- Selectable 16 bit or 24 bit High Resolution audio
- 24 bit A/D and D/A converters
- >104dB Dynamic range
- 20Hz - 20kHz frequency response ±.5dB
- 1 hr. 48 min. recording time on a single 120 tape
- On-Board SMPTE synchronizer - chase or generate timecode
- On-Board support for MIDI Machine Control
- Internal digital mixer with level and pan for internal bouncing, or for quick mixes
- Track slip from -200 to +7200 samples
- Expandable up to 128 tracks (16 machines)
- Word Sync In/Out/Thru
- Analog output on DB25 balanced or RCA unbalanced
- Digital output on TDIF or 2 channels of S/PDIF

A TO D CONVERTERS

APOGEE Rosetta 24-bit A to D Converter

The high-end quality analog to digital solution for the project studio. With support for both professional and consumer digital formats you can now record your audio at a higher resolution and with greater detail than standard converters found on MDM's, DAT's and DAW's. Ideal for mastering or tracking.

FEATURES–

- 24-bit, 44.1-48, 88.2-96 kHz Sample Rate (±10%)
- 116dB dynamic range (unweighted)
- Improved UV22HR for 16 and 20-bit A/D conversion

FRONT PANEL:

- Power switch • Sample Rate (44.1, 48, 88.2, 96kHZ)selector • 16-bit (UV22), 20-bit (UV22) and 24-bit resolution selector • S/PDIF-ADAT optical selector • Soft Limit on or off • 12-segment metering w/ over ondicator & Meter Clear switch • Level trim

REAR PANEL:

- XLR balanced inputs • 2 x AES/EBU for 88.2/96kHz 2 channel path, Coaxial S/PDIF, switchable S/PDIF or ADAT optical outputs • Wordclock out

LUCID AD 9624 24-bit A to D Converter

Transparent analog to digital conversion designed to bring your music to the next level. XLR balanced inputs feed true 24-bit converters for revealing all the detail of the analog source. 16-bit masters can take advantage of the AD9624's noise shaping function which enhances clarity of low level signals.

FEATURES–

- 24-bit precision A/D conversion • Support for 32, 44.1, 48, 88.2 & 96kHz sample rates • Wordclock sync input• Selectable 16-bit noise shaping •

Simultaneous AES/EBU, coaxial and optical S/PDIF outputs • 20-segment LED meters w/ peak hold & clip indicators • **ALSO AVAILABLE:** DA9624 24-bit D/A converter

DIGITAL MIXERS

Roland VM Basic 72 Digital Mixing System

The all digital Roland V-Mixing System, when fully expanded, is capable of mixing up to 94 channels with 16 stereo (32 mono) onboard multi-effects including CDSM Speaker Modeling. Utilizing a separate-component design, comprised of the VM-C7200 console and VM-7200 rackmount processor, allows the V-Mixing System to be configured to suit your needs. Navigation is made easy via a friendly user interface, FlexBus and EZ routing capabilities as well as a large informative LCD and ultra-fast short cut keys.

- 94 channels of digital automated mixing (fully expanded)
- Up to 48 channels of ADAT/Tascam T-DIF digital audio I/D with optional expansion boards and interfaces
- Separate console/processor design
- Quiet motorized faders, transport controls, total recall of all parameters including input gain, onboard mixer dynamic automation and scene memory
- 24 fader groups, dual-channel delays, 4-band parametric channel EQ + channel HPF
- FlexBus and "virtual patchbay" for unparalleled routing flexibility
- Up to 16 stereo (or 32 mono) multi-effects processors using optional VS8F-2 Effects Expansion Boards (2 stereo effects processors standard)
- COSM Speaker Modeling and Mic Simulation technology
- 5.1 Surround mixing capabilities
- EZ Routing allows mixer settings to be saved as templates
- Realtime Spectrum Analyzer checks room acoustics in conjunction with noise generator and oscillator
- Digital cables between processor and mixer can be up to 100 meters long- ideal for live sound reinforcement.

- **VS8F-2 Effects Expansion Board** -- Provides 2 stereo effects processors including CDSM Speaker Modeling. Up to 3 additional boards can be user-installed into the VM-7200 processor, for 8 stereo or 16 mono effects per processor.
- **VM-24E I/O Expansion Board** -- Offers 3 R-Bus I/Os on a single board. Each R-Bus I/D provides 8-in/8-out 24-bit digital I/O, totalling 24 I/D per expansion board.
- **OIF-AT Interface Box for ADAT/Tascam** -- Converts signals between R-Bus (VM-24E expansion board required) and ADAT/Tascam T-DIF. Handles 8-in/8-out digital audio. 1/3 rackmount size.
- **VM-24C Cascade Kit** -- Connects two VM-Series processor units. Using two VM-7200 processors cascaded and fully expanded with R-Bus I/D, 94 channels of audio processing are available.

EFFECTS & PROCESSING

Lexicon MPX-500 24-Bit Dual Channel Effects Processor

The MPX 500 is a true stereo 24-bit dual-channel processor and like the MPX100 is powered by Lexicon's proprietary Lexichip and offers dual-channel processing. However, the MPX 500 offers even greater control over effects parameters, has digital inputs and outputs as well as a large graphics display.

- 240 presets with classic, true stereo reverb programs as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, 5.5 second Delay and Echo
- Balanced analog and S/PDIF digital I/D
- 4 dedicated front panel knobs allow adjustment of effect parameters. Easy Learn mode allows MIDI patching of front panel controls.
- Tempo-controlled delays lock to Tap or MIDI clock

t.c. electronic M-One Dual Effects Processor

The M-One allows two reverbs or other effects to be run simultaneously, without compromising sound quality. The intuitive yet sophisticated interface gives you instant control of all vital parameters and allows you to create awesome effects programs quickly and easily.

- 20 incredible TC effects including, Reverb, Chorus, Tremolo, Pitch, Delay and Dynamics
- Analog-style user interface
- 100 Factory/100 User presets
- Dual-Engine design
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

D-Two Multitap Rhythm Delay

Based on the Classic TC2290 Delay, the D-Two is the first unit that allows rhythm patterns to be tapped in directly or quantized to a specific tempo and subdivision.

- Multitap Rhythm Delay
- Absolute Repeat Control
- Up to 10 seconds of Delay
- 50 Factory/100 User presets
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

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MICROPHONES

AKG C414 TLII "Vintage TL"

Combines the best of old and new: legendary C12 acoustics and the latest generation of C414 transformerless FET electronics. Although similar in design and shape to the C414BULS, the TLII features a capsule that is a faithful sonic recreation of the one used in the classic C12 tube mic combined with computer-aided manufacturing techniques that assure greater uniformity in response from microphone to microphone.

FEATURES-

- Cardioid, hypercardioid, omnidirectional and figure 8 polar patterns
- Warm, smooth microphone that is suitable for high-quality digital recording.
- **Frequency response 10Hz to 20kHz**

C4000B ELECTERET CONDENSER

This new mic from AKG is a multi polar pattern condenser micropone using a unique electret dual large diaphram transducer. It is based on the AKG SolidTube deisgn, except that the tube has been replaced by a transistorized impedance converter/ preamp. The transformerless output stage offers the C4000B exceptional low frequency

FEATURES-

- Electret Dual Large Diaphram Transducer (1st of its kind) • Cardioid, hypercardioid & omnidirectional polar patterns • High Sensitivity • Extremely low self-noise • Bass cut filter & Pad switches • Requires 12, 24 or 48 V phantom power • Includes H-100 shockmount and wind/pop screen
- **Frequency response 20Hz to 20kHz**

RØDE NT-2 Condenser Mic

The RØDE NT2 is a large diaphragm true condenser studio mic that features both cardioid and omnidirectional polar patterns. The NT-2 offers superb sonic detail with a vintage flavor for vocal and instrument miking. Like all RØDE mics the NT-2 is hand-assembled in Australia and is available at a breakthrough price.

FEATURES-

- Dual pressure gradient transducer • 1" capsule with gold-sputtered membranes • Low noise, transformerless circuitry • Omni and cardioid polar patterns • 135dB Max SPL • High pass filter switch and -10dB pad switch • Gold plated output connector and internal head pins • Shockmount, Flight Case, and Pop Filter included
- **20Hz-20kHz frequency response**

audio-technica AT4047 Cardioid Condenser

The AT4047 is the latest 40 Series large diaphragm condenser mic from Audio Technica. It has the low self noise, wide dynamic range and high sound pressure level capacity demanded by recording studios and sound reinforcement professionals.

FEATURES-

- Side address cardioid condenser microphone for professional recording and critical applications in broadcast and live sound • Low self noise, wide dynamic range and high SPL • Switchable 80Hz Hi Pass Filter and 10dB pad
- **Includes** AT8449/SV shockmount

MICROPHONE PREAMPS

AVALON DESIGN VT-737SP Mono Class A, Vacuum Tube-Discrete Preamp-Opto-Compressor-Equalizer

The VT-737SP is a vacuum tube, Class A processor that combines a mic preamp, instrument DI, compressor and sweepable 4-band equalizer in a 2U rack space. Like all Avalon Design products the VT-737SP utilizes a minimum signal path design with 100% discrete, high-bias pure Class A audio amplifiers and the best active and passive components available. Used by renowned artists and studios world wide and the winner of the Electronic Musician 1999 Editors' Choice Award for Product Of The Year.

FEATURES–

- Combination of TUBE preamplifiers, opto-compressor, sweep equalizer, output level and VU metering in a 2U space
- Four dual triode vacuum tubes, high-voltage discrete Class A with a 10 Hz to 120kHz frequency response ±0.5dB
- **The Preamp** has three input selections- The first is a high performance XLR balanced mic input transformer with +48v phantom power, the second is a high impedance instrument DI with a 1/4" jack located on the front panel and the third is a discrete high-level Class A balanced line input.
- High gain switch boosts overall preamp gain and a passive- variable high pass filter, hardwire relay bypass and phase reverse relay is available for all three inputs
- **The Opto-Compressor** uses a minimum signal path design and features twin Class A vacuum tube triodes for gain matching. A passive optical attenuator serves as a simple level controller. Variable threshold, compression ratio and attack and release offer dynamics control from soft compression to hard-knee limiting.
- The dual sweep mid-EQ can be side chained to the compressor allowing a broad range of spectral control including de-essing. The EQ can be assigned pre and post compressor from the front panel to add even greater sonic possibilities.
- Two VT-737 SPs can be linked together via a rear panel link cable for stereo tracking
- **The Equalizer** utilizes 100% discrete, Class A-high-voltage transistors for optimum sonic performance.
- The low frequency passive shelving EQ is selectable between 15, 30 60 and 150HZ with a boost and cut of ±24dB
- The high frequency passive shelving EQ is selectable between 10, 15, 20 and 32 kHZ with a boost and cut of ±20dB
- The low-mid frequency is variable between 35 to 450 Hz while the high-mid frequency is variable from 220Hz to 2.8 kHz. Both mid-hand frequencies offer a boost and cut of ±16 dB and a hi-Q/lo-Q switch.
- When the EQ to side chain is used, the low and high EQ is still available for tonal adjustment
- **The Output** level is continuously variable and utilizes an another dual triode vacuum tube driving a 100% Class A, high-current balanced and DC coupled low noise output amplifier.
- Sealed silver relay bypass switches are used for the most direct signal path

POWERED STUDIO MONITORS

VERGENCE A-20 Studio Reference Monitor System

Incorporating a pair of 2-way, acoustic suspension monitors and external, system-specific 250 watt per side control amplifier, the A-20 provides a precise, neutral studio reference monitoring system for project, commercial and post production studios. The A-20's control amplifier adapts to any production environment by offering control over monitoring depth (from near to far field), wall proximity and even input sensitivity while the speakers magnetic shielding allows seamless integration into today's computer based studios.

- Type Modular, self-powered near/mid/far-field monitor.
- 48Hz - 20kHz frequency response @ 1M
- Peak Acoustic Output 117dB SPL (100ms pink noise at 1M).
- XLR outputs from power amp to speakers
- Matched impedance output cables included,
- -6dB LF Cutoff 40Hz
- 5 position wall proximity control
- 5 position listening proximity control between near, mid and far-field monitoring
- Power, Overload; SPL Output, Line VAC and Output device temperature display

Amplifier

- Amplifier Power 250W (continuous rms/ch), 400W (100ms peak).
- XLR, TRS input connectors
- Headphone output
- 5-position input sensitivity switch with settings

Speakers

- 2-way acoustic suspension with a 6.5-inch treated paper woofer and a 1-inch aluminum dome tweeter
- Fully magnetically Shielded with an 18-inch recommended working distance

EVENT ELECTRONICS PS-5 Bi-Amplified Project Studio Monitors

The PS-5s are small format, full-range, non-fatiguing project studio monitors that give you the same precise, accurate sound as the highly acclaimed 20/20 series studio monitors. The use of custom driver components, complimentary crossover and bi-amplified power design provides a wide dynamic range with excellent transient response and low intermodulation distortion.

FEATURES-

- 5-1/4-inch magnetically shielded mineral-filled polypropylene cone with 1-inch diameter high-temperature voice coil and damped rubber surround LF Driver
- Magnetically shielded 25mm diameter ferrofluid-cooled natural silk dome neodymium HF Driver
- 70 watt continuous LF and 30 watt continuous HF amplification per side
- XLR-balanced and 1/4-inch (balanced or unbalanced) inputs
- 52Hz-19kHz frequency response ±3dB
- 2.6kHz, active second order crossover
- Built-in RF interference, output current limiting, over temperature, turn-on transient, subsonic filter, internal fuse protection
- Combination Power On/Clip LED indicator
- 5/8" vinyl-laminated MDF cabinet

Hafler TRM-6 Bi-Amplified Studio Monitors

Offering honest, consistent sound from top to bottom, the TRM-6 bi-amplified studio monitors are the ideal reference monitors for any recording environment whether tracking, mixing and mastering. Supported by Hafler's legendary amplifier technology providing a more accurate sound field, in width, height and also depth.

FEATURES-

- 33 Watt HF & 50 Watt LF amplification
- 1-inch soft dome tweeter and 6.5-inch polypropylene woofer
- 55Hz - 21kHz Response
- Magnetically Shielded
- Electronically and Acoustically Matched

Also Available- TRM-8

- 1-inch soft dome tweeter and 8-inch polypropylene woofer
- 45Hz - 21kHz frequency response ±2dB
- 75 Watt HF, 150 Watt LF amplification

TRM-10s And TRM-12s Active Subwoofers

Combining Hafler's legendary amplifier technology with a proprietary woofer design, the TRM10s and TRM12s active subwoofers provide superb bass definition required in today's studio and surround sound environments.

TRM-10s

- 10-inch cellulose fibre cone down firing woofer.
- 200 watt low frequency amplifier
- 30Hz to 110Hz frequency response ±2dB
- 24dB/octave Linkwitz-Riley crossover variable (40Hz to 110Hz)

TRM-12s

- 12-inch cellulose fibre cone down firing woofer.
- 200 watt low frequency amplifier
- 25Hz to 110Hz frequency response ±2dB
- 24dB/octave Linkwitz-Riley crossover variable (40Hz to 110Hz)

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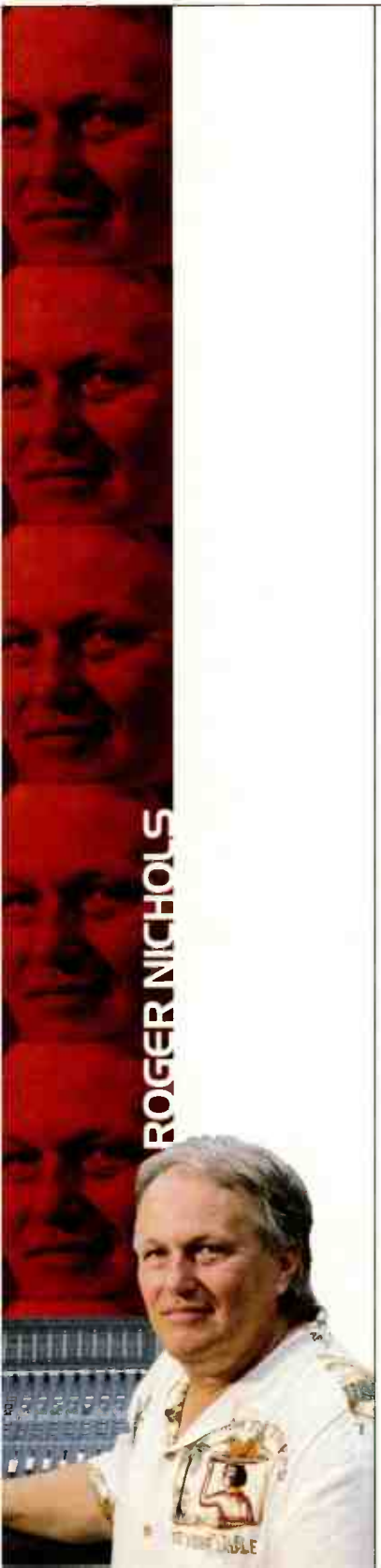

It is possible to hear something and tell whether or not you like it; we do it all the time. What you *can't* do is tell whether or not what you're listening to is the best representation of the original.

If you're recording in the studio, you put down a take and listen to the playback to decide if everything is in order. You can determine whether all the musicians played their part correctly, if all the instruments are in tune, if the tempo is correct, if the groove is what you were looking for. You can also tell if the sound of the song as a whole is good enough to go on the finished CD. What you can't tell is if the sound you hear on playback is the same as the sound that was heard in the control room during the recording.

The only way to determine if what was recorded is the same as what was played is to compare the two directly. To do this comparison, you need the band to play along with the recorded material. You can then switch between then live material and what's coming back from the recorder. To do this A/B comparison correctly, you have to change sources at the recording console, switching between monitoring the live console outputs and monitoring the tape playback. If you do the comparison by switching the recorder from playback to input monitoring, you'll still be hearing the converters and electronics of the recording device, which add coloration to the sound.

If you're still recording analog, you'll hear the biggest difference. I'm not talking about whether you like the sound of analog or not, but whether the input sounds like the tape playback. They're not the same, and that's just the law of physics. Digital recording doesn't necessarily sound exactly the same as the input signal, but it is measurably closer to the input than analog recording.

Every process does something to the sound. Some of the processes are done to make the sound more pleasing to the artist, engineer, or producer. A process can improve the sound or make it worse. These processes include mic choice, mic position, instrument choice, instrument placement, instrument tuning, player performance, guitar amp settings, drum head mutes, and literally hundreds of other variables that color the sound on the way to the recording machine. The recording method is one of those variables.

Shootouts

Whenever a bunch of equipment gets accumulated in one place, there will inevitably be a shootout to determine which piece of equipment is better than all of the other pieces of equipment. I was involved in one of these shootouts in Nashville a few years ago involving a swarm of digital converters. All of the converters were calibrated exactly so that there would be no level change when switching between them. As a control, a straight wire connection was one of the choices.

About a dozen engineers showed up to evaluate the converters. The source material varied between live acoustic guitar, live vocal, multitrack analog and multitrack digital material mixed through an SSL with Ultimation. Remember that one of the choices was the original signal right from the console. The other seven choices were the six converters and the straight wire. Only two of us picked the straight wire as sounding the most like the original. Everyone else picked converter #3. They said that they liked converter #3 because it made their mixes sound better than what was coming directly out of the console. They said *it made their mixes sound better than what came out of the console.*

Well, that wasn't exactly what I was looking for. I wanted a converter that sounded just like the straight wire. I wanted a converter that did nothing of its own to my mixes. I wanted to mix by listening to the output of the console, print the mix to the mix machine, and when I played the mix back, I wanted to hear exactly the same thing. If I wanted anything different, I would change it in the mix, or do it in mastering.

I want the same thing in my multitrack recording. I don't want the recorder to do anything to my recording. If I want an analog sound on the drums, I will record them onto an analog machine and then put them back. If I want the hi-hat brighter, I will record it brighter. If I want some compression on the vocal, I will compress the vocal.

During the shootout, if I had not had the original for the comparison, I probably would have selected the same converter that everyone else selected. It sounded good. If there was no way to tell what it was supposed to sound like, then converter #3 would have been the clear winner as the best converter of the shootout. Only by comparing it with the original source could we know whether the converter was doing a transparent job or not.

CD Formats

This leads me to the latest confrontation on the

► *continued on page 155*

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The Neumann M 147 Tube

For years, vintage Neumann tube mics, such as the venerable U 47, have been high-priced, highly prized commodities. Why, when advances have created mics with near-perfect, virtually transparent reproduction, have producers and engineers travelled to the ends of the earth in search of these vintage relics? Because of the way they sound (especially the way the sound sits in the mix).

Enter the M 147 Tube.

Using the same capsule as the classic U 47 and its smaller cousin the U 47 FET, the M 147 Tube microphone brings a warmth, presence and detail to vocals that is simply unattainable from any other mic being produced today, regardless of how much it looks like a Neumann. The fact is, there is really only one way to get that classic sound you seek. Fortunately, it's priced well within your reach.

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What The Professionals Are Saying About The Neumann M 147 Tube:

"So far, I'm thrilled to pieces with the Neumann M 147 Tube. I don't think there's any instrument that I wouldn't try them on. Whatever instrument I used them for, I was very impressed with the sound. I wish I had about five or six of them!"
- Al Schmitt, as quoted in EQ, March 1999

"I would recommend the M 147 highly for rock, rap, pop, jazz or blues vocals; drum room and/or kick drum miking; all tube and solid-state instrument amplifiers; nylon string guitar; and low-volume or indistinct sound sources that need some extra presence. and for any type of digital recording. In short, I like the M 147 a lot -- so much so that I bought one."
- Myles Boisen, Electronic Musician, August 1999

"The particular kind of presence it adds is really unique and desirable, and it's really not available from any other mic or easily obtainable with an equalizer. Typically, condenser mics that have a forward character are really just brittle and edgy, and the M 147 is completely different from that."
- Monte McGuire, Recording, July 1999

"I asked the singer on my session which mic she preferred and, when presented with a finite budget, her pick (and mine) was the M 147. Classic Neumann sound, tube electronics, the U 47 legacy, and a price that won't savage your bank account. Gotta love it!"
- Rick Chinn, Audio Media, February 1999

"The M 147 proves again that however close the imitators get, there is no substitute for the genuine article. This is the real McCoy and although it cannot be called cheap, its simple approach means that it is far more accessible than a valve Neumann would normally be expected to be. Another classic in the making."
- Dave Foister, Studio Sound, February 1999

"It's my opinion that the tone of the Neumann would not require much EQing during mixdown; a decided advantage. Its high end would sit nicely in a mix, and its round but controlled low end would not have to be cut to provide room for other instruments."
- Mitch Gallagher, Keyboard Magazine, June 1999

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