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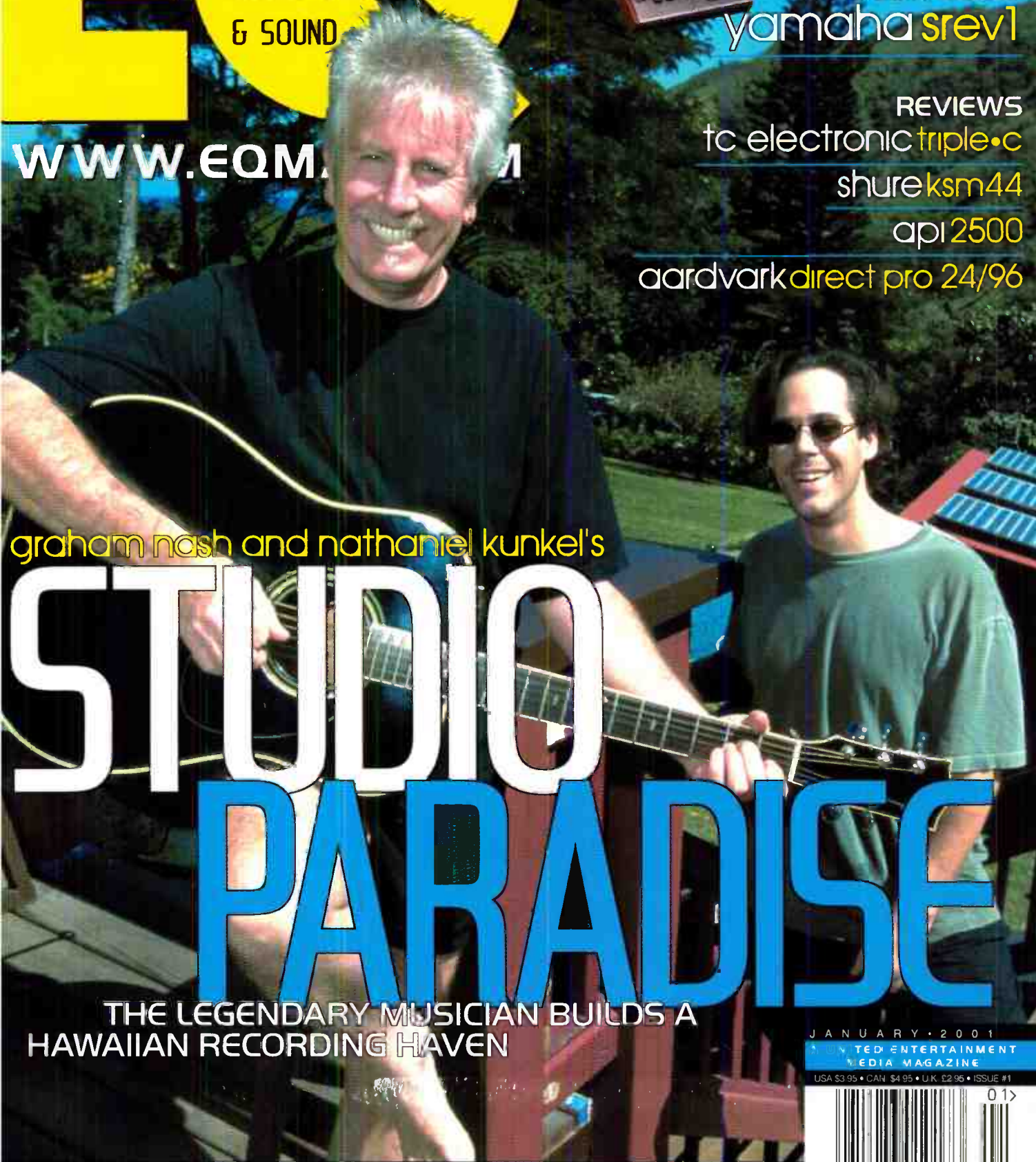
graham nash and nathaniel kunkel's

STUDIO
PARADISE

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HAWAIIAN RECORDING HAVEN

BRANCHING OUT INTO VIDEO GAME AUDIO

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THE RACK

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- SENSATIONAL SEQUENCING
- DUAL POLYPHONIC ARPEGGIATORS
- USER-INSTALLABLE EXPANSION

THE KEYBOARD
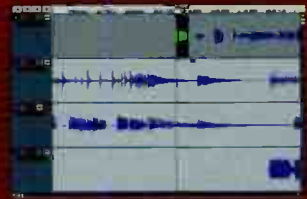
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WORLD CHAMPIONSHIP HEAVYWEIGHT ACTION

KORG

CIRCLE 21 ON FREE INFO CARD

VOLUME 12 • ISSUE 1
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Graham Nash and Nathaniel Kunkel in Hawaii.
Photo by Mr. Bonzai

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Force of Habit

About a month ago, as I write this, I underwent laser surgery on my eyes. Pretty amazing, really; after almost 30 years of wearing glasses, seven minutes under a beam of light per eye and I can see perfectly. Naturally, after all that time wearing specs, I developed a few hard-to-break habits. I still reach for my glasses when I wake up in the morning in order to see what time it is. Every time I bend over or tilt my head forward, I feel an urge to push my glasses up. Since I'm not wearing glasses, I catch myself reflexively squinting to see clearly, when in fact I can see just fine with my eyes wide open.

The experience got me thinking about habits, and naturally, when I'm thinking about something it ends up as fodder for an editorial (convenient for me, since I have to write these things each month). We all know habits can be good things; often they allow us to accomplish tasks with increased efficiency, they can free our minds from dealing with the mundane, thus letting us put our gray matter to work on higher-level or more creative tasks, and so on.

However, habits can also carry a negative aspect — the dreaded *bad habit*. (Not that I have any of those myself, but I've heard that people occasionally suffer from them.) But the point of this editorial isn't to get you thinking about your bad habits. (Assuming you have any — if you're in doubt, ask your significant other or spouse; surely they can put together a list for you.) No, the point here is to consider a more subtle kind of habit: responding to various situations through habit or by always doing what you normally do.

Specifically, let's talk about this as it applies to creative endeavors such as engineering, producing, composing, and playing an instrument. How many times do you find yourself "automatically" doing something in your studio? Miking a vocal? Throw up this mic; it always works. Need to compress electric bass? Patch in the box you always use for that application. Need a reverb? Preset 103, "Bright Vocal Hall" always works. On and on...in some ways, making automatic choices through habit is cool — it lets you work faster, and you can devote your mind to other aspects of the job at hand. However, it can also lead to "doing things in the way you've always done them," which ultimately isn't conducive to enhanced creativity.

Since I don't have any bad habits to break (depending on who you ask), here's my New Year's resolution: I'm going to take a brief instant to think before I make each move in the studio. Maybe after thinking about it, I'll make the exact same decision I would have made automatically, but at least I'll be actively controlling the creative decisions I make and therefore more actively controlling the quality of the music I'm making.

Happy New Year!

—Mitch Gallagher
mgallagher@uemedia.com

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> "IN THE NAPSTER AGE, WHEN THE PUBLIC IS BEGINNING TO THINK THAT FREE MUSIC IS A GOD-GIVEN ENTITLEMENT (LIFE, LIBERTY, AND THE PURSUIT OF RIPPING), WE AS A MUSIC COMMUNITY MUST SPEAK LOUD AND CLEAR ABOUT WHAT'S RIGHT, ESPECIALLY AMONG OURSELVES."
>
> — *BILL MULLIN, SIGNATURE MUSIC*

JUST SAY NO

You can not only have an *amen,* but a *hallelujah.* ["Just Say No," November 2000] It was a breath of fresh air to read how you are encouraging all music makers to take a stand against ripping off fellow musicians' compositions.

In the Napster age, when the public is beginning to think that free music is a God-given entitlement (life, liberty, and the pursuit of ripping), we as a music community must speak loud and clear about what's right, especially among ourselves. I have spent sixteen years educating people about not stealing music off commercial records for their productions with my music library, Signature Music. I've heard all the excuses and phony justifications. ("Hey, I'm only going to use fifteen seconds of it.")

Years ago I picked up a cue from my associate Doug Wood at OmniMusic (now on ASCAP's board of directors and doing great things for commercial instrumental composers). He decided to clear his library of all sound-alikes for self-preservation against lawsuits.

In the quest for having their music heard or being paid anything just to have a music gig, I'm afraid some producers will stoop to aid and abet copyright infringers. It's too bad. Some of them are just young and naive. Your article will help them develop awareness and professional conduct. Others will just be whores, but it's the responsible musician's duty to keep that to a minimum. Testify, brother Jim!

Bill Mullin
Signature Music

THE NERVE

This column struck a nerve! ["Just Say No," November 2000] The illegal use of copyrighted music has been a real problem in the radio industry for years. Especially in smaller markets where the advertisers have very limited budgets and can't even consider having a studio produce a custom music package. Over eighteen years I don't know how many times I've had heated words with the sales department, refusing to do an ad with copyrighted music in it.

ASCAP has been aggressive in combating this and (maybe) the stations and advertisers are finally getting the idea. But a lot of musicians have missed a lot of royalties because of it.

Robert Moulton
via email

THE DONGLE DILEMMA

I was just reading your review on Nuendo. [Nov., 2000] My email is about your comment on the copy protection of Nuendo. The dongle has started many pointless arguments in the past — people seem to disagree with it "just because." However, your point about the dongle as a weakness is that it "can be a hassle if you work on multiple computer systems (such as a laptop and a desktop)." I'd have to say that this comment holds no grounds. If you read the software license agreement, you can only install Nuendo on one machine anyway. So, if you were to install it on a desktop and a laptop you would be breaking the software license agreement. Just something to think about....

Christopher Hawkins
Streamworks Audio

NATURAL HIGHS

In the November 2000 issue, Tony Visconti states that, "You're not going to get clarity out of a bass amp." I've been tracking my Wal basses via a DI and an SWR bass head with Goliath cabinet for some time now. Never completely satisfied with combining, I would use one or the other depending on the song — usually the DI.

One day I tried miking the cabinet's speaker and tweeter separately and discovered that, once properly tweaked, the essential tone, as well as the attack characteristics, were virtually identical to the DI but with preferable warmth and character. You can add more or less high end without EQing, and also overdrive sounds better through the amp than the DI. The Wals have wonderful fidelity and presence, and every bit of it is now translated through the SWR rig into my sessions. I guess you really can get clarity out of a bass amp after all!

Steve Miles
West Palm Beach, FL

AT LEAST HE KEPT AN OPEN MIND

Please cancel my subscription — it went into the trash as soon as I saw Marilyn "Moron" on the cover. I have no use for any of your magazines ever again.

Rick
Bassax Music Studio
Dallas, TX

MORAL DECLINE

I have been a loyal fan of your magazine for several years. I enjoy the informative reviews and other regular columns in your magazine. I must, however, take issue with this month's cover featuring Marilyn Manson. I don't believe that showing his horrid face on your magazine shows any moral consideration on your part. This self-proclaimed anti-Christ and others like him, along with major record labels, are the leading factors in the moral decline in the United States and the world. As long as idiots like him have a way of getting their face and work shown in the media, Satan will continue to have a death grip on the youth of the world.

Bill Orr
Edgemoor, SC

RAP AS SCAPEGOAT

In response to the "Use It, Don't Abuse It" letter in the November, 2000 issue: Why does rap always have to be a scapegoat? I don't like rap, but, quite frankly, I'm sick of hearing people blame it for the "downfall" of modern music. It takes plenty of talent to program beats and whatnot into drum machines. I mixed a few demos for a friend of mine; he wrote some hip-hop music and one of his friends rapped lyrics

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ALESIS

> ### "IF YOU REALLY WANT TO DO MODERN MUSIC A FAVOR, WRITE A LETTER TO ALL OF THE WORTHLESS STAFF PRODUCERS AT THE MAJOR LABELS. THEY'RE THE ONES CRANKING OUT THE GARBAGE YOU SEE ON MTV."
>
> *—CHRIS BROWN, POWER PYE STUDIOS*

over the top. They sat for hours working with their numerous samplers, synths, and modules — as long as someone would sit with a guitar strumming chords to write a song.

I don't like the music they're coming up with, but you can't say it doesn't take talent, or that people who make rap music are less skilled or intelligent. If you really want to do modern music a favor, write a letter to all of the worthless staff producers at the major labels. They're the ones cranking out the garbage you see on MTV.

Chris Brown
Power Pye Studios
Pittsburgh, PA

COULD YOU ELABORATE?
I read Ed Bialiach's letter in the November 2000 issue with much interest. I was surprised at how similar my reaction to EMTEC's advertisement was. I too "ripped the ad out of the new issue" — it made a great pin-up! I too "considered canceling my subscription" — not to *EQ* but to *Playboy!* I mean, with the high-quality *articles* in your publication, I no longer need to read the pictures...I mean *articles*...(sorry for the slip) in *Playboy.* Is there any chance the Media Goddess will appear in a fold out?

Then, in the same issue, I saw EMTEC's vice president of sales and marketing, Studio and Broadcast Products — her smiling face was just above the Media Goddess — is she the next Media Goddess? Maybe she could elaborate, for our "convenience and pleasure," a little on what the Media Goddess has to teach us.

Bill Prentice
via email

LOVE/HATE RELATIONSHIP
There are six regular employees at our studio who are loyal fans of your magazine. We've been reading you since the first issue and have largely built the studio using the magazine as a guidebook. We love you.

However, independently and unanimously, all six of us began whining about the new format and font. It's very hard to read. We all hate it.

Sonny Gentry
Exit 56 Studios
Memphis, Tennessee

IT'S A TREAT
A beautiful job on the redesign of *EQ.* The layout/format is fresh, clean, easy to navigate, easy on the eyes, and a real treat.
Dan DeMars
Norwich, VT

NOT PORN
Regarding letters to *EQ* in the November 2000 issue, where exactly is the pornography in the BASF advertisement? Is the female form pornographic in and of itself? Is every statue of the female form pornography? I'm baffled by this line of thought. To the person who wrote the letter titled "Porn Ads," I would suggest that he open his eyes. What he will see is the interior of his own colon! I guess if you're gonna pull the BASF ad, you better pull the JoeMeek ad, too. After all, he did kill his landlord and *she* was someone's wife and/or daughter.
Hans Schneider
via email

CONSPIRACY THEORY
Howard Massey starts with a false premise in Part I of "Surround Sound in the Project Studio." [October 2000] He writes that analog technology "didn't allow" surround, "which is one reason Quad didn't take off in the '70s." I can think of at least three analog formats of that era that could deliver Quadrophonic sound: LP, eight-track cartridge, and 1/4-inch open-reel tape. The last is considered a very good — in fact, "audiophile" — delivery medium.

The reason Quad didn't take off is that consumers didn't perceive the benefits as outweighing the extra costs and burdens involved. Hmm — aren't those even greater with six-channel sound?

Then Mr. Massey accuses those of us who see surround as being driven from the top down of being "conspiracy theorists." It doesn't take a conspiracy or a theory to see the fact that surround is being promoted despite objections that no one knows what to do with the LFE speaker or the "dialog" channel, nor how to design a control room for surround. This suggests that something besides demand is driving surround.

I hope the "surround hounds" understand that what they are advocating is the elimination of music as an aural experience. Joe Public will sit on his "home theater" chair and listen to 5.1 music while staring at a blank screen. A blank screen! Nature abhors a vacuum. An empty canvas *will be filled.* If they have a screen, they will soon demand that we put a picture on it. The inevitable end result will be a *visually* driven presentation instead of music. Can any audio professional believe that this is a worthwhile goal?
Dan Popp
Colors Audio

PASSION
Great editorial! ["Word For The Day," November 2000] I'd like to let you know of a project that I recently finished. My latest CD, *Fluid,* was the product of my own challenge to create a collection of new music from the manipulation of a single sound. It worked. I took a 14-second sample, and, through the magic of recording technology — using a modest project studio setup — turned that sound into a 54-minute CD of music. I spent two solid months on this project, even turning down paying work in the process. Why? Well, you said it in your editorial — *passion.* Sometimes just stepping back, taking a moment, and realizing what drives us to be a part of this business can make a significant impact.
Michael Oster
F7 Sound and Vision

CLARIFICATION
In our November issue, the studio used to illustrate the second part of Howard Massey's "Surround Sound in the Project Studio" was Margarita Mix de Santa Mónica, a full-blown commercial production facility with 5.1 surround capabilities. It was not our intention to imply that Margarita Mix is project studio; the photo was used to exemplify what a well-done surround studio could look like. We apologize for any confusion we may have caused.

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Well? Is it real-time or nothing? Do you guys ever let a fast one go out?

—Jules

The best speed to use is 2x. If you use 1x, the laser can burn pits that are too deep, resulting in errors.

—Irbreez

1x got me in trouble. I only burn at 4x now, never had a single problem at that rate. Besides, you don't have to wait nearly as long.

—alphajerk

I have a LaCie 12x (Sanyo) and a LaCie 8x (Yamaha). Both writers are Firewire. No problems at 12x or 8x.

—InSights7

The recommended speed from most duplication houses is 2x. Use 2x for final master CDs that are going to replication. Use the faster speeds for data archival and for test discs (if you're in a hurry).

—David Frangioni

VOCALS SITTING ON TOP OF THE MIX

How do I get male tenor vocals to sit nicely inside the mix? I don't know which EQ bands to cut in the music to make it fit, and so the vocal seems to be on top and separated from the music.

—scottk184

This is always a common problem. Especially when I have worked in Nashville, the word from the top has been, "Make sure that vocal is loud."

It sounds like you are getting the music to balance, and then trying to place the vocal. Try this instead — it will feel weird at first, but trust me, it will work eventually: The first thing to push up is the vocal. Put a little reverb or whatever on it, and then concentrate on building the music around the vocals, not the other way around. You'll intuitively

shape the music to wrap around and embrace the vocals — and when you're going for that final balance, turn the monitors way down, even mono it and turn off one speaker, and see what you have — that's where the truth should be. Then, you may want to use overall compression (and EQ, perhaps) to really get it to glue together.

—Ed Cherney

That works for anything you have trouble placing. I just had a mix I was working on with a double-tracked baritone Danelectro guitar, and while I thought the mix sounded nice and full, the bass was lost, and competing with the guitar in definition. Well, the "leader" of the band was the bassist, so you had to be able to hear the bass at all times (politics). After bringing up the bass fader, then the guitar again to match it, the drums were getting drowned out, so pulling them up became necessary (the nasty cycle of competitive levels).

In a fit, I pulled every fader down and put up the bass, comparing the level to pre-released CDs and then never touched it after that. All level changes were made by lowering the other faders or with EQ. Next, I brought up the drums, guitar, and vox, all around the bass. Needless to say, the mix ended up sounding far better than it had. You can also use this technique for any part that is the hook of the section.

—alphajerk

CREATING AND MARKETING MY OWN CD

I'd like to know how hard it would be, after recording my CD, to market it and get it to sell. I need to know what the steps would be to get it to the public. I'm thinking things like copyright, ASCAP, royalties, etc. Where could I find this type of information? Are there books available about this?

—Finger Picker

To get copyright forms, go to the Library

of Congress Web site (http://lcweb.loc.gov/copyright/forms/). You need Adobe Acrobat Reader, then you can download and print the forms right from the site. You want form SR for "sound recordings." Then you can do one of two things: you can submit each song individually for either 20 or 30 bucks a pop, on a cassette or some other sound recording medium, or you can submit a string of tunes as "The Collected Works of (your name here)."

At least one other thing you'll want to do is to make sure you get a UPC symbol on your CD before you have everything finalized. This will enable you to market your CD in stores.

As for ASCAP, BMI, etc., I think you have to be in a position to be getting some airplay before you need to worry about this.

—tedsblues

I'm pretty clear on the copyright forms, but what is a UPC symbol? Also, I'd like to be able to get airplay because that's the best way to get the public to hear your stuff. Since I do a lot of solo guitar arrangements on other people's songs, do I go through the same copyright procedure on those tunes as I would on the ones I write myself? If I use someone else's music that I arranged for guitar, how do I go about getting permission to use their music and pay the royalties, etc.?

—Finger Picker

To use music composed by someone else, you need to find out who the publisher is and get permission to record the material. You do not have to copyright your arrangement of that material. To find these publishers, go to www.ascap.com and/or www.bmi.com, depending on the songwriter's performing rights affiliation, and search the database for the song you want to record. It will list the writers and the publisher. Contact the publisher and find out what is required to use the song on your CD. You will have to pay mechanical royalties to the writer/publisher for every song on the CD that you didn't write yourself. If there will be several on the CD, it will require accurate

AL KOOPER
GUIDE TO THE MUSIC BIZ

ED CHERNEY
BEHIND THE BIG BOARDS

ROGER NICHOLS
DIGITAL RECORDING FORUM

GEORGE MASSENBURG
ASK GEORGE MASSENBURG

DAVID FRANGIONI
STUDIO TECH

bookkeeping on your part to ensure everybody gets paid.

If you manage to get airplay, AS-CAP/BMI handle that end of the royalties (performance royalties) for your songs as well as any of the covered tunes you release. Check out the ASCAP and BMI Web sites; they have pretty extensive info regarding royalties. You can only be affiliated with one of them — either AS-CAP or BMI. There is another one called SESAC you can check as well (www.sesac.com).

—Songbuilder

Not to mention CAPAC, if you live in Canada. By the way, a UPC symbol is just the "bar code" that you see on all retail products. As there are a few retailers that will sell local product, you want to make sure that they will be able to scan the CDs. They probably won't consider selling them at all without a bar code (UPC symbol).

—Tedster

Here are a few books that may be of help to you:

• Start and Run Your Own Record Label by Daylle Deanna Schwartz
• Tim Sweeney's Guide to Releasing Independent Records by Tim Sweeney and Mark Geller
• This Business of Music by M. William Krasilovsky and Sidney Shemel
• Everything You'd Better Know About the Record Industry by Kashif
And for a great read on the real stuff, Backstage Passes & Backstabbing Bastards by none other than Al Kooper!
The list is certainly not complete, but any of the above will help you. All of these books are available at Amazon.com.

—skp

I thought you didn't need permission if it's a previously recorded work. I thought you just had to pay the statutory rate. Am I confused?

—StrangeGlow

Basically, you have to get a license for the song from the publisher, so it's like getting permission.

—Al Kooper

AUXES/BUSSES/
SENDS/RETURNS

In the analog world, I bring my FX back through a channel (FX out to channel in). In Pro Tools, how do I get this to happen? Using inserts on each channel would be very CPU-intensive. Can you explain the proper use of busses, sends, and returns?

— themillenniumartist

Pro Tools uses auxes, sends, and busses just like an analog console does. In the analog world, to add a general reverb across several tracks, you'd first patch a send from your desk to your favorite reverb, then you'd patch the output of that reverb to either a channel input or an FX return. Then you'd turn up the corresponding send on each of the channels that you want to put the reverb onto.

You don't say whether you're using a plug-in or a hardware piece, but the process is similar. First, create an aux, then assign its input to any available bus. Next, instantiate your favorite reverb plug-in as an insert on the aux, or you can choose an I/O as an insert if you want to use a hardware piece (you'll need to patch the hardware into the corresponding I/O on your interface). Create a send (it can be either pre/post fader, mono/stereo depending on your needs) on all of the tracks that you want to send to the reverb. Assign the sends to the same bus that you've assigned to the aux input. Adjust levels to taste.

—Jan Folkson

Great explanation, Jan. Here's some additional information if you want to emulate a traditional recording console layout:

• Create an aux return (either mono or stereo)
• Insert your effect on that aux return
• Set the input of that aux return to an open bus
• Set the output to your interface main outputs
• Then, whatever track that you want to send to that particular effect, instantiate an aux send (there are five per track) using that particular bus as the send (that routes to that aux return).

—David Frangioni

THE PRODUCT: Tannoy Universal SuperTweeter
THE BASICS: Delivers those ultra-high highs
THE DETAILS: With DVD-Audio and SACD just around the corner, it was just a matter of time. Operating between the roll-off point of existing monitor loudspeakers and 54 kHz, the Tannoy Universal SuperTweeter is designed to accurately reproduce the frequencies that you thought only your dog could hear. Connected in parallel with existing monitor speaker input terminals, the SuperTweeter's high frequency diaphragm is a gold vapor-deposited titanium design, using a neodymium motor system. A special mounting system is included to ensure isolation from loudspeaker cabinet or mounting surface vibrations. **CONTACT:** Tannoy at 519-745-1158 or visit **www.tannoy.com**. Circle *EQ* free lit. #105.

THE PRODUCT: AppianX graphics cards
THE BASICS: Drive up to four digital or analog monitors from one CPU
THE DETAILS: The AppianX is a single slot card (available in two-port or four-port configurations) that can power multiple analog or digital displays from any Windows 2000 computer. Both models incorporate Appian's workstation-class graphics accelerator (AGX) and HydraVision desktop management software. The four-port version comes with 64 MB SDRAM, video-in support for baseband NTSC and PAL signals, and quad DVI-I support for digital displays. **CONTACT:** Appian at 425-882-2020 or visit **www.appian.com**. Circle *EQ* free lit. #106.

THE PRODUCT: HHB CDR830 BurnIT
THE BASICS: CD recorder
THE DETAILS: The latest in HHB's line of CD burners, the CDR830 BurnIT features 24-bit A-D and D-A converters and both coaxial and optical S/PDIF I/O, with an onboard sample-rate converter accepting frequencies from 32–48 kHz. CD Text support allows you to name discs, as well as add artist and track names. Also included are digital record gain and balance controls, input monitoring with track increment rehearsal, adjustable fade in/out times, multiple CDRW erase modes, track skip ID recording, track index search, and random and repeat playback modes. **CONTACT:** HHB at 310-319-1111 or visit **www.hh-busa.com**. Circle *EQ* free lit. #107.

THE PRODUCT: Aardvark LX6
THE BASICS: Computer hard-disk recording interface
THE DETAILS: The latest addition to Aardvark's Direct Pro Series of professional hard-disk recording interfaces is the LX6, a four-in/six-out, 24-bit/96 kHz audio interface. Onboard DSP allows for monitoring with effects (compression, three-band EQ, and reverb) without latency. Software features include a drag-and-drop digital patchbay, digital peak metering, VU meters, and snapshot recall of all settings.
CONTACT: Aardvark at 734-665-8899 or visit **www.aardvark-pro.com**. Circle *EQ* free lit. #108.

THE PRODUCT: Hardware option and software update for the SSL Axiom-MT
THE BASICS: New goodies for a world-class mixing console
THE DETAILS: The new SuperAnalogue mic amp option boasts improved digital linear motor faders that are individually hot-swappable and includes a four-character LED and virtual detents for easier control of null points and level matching. Version 2.2 software enhancements include new EQ processing setup options and signal latency management tools to time-compensate when using external inserts. There are also new automation features and metering options for stereo channels as well as both main and banked signal paths.
CONTACT: SSL at 44/1865-84-2300 or visit **www.solid-state-logic.com**. Circle *EQ* free lit. #109.

THE PRODUCT: StageSmart.com
THE BASICS: Online database of audio gear
THE DETAILS: Looking to buy a new or used piece of audio gear, but don't know where to start looking? You might want to begin by pointing your Web browser to StageSmart.com. Said to incorporate a database of "all available pro audio gear, from every manufacturer on the planet," this new online service is designed to connect buyers of audio gear with sellers, and even allows you to send an RFQ (Request For Quote) to retailers nationwide, allowing you to instantly receive the best prices, as well as detailed availability and shipping info.
CONTACT: StageSmart.com at 415-643-3000 or visit **www.stagesmart.com**. Circle *EQ* free lit. #110.

THE PRODUCT: Korg Triton-Rack
THE BASICS: Rack version of the Triton synth
THE DETAILS: If you love the sound of the Korg Triton but simply don't have room for another keyboard, you'll want to check out the new Triton-Rack. All the great sounds are here, but there's an S/PDIF input and output and even more memory (room for up to a whopping 2,057 programs) and expansion capability (eight PCM expansion slots, plus three SIMMs slots for up to 96 MB of sample memory).
CONTACT: Korg USA at 631-333-9100 or visit **www.korg.com**. Circle *EQ* free lit. #111.

THE PRODUCT: CLM DB500s Expounder
THE BASICS: Dynamic equalizer
THE DETAILS: The secret of the Expounder is its "adaptive circuitry," which responds to incoming program material, automatically expanding the dynamic range of selected frequencies by up to five decibels — a unique way to add extra low-end punch and lift high-frequency detail out of the mix. Each of the two channels offers four overlapping frequency bands and both Hi-Cut and Lo-Cut filters, with up to 24 dB per octave slope.
CONTACT: Wave Distribution at 973-728-2425 or visit **www.wavedistribution.com**. Circle *EQ* free lit. #112.
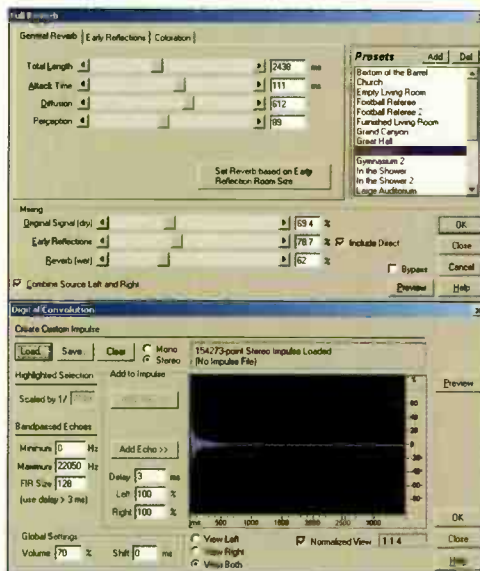
THE PRODUCT: Syntrillium Phat Pack and Tweakin' Toys
THE BASICS: Plug-ins for Cool Edit 2000
THE DETAILS: Phat Pack adds four delay-based effects: Full Reverb, Multitap Delay, Chorus, and Sweeping Phaser. Tweakin' Toys offers four special-effect and mastering tools: Hard Limiter, Pan/Expander, Pitch Bender, and Convolution.
CONTACT: Syntrillium Software at 480-941-4327 or visit **www.syntrillium.com**. Circle *EQ* free lit. #113.

THE PRODUCT: 2000 Series Speaker System
THE BASICS: Lightweight PA system
THE DETAILS: The Wright Bros. 2000 Series combines a two-way speaker system with a subwoofer. The EQX 2015 is equipped with 25 mm compression drivers and 15-inch LF drivers. The EQX 2018 subwoofer features an 18-inch cast frame driver, 3rd-order passive crossover, top pole mount, and multi-position design (front and downloaded).
CONTACT: Wright Bros. at 802-362-5808 or visit **www.wbsound.com**. Circle *EQ* free lit. #114.

By Steve La Cerra

JBL EVO Sound Reinforcement System

Anyone who has endured the challenging task of setting up a sound reinforcement system knows that it can be a difficult task. JBL's EVO Intelligent Sound Reinforcement System was designed not only for painless installation, but also to monitor itself and make changes according to what's happening in its environment. Using JBL's Distributed Intelligence, an EVO System can self-adjust for potential feedback, transducer stress, and changes in room response.

Distributed Intelligence allows system components to communicate over a network. By locating the "brains" (transducer control and DSP functions) inside the EVOi.324 Intelligent Loudspeaker cabinet, data transmission is reduced to a minimum. The system controller is the EVOi.net, a single-space rackmount unit that comes with a measurement microphone. Through the use of BiDAT (Bi-directional Data over Audio Transceiver) technology, the controller communicates with the loudspeaker via (inaudible) Frequency Shift Keying over the same cable that carries the audio signal to the loudspeaker cabinet.

Components

JBL's EVOi.324 loudspeaker is intended for use in fixed-installation sound systems. Each EVOi.324 contains two 14-inch, carbon fiber cone transducers and one neodymium compression driver. Three amplifiers provide 600 watts in Class-D operation for each of the 14-inch drivers, and 100 watts for the compression driver. JBL's Filtered Array Technology (FAT) allows independent control over each low-frequency transducer for amplitude and phase response, resulting in smooth power response and constant coverage. The compression driver uses a three-inch voice coil with a 1.5-inch exit; it's acoustically coupled to an aluminum waveguide horn with an 80x80 coverage pattern.

Output from any line-level source can be connected to the EVOi.324's XLR input. To take full advantage of the speaker's "intelligent" networking abilities, the EVOi.net controller can be added as an interface between the source and (up to) four EVOi.324 speakers. Using the EVOi.net controller, the person operating the system can choose whether EVO optimizes parameters for speech, live music, or recorded music, in full or empty rooms. Each EVOi.324 then generates a signal called a Maximum Length Sequence (MLS), which is captured by the mic. This signal is analyzed and compared to a factory response curve; by comparing these two curves, EVO generates a response curve for the room and applies complementary equalization.

In larger venues where delay stacks may be required, the EVO System has an automatic Delay Set mode to calculate delay times for each cabinet. EVO is capable of running two Anti-Feedback engines: one that "rings out" the room ahead of time (for fixed microphone positions) and a second that operates in real time, constantly monitoring the system and making adjustments when a microphone is moved and feedback is initiated. EVO's Enhanced Total Thermal Management System (ETTMS) combines Thermal Modeling with Thermal Driver Protection to compensate for changes in transducer performance due to driver temperature fluctuations. ∎

WHAT IS IT? Active loudspeaker system with onboard DSP and networking capabilities.

WHO NEEDS IT? Anyone who wants aggravation-free PA system setup and adjustment.

WHY IS IT A BIG DEAL? EVO "piggybacks" the computer network data on the same cables that carry the audio signal.

SPECIAL NOTES: In addition to self-adjusting for room response and potential feedback problems, the EVO system can compensate for transducer stress.

SHIPPING: Now

SUGGESTED RETAIL PRICE: EVOi.324 Intelligent Loudspeaker: $3,499; EVOi.net system hub and controller: $999. Complete EVOi.sys System: $11,499.

CONTACT: For more information, contact JBL Professional at 818-894-8850 or visit **www.jblpro.com**. Circle EQ free lit. #101.

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By Steve La Cerra

Yamaha SREV1 Sampling Reverberator

With the introduction of the SREV1 Sampling Reverberator, Yamaha has added a high-end reverb processor to their well-established line of processors. Unlike "conventional" digital reverbs that use algorithms to generate their effects, the SREV system uses what Yamaha calls "digital convolution" processing. Convolution sampling technology takes impulse response samples of real acoustic environments and imposes an acoustic "fingerprint" from the measured environment upon the audio signal to which the reverb is being added. The result is claimed to be a faithful recreation of the original room's reverberant characteristics. In addition to providing a CD-ROM library with presets simulating well-known venues around the world, Yamaha has developed PC-based software that may be used by engineers in conjunction with a Time Stretch Pulse (TSP) signal to capture any room and create custom reverb programs emulating that room.

The SREV1 system is based around the SREV mainframe, a three-space rack chassis with 24-bit, 48 kHz audio I/O. Inside the SREV mainframe are 32 CNV3 Convolution Chips — custom ICs developed by Yamaha to perform the calculations for generating reverb. Rear-panel XLR jacks are provided for two AES/EBU digital I/Os (four channels in, four channels out). These I/Os allow the SREV1 to be used in several different configurations such as a two-channel mode with 5.46 seconds of reverb per channel, a four-channel surround mode with 2.73 seconds per channel, or dual-stereo processors with 2.73 seconds per channel. In dual-stereo mode, each stereo engine has its own inputs, outputs, and program settings. An optional DB-SREV1 DSP expansion board (which contains another 32 CNV3 IC's) may be added to double the reverb time to 10.92 seconds per channel in two-channel mode, or 5.46 seconds per channel in four-channel or dual-stereo modes. In addition to the AES/EBU digital I/Os, two rear-panel slots on the mainframe accept Yamaha mini-YGDAI expansion cards for installation of analog, or ADAT lightpipe or TDIF digital I/O.

The SREV comes with an assortment of simulated environments, each allowing control over reverb time, initial delay, reverb balance, and four-band parametric EQ (both pre- and post-processing). PC-based software included with the SREV system makes it possible to measure the impulse response or "sample" an existing room, and then create an SREV program based upon that space. Since four-channel sampling is possible, surround programs as well as stereo programs may be created. The software also provides graphic program editing. Program data may be stored internally, on standard PC memory cards, or on CD-ROM, and may be recalled via MIDI; the SREV mainframe includes a memory card slot as well as a CD-ROM drive.

An optional remote control — the RC-SREV1 remote controller — may be used for operation of up to four SREV mainframes. A 20-meter cable is provided with the RC-SREV1; power for the remote is transmitted from the mainframe through the cable. For longer cable runs (up to 100 meters), the RC-SREV1 may be powered with an external supply. ∎

WHAT IS IT? The new "flagship" high-end digital reverb from Yamaha.

WHO NEEDS IT? Anyone requiring the highest quality in reverb effects.

WHY IS IT A BIG DEAL? Unlike conventional digital reverbs, the SREV1 employs digital convolution processing to impose the acoustic fingerprint of a room onto any audio signal.

SPECIAL NOTES: In addition to being able to produce multi-channel reverb for surround applications, the SREV1 also has the ability to "sample" the reverb signature of a room and create a reverb program of that room.

SHIPPING: First quarter of 2001.

SUGGESTED RETAIL PRICE: Mainframe: $5,499; DB-SREV1 DSP Expansion Board: $1,999; RC-SREV1 Remote Controller: $1,499.

CONTACT For more information contact Yamaha Corporation of America at 714-522-9011 or visit **www.yamaha.com/proaudio**. Circle *EQ* free lit. #102.

BILLY BUSH

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

Making the transition from onstage guitar tech to recording engineer is difficult enough, but imagine doing that and having your first recording project go platinum! That's just what happened to Billy Bush, who found himself debuting behind the board for Garbage's acclaimed sophomore album, *Version 2.0*.

Bush — the band's self-described "resident engineer and gearhead" — is currently holed up at Smart Studios in Madison, Wisconsin, recording the third Garbage record. In addition to the engineering and editing involved, he's in charge of maintaining sample libraries, acquiring gear, archiving, and keeping everything in the studio running smoothly. "We usually work ten to twelve hour days, six days a week," Bush explains, "although. as we get deeper into mix mode, that will stretch to 16 hour days, seven days a week."

EQ: How did you land your current gig?

Billy Bush: I was contacted by Garbage to help them out on their first tour. Toward the end, they started discussing how they wanted to approach making their second record and decided to explore using a digital audio workstation. Since I had been keeping all their sequencers, samplers, and laptops updated and running, the band left the design and purchase of the system in my hands. I was supposed to find the best system, buy it, learn how to use it, and then show them how to work it in a matter of six weeks. After a month of recording, the band turned to me and told me that I had to stay on and engineer the record — and I've been here ever since!

How did you get started in engineering?

Originally I started recording demos of my own band on a Vestafire four-track, and when I started touring with bands as a guitar/keyboard tech I was often required to come into the studio to help get the guitar and keyboard sounds together. That was when I caught the bug seriously. Having the opportunity to watch and learn in that environment was priceless.

Where do you see yourself in five years?

Hopefully continuing to make records that engage and interest people.

Who are your heroes in engineering and record production?

George Martin, Flood, Timbaland, and Butch Vig among many others.

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

Digidesign Pro Tools Mix Plus, Focusrite Producer Pack. Eventide Orville, Brauner VM-1, and a '62 brown-face Fender Vibroverb with one 15-inch speaker.

What's the coolest recording technique you've discovered?

If it's broke, it sounds better.

What's the best piece of advice anyone ever gave you?

Listen.

What's the best piece of advice you can give our readers?

Remember what Hunter S. Thompson once said: "The music business is a cruel and shallow money trench, a long plastic hallway where thieves and pimps run free, and good men die like dogs. There's also a negative side." ∎

WEBLINK

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World Radio History

Red Rock Studios

By Steve La Cerra

Red Rock comes to Miami

STUDIO NAME: Red Rock Studios

LOCATION: Miami. FL

KEY CREW: Clay Ostwald

CREDITS: Clay Ostwald has been collaborating with Gloria and Emilio Estefan for almost 14 years, both in the studio and on stage as a member of Miami Sound Machine. Ostwald also produced Jon Secada's hugely successful debut CD. His songwriter credits include songs for Gloria Estefan/Miami Sound Machine, Jon Secada, Raul DiBlasio, *The Specialist* soundtrack, Laura Brannigan, and Masayoshi Takanaka.

MIXING CONSOLE: Yamaha 02R mounted in an Argosy desk, 01V (used for live concerts and submixing keyboards to the 02R)

MONITORS: Westlake BBSM-8. Genelec 1029A, 1091A subwoofer: Fostex T20 headphones

AMPLIFIERS: Hafler P400, P120: Carver M500

RECORDERS: TASCAM DA-88. DA-38. DA-30: JVC TD-W718 double cassette deck

OUTBOARD: UREI 1176. DigiTech RPM-1 Leslie simulator. Yamaha GC2020 compressor/gate, dbx 166 compressor/gate

EFFECTS: DigiTech TSR24, Yamaha REV5. SPX90II

MICS: Audio-Technica AT4060, AT4051 [2]. AT4041 [2], AT3525 [2]: CAD Equitek E-200. AKG D112

MIC PREAMPS: True Precision 8

SAMPLERS/KEYBOARDS/MIDI: Roland A90EX Keyboard Controller. JV1080. D550: Yamaha C7 Grand Piano, TG77, TX816 (with 6 modules): Akai S1000 [2], Korg X5D, Emu Orbit

COMPUTERS: Power Computing PowerTower Pro with Sonnet Technologies G3/400 upgrade, Atlas II drive for audio, Mitsubishi CD-R drive; Apple PowerBook G3

WEBLINK

Check out Clay Ostwald's Web site at www.clayostwald.com

DAW EQUIPMENT: Lexicon Studio Core32. PC-90

SOFTWARE: Steinberg Cubase VST v4.1; Steinberg Free Filter. Model E, LM4, Pro-Five, and Loudness Maximizer: SPL De-Esser. Antares Auto-Tune

STUDIO NOTES: Ostwald states: "Built in 1998. Red Rock Studios is my home base for writing. production, and recording. First and foremost, Il set out to create an environment with music being the priority — a comfortable place where writing and recording can be relaxed, spontaneous, and productive. This meant that the design and technology had to be first-rate, but must never intrude on the creative process. I wanted a design that was flexible enough to accommodate everything from writing a song with a guitar and piano to a full rhythm section playing live together."

EQUIPMENT NOTES: Ostwald continues: "Jan Tholenaar of Wizard Wire Works (**www. wizardwireworks.com**) and I developed a wiring scheme for my keyboard racks that has worked out very well. We set up a system of ELCO connectors that accommodates multiple situations (live. studio, sub-mixing, etc.). I can patch my keyboard racks in and out of the studio very easily. Since I use the same keyboard racks for live concerts, Jan worked it out so that the keyboards are normalled to the 02R, but the connections to the 01V remain intact — leaving me a choice of routing them through either mixer at any time without repatching.

"I just got a B3 organ simulator called 'B4' from Native Instruments, and it's *excellent*. I'm using more and more software synthesizer plug-ins.

"My favorite piece of equipment, sonically-speaking. is always the acoustic piano. Technically, I spend a lot of time with my Macintosh computers; I wouldn't have it any other way! My Cubase setup works like a Synclavier-type system that integrates audio and synthesizer tools

Porcelain God Studios

By Steve La Cerra

Recording tribute CDs and more

PHOTO BY MARIE PENELLO

STUDIO NAME: Porcelain God Studios

LOCATION: Lynbrook, NY

KEY CREW: Steve Booke

CREDITS: Among Steve Booke's releases are a Christmas CD, an acoustic instrumental CD of Carpenters songs, and a solo CD entitled *Rare Earth.* He has produced, arranged, and/or played guitar on tracks for a number of tribute CDs released on the Progressive Arts label. Steve scored and recorded music for the independent movies *Open Mic* and *Evil Streets,* and engineered music for the television series *Golfin' Buddies.* He also worked on projects from Mike Chlasciak (from Rob Halford's band), Mike Knight, Carl Roa, and WRCN radio.

MONITORS: Yamaha NS-10M, JBL Control 1; Headphones: Sony 7506, AKG 141, Technics HT86

AMPS: Soundtech PL150

RECORDERS: Digidesign Pro Tools Project, Sony A7 DAT, TASCAM 112 cassette deck, 202 mkII dubbing deck

OUTBOARD: DBX DDP, Behringer Composer, Rane PE15 parametric EQ

EFFECTS: Lexicon LXP-15, Alesis Wedge, Quadraverb; DigiTech Studio Quad

MICS: AKG C414 B/ULS, D3800; Shure SM57, SM58

MIC PREAMPS: dbx 586 Dual Tube Mic Pre

SAMPLERS/KEYBOARDS/MIDI MODULES: Korg X5DR, Roland Sound Canvas, MOTU MIDI Timepiece II

COMPUTERS: Apple Quadra 650 ("at a blazing 25 MHz"); Seagate 2.1 GB, 9.1 GB external drives; Yamaha 4260 CD-RW drive

SOFTWARE: Pro Tools Project v3.2, MOTU Performer v5.02, Adaptec Jam v2.5, Toast v3.5.7

DAW EQUIPMENT: Digidesign Project II Card, 882 audio I/O

INSTRUMENTS: Acoustic, electric, and bass guitars from Fender, Ernie Ball/Music Man, Ibanez, Valley Arts, Danelectro, Larrivee, Yamaha, Guild, Takamine, Samick; various percussion instruments

INSTRUMENT AMPLIFIERS: Fender, Marshall, Peavey, Mesa/Boogie, Randall, Tech 21

STUDIO NOTES: Booke states: "My studio is mainly a private studio. I'm always recording for a project I'm involved in — whether it be my own music or something I was commissioned to write and/or record. Occasionally I record friends here. When I first got started seriously recording about six years ago, my studio was better than I was at being an engineer/producer. Eventually I outgrew my studio, but didn't have the money to upgrade my computer and software. This turned out for the better. If I had upgraded earlier, I might not have developed the production and engineering skills I gained from recording complex songs on only eight tracks."

PRODUCTION NOTES: Booke continues: "Anything I record direct almost always goes through the SanAmp Acoustic DI box, then to the tube mic pre and the dbx digital compressor processor. This chain fattens and warms up the sound of anything I put through it. I mic everything with the AKG C414, which also goes through the tube mic pre and the digital compressor (I usually put minimal compression on the signal as it's being recorded). When I record my acoustic guitars I generally mic them around the twelfth fret, and record the direct signal on another track. Typically the mic captures the lower frequencies and gets more of a bassy sound. The direct signal captures more high end, giving me a natural EQ between the two tracks. I then bounce the two tracks together digitally. The system I currently own only allows four EQs to be used — not per track, but for the entire session. After I do my bouncing and/or straight recording, I route the signal out of the computer where I EQ and add heavier compression if needed. Once I get the desired sound, I then re-route it onto a new track."

PHOTO BY EDWARD COLVER

MBHO MBNM648 Microphones

MICROPHONE NAME: MBHO MBNM648 Microphones with Schneider Disk

PRICE: MBNM648 including KA-100 capsule: $501.00 each; Schneider Disk: $292.00

TYPE OF MIC: Condenser

POLAR PATTERN: Omnidirectional (see notes)

FREQUENCY RESPONSE: 10 Hz to 20,000 Hz

SENSITIVITY: 6 millivolts/Pa

SIGNAL-TO-NOISE RATIO: 67 dB CCIR; 80 dB DIN/IEC

MAXIMUM SPL: 132 dB

POWER REQUIREMENTS: 48-volt phantom, 1.7 milliamps (per microphone)

ELECTRICAL IMPEDANCE: 200 ohms

DIMENSIONS: MBNM648 with KA-100 capsule: 0.84 diameter x 5.68 length (inches); Schneider Disk: 12 inches (diameter)

MIC NOTES: What you're looking at isn't an extraterrestrial spacecraft but a pair of MBHO's MBNM648 microphones mounted on a Schneider Disk. The Schneider Disk is a variation of the Jecklin Disk developed by Jürgen Jecklin for his stereo mic technique known as Optimal Stereo Signal (OSS). The idea behind OSS is to use a pair of spaced omnidirectional microphones approximately six inches apart (roughly the distance between the human ears) with a flat disk placed in between the spaced pair. The disc is covered with a thin layer of foam intended to absorb high frequencies, thus enhancing stereo separation between the mics. The difference between the two types of disks is the foam sphere at the center of the Schneider Disk, which reduces the amount of high-frequency energy reflected from the disk, resulting in increased stereo separation. Below approximately 200 Hz, the Schneider Disk has little (if any) effect on the stereo signal because audio wavelengths are large enough to bend around the disk and reach both microphones. However, as frequency increases above 200 Hz, the Schneider Disk's foam sphere helps increase stereo separation between the spaced pair.

USER TIPS: The Schneider Disk may be used in OSS technique for stereo recordings of orchestral or chamber music, as well as for stereo miking of solo instruments. To properly implement OSS, the microphones should be identical, omnidirectional, and should be set for equal output to avoid unbalancing the stereo image (this can be checked using headphones).

Technical information furnished through the courtesy of Arthur Garcia and Bob Paquette.

World Radio History

Malcolm Luker shares his techniques
for recording orchestras in surround

In Love with
the Thunder

by Bobby Owsinski

Scoring mixer and recordist Malcolm Luker has seen his share of success. With Emmys for television movies such as *Peter the Great, Anastasia, Bourne Identity,* and *Young Indiana Jones,* 14 gold and platinum records, and feature films such as *The Piano, The Man In The Iron Mask, Chill Factor,* and the upcoming *Dungeons and Dragons* and *Attila The Hun* under his belt, his credits speak for themselves. Starting his recording career in the maintenance department of London's Morgan Studios, Luker soon migrated to Munich, where he graduated to chief engineer at Munich's Union and Arco Studios. There he racked up an impressive 18 number one hits with the likes of Falco, Placido Domingo, and even the Count Basie Orchestra. But the hand of fate eventually led Luker from the pop world into the world of orchestral recording and mixing.

Recently Malcolm has been witness to orchestral recordings passing from a purely stereo format to the current realm of 5.1 surround.

EQ: How did you get into doing film music?

Malcom Luker: I First got into film in 1983 via a Georgio Moroder TV project called *Mussolini,* but the thing that sort of switched the light on for me about film music was when I did a movie with a guy named Dan Wallen (who's a bit of a mentor to me) called *Body of Evidence.* He asked me to help him record the orchestra, and, in doing so, made me question everything I'd ever thought about recording and about how microphones really work. So, I took his philosophy, built on that, and then went my own direction afterwards. I was very fortunate to have someone like Dan to help me, because most people don't sit down and tell you why things happen.

Do you have a specific approach for recording an orchestra in surround?

In the early days I used the Deutsche Grammophon approach of X/Y, M/S, and pointer microphones. Then I discovered the Decca Tree (see sidebar) and realized just what can be done with it. It isn't as easy as just hanging it though, since the distance of the microphones from each other, as well as their height, is the key depending upon the room that you're in. Setting up your microphones is really important — rather like being a mathematician and an architect at the same time.

I now use the Decca Tree all the time, and then I stick a couple of extra mics up in the room for my surrounds. I also record a separate dedicated subwoofer track, and that's the key to the whole thing.

How many mics do you use on a scoring date?

I use five Brauner VM-1's (a couple that are hung up for surrounds) and a [Neumann] TLM 170 for the sub (it's where you put it that makes it work). Then I use a mic on the front and another halfway down on the violin section (you have to phase correlate those for height and distance or else you get into mud), and a stereo mic on the violas so I don't get into any phase problems with the tree. For the woodwinds, I'll use four Sennheiser MKH-80's. Then I use a Neumann SM 69 stereo mic on the horns.

How do you deal with samples?

Nowadays a lot of guys are using "19-inch Percussion" [samplers] and, unfortunately, it really sounds like a sample. The samples actually take away the depth of your score, and then you begin

World Radio History

CLIMBING THE DECCA TREE

Developed by Decca for recording in stereo, the "Decca Tree" is a three-microphone technique designed to capture orchestral and operatic performances. Because of its natural sound and the fact that it doesn't normally cause problems with Dolby and other surround processes, the Decca Tree mic configuration is useful for recording orchestral film scores.

Three large-diaphragm omnidirectional microphones are placed above and behind the conductor in a triangular arrangement — cardioid-pattern mics are sometimes used, but most engineers favor omnis. The size of the triangle varies with the stereo width desired, but generally one side's mic is placed in front of the cellos, while the other is in front of the first violins; often the two mics end up about two meters apart. The third mic is centered between the outside pair, and usually positioned about 1.5 meters in front of them.

The two side mics are panned hard left and right in stereo, while the middle microphone is centered across the two channels. "Spot" mics dedicated to picking up certain instruments or sections are commonly used in conjunction with the Tree mics.

—*Mitch Gallagher*

to fight against the sound effects. In order to make a sample sound bigger, I'll send it out to a speaker in the studio and set up left, center, and right room mics and in some cases even a pair of surround mics to give it some size. This way you get more air and the two [sample and sound effects] can then live together rather well. The fewer samples you actually use the more air you get, and the result is you can actually still hear the music against the dialog and effects.

Since you've done a lot of both, what's the difference between doing records and film dates?

It's a whole different approach. I think you even go to work dressed differently. I started in the business as a guitarist in a rock 'n' roll band, and I still love turning an amp to 11. I've done all sorts of rock and jazz records over the years, and I loved it all, especially when it's done well. But I just fell in love with orchestral recording because when you get 80 guys out there and they're all thundering away, it's just like standing in front of two Marshall stacks turned to 11. I love doing what I do. It's a passion, not a job. ∎

Managing and controlling bit
rates for surround sound delivery

Adventures in
Bit Budgeting

By Rich Tozzoli

As surround sound becomes an increasingly common delivery requirement for many projects, more and more audio professionals will soon be traveling into previously uncharted waters. As if the complexity of providing a six (or more) channel master weren't enough, the "back end" issues of surround are often even less understood.

By "back end," we're talking about the delivery of encoded tracks at different bitrates to create the multichannel master. The data "pipe" for delivering the content on the DVD is only so large, and all the various video and audio tracks must accordingly have their bitrates budgeted and controlled or there will be problems. This article will explore a real-world surround project,

and describe what happens to the audio once the mix is done.

StarGaze — Hubble's View of the Universe is the creation of Ralph LaBarge of Alpha DVD, based in Gambrills, MD. Having authored literally hundreds of DVD projects, including the award winning *MARS — The Red Planet*, LaBarge was responsible for deciding how the surround mix for this DVD-video should be encoded. In making that decision, he had to first look at the project's overall bit-budget. "The first step of the process is determining how many audio tracks you're going to have and what their bitrates are," he explains, "because they're fixed; you can't play around with them. If you have an AC-3 5.1, it's normally 448 kbps, and DTS is normally at 1,509 kbps, although there is a half-bandwidth DTS rate (754.5 kbps) that I have never used. Dolby AC-3 stereo tracks are generally 192 kbps." [Note that Dolby offers other bitrate options; see the chart.]

"On *StarGaze*," he continues, "I planned on having seven audio tracks: a full 5.1 AC-3 surround, a 5.1 DTS surround, an AC-3 stereo music track, and four narration tracks in AC-3 stereo. What I would do is simply take all of the bitrates and add them together, giving me the total of what I'm going to use for audio." The rest of the available bit-budget would be used for the high-resolution video images.

A DVD's bit-budget is normally, though not always, decided by the producer. "It directly falls out of how many audio tracks, what type of video, how many angles and subtitles, what level of quality you want, and whether it's a dual- or single-layer disc," LaBarge says. "Those are usually client/producer questions, and once you've made those decisions, everything else just falls into place." Actually, two bit-budget calculations have to be made: one for the bitrate, and one for the total disc storage. "You then run the numbers both ways, make sure everything
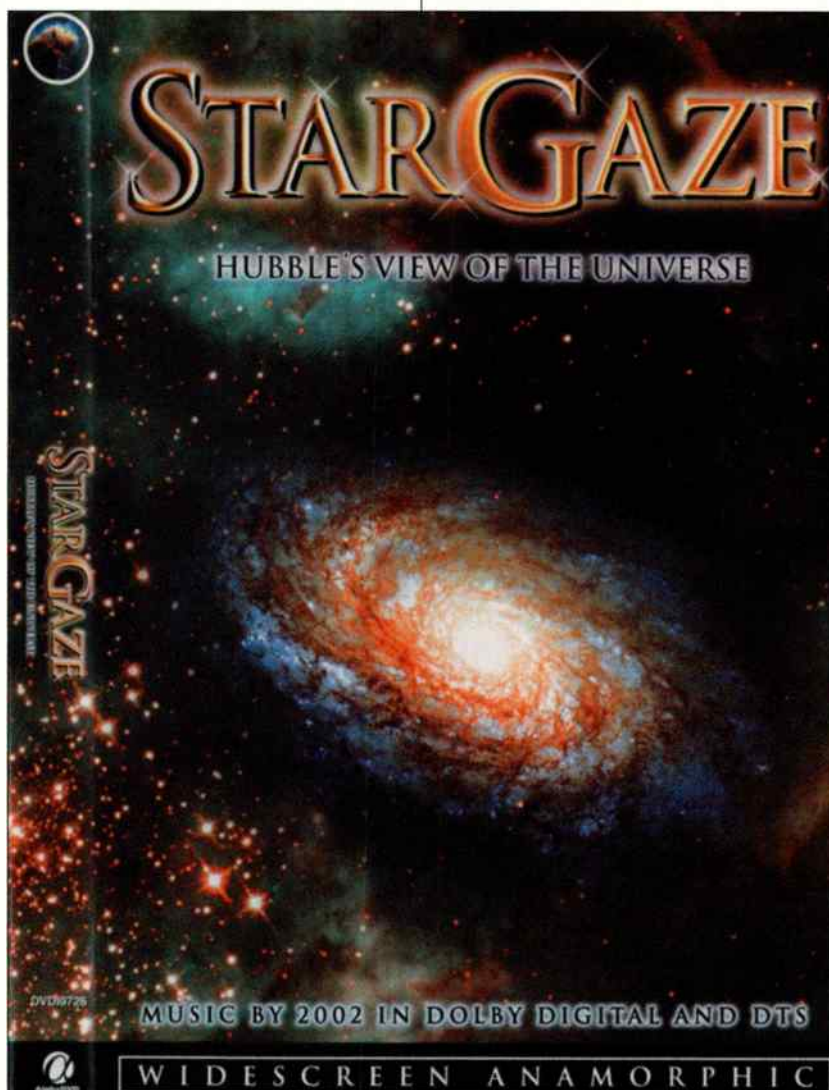
The *StarGaze* DVD includes seven audio tracks: a 5.1 AC-3 surround, a 5.1 DTS surround, an AC-3 stereo music track, and four narration tracks in AC-3 stereo.

DOLBY DIGITAL (AC-3) VS. DTS: THE SPECS AT A GLANCE

	DOLBY DIGITAL	DTS
Sampling Rate	32, 44.1, 48 kHz	32, 44.1, 48, 88.2, 96, 192 kHz
Bit Resolution	16, 20, 24*	16, 20, 24*
Bitrates	224-640 kbps	1.509 kbps** (1.509 mbps)
	384 kbps***	754.5 kbps ("half speed")
	448 kbps****	1,234 kbps (CD only)

NOTES:

* 24-bit is theoretical maximum; both Dolby Digital and DTS currently operate at 20-bit resolution

** Standard for DVD encoding @ 48 kHz (legacy 44.1 kHz DTS files use 1,411 kbps bitrate)

*** Standard for DVD encoding

**** Recommended maximum bitrate for DVD encoding

fits, and stay under the magical 10.08 mbps maximum. The pipe for a single-sided DVD-video is 10.08 mbps maximum, and for video and for DVD-audio, the max is 9.8 mbps."

Armed with this information, LaBarge sent 24-bit stereo WAV files to us at Gizmo/333 Entertainment in New York for surround mixing. The music, from the group 2002, was to be re-purposed into surround and delivered in the form of AC-3 files at 448 kbps and DTS files at 1.509 mbps. LaBarge was going to handle the 192 kbps AC-3 stereo files himself back at Alpha DVD. I completed the mix in Pro Tools at 24-bit/48 kHz using Kind of Loud's Smart Pan Pro for surround panning and a Sony DRE-S777 reverb and Kind of Loud's RealVerb 5.1 plug-in for depth and ambience. With the mixes completed, it was time to take the six-channel master (left, right, center, sub, left surround, right surround) and create the encoded files required for delivery.

We started by assembling an encode/decode station, consisting of a PC with a SEK'D ProDif 96 sound card, a Dolby DP-569 encoder, a Dolby DP-562 decoder, a DTS CAE-4 encoder, and a DTS CAD-4 decoder (see fig. 1). Brian Mackewich and Matt O'Conner handled all the encoding tasks after the mix was completed. The six-channel final master was output to a TASCAM DA-78HR for backup at 24-bit/48 kHz, although it could have been sent to the encoders directly from Pro Tools. The basic signal path for an encode/decode session in our studio is as follows: From the multitrack or workstation into the encoder via three pairs (stereo) of AES/EBU

connections at 24-bit/48 kHz, with separate timecode (Note: we did not use timecode on *StarGaze*). The six surround channels are routed differently for DTS (L, R, Ls, Rs, C, LFE) than they are for Dolby (L, R, C, LFE, Ls, Rs). Parameter choices are set using the soft-

STARGAZE'S AUDIO AND VIDEO WERE BOTH ENTIRELY AUTHORED AND MASTERED IN THE DIGITAL DOMAIN, AND NOTHING ON THIS DVD EVER WENT ANALOG. THE AUDIO FILES WERE 24-BIT/48 KHZ THROUGHOUT.

ware programs provided with each type of encoder (DTS and Dolby).

A single AES/EBU output runs from each encoder, complete with embedded metadata and control information, into the SEK'D card in the PC. The encoded

DTS E-175-01 TIMECODE CONTROLLER

DTS CAD-4 DECODER

AES/EBU

L, R
Ls, Rs
C, S
AES/EBU

DTS
BITSTREAM
AES/EBU

PRO TOOLS WORKSTATION

DTS CAE-4 ENCODER

SEK'D PRODIF 96 CARD

CD BURNER

FIGURE 1: The DTS encode/decode station.

signal from the PC is then sent back via AES/EBU to the decoders. The outputs of the decoders are hooked up to our Pro Control in the analog domain, allowing us to monitor the encoded mixes and check fold-down compatibility.

At last, the encoded *StarGaze* surround audio files captured into the PC's hard disk were ready for delivery. The Dolby files were each given an AC-3 suffix, and the DTS files a CPT suffix, which stands for compacted (these files have the embedded timecode stripped away for compatibility with certain DVD authoring stations; more on this in another article). Next, the encoded files were burned to disc and sent back to LaBarge. The files were then transferred into his Daikin Scenarist NT 2.0 DVD authoring system, and laid back manually against the images.

StarGaze's audio and video were both entirely authored and mastered in the digital domain, and nothing on this DVD ever went analog. The audio files were 24-bit/48 kHz throughout, and the Hubble Telescope video images were actually created on computers. There was an audio change made at one point, but we simply uploaded the new files to LaBarge's FTP site for quick turnaround. Kind of makes you long for the days of simple 16-bit CDs, doesn't it?

StarGaze used up much of its precious "data pipe" providing maximum audio quality and multiple surround mixes. LaBarge concludes, "If you look at the final disc, you'll see the average bitrate is around 9 mbps, so it's pretty close to capacity."

Rich Tozzoli is a contributing editor to both *EQ* and *Surround Professional* magazines. He has done surround mixes for Al DiMeola, Vernon Reid, Foghat, and Joni Mitchell, among others.

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From the
Trenches

by Dave Aron with Steve La Cerra

As a producer, engineer, and remixer, Dave Aron has worked on projects with 2 Pac (All Eyes On Me), Snoop Doggy Dogg (The Dogg Father), Eastsidaz, Sublime, Xzibit, Daz Dillinger, and Priest "Soopafly" Brooks. Definitely one who adheres to the "it's the man, not the machine" proverb, Aron has mixed great records on consoles varying from Mackies up to SSLs. Along the way he's come up with some cool tips and techniques that can make life easier during any production. In this article, Dave shares some of his ideas.

Home studios are the latest, greatest weapon a producer or songwriter can own on his quest for a great-sounding production. Today's technology enables us to do multitrack recordings good enough to mix and release. Even if your plan is to mix your song in a commercial studio, every songwriter wants a rough mix good enough to play for people without excuses. But getting a good mix can sometimes be a problem, especially without automation or a mix budget. Sure you can spend your own money on a mix room, but the dilemma persists: how do you get a good mix without breaking the bank?

After dealing with this problem for years, I've developed an extremely efficient and cost-effective method of pseudo-automated mixing that works in any multitrack studio. It's part of what I call "Ghetto Engineering" — working with what you have on hand to get the job done quickly. The best part about it is that you already have everything you need. The way to do this is to "save" two tracks of the multitrack, and record your mix to those two tracks. On a two-inch 24-track tape this could be tracks 23 and 24, or, if you're using DA-88's or ADATs, you can use the last two tracks on the last machine.

Get Out Yer Patch Cables

Patch the stereo mix output from the recording console straight to the two mix tracks of the multitrack. You have a couple of options to monitor your mix tracks. I like to patch the output of the two tracks directly into my DAT machine. Then I connect the headphone output of the DAT machine to the input of my monitor amp. This allows me to use the DAT machine's headphone volume control as a monitor level control (it also allows me to quickly drop the mix to DAT when I'm ready). Another possibility is to patch the mix tracks back to a stereo two-track return of your console. *Don't* patch the mix tracks back to a pair of channels on the console because there's

FIGURE 1

potential for a feedback loop. Here's why: At some point you'll need to put the two mix tracks into "input" to set a record level. If those tracks are coming back to the console on channels, and the channels are assigned to the L/R mix bus, you'll have a loop. Not fun.

Record your mix to the multitrack as you would if you were mixing to a DAT or two-track analog machine. At some point you'll want to make a change — such as adjusting the level of a bunch of different tracks when the song goes from the chorus to the bridge section. At this point, stop the machine(s) and rewind the tape (or disc). Make whatever changes to the mix you need to make. Rewind again and punch in on the two mix tracks. I can't speak for other machines, but on my DA-88's the punch is absolutely seamless and I defy anyone to hear the punches. As long as you don't change the overall level of the mix, and as long as the machines are synched up, you won't hear the punches. You can punch anywhere as many times as you like. The better your punching and mixing skills, the further you can take this concept.

Now if I'm mixing and I hear something like a popped "p," I go back and punch that spot with the low frequencies filtered out on the vocal. De-essing is possible just by rolling out some high end (between 2 kHz and 8 kHz, depending on the vocal). Or, if I miss a spot where a cool delay would work, I can go to the spot and punch the delay like a normal delay throw (FYI: the old-school way of doing this was to mixdown to two-track tape section by section, and then edit the sections together).

Catch The Bus And Double Your Channels

I own a 24x4x2 console, and having only four busses makes it a challenge to record to more than four tracks at a time. But with a little planning I can easily record a five-piece band. Of course, I can bus four instruments to four different tracks such as bass, keys, guitars, and vocals. But what about all of those drum mics or the extra guitar? It'd be great to use direct outputs from each channel to put the kick, snare, hat, etc. on separate tracks. Unfortunately, my console doesn't have direct outs. It does have inserts on each channel, and they can be used as direct outputs by inserting a 1/4-inch plug halfway into the insert jack (one "click"). This will tap the "send" of the insert and give you an output directly from the channel; you can patch this to the input of a tape track.

Another problem you might run into is not having enough channels to accommodate all the inputs and tape track returns needed for a big session. On an inline console this

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usually isn't a problem — just use the input section for your mics and DIs, and then use the tape returns to monitor output from the tape tracks. My 24x4x2 isn't an inline console, so it doesn't have a separate tape return path. One possibility is to split the console in half, using the first 12 channels for inputs (mics, DIs, etc.) and the last 12 inputs as tape returns. That's fine, but it won't allow me to record and monitor 24 tracks simultaneously. The solution: Use the insert return as your tape return. Patch an insert cable into the channel. Use the insert send to route signal to the tape track. The input trim pot will set level to tape. Patch the output of the tape track to the return of the insert cable. Since the return dumps the signal to the straight fader, you can do a monitor mix on the faders without changing the level going to tape. You probably will have to use the EQ for play-back-only because EQ usually follows the insert patch (see fig. 1). Along the way it's easy to route the send signal through an outboard EQ, compressor, delay, or gate. If the level at the tape machine is too hot — even with a mic pad switched in — you can use any outboard processor to pad the level down. I've even used digital delays to pad the level down: Set the delay time to 0, the mix control to dry, and lower the output level.

Add Life To Your Sequencer

While you might like the "feel" of a particular drum machine or sequencer, there's no sub-stitute for true spontaneity to liven up your tracks. Don't get stuck in a formula for writ-ing songs. Try building around the bass or guitar instead of the drums, then vice-versa to change things up. Try starting a song with a loop, but also try adding loops later on. Cre-ate distinctly different sections when se-quencing. Have the artist play a live instru-ment, or add live fills and crashes to the drum sequence. And last but not least — keep first takes. Chances are that no later take will cap-ture the energy of the first performance.

Just Do It

Some engineers feel that the lack of a par-ticular piece of gear is what's holding back their music from success, but you have to remember that gear only makes tasks more convenient. The most advanced things can be achieved in the most basic studio with just a little thought.

Dave Aron is currently working on Snoop Dog-gy Dogg's Last Meal, as well as the debut from Doggie's Angels Pleazebeliveit. He is also work-ing on Godwheel — the first release on his own label, Sticky Green Hits. Dave may be reached at 877-581-9065@skytel.com.

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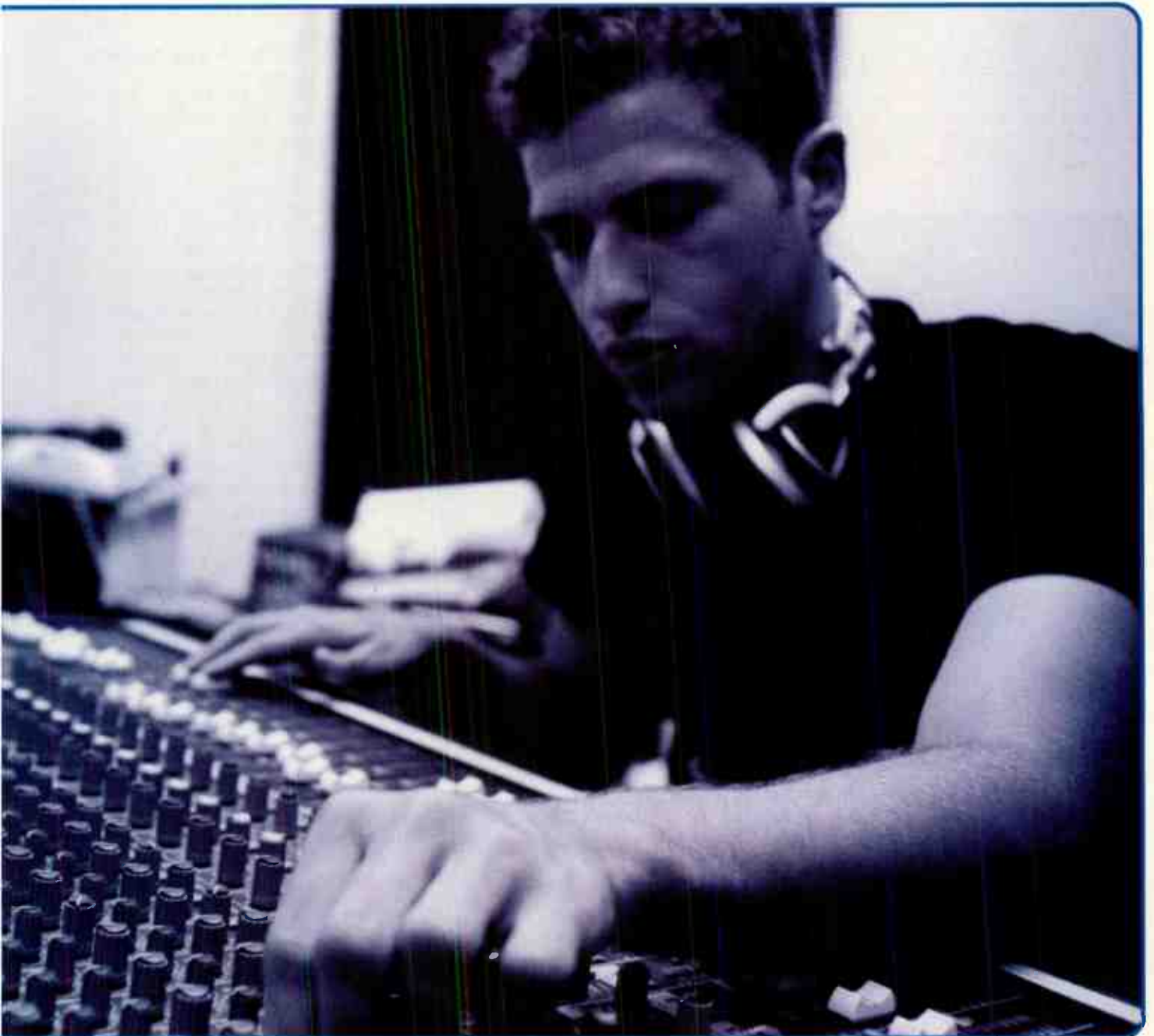
Music
Editor

by Bobby Owsinski

To the average musician or engineer not working in the film or television industry, the job of music editor is mysterious indeed. Most have heard of the position, yet few know anything about it. Daryl Kell, one of the most sought-after music editors in Hollywood, with recent credits including *X-Men*, *U-571*, *Whatever It Takes*, *Charlie's Angels*, and the upcoming *Miss Congeniality*, was kind enough to take us behind the scenes and explain just how integral to the process a music editor actually is.

EQ: What exactly does a music editor do?

Daryl Kell: The main part of my job is to make sure that everything that gets recorded and conceived by the composer ends up in correct synchronization with the movie. Along the way we deal with all of the changes that happen with the picture. For instance, the composer writes something for the original picture, but along the way the picture gets edited. If it's something that's already been recorded and mixed, I'll make the changes to keep everything in sync.

Another aspect of my job deals with recording the orchestra, if one is used. I provide proper click tracks and visual synchronization to help the conductor, and I make sure that everything stays in sync so that it locks up later in the intended manner.

At what point do you begin your job on a movie?

Normally I'll start before the music has been written and in some cases even before the composer comes on. I'll provide a temporary score for the movie using either existing soundtracks or classical recordings to help the director determine what direction he wants to go with the score. That temp score might later be used stylistically by the composer or he might go in a completely different direction.

Are you hired by the director?

Usually the studio hires me, but I'm asked for by the director or composer.

How did you get into doing this?

Originally I studied film scoring at Berklee College of Music. When I got out of school I started out by helping composers with sync problems they were having. I also helped them do some recording or made them click tracks. Because of my music background, I'm able to keep the integrity of the structure and harmony together as much as possible when editing.

How many films have you done?

I've been doing it for about 10 years now, and during that time I've done about 20 features as well as a bunch of television stuff. The average time that I work on a film is about three months, although the longest was nine months.

What's the difference between music editing for television and music editing for features?

Film is basically television expanded. There's more money involved, so you get more time to do things as a result. In television, schedules are very short, but features give you more options to fine tune things a little more.

Was there anything different about doing *Charlie's Angels?*

PHOTO BY EDWARD COLVER

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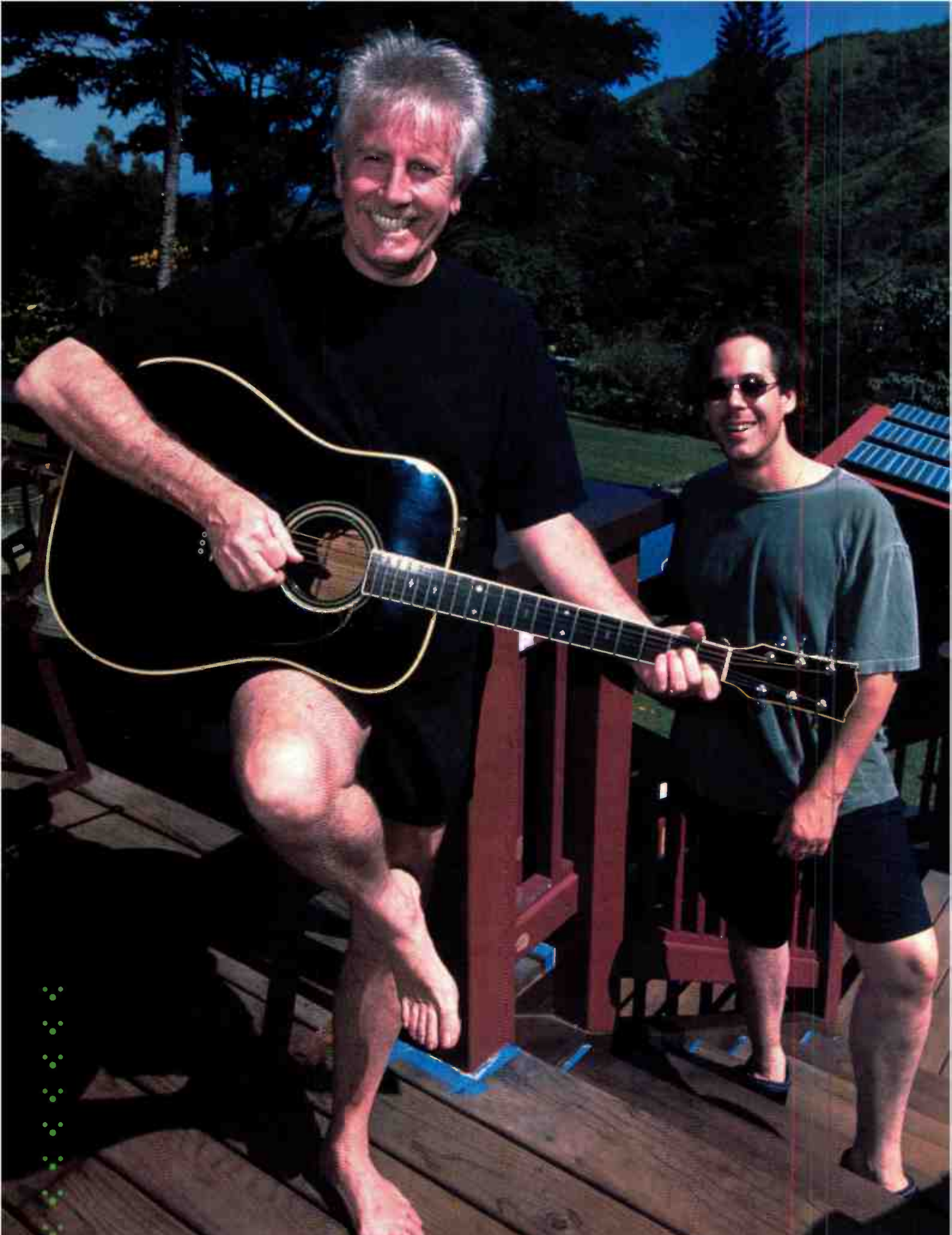
GRAHAM NASH AND NATHANIEL KUNKEL

PARTY IN
PARADISE

WORDS & PHOTOS BY MR. BONZAI

GRAHAM NASH AND NATHANIEL KUNKEL
DESIGN AND BUILD A STUDIO IN PARADISE

Imagine visiting Hawaii and being the guest of Graham Nash. You find yourself in a tropical valley carved from ancient volcanic rock. The main house looks out over palm trees, manicured gardens, and down to the rolling Pacific surf. Down the hill is a rock-solid new building designed by Graham and Susan Nash with their friend Bill Long, who has designed and built several homes for them. The new recording spot is dubbed "Camp Nash."

We arrived just a few days after the equipment package was flown in from L.A., where engineer/producer Nathaniel Kunkel had supervised the acquisitions, the wiring, and the sturdy road cases. Everything was fired up and worked perfectly. Within a few days, Kunkel was already recording Nash.

While visiting, we ate fresh oranges and listened to rough mixes from his upcoming solo album, which was tracked last October at O'Henry Studios in Los Angeles. Russell Kunkel is co-producing with Nathaniel, and providing his special drummer's signature, with Dean Parks and Steve Ferris on guitars, Matt Rawlings on keyboards, Victor Krause on bass, Dan Dougmore on pedal steel, and Lenny Castro on percussion. Mixdown took place at L.A.'s Conway Recording.

We've been touched by Graham Nash's music for many, many years. He was a founding father of the English group The Hollies, who scored more than a dozen hits. Perhaps the most enduring soldier of

the British Invasion, he pulled up his roots and changed his life at the Woodstock debut of Crosby, Stills, and Nash.

It's appropriate that CSN is three names; three distinct musical personalities. What's remarkable is the harmonious blend that became a three-headed chimera, and sometimes four-headed, as in last year's CSNY album and major tour. But this group has never been a "group" in the traditional musical sense.

"The individualism became a dominant part of what it was that we wanted to say," Nash explains. "We wanted to let people know that we were no longer going to be in a group; that we'd be individuals that came together to make music and could make music with whomever and in whatever form we wanted."

Nash has had great success as a solo artist, as a member of a group, and as a member of a non-group. He's living proof of strong individuality working in harmony with others — a man who knows how to make great records.

EQ: *Aloha.* Here we are at Camp Nash. How long has this project been on the drawing board?

Graham Nash: I used to come to Hawaii to escape the world of recording. I had relative control over what I was doing on the Mainland, and for many years I didn't even bring my computer here because I wanted to spend time with Susan and the kids. I tried to keep everything else away

PARTY IN PARADISE

from this place, but in the last couple of years I've been spending more time here, and my kids are now grown and out in the world. They don't spend as much time here now because they have jobs and careers they're dealing with. I began to get a little antsy while in Hawaii — if I got an idea at three in the morning I wanted to walk over to the studio and put it down, which I can do in Los Angeles. I began toying with the idea of bringing the studio here — duplicating the scene I have in L.A.

When Nathaniel and Russell and I were working on this solo record, I mentioned to Nathaniel that it would be great to have a studio in Hawaii. He immediately jumped on the idea because he loves it here as much as I do. He designed a studio for me that an idiot could use. I'm not really technically-minded, even though I've worked for many years in many, many studios.

Is it easier now to record on your own?

Nash: It's easier if you have someone like Nathaniel to put it together. I told him that if I came down here in the middle of the night, I didn't want to wonder where the kick drum was. I want to be able to plug my guitar or my piano in — and rock. If you can't find that kick drum, or the button to hear your voice, you get pissed off. By then the creative moment is gone. My studio at home, which was set up by Joe Vitale, is ultra-simple. A monkey could record. That is, if a monkey wanted to.

For this new room here in Hawaii, Nathaniel said that I could do it the old-fashioned way, and hence we have three TASCAM DA-78's for 24-tracks, and a Mackie board. But Nathaniel seems to think that, with a little concentration and training, I would do away

come into the studio to listen has had no idea that it isn't two-inch analog tape. They don't have a clue.

Nathaniel, how did you do that?

Nathaniel Kunkel: I masterclock everything with an [Aardvark] Aardsync, and I use the new dB Technologies "blue" converters. We had both their A-to-D and D-to-A. Sixteen channels of A-to-D for recording, and 32 channels of D-to-A for the mixing.

What other equipment is surrounding us here?

Kunkel: There's one rack that has 24 tracks of DA-78 — the HR 24-bit model. Another rack is basically two-track: two DAT machines, a CD player, cassette decks, and a CD duplicator. Another rack has the synthesizer equipment: a sampler, a Roland XV-5080, two Emu Proteus synths, a Mackie mixer, and a MIDI interface. [See sidebar for complete gear list.] Pretty straight-ahead, and also emulating what Graham has at home in L.A.

I see that everything is packed in boxes on wheels.

Kunkel: Yes, it's all in road cases so it can move. All the power distribution and patching is modular and can be pulled out at anytime. The studio could leave here in five minutes.

In this remote location, were there any unusual challenges you had to face?

Kunkel: No, nothing really out of the ordinary. I just made sure I used the best stuff — the patchbays are all nickel-plated, all the Alpha pins are gold-coated. Mostly, it was just a

with that system and go straight to hard drive. The record we just recorded went directly to [Digidesign] Pro Tools — there was no tape involved. It was the first time for me, and I found it very, very interesting.

How would you compare the quality?

Nash: I think it's better. I think the A/D converters that Nathaniel found and used in the system made it. With all due respect, I think you can detect the fingerprint of Pro Tools; it's a little bit brittle — you can hear it if you really know what you're listening for. But, every single person who has

matter of using the best quality materials. I have a really good wiring crew, so when everything arrived here last week it worked perfectly the first time.

Our concerns here were obvious: humidity and sunlight. We have a de-humidifier here in the room, and we tried to use gear that would be the most tolerant of moisture. Everything has lots of air space, especially things that generate heat. This room also has air-conditioning, which removes moisture. The most important thing was to make an environment where Graham felt comfortable, coming in and working and having the equipment survive as best as possible.

Nash: And for years, we've known and worked with the people who designed and built the road cases.

Yes, you have been on the road a bit, haven't you?

Nash: Once or twice.

How many days a year are you on tour these days?

Nash: It depends. In 2000, CSNY did 41 shows between January and April. And we also did our usual fair share of benefits. Last year there was no touring because we were recording.

Benefits — you're strongly associated with social activism — what's on your mind these days?

Nash: It runs the gamut from personal stuff to things I believe in. For instance, we recently did the Fred Waleke benefit at the Santa Monica Civic Auditorium, to help out an old friend of many, many musicians. A few years ago we did the huge concert for Nicolette Larsen, to preserve her memory and have a good time that night. Last November, I did a benefit with David [Crosby] for the Grace Foundation, which deals with child abuse and domestic violence.

A benefit concert might be for a small group that wants to remove broken glass from the beach, which is pretty important if you're a parent. And, over the years, I've worked with groups like UNICEF and the No Nukes movement. Peace Sunday at the Rose Bowl. It's a wide spectrum, but they are all connected by the word "help." As an example, it enables other people to say, "I can do something, too." If you want to help, you can, whether it's answering phones, pouring coffee, or getting on stage.

Are you an optimist?

Nash: I have to be. If I wasn't an

World Radio History

PARTY IN PARADISE

optimist I wouldn't bother being here making a record. I'd just be watching TV with a beer. It's easy to do, because the problems that face us are overwhelming.

Let's jump back to your early days. You're from Manchester, aren't you?

Nash: Yes, in the north of England.

Were you musically inclined as a child?

Yes, I was. Allan Clarke and I met when I was five years old and we started singing together immediately. I don't know why, but we were singing school prayers, and harmonized in school choirs and minstrel shows. Then, in the late '50s, skiffle music came to England via Lonnie Donegan from America. Skiffle was a simple form of folk music, basically three chords. It was fast and easy, and it was fun. We got into skiffle in a big way.

Then with the coming into our lives of The Everly Brothers, Elvis Presley, Jerry Lee Lewis, Buddy Holly, Fats Domino, The Platters, Gene Vincent, etc., we began to really realize that: A) we needed drums and bass; and B) the two-part singing we had been doing for many years was coming of age. So we found ourselves in the early '60s forming a band called The Hollies, with basically Allan and myself singing lead, occasionally three-part with Tony Hicks. We cut our first hit record in 1963 and I haven't looked back since.

I left the Hollies in 1968 because of musical differences and personality conflicts. We had been coming to America since 1965 and this was an entirely different environment for me. It was one in which I thrived, one which I loved. I got turned on to smoking marijuana, hashish, doing LSD, and expanding my consciousness. I gradually grew apart from the rest of The Hollies, who were basically more interested in the drug of their choice — beer.

At that time I had been hanging out with David [Crosby] and Stephen [Stills], so I knew that we had something tremendously vital. I was writing songs that The Hollies were not interested in recording, songs that I felt were deeper, more meaningful than what we had done earlier. All those factors came together in late 1968 and I decided to live in America.

So, 1968 — Hollies behind you, CSN ahead of you.

ON THE AIR

PARTY IN PARADISE

BUILDING CAMP NASH

Nathaniel Kunkel began apprenticing with George Massenburg before entering high school. Years of hard work as an assistant led to gigs with James Taylor, Linda Ronstadt, Little Feat, and his first solo recording and mix: Lyle Lovett's I Love Everybody *in 1994, followed by Lovett's Grammy-winning Country Album of the Year,* The Road To Ensenada.

Among numerous other projects, Kunkel has recorded Kenny Loggins's last three albums, recorded and mixed the 5.1 soundtrack for Robert Altman's Dr. T. and the Women, *and was nominated for a Grammy for engineering* Trio 2 *with Linda Ronstadt, Dolly Parton, and Emmylou Harris, produced and mixed by George Massenburg.*

What I Do

The process always starts by asking my client what their dream studio would consist of. We'll design that together and then scale it down, usually to meet a predetermined price point. That way I won't do anything building the smaller system that couldn't, for the most part, be used in an expansion on the way to the larger dream studio later.

I use the best materials possible, as the major studios do, when I build a system. You see, even though my clients request a small number of tracks, they're still in need of total reliability. That only comes with exceptional build quality. In truth, one of the things that makes that possible is my wiring crew. Kurt and Tess Stein have been doing all of my wiring for a while, and they're artists — they consistently give me the best wiring I've ever seen. Providing the level of quality I provide is only possible because my delivered wiring has a zero percent failure rate. In contrast, most of the hookup cables out there are garbage. Yet most people won't go through the trouble of placing a custom cable order if it's only for a single eight-pair snake — it's expensive and requires an overview of the complete system to not be a waste of money. I understand, but it still makes for less than perfect interfacing. So I'll try to develop the proper grounding, power, and wiring scheme for each install and then implement them in such a way that the studio expands easily without modification to the existing installation. For instance, Nash's studio is already pre-wired for the Sony DMX-R100 mixer and surround monitoring.

I also do everything I can to eliminate the possibility of mechanical failure — which also means making repairs easy — so I work very hard on documentation and cable management. Then, when there is a problem, you know where to look, and when you do, it's clean enough to see what's going on. I'm not doing anything that's groundbreaking, really — just not making any compromises and being super tidy.

So You Build Studios?

The only odd thing here is that I'm a recording engineer. Why am I building studios? Like I've got time to build a recording studio! I guess this all really started with my own portable remote setup that my friend Stephen Jarvis dubbed "Studio Without Walls." In the course of several live records, we had really figured out how to make a studio move, remain flexible, and be reliable. Some of our friends saw the systems and wanted to implement our designs into their new builds. So then I was building studios.

But I'm an engineer. Well, so what — I love building studios. (Someone must be wincing at that one.) Not only are they fun to put together, it's also really great to engineer in a room you put together. You know where everything is and it's all your favorite stuff.

—Nathaniel Kunkel

Singles had been the style of the record business, and your forming CSN was around the time of album domination and FM radio.

Nash: Well, *Sgt. Pepper* helped change that a lot. But it began to be obvious that the art form of albums was much more interesting than singles. The singles were like ads for albums. I knew when I sang with David and Stephen that that was what I wanted to do. We also knew we had tremendous song potential because we were three reasonably strong writers.

Our first record was brought out in a time of pre-heavy metal, stacks of Marshalls. We came out with this acoustic-feeling album and it threaded right through everything and made its mark. We knew when we left the studio with that two-track master that we had a hit record.

You've also had solo success, such as *Songs for Beginners*, which still sounds so fresh and alive.

Nash: I was very pleased with that record. I think I'm most pleased with its longevity. I get kind comments from people all the time about it being one of their favorite albums. I've tried to think what it was about that album that was so attractive and I can't really figure it out. I think it was very simple, very straightforward, and it had a very live feeling. There are some good songs, but I've never been really able to pin what it was about that album. I'm kinda glad, because I certainly don't want to repeat it. But I do wonder what it was that made it so attractive.

So much of your solo music, and the music you make with your compadres, captures a period. It's been said that if you want to go back and feel the Woodstock time, you listen

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to CSN and The Grateful Dead.

Nash: Kind of an awful place to be stuck, isn't it? [*Laughs.*]

But let's talk about your music as historic anthems in times of social upheaval.

Nash: One of the things that was upsetting me during my association with The Hollies was that there were social things happening that were far more relevant than the kinds of singles we got into — the "moon, June, spoon" in the back of the car, "summer days" types of songs. I'm not against that, but there is certainly more to be concerned with.

So when I joined David and Stephen, I knew that we had three writers that felt similar to the way I felt. Things were happening sociologically that were extremely important from our point of view, as artists, to make comments upon. I think it's been the role of the artist throughout history to bring news from one village to another, to spread tales of what's happening. In many ways, you're not getting the real truth from the mass media. You're only getting a part of the truth. What we have always tried to do as artists is to reflect what is going on around us, reveal what is going on inside of us, and express how we feel about certain subjects.

I'm not so sure that we're a political group. When you shoot four kids down at Kent State, is that political or is that just a tragedy? When you jail the Chicago Eight for allegedly trying to wipe Chicago off the map, is that political or is that a tragedy? And when we talk about the nuclear power industry, are those political issues or are they human condition issues? As artists we've always wanted to be our own psychiatrists in a way. We have the ability, because we are writers, to internalize situations and then bring them out as music. I'm not sure what enables us to do that, but I guess that's what an artist is.

Could you tell me a little about this new solo project you're working on?

Nash: I go into the studio to have fun, and I've been doing that since I cut my first hit record in 1963. This particular project has been better than most, except for maybe the first CSN record. I like to get work done and I like to remain open to everyone's ideas. When you're asking great musicians to contribute, you don't want to tell them what to play.

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ED CHERNEY

Ed Cherney is one of the most popular engineer/producers in the industry, and it's easy to see why. Loose, relaxed, and witty, he's got exactly the kind of laid-back personality required to put even the most neurotic artist at ease — but make no mistake: when that red light goes on, the man is all business.

Cherney began his career in the mid-1970s, assisting legendary recordist Bruce Swedien. Before long, he was working with the likes of Barbra Streisand, Ry Cooder, the B-52s, and Bob Dylan. In the mid-'90s, his career took off like a shot. His work on Bonnie Raitt's Longing In Their Hearts *netted him the 1994 Grammy for Best Engineered Album; shortly thereafter, the Rolling Stones hired him to man the desk for their* Stripped *and* No Security *live albums, as well as their* Bridges To Babylon *studio release. Somehow, he still found the time to become one of the founding directors of the Music Producers Guild of America (now the NARAS Producers and Engineers Wing). Cherney recently chatted with us at Ecstasy Studios in L.A., where he and co-producer Mark Goldenberg were putting the finishing touches on a new Jann Arden release.*

EQ: What are the common mistakes you're hearing in the recordings that people are making in their home studios?

Ed Cherney: Bad songs. You can tell when a song is amateur hour, but anything goes with recording. Look at what people are buying — you've got stuff that sounds like a tin cup on a string, and you've got things that just fill the speakers. Certainly, we're living in a scratchy time [*Laughs.*], but nonetheless, it's still music. Unless you're listening to the symphony or something, it's not an issue.

BY HOWARD MASSEY

FROM THE
Heart

ED CHERNEY

If it doesn't matter, then what's the justification for working in a professional studio?

Primarily, the service. There will be equipment in a professional studio that you won't find in a project studio, and some equipment is tailored for the artist. For some artists, only a Neve 8078 will do them justice; others can create their tracks on a laptop in a garage, and it will be fine.

How do you approach working with an artist that may have talent and good songs, but doesn't have a clear concept, technically, of what they want to do?

I don't know if I'm necessarily qualified to work with an artist like that. Typically, I work with an artist that has a vision, and it's my job to deliver that, and hopefully make it palatable to the record company. But it's not my responsibility to provide the vision. I generally work with artists that have an idea of who they are and are expressing themselves rather than manufacturing a teen idol or something like that.

What approach do you generally take to recording guitars?

I've been using these new Royer ribbon microphones, kind of all around the room, and that's been working really well. Typically, the kind of guitar players I record come in with their own rig and their own sound, and it doesn't call for much except for the usual miking down the hall, finding a good corner of the room, and blending microphones together. The guitar players will have messed around a lot at home and they know what they expect to get out of their amp, and usually what they give me is pretty darn close [to the final sound].

So you're talking about simply capturing the sound that they craft at the amp?

Yeah, I'm a pretty straight-ahead guy in terms of that, though I'll go chase some things around. With Ry Cooder, we hung an old ghetto blaster, an old SuperScope that had a microphone that was really limited, and we took a line feed out of that. I've recorded guitars out of those kinds of things to get a certain "bing!" kind of sound.

Do you often double guitars?

It depends on the music, but usually not. If I do it, I'll usually use different instruments for different textures. It can be nice to double an acoustic guitar that's strung normally with another one that's strung differently or capoed up. That adds some harmonics to the sound,

which works well. The kind of records I'm making these days tend to have a bunch of individuals playing as opposed to a big wall of sound, so you can pick out each musician; each has his own character, and you're trying to invent a new sound each time in order to develop that character. But you're typically stumbling across it — there's not a whole lot of premeditation involved. Making records is

different every time — there's really no "typical" way of doing things.

One of the biggest problems facing recordists in project studios is getting the bottom end right — big, but tight and not woofy or flabby.

There are a couple of things working against guys in project studios. One is that they're probably not working in a space that's acoustically beneficial to getting a good, tight low end. There're typically parallel walls and standing waves, so they don't know what the heck they're getting, and they may be listening through speakers that maybe aren't that great, so they can't turn them up to shake the low end. And it takes years just to learn how to use your EQ correctly. What I see most people at home doing a lot is just boosting frequencies. They give it a boost at the kick drum fundamental — at 30 Hz, with maybe shelving, so it starts to go all the way up and begins to fill up the sound at 250, 300 [Hz], where it starts to murk up your record. Then you've got no place to put the bass. So people need to know how to use their EQ — learning how to dip some frequencies and move the fader up instead of boosting all the time. It can take ten or fifteen years to learn how to do that. That's the biggest mistake that rookies make — they just reach down and boost the EQ and say, "That's better because it's louder now," but they're not necessarily

listening to how the sound is being shaped. That's just experience, and that's one of the hardest things to learn. It's very difficult to shift your thinking that way.

A lot of novices also have trouble dealing with dynamics processing.

Well, the thing is that guys like me and a lot of my contemporaries served apprenticeships in studios. We worked un-

> "SOMEBODY WHO KNOWS WHAT THEY'RE DOING CAN MAKE A GREAT RECORD WITH A TWO-BY-FOUR, AND SOMEBODY WHO DOESN'T IS GOING TO GET A POOR RESULT REGARDLESS OF HOW GOOD THE GEAR IS THAT THEY USE."

der engineers that started in the '40s and '50s and built the gear that they used and went from big bands to symphonies to rockabilly and country, then eventually to rock 'n' roll; guys who saw the technology evolve. Sitting behind the Bruce Swediens, the Phil Ramones, the Al Schmitts — we learned how to listen. We learned what worked and what didn't and how to use that stuff — more than just reading the manual. But when you're at home, you're on your own — you usually don't have a mentor, and you never really sit behind somebody who's been there, done that, has 30 or 40 years experience, and knows how to make great records. Like they say, it's the nut behind the wheel. Somebody who knows what they're doing can make a great record with a two-by-four, and somebody who doesn't is going to get a poor result regardless of how good the gear that they use is.

When you're recording bass, do you take an amp signal as well as DI?

I do, to shape it. The truth is, with a great player that has the touch, you can stand his bass anywhere near a recording machine, and it's going to sound great. A great musician is really easy to record.

What mic will you typically put on the bass amp?

I've had a lot of experience with [Neumann] FET 47's; they work really well if

you're careful to pad them down and are aware of what the proximity effect is doing so you find the sweet place to put them. But I always listen to a good direct box first, just to see what's coming out of the bass. Then you go out into the studio and listen to the speaker; it's fairly simple.

Do you use the amp signal for the low end or for the bite?

Most signals will only take so much EQ, so I'll use the amp as filler for the low end, and I'll use the direct more for the string noise and finger-popping for the attack. Then, without using a lot of EQ, you can make it brighter or duller or fuller just by moving one fader up and down against the other. You can fill it in that way rather than ruin the phase of the thing by twisting EQ knobs.

Do you compress both signals as you are recording?

Typically, I'll put just a little compression on the direct signal as I'm recording it and leave the amp compression-free and EQ-free if I can. I don't want to lock myself into something that I can't undo or change later. But, with a great bass player that has the touch, you don't need a compressor, because he makes the instrument speak. Certainly, having a compressor with variable attack and release times is important. That's a lot of the big trick about compressing bass: using the right attack time so you can get the attack of the thing and using the right release time so it's not pumping — unless you want it to pump, of course. Basically, you're trying to get all of the notes to speak evenly so there are no holes, nothing sticking out.

What sort of compression ratios would you use on bass?

4:1. But you might go 12:1 or even 20:1 — whatever it takes. It's going to be dependent on the way the guy plays it and what the music is calling for.

Do you typically compress the bass again when you mix?

Lately, I've been adding a little bit of compression on just one of the tracks; sometimes I'll compress it two, three, or even four times.

Project studio recordings often lack "ear candy" — the little, almost subliminal elements that seem to make a record a record. How is the decision made to add a little percussion here, a little sound effect there?

If we've got an open track, let's put something on it! [*Laughs.*] You just lead with your heart. Your instinct tells you what needs to be there and when to leave something alone. You listen to stuff over and over again, and you may find yourself thinking, "We need a little

action over here, something to set this part up a little bit better." You just go for what the song asks for, so you try it, you put it on, and if it doesn't work, you get rid of it. When you come to a fork in the road, take it!

Which leads to the next question: How can you tell when something's done, as opposed to not yet complete or overdone?

You just know; it comes from the heart. There's nothing quantitative about being an artist or making a record. That's why not everybody is rich and famous! [*Laughs.*] I think it's down to hearing every individual musician somewhere in the soundscape, somewhere in the band, rather than necessarily having every color and every space filled in.

That's probably the hardest thing for the person in a project studio working on their own — knowing when to stop.

Yeah, but for people who are just doing music for the enjoyment of it, that's the whole point of it — to just keep sitting there working on it, messing around and trying stuff. If it's not going to find its way onto a rack at Tower Records — if it's just going to find its way into cyberspace, maybe that's the point of the thing, to just keep working on it, to keep it as a work in progress. But if you have a deadline and there are record executives counting on you to fill in the numbers on their quarterly reports, you're certainly dealing with a different set of problems.

Some artists, it seems, are never satisfied with their work; others can let it go after the record has been released.

Well, yeah, and I know both kinds of people, though I can't name names while they're still alive! [*Laughs.*] What happens, too, is that a lot of times an artist records a song before they have a chance to perform it before a lot of people, and as you perform it dozens and dozens of times, there's a certain life, a certain texture that you discover. You don't get that if you're making a record and you just do a few takes.

That's a good point, and it probably accounts for the sophomore curse of a lot of second albums.

Very true.

What can you, as a producer, do when it's clear to you that a song just isn't ready to be recorded?

You have to say so — in the nicest way, of course. If you're working with artists that write for themselves, you have to inspire them to write by playing them lots of songs, perhaps pointing out examples of what you're talking about. But you have to do it in a very decent, very kind way. These days, though, there will be a lot of people listening to the songs, and there will usually be a consensus of opinion.

Producing by committee, almost.

Yeah, there's a lot of that, unless you have an artist that contractually has the right to have things done their way.

How long do you typically spend on a mix?

As long as it takes. I have a reputation for being really slow and driving everyone crazy.

Is that because you're a perfectionist?

No, it's definitely not that — I'm really a slob! [*Laughs.*] But you delve into a mix and you get led down different paths — you want an opportunity to explore: Where's the groove, where's the

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pocket, where's it at? Having said that, if the stuff's really well-recorded, it's largely a matter of putting the faders up and figuring out how to pan it, where to put a stunt here and there, getting the vocals sounding right and balanced, and going from there.

Which faders do you bring up first?

All of them. I start out by pushing everything up and getting a feel for what's on tape, where everything is. I rarely solo stuff unless I'm looking for a problem. I balance from the perspective of looking at it as a whole instead of looking at each sound individually.

Once you've got all the faders up, what do you do next?

At that point, you can make arrangement decisions — maybe you don't need the four trombones in the verses, and maybe we can empty this out or use that part to build the chorus. But the shaping of the sounds is done with all the faders up. That's the best way to do it, though it takes discipline — you've

have to keep wrestling with it. Vocals are usually the trickiest thing, especially in country records, where everybody wants to be able to hear every single word. In modern rock or other genres, you can get away with things like burying it or moving it to the side. I usually start by putting the vocal up dry and see if I can shape it around the music. If it doesn't sit right — if you have to make it too hard to get up over the music — I might try cooling out some of the music around it by cutting some of the upper midrange. It's all about creating a space; you only have so much 2, 3, 4 kHz that you can fit on a record before curling everybody's eyebrows. So instead of stacking it up, I'll try to dip some out and see if I can fit it in there, frequency-wise. A lot of people who haven't been doing this a long time, the first thing they'll do is start reaching for delays and reverbs and all that kind of stuff, but that's missing the point. You want to shape it so the music is sitting around it and the vocal is telling the story like it should, so it's in the right

"LEAD WITH YOUR HEART. YOUR INSTINCT TELLS YOU WHAT NEEDS TO BE THERE AND WHEN TO LEAVE SOMETHING ALONE."

got to concentrate and listen inside the music instead of using your solo button and listening to one thing at a time.

"Listening inside the music" is an interesting concept...

Well, I'm talking about things like finding that guitar line that's inside and then finding its counter-line — being able to isolate and identify the different things that are happening in the context of the entire spectrum.

Are you talking about achieving maximum separation?

Not necessarily. It depends on what is called for — it might be achieving a perfect mess! [Laughs.] You have to listen with your heart and do what's called for. That's what makes some people great artists and some people not. Some people can throw the ball and some people can't; some people can run and some people can't.
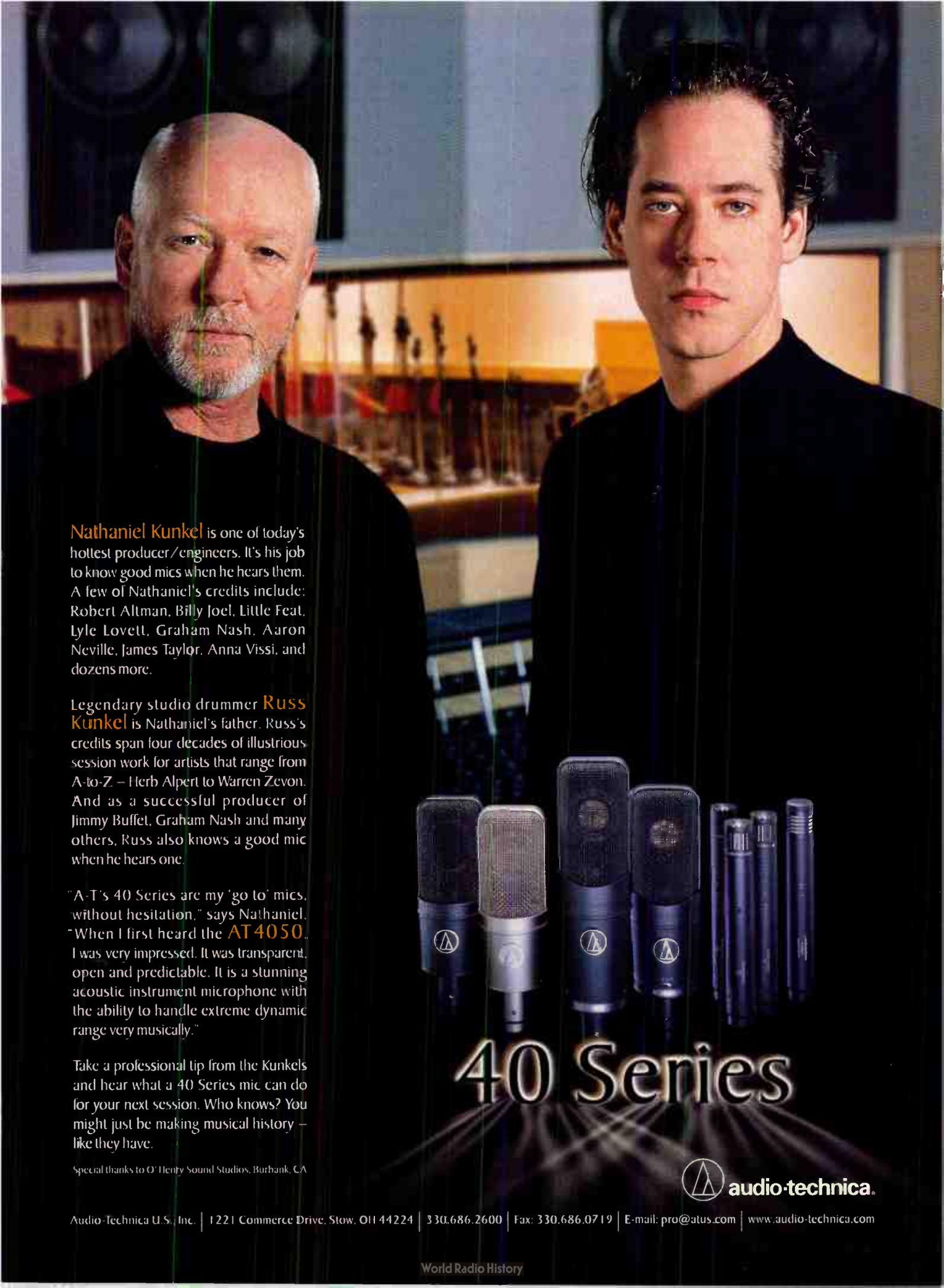
Any tips for getting a vocal or a solo to sit right in the track?

It's a major pain in the ass. You just

place. And then if you want to add something to it to put it in a space, go ahead, but you have to shape it first.

Do you typically use one or two reverbs for all the instruments, or do you favor using discrete effects for each instrument?

From experience, I know that I'm going to want something that's short and bright — a room reverb set to a second, second-and-a-half kind of thing — and I know I'm going to want something a bit longer than that, maybe 2.5, 2.6 seconds. I'm going to have effects for maybe three acoustic spaces: something short and bright, something medium, and then maybe a long hall with a long predelay. I'll definitely start with those, and those will be off echo sends. Then I'll have an eighth-note or quarter-note delay, and I'll have some Harmonizer left/right stuff. But I won't send much stuff to it at first; I try to balance the music dry and then use the echo sends to put it in an interesting

space. I just like to have the options, and then I turn things up and down and mess with it until I stumble upon the truth.

But you'll send multiple instruments to each of these effects?

Sure. You're putting together a cocktail, so some things will go into others — it can be quite an extravaganza. Usually you don't have enough echo sends.

When you talk about setting up "stunts" in the mix, what kinds of things are you talking about?

It could be little lo-fi things, stuff like that. Different vocal treatments in the chorus, any kind of event that will add excitement. Changes that will catch your ear a little bit, up the ante. That's the kind of musical world we live in now — you have all these digital and analog tools, workstations, all this different stuff — and that's what's making it fun, combining all the different elements and using all the different tools. It's a broad palette.

What's your typical vocal chain?

I have some Class A [Neve] 1073 mic pres in my rack, and I've got a very nice Mastering Lab tube mic pre that I use. As far as compressors, it could be an 1176 or a GML; for some kinds of singers, a dbx 160 works really well if you just want to protect the tape.

How would you sum up the audio state of the art in the year 2000?

In a lot of ways, it's the worst of times and it's the best of times, in terms of the way a lot of records sound now. It's a fairly graceless age — especially 16-bit, 44.1 — but it's also a time when everyone can be an artist and make music. And why not? Even if 90 percent of it is crappy, there's always that other 10 percent. I'm waiting for the time — and it could happen any day — when somebody at home is going to make some sort of record and put it on the Internet and a hundred million people are going to download it. When they figure out a way to get paid for it, it's going to sweep the world, and it's going to shake things up about the way people do things. It could happen anytime, and I want to be there for that — it's an exciting prospect.

BY JIM BORDNER

When the venerable Atari 2600 breathed its last rattling breath, conventional wisdom said that video games were *kaput*. So much for conventional wisdom. This year, video games actually eclipsed movies in annual revenue.

Most game developers are working on improved playability, pushing the envelope on numbers of polygons generated, or tweaking the movement of mo-capped player avatars. But a few are defining the potential for music and audio for video games, and *EQ* recently talked with two producers at the forefront of this movement.

Chance Thomas, a classically trained composer, came into the game business at the beginning of its renaissance. He now heads Huge Sound, a group of independent project studio pros responsible for the sound of some of the decade's most influential video games. Thomas compares game music to film music, and feels the industry is entering a Golden Age.

This makes sense to Duane Decker, a composer and producer for Microsoft. Decker's latest achievement says something about the current state of game music: his soundtrack for *MechWarrior 4: Vengeance* will be the first video game soundtrack released on Varese Sarabande records, a label well-known for film scores. We talked with each of them in their respective studios: Duane Decker's within the megalopolis that is Microsoft's Redmond, WA headquarters, and Chance Thomas in his home project studio outside Yosemite National Park.

TODAY'S GAME MUSIC RIVALS MAJOR FILM SCORES

SCORING IN THE BIG GAME

big game

taking a chance

Chance Thomas began specializing in game music because it was fun. "I started out in advertising," he says. "Before long, I was burned out — so much of the work was the same. But when I got a chance to do my first game, I was asked to do fifty different pieces of music, in all different styles. I loved the challenge of it. As I finished that project, a friend saw an ad from Sierra Online, looking for a composer. So I applied."

Thomas moved to California to work full-time for Sierra. But it wasn't long before his entrepreneurial spirit kicked in, and he began seeing possibilities for independent game composers. As Sierra began development of the *Middle Earth* series of games, Thomas struck a deal with them to open Huge Sound and make the *Middle Earth* music the new company's first project. Since then, Huge Sound has created audio tracks for *Tomb Raider, Unreal, Deus Ex,* and many more.

Huge Sound now offers a complete one-stop audio solution for developers, providing music, sound design, voice casting, and recording. But Thomas hasn't grown by hiring staff. In a unique approach to the business, he's assembled a network of project pros reaching from California to London, each with expertise in specific areas and tied together by high-speed Internet access. This business model gives Huge Sound the ability to respond quickly to clients and to the continual changes of the industry. "And we avoid all the administrative costs and legal problems associated with a partnership," he adds. "We're each running our own business, but working together."

"There aren't a lot of independents in the industry right now," he says. "Major game

CHANCE THOMAS RECORDS A FULL ORCHESTRA FOR ONE OF HIS VIDEO GAME RELEASES.

> USING INDEPENDENT COMPOSERS IS NOT ONLY COST-EFFECTIVE BUT OPENS THE DOOR TO A WORLD OF STYLISTIC POSSIBILITIES.

publishers are very much in the place where movie studios were in the '50s, with all the music for films being done by in-house guys. But they're beginning to see, as the movie studios did, that using independent composers is not only cost-effective, but opens the door to a world of stylistic possibilities."

Thomas sees a great potential in the game market for project pros. But, if you're interested in this market, get involved now, and be prepared to work cheap or on spec. "You may not get rich right away, but there's a lot of opportunity to get involved. Hundreds of developers are constantly cropping up, and someone working in their garage right now might be the next big thing. It's almost like a rock 'n' roll band…if they have a hit, they can rocket to the top. If it's your music on that hit, you get a lot of attention. But the flipside is that it may never see the light of day."

"And budgets are all over the map, even with the major developers. They may come in and try to get you to do a job for $65 an hour and tell you how many hours it's going to take you. Or they may say, 'Here's our huge project and our glorious vision for this project,' and leave it to you to set the budget."

Thomas's best advice for composers who want to work on games is to get inside the industry. "My job with Sierra led to my starting Huge Sound," he says. "There's a revolving door at the publishers, and opportunities open up. New companies are starting up all the time. It's not six-figure money — $45 to $75 thousand a year is average. You can subscribe to the Gamasutra online newsletter and see job openings continually."

But the best thing you can do as a composer is to hone your craft. As game scoring and film scoring become more similar, there's greater demand for people like Chance Thomas with traditional orchestrating skills. "It's getting more sophisticated all the time. Full orchestrations, acoustic scores, surround mixing so you can hear which direction the bad guys are coming from — they're all getting more popular. It won't reach its peak until the average living room has the giant full-service Visio-Screen in the center…" He chuckles. "…but it's coming."

the upper decker

Duane Decker has lived music all his life, starting at age eight as drummer. He paid his way through college doing the bar grind, and then hit the road for thirteen years, playing keyboards with a string of rock bands. He became a product specialist for Emu and Kurzweil, and eventually was attracted to game music, starting out making tunes for video gaming's high-touch analog cousin, the pinball machine. "I created music and sound for thirteen games at Gottlieb before a headhunter recruited me for FASA International, the company that created the *BattleTech* engine that's the basis for the *MechWarrior* universe," he says. "They were looking for a publisher, and Microsoft liked them so much that they bought them. So, I moved to Redmond."

Decker's first big surprise at Microsoft was how little the software giant tinkered with FASA's formula. "I thought we were gonna get sucked into this corporate void, but we got here and

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found people with the same passion we had. It's just like it ever was, except we didn't have to worry about where the next check was coming from. And, of course, the resources are incredible. Getting new gear used to be like pulling teeth, but now it's like, 'Why don't you buy two?' It's great."

Duane had scored three previous *MechWarrior* games, but as he progressed on the music for number four, it was obvious that this was going to be something special. In March of 2000, he started talking to Bob Townsend at Varese Sarabande about releasing the soundtrack as an album. Townsend was interested, but a realist. "He kind of rubbed our noses in the level of production," says Decker. "Bob played us records by Jerry Goldsmith and told us it was going to have to sound that good. And I said, 'Hey, I love a challenge.'"

The score is a mix of orchestral and rock elements, and Decker was unsure about contracting symphony players until visiting a film session where Elmer Bernstein was conducting the Seattle Symphony, which inspired him to hire them for his sessions. "The orchestrations were written by Stan LePard, based in my MIDI parts," he recalls. "We had two exceptional wind players on English horn and flute. The horns are about 80% live, with some synths, and the strings are about 60% live. That's what I love about Pro Tools — you can go over and

there's more opportunity than ever for project pros in the game industry. And Chance Thomas and Duane Decker are proving it.

mask a lot of things, and make it sound so real."

The final element of the score, the rock guitar parts, came from an unlikely source. "I hadn't planned on hiring a guitar player. But, my manager here is a guitar player, and when he listened to the tracks, he asked for a real guitar and suggested a Microsoft employee, one of the guys on the X-box team, Cliff Garrett. I had my doubts, but he showed up with this rack of gear and a couple of Marshall cabs and just *wailed*! He nailed the parts. We did a nine hour session, finished at 3 AM, and it was the most fun of the whole project."

The MechWarrior 4 score was recorded in several studios, and assembled and mixed in Decker's studio, where the gear list would make most *EQ* readers drool (see sidebar, "Tools of the Trade"). "Digital Performer was used with Digidesign Pro Tools MixPlus hardware to integrate MIDI and digital audio. We upgraded my studio two months before live recording to accommodate more tracks. We added the Mackie d8b, G3's, and lots of other stuff. And we had the typical problems getting the system tuned. Transferring data to and from different studios and different Pro Tools systems was very time consuming; delivering reliable data to the studios for live sessions, bringing back the newly recorded tracks, distributing tracks across various drives, and formatting drives in the middle of final mixdown! But once I had all the tracks and devised a system to do the mixes, it was a *great* way to work."

In the end, the tracks were sufficiently impressive to warrant release on Varese Sarabande. "I've got some people to thank," says Decker. "Nancy Figatner at Microsoft was instrumental in getting the deal done, and Erik Labson at Universal Mastering – West really brought the tracks to life. He has such great ears, and knows the [Waves] L2 backwards and forwards." The album will be released concurrently with the game in every market where the game is available, and Duane Decker has high hopes for its success. "Fans of the *BattleTech* universe are just rabid — twenty million people have played, seven million computer games sold, over 5,000 *BattleTech* fan sites are on the Web...." he pauses and laughs. "But the whole concept is so new, I don't know if it'll sell or not!"

There's more opportunity than ever for project pros in the game industry, and Chance Thomas and Duane Decker are proving it. Hit the *EQ* Web Links for more information on these composers, the game industry, and your place in it. ∎

WEBLINK

Check out Chance Thomas's audio samples and his unique business model at www.hugesound.com. Learn more about the catalog of film music or order Duane Decker's CD at www.varesesarabande.com. Gamasutra has the latest on movers, shakers, and players in the game business: www.gamasutra.com.

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World Radio History

THE ENDS
JUSTIFY
THE MEANS

MIKE SHIPLEY GOES TO EXTREME
LENGTHS IN PURSUIT OF THE ULTIMATE MIX

BY HOWARD MASSEY

As if you didn't need more evidence that the world is shrinking, consider that the top mixing engineer in Los Angeles was born in Australia, received his training in British recording studios, and has made his name crafting the smooth-as-silk sound of Nashville divas Shania Twain and Faith Hill.

Mike Shipley is the Aussie in question, and you'd be hard-pressed to meet a nicer guy from "Down Unda." First cutting his teeth in the '70s at Wessex Studios under legendary engineer Bill Price, Shipley worked with many of the post-punk bands of the era, including the Sex Pistols and The Damned. He also manned the board for famed producers Roy Thomas Baker and Chris Thomas, and began a long association with producer Mutt Lange, engineering and mixing a succession of albums for Def Leppard and Lange's wife, Shania Twain.

Today, Shipley is enjoying unparalleled success as a mixing specialist supreme, bringing his unique touch to hit after hit with a diverse crop of artists ranging from The Corrs to Aerosmith, from Neve to Semisonic, and from The Black Crowes to The Catherine Wheel. Though soft-spoken — even shy — Shipley noticeably came to life when seated behind a conveniently located mixing console, often illustrating his point by moving faders and turning knobs, even though there was no tape playing. Clearly, he's more comfortable doing than talking, but his insights into the process of mixing were fascinating nonetheless.

EQ: Some engineers decry the whole idea of a mixing specialist, questioning the whole concept of having somebody who was not involved at the outset coming in and doing the final work on a project.

Mike Shipley: I understand that sentiment, because I've been on both sides of that situation. Certainly there's an argument to be made for having

the person who recorded the album mix it. But, there also seems to be a fair number of people who like the idea of having someone with a fresh perspective come in at the end of the process.

Is it always an advantage to come in fresh?

In certain cases, yes; in certain cases, no. A lot of records these days are recorded into huge Pro Tools systems or in all kinds of different formats, and basically the tracking engineer is doing the best he can just to keep up; he doesn't really have the time to be planning ahead on how he's going to put everything together. That's the case with producers like Mutt Lange, who record a lot of information, with quite a lot of shifting going on in the mix process. There are a lot of times when it's great to have someone come in fresh at the end of a record, at the point when the people originally involved have lost sight a little bit or are tired.

Personally, I always make sure that I listen to whatever they've got. It's in my interest to hassle the producer for rough mixes of what they've done so I can see the direction they're going, how everything is feeling, what I think I can add to it. I like to talk about it beforehand rather than just being this person that gets handed a tape ten minutes before the session starts. I'd much rather drive around in my car for a couple of weeks listening to roughs so we can all be on the same page.

Do you ever find yourself in a situation where you're chasing the rough mix instead of trying to inject a fresh approach? Some remixers I've talked to refuse to listen to rough mixes for that reason.

I make a point of it, actually. I kind of hound whoever I can to get a copy of the roughs because I really like to think ahead; I really like to get a vibe for a song. Do I try and chase the rough mix? No, I don't think so, but by that point, having listened to it in different situations, like in my car, for example, I've got a fair concept of what they're going for. That doesn't mean to say that what I'm going to do isn't going to be 180 degrees different, but I want to at least give all the people involved in making the record in the first place the benefit of listening to what they've done. Sometimes you find yourself going in a completely different direction, but there're a lot of times when they've done a lot of things right.

Do you usually have the producer or the artist in the mix sessions with you?

It's different every time. With a couple of the records I've just done recently, the producers are in Nashville, so I just mix away, and it would be an ISDN thing — I'd give them a call when I was ready, and say, "Zap over to the studio and let's see what you think."

On the other hand, some people like to be there most of the day. Generally, you hope that someone's going to be around; I like it when someone's there — it's much more fun.

Do you like to have the artist's input at the mix session as well?

Sure, if the artist wants to be in on it, absolutely. There are some records where the artist is very, very involved, and some where the artist is less involved.

There are many mixers that really don't like to have the artist present, at least not until the very end.

I like to get started on my own, but at some point it becomes a team effort. It really is; it's a part of the process. It's not, "Get out and stay out," it's all for the common cause of what's best for the record. Any input is great; sometimes you might not see eye to eye, but it's a matter of making it work for everybody.

In those situations where you don't see eye to eye, how are those kind of conflicts generally resolved?

There are plenty of times when a producer or an artist might have a fixed idea about what they think they want to hear, and it's up to me to do my best to give them what they want. But there are other instances when I personally think things would be better done a little differently. All it means is putting down another pass. I'll do it both ways, and both parties can go away and live with it.

I can't think of any instance where I couldn't find a compromise to make it work for everyone. You just put down multiple passes, and when you get out of the studio and listen to it from a different perspective, when you're not so totally attached to what one person's idea is, often you go, "Well, what they said was a great idea," or you might say, "We'll use the verse from this one and the bridge from that one." It's very easy to have a meeting of the minds, and that's the important thing.

Do you miss doing tracking?

Yes and no. I'm happy mixing — put it that way. I did a lot of tracking for a long time and even during mixing there are quite often times when overdubs need to be done and so forth. I can't say that I miss tracking that much, to be honest. It's certainly a pleasure when I do it, but I just love mixing.

But there have to be times when you pull up the faders and you say to yourself, "I wish I'd recorded those drums and saved myself ten hours of work fixing them."

Yeah. [*Laughs.*] At the same time, I have no problem spending the ten hours fixing them because these days there are so many tools that you can use if it's not a great sounding track.

And you can look at it both ways. Quite often people will shoot for a certain kind of drum sound because that's just the style of the drummer; that doesn't give me license to change it — that's exactly how it should be. I just make it sound as best I can with respect to the song.

The thing is, it used to be even harder to mix the stuff that I tracked; I was never very good at putting a value judgment on what my stuff sounded like. We might have spent an awful long time getting the sounds — like on the records I did with Mutt [Lange] — but by the time it came around to mixing, you almost wished that they were different sounds, anyway. I really enjoy getting other people's tapes; I find it kind of interesting to mix other people's stuff, more than my own.

Let's talk a little bit about the mechanics of how you start a mix. What faders do you bring up first? What's the thought process?

First I do a quick rough mix for myself. I find out what's on tape, how it fits together, what the overall character of the sound is like. So I'll start by putting up all the faders and making very quick decisions about the next mode that I'm going to go into. Once I've figured out what the overall sounds are like — and I know pretty much after spending a few minutes in that mode — then I've made my decision about how I'm going to go about what I need to do.

I'll balance out the rhythm section — the drums and the bass — to find out what the overall bottom end is like. Then I'll bring up the guitars and all the instruments and

do a balance, then put the vocal in and balance the instruments around the vocal as a rough, just so I can get the feel in the control room for how things are coming out on the speakers.

What happens after you get the rough done?

I'll pull all the faders down, then I'll usually start with the drums and bass, same as everybody. In the midst of tweaking them, I'll still be zapping in the guitars just to see how they fit. You don't want to work solely on one particular sound for endless amounts of time, so I'll try and work things in and around them. It depends entirely what the music is — it depends on the consistency of the guy's foot as a drummer, and how it connects with the bass; everything is so interdependent. So I start building the mix; I figure out what needs to be multed up, how I can make things better, how I can make the drums work better.

Are vocals generally the last thing you bring in?

No, vocals are in and out the whole time. Because, again, everything is so interdependent. I'll have the vocal in and I'll be working on a vocal sound at the same time, then I'll slap it out again once I'm working on the parts around it, but I'll keep putting it in and see how it sits. I don't just try to pile it on at the end, that's for sure. As soon as I bring the vocal in, the whole perspective changes so much; it'll change the interbalance. I'll get a feel for the vocal; find out if it needs to have different EQ in different sections of the song. Usually I'll mult the vocal up many times so I can keep the perspectives right in terms of different compression in different parts of the song, different EQ, different effects.

So you actually apply different EQ to different sections of the vocal?

Oh, yeah.

Do you do a lot of manual level riding? Or is it basically done with compressors and limiters?

A lot of manual level riding, though it depends on how it's been recorded; it depends how it feels, how it should sit. There's definitely outboard stuff used; one size doesn't fit all. You can't set a compressor for a verse and expect it to work for a chorus when the person's belting and the level is a lot hotter on tape, so I tend to mult stuff up and have different-sounding compressors or different ratios or different amounts of compression for different sections. Again, one size does not fit all.

Do you have any tips or techniques for getting a lead vocal to "sit" in a dense backing track?

It's down to the kind of compression that you use. Generally, the more apparent loudness the lead vocal has, the better, so you have to consciously work with the compression so it doesn't sound

THE ENDS JUSTIFY THE MEANS

squashed. And a lot of it is down to manual riding; the perspective of how a vocal sits in a dense track has a lot to do with how loud the ends of words sound. If you can ride the vocal so that there's still power at the end of a line rather than it tailing off, then that gives a perspective of being just loud enough. You need to ride it so you can hear every part of the line. That's what I'm used to doing — even when I'm recording — and that's what gives the apparent loudness to it.

It's a very fine line, though; if you go over that line, the backing tracks start to sound really small.

Yeah, it goes wimpy. It's a fine balance, but you know instinctively when you've got it. But it's a lot of work; there's an awful lot of riding you have to do. Compression can do only so much; at a certain point, it starts to be an unpleasant sound, so the rest of it is down to physically how you ride the vocal. It's also down to the EQ; in certain sections, if someone's belting a chorus, there will be an upper-mid frequency that is really masking, and it'll make the voice seem smaller. That's why I like to mult up the vocals so many times. You want a full-bodied sound, obviously, and there's no way you can do that if it's coming up one fader with one compressor.

As more energy goes into the vocal, if you still need to place it so it has lots of apparent loudness but isn't making everything else sound small, you might need to scoop out [EQ] with a very fine Q some area of the vocal that sounds a little offensive. You might scoop out a little bit of 1.6k or 2k or something like that, just so it's not quite so strident. It doesn't work that way for every song, but as someone belts harder, if you can just notch a little bit of that frequency out, it keeps the body in the voice; it'll keep the warmth of the voice still there.

"IT'S VERY RARE THAT ONE SOUND — OF ANY INSTRUMENT — WILL WORK THROUGH A WHOLE SONG."

The key to your mixing technique seems to be lots of multing.

Yeah, I do a lot of multing because, to get the apparent energy to change, sometimes you have to kind of fake it a little bit. I like to make allowances on the board so that there are at least eight channels free right next to the drums where I can mult stuff up, because in one section of a song, I might want to hear more energy out of a snare or a bass drum. It's very rare that one sound — of any instrument — will work through a whole song. Sometimes it's down to treatments, as well — what you use as outboard gear. For example, the snare drum can get swallowed up in the chorus if there are lots of power chords, so you'll need to add something that gives it a bit more width, so it keeps the size there. That doesn't mean it has to feel like you're pushing the level up, but the perspective keeps changing. You don't want to do it in a dramatic way — it can be quite subtle, but achieve the same results. So I find myself multing up a lot of instruments, actually.

It sounds like, wherever you go, you need a console with a lot of input channels.

Well, I usually work on a hundred-channel console, which is brilliant. [*Laughs.*] Some people would probably go, "That's a bunch of hooey; you could just get the sounds right and put the faders up," but, especially after working with Mutt all these years, I'm used to riding every nuance and creating a picture that is constantly changing. You can't just say, "Oh, just put a ton of compression on the snare for the chorus." You need to mix in a slightly different version of the same snare in the chorus to try and keep the energy building.

What's the silliest thing you've ever done in a mix that

actually made it on to the record?

There have been rare occasions where, through necessity, we've needed to EQ every syllable and every consonant of a lead vocal. That can be termed by some people as being silly, but in certain instances, that's what you do; it's a matter of trying to keep the energy of the vocal staying at a certain place. A lot of people don't understand that. A lot of people just think you can put the vocal up, do a bit of a filtering, compress the crap out of it, and move on to the next thing.

So your philosophy is that the mix should change constantly as the track is progressing?

That's the only way *I* can be satisfied by a mix at the end of the day.

Hey, I think we've stumbled on your secret formula....

Maybe. All I know is that you have to reach a certain point when I can actually sign off on a mix, and it can't be because of any kind of time constraint. It has to be because something happens inside you that goes, "This is a good place, where I feel comfortable enough that I've experimented with everything I can." I can't be going, "Oh, damn, the vocal in the chorus just isn't sitting right, but I've got to put this mix down." That just doesn't hold any water for me. There's no excuse for not being able to get what's in your mind out of the mix. I have to be able to walk out of the control room thinking I've done the absolute best that I can. If that means multing up something a bunch of different ways and changing the panning of it, or putting a flanger on for just this section or a different room thing on it for that section…I can't *not* go to the lengths of doing those things, even to satisfy my own curiosity about it. I just don't think I can mix any other way.

Have you ever been presented with a project that you just couldn't get a good mix out of? One that just wasn't ready to be mixed, perhaps, or was unsalvageable for some reason?

There are some that are harder than others. There are some that you'd like to take further than what you've been able to. The question is, does someone else hear that difference or not? That's the thing.

Have you ever walked away from a project and said "I can't do this justice," for whatever reason?

[*Pause.*] No, I don't think so. These days there's always a way around everything. There are some things I've done that I might listen to later and think, "Oh, I wish I'd done that differently," and sometimes that gives you a feeling in the pit of your stomach; some days you don't get it perfectly right. But there's never anything that's more than just a challenge to do the best that you can with what you've got. And you can't walk away. There have been situations way back where I didn't finish a couple of records off due to personality conflicts. But these days it's a total challenge. If it ever became rote, then I'd find something else to do.

I'm still thankful every day I can go to work. I mean, look at this! [*gestures at mixing console*] I remember getting the feeling when I first walked into a studio in London, where I knew instantly I didn't want to do anything else, ever. I still have that same feeling today. I just do the absolute best I can; there's no excuse for not doing that. You have to live with it yourself; it's as simple as that.

> ## "YOU CAN'T SET A COMPRESSOR FOR A VERSE AND EXPECT IT TO WORK FOR A CHORUS."

Howard Massey's latest book, *Behind The Glass*, is a collection of interviews with record producers, now available from Miller-Freeman Books.

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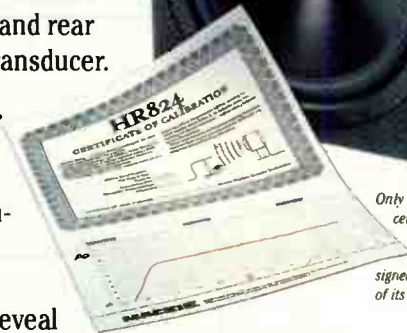
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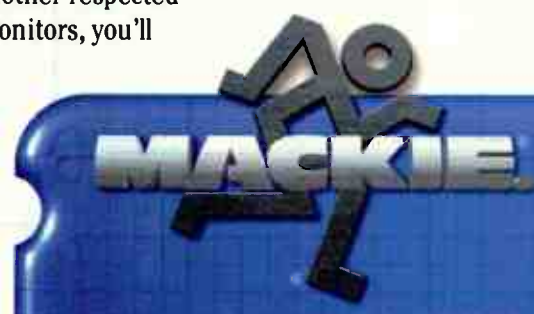
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API 2500 Stereo/Dual-mono Bus Compressor

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API (Automated Processes, Inc.) has long been known as a purveyor of fine analog audio products including large-format mixing consoles and several series of rackmount and modular mic preamps, EQs, and processors. Among the latest offerings from API's designers is the model 2500, a stereo/dual-mono compressor offering a number of unique features.

The Tour

The 2500 is a single rackspace unit. The unit runs cool, but since there's venting on the top and bottom of the enclosure, I wouldn't sandwich it tightly into a rack. Its back panel is clean and straightforward. In addition to the detachable AC power cable and voltage selector, the I/O complement includes balanced XLR's for each channel's input and output as well as 1/4-inch balanced sidechain inputs for each channel. API's 2503 transformers are used in the channel output stages.

The front-panel layout is similarly straightforward, although some controls may be unfamiliar to first-time users. Most of the rotary controls (except the manual makeup gain control and a "fine" control for dialing in release times) are stepped switches as opposed to continuous pots. This allows you to easily return to previous settings, but could be a problem if you really need a setting that falls between the steps. However, the step values are wisely chosen for usable results, and I didn't find myself looking for "in-between" settings.

The front panel is divided into five sections. On the left are the standard compression controls: threshold, attack, ratio, and release. Attack times vary from a fast 0.03 ms to 30 ms, while release times can vary from 50 ms to three seconds. Ratios can range from 1.5:1 to ∞:1 (limiting).

The next section, labeled "Tone," contains a switch for setting the compression knee. The 2500 is more flexible than other units: You can switch the "knee" (how fast compression becomes active after the input signal crosses the threshold) from soft to medium to hard, as opposed to the more usual soft and hard. I found the medium setting to be a nice addition for fine-tuning the compression response.

Also in the Tone section is a switch for setting the unit's proprietary "Thrust" circuit, which has long been a standard feature on A.T.I.'s Paragon series live sound consoles. Thrust is a highpass filter that can be thought of as a "punch" control. It can be switched from "Normal" (Thrust bypassed) to "Loud" (4 dB/octave added) to "Med" (2 dB/octave added). The nice thing about the Thrust control is that it can add significant presence and punch to the signal seemingly without adding significant gain to the output: *i.e.*, switching it to the Loud setting doesn't add 4 dB of gain to the overall output.

Last up in the Tone section is the Type switch, which determines the compression style used. The 2500 can operate in "Old" or "New" compression modes; Old is said to be similar to the type of "feed-back" compression found in the API 525, the Universal Audio 1176, and the Fairchild 600, while New is said to be more similar to dbx and SSL "feed-forward" compression. To my ears, Old was a gentler sound, while New was more aggressive and obvious.

The Tone controls interact to a large degree, and their effect will also depend on the source signal being processed. For example, when lightly compressing a stereo mix with soft knee and Old mode selected, the effect of the Thrust control was only barely audible. But when I ran a stereo recording of a finger picked steel-string acoustic guitar through the 2500 set for medium knee and Old mode, switching on Thrust made a dramatic difference in the low-mid region.

The next front-panel section, "Link," offers controls that, to my knowledge, are unique in a compressor. (API has a patent pending on this circuit.) In addition to normal stereo linking, the unit's channels can be set to operate independently (dual mono), or a link "percentage" can be specified (50, 60, 70, 80, 90, or 100%). You can also choose to insert a filter (high-, low-, or bandpass) into the channel coupling circuit. The idea is that you can really dial in the stereo linking response for the best possible stereo imaging.

The effect of linking the channels by percentage is easily visible when viewing gain reduction levels on the meters. When set to 60% linking, one meter tracks at about 60% of the level of the other, and so on. The effect of the link filter is also readily apparent in the meters; in lowpass mode with a full-range signal, the meters move together; in highpass mode, the two meters tend to be more independent, only having interaction when there are strong transients.

World Radio History

However, as much as these effects are visible on the meters, to my ears the audible effect of percentage stereo linking is very subtle. If you're going for that last iota of stereo-imaging control, you may find percentage linking useful, but for most of my applications, I ended up sticking with the standard 100% linked setting. Conceivably this feature could also be used creatively when feeding non-stereo (dual mono) signals through the 2500, and I did experiment with this aspect. The drawback here is that in dual-mono mode, the 2500 still uses one master set of knobs to control both channels. I generally had better results using the sidechain inputs rather than percentage linking in this type of application. And for applications such as de-essing and ducking, you'll certainly want to use the sidechains, which function as expected. It would be nice if the 2500 allowed you to monitor the signal feeding the sidechain inputs.

The next section of the front panel contains the bypass switching (the unit can pass signal while powered down) and makeup gain controls. You have the option of using the 2500's Auto makeup gain circuit, which works quite well for maintaining consistent output levels while you adjust the threshold and other controls, or you can choose to manually set the output level using the front-panel makeup gain control.

Wrapping things up on the right side of the front panel is the metering section. The two lighted mechanical (no LED ladders here) VU meters can be switched to display input, output, or gain reduction levels.

► **API 2500 COMPRESSOR**

MANUFACTURER: API a subsidiary of the A.T.I. Group, 9017-C Mendenhall Court, Columbia, MD 21045. Tel: 410-381-7879. Web: **www.apiaudio.com**. 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: A great-sounding stereo/dual-mono compressor offering several unique features.

STRENGTHS: Flexible. Three knee settings. Thrust tone circuit. Selectable feed-forward or feed-back compression. Auto makeup gain. Variable stereo linking.

WEAKNESSES: Can't monitor sidechain inputs.

PRICE: $2,649

EQ FREE LIT. #: 115

Conclusions

The 2500 packs a ton of dynamics processing power into a single rackspace. The unit can be made to be very aggressive sounding, or can be used very subtly. Its output is clean and noise-free, but adds a pleasant round warmth to the sound. I was extremely pleased with the results it provided, whether I was processing tracks such as vocals, electric bass, flugelhorn, acoustic guitar, or keyboards or complete mixes.

But in addition to dynamics control, the 2500 offers a great deal of flexible sound-shaping control. The Thrust circuit, in particular, can be incredible for punching up anemic mixes and tracks, and being able to choose between feed-forward and feed-back compression styles provides even broader tonal possibilities.

At $2,649, the 2500 isn't cheap, but it easily offers performance and capability to match its price tag, and unique features such as Thrust, flexible stereo linking, switchable compression type, and soft, medium, and hard knee operation differentiate it from the competition. Check this one out, you won't regret it. ∎

Native Instruments Pro-52 Software Synthesizer

Now you can own your own (virtual) Prophet-5

Back in the '80s, I lusted after a Sequential Circuits Prophet-5. After years of playing monophonic synths and writing down patch settings, the concept of a synthesizer with an astonishing *five voices* of polyphony, and the means to remember patches in memory, definitely rocked my world. Although I've played a ton of Prophet-5's in my time, I never did gather enough disposable income to actually own one.

Well, at least I now have something that sounds like one — but it never goes out of tune, its chips don't fail, there's no hiss, and I can actually afford it! Welcome to the wild and wonderful world of virtual instruments, where models of vintage synths come alive on your computer screen. If you've been wanting to add that vintage vibe to your latest recordings, this is one of the better options.

Getting Started
You can install Pro-52 as a stand-alone app, a VST 2.0 compatible plug-in, or both (choose the "typical" installation option, as opposed to "custom"). I'm pretty familiar with installing VST instruments, so when it didn't show up in the Cubase 5.01 VST Instruments panel, I wasn't too concerned. I found the VST plug-ins folder used by Cubase, copied the appropriate bits over to it, and all was well.

As with the Native Instruments B4 (a Hammond B3 software synth reviewed November, '00), you need to insert the CD periodically to verify ownership. I used to like this method of copy protection until I started taking a laptop on the road capable of doing audio; now I have to carry a bunch of CDs for the programs that require ownership verification. Oh well.

Anyway, I opened Cubase, called up the VST instruments panel, selected the Pro-52, hit edit so I could see the controls (see fig. 1), turned it on, clicked on a key with the mouse, and — wow! There was no mistaking the filter sound and note attack: Native Instruments had indeed nailed the Prophet's distinguishing characteristics. I was hooked.

Drivers, Sound Cards, And Latency
In stand-alone mode, I tested MME drivers and

▶ NATIVE INSTRUMENTS PRO-52

MANUFACTURER: Native Instruments USA, 6477 Allmaden Expy., Suite D2-F8, San Jose, CA 95120. Web: **www.native-instruments.com**. Distributed by Steinberg, 21354 Nordhoff Suite 110, Chatsworth, CA 91311. Tel: 818-678-5100. Web: **www.us.steinberg.net**.

SUMMARY: The Pro-52 adds several enhancements to its predecessor, the Pro-5, while remaining faithful to the vintage synth it models.

STRENGTHS: Sounds like the real thing. Excellent MIDI control. Low latency with the right sound card and drivers. Delay section is a useful addition. Accepts original Prophet-5 sys ex. Useful presets. Useable as VST insert effect.

WEAKNESSES: No numerical equivalents or calibrations for controls. No typing in of values.

PRICE: $199

EQ FREE LIT. #: 116

ASIO with a Frontier Design Dakota card. Surprisingly, latency was 20 ms with the MME drivers — I expected far worse. ASIO worked fine, and while I couldn't measure the latency, it seemed better than what I was getting with MME. In either case, latency certainly was not an impediment to musical enjoyment. When running under Cubase with ASIO drivers, the latency was 12 ms — fast enough to be plenty responsive, and musically satisfying.

Polyphony
Polyphony is a function of processor speed; a "voice" parameter, storable with each preset, places an upper limit on voices, from five to 32. Pro-52 could generate 32 voices without the computer breaking a sweat, but the more voices, the greater the VST performance hit. At maximum polyphony, the Pro-52 demanded close to half of the available processor power. A more reasonable 10 voices used about 15%. Any host-based digital audio system requires making these kinds of tradeoffs, but remember that you can always bounce the output to an audio track, thus freeing up resources.

If you instead want to use up resources, check out unison mode. You can stack as many voices as

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- 20 bit/96 kHz analogue I/O.
- ADAT 8 channel lightpipe I/O.
- ASIO drivers.
- Windows and Mac compatible.

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- Windows compatible.

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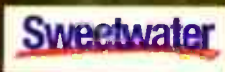
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you've specified on one key — talk about phat. Also talk about using up processor power! But it sure is fun.

Compatibility

Unlike the B4, Pro-52 works with no plug-in format other than VST. However, it is compatible with the fxpansion VST-to-DirectX adapter, and there is a driver included that lets it show up as a MIDI port for use with various MIDI sequencers.

You can also use the Pro-52 as a post-oscillator VST insert effect. This lets you take advantage of the filter and onboard effects, and adds a cool "freebie" signal processor to your setup. (According to the ReadMe file included with the CD, this feature is not available with Logic Audio, which also has some issues with MIDI controller assignment to the Pro-52 controls, and cannot accept program changes or tempo information while running under VST. These limitations will be addressed in an upcoming Logic update.)

Programs

The Pro-52 reproduces the Prophet-5's method of program numbering and selection. I thought the days of octal numbering were behind us, but no — you can have eight files, each with eight banks, and each bank can have eight programs, for a total of 512 presets. But the highest numbered preset is 888, because it's numbered in octal (512 decimal = 888 octal).

So why didn't they change this? I presume it's because the Pro-52 is compatible with sys ex data from the original Prophet-5. By keeping the octal numbering, your patches will show up where you expect them to. (One VST quirk: you can't load sys ex while the instrument is running in VST mode. You need to load any Prophet-5 data in stand-alone mode, save the programs, then load the programs when you switch over to VST mode.)

The User Interface

The keyboard can be retracted if you're tight for screen space, although that means you can't watch the mod and pitch bend wheels rotate as you tweak them on your main controller. The knobs can either move in response to linear motion (move mouse up to increase, down to decrease), or be "turned." You can also change those modes momentarily if desired, and obtain a finer degree of control by holding down the shift key as you move the mouse. Unfortunately, there's no numerical representation of the control position, so you can't type in values

SYSTEM REQUIREMENTS

Mac: 120 MHz PowerPC, 32 MB RAM, Sound Manager-compatible audio interface, OMS-compatible MIDI interface, 60 MB free hard disk space, Mac OS 7.6.1 or higher, Opcode OMS. **Windows:** 133 MHz Pentium, 32 MB RAM, 60 MB free hard disk space, Windows 9x/2000/NT 4.0 OS, OS-compatible soundcard, MIDI interface.

Test Conditions: Version 2.0 was tested in both stand-alone mode and as a VST plug-in within Cubase VST/32 v5.01. The computer, by Q Performance Systems, was running Windows 98SE on an overclocked Celeron processor (850 MHz) with 256 MB of RAM.

if you want (for example) a specific filter frequency.

Details

There are some major changes compared to the original Prophet-5: the front panel is laid out differently to accommodate the addition of the delay section, which features four delay lines. There are 512 programs instead of the 120 found in later revs, and velocity response. But there's also the question of which version is being emulated, as the rev 3 Prophets used different chips compared to the first 2 revs. (It seems like the bass sounds more like the earlier revs, but the high end is brighter, and sounds more like the later revs. However, as I didn't have the two versions available for comparison, I can't confirm whether my auditory memory is functioning correctly or not.) Incidentally, if you're an analog diehard, there's a control to introduce variable amounts of the kind of tuning anomalies caused by component drift in an analog system.

MIDI is pretty straightforward; you can load multiple instances (up to 8) if you want multi-timbral operation, and almost all front panel controls can be controlled by MIDI continuous controllers. This is great for those with hardware fader boxes. Furthermore, the on-screen controls also send MIDI controller data.

The manual is well-done, and even includes a short guide on analog synth programming and a useful tutorial.

So what's there not to like? Well...hmmm...let's see...uhh...okay, I give up. VST instruments are a gas, and Native Instruments has the recipe down. Like the B4, this baby deserves a major thumbs up. ∎

Shure KSM44 Condenser Microphone

Shure s successful KSM32 gains a sibling with the large-diaphragm three-pattern KSM44

The first microphone I ever bought was a Shure SM58. The guy at my local music store told me that Shure microphones were the industry standard, and that I wouldn't be disappointed its sound. He was right — 30 years later, I still use SM58s and I'm rarely disappointed. But Shure does much more than simply make a variety of dynamic microphones — numerous condenser mics, such as the SM81, a small-diaphragm electret condenser, have been in production for many years. More recently, the company developed the KSM32, a condenser with a medium-sized diaphragm. Building on the success of the KSM32, Shure has introduced the KSM44, a large-diaphragm, multi-pattern condenser mic.

The KSM44 has three patterns: cardioid, omni, and bi-directional (or, if you prefer, Figure-8). The microphone also features a three-position highpass filter and a 15 dB pad The dual capsules of the KSM44 are 2.5-micron gold-layered Mylar, and the electronics comprise a Class A transformerless design. The KSM44 looks very similar to the KSM32, and weighs a little over a pound. At 7 dBA (10 dBA in omni or bi-directional patterns), the self-noise of the KSM44 is very low. The microphone has a maximum dynamic range in excess of 120 dB, and with its 15-dB pad engaged, can handle maximum SPL in the 150-dB range. The highpass filter has two active settings (as well as off); one is a low-frequency cutoff, an 18 dB per octave filter at 80 Hz, and the other is a low frequency roll off — a 6 dB per octave filter at 115 Hz. The steeper 80 Hz filter is designed to help reduce ambient low-frequency noise (stage rumble or HVAC generated room noise, for example), while the milder (but higher frequency) roll-off switch can help compensate for the proximity effect or be used for tone shaping. The KSM44 requires 48-volt phantom power and typically draws 5.4 mA. The output impedance of the microphone is 150 Ohms. The KSM44 comes packaged in a locking aluminum hardshell case, which also holds the included swivelmount and a very nice elastic "spider"-style shockmount.

As it does with all multi-pattern mics, the frequency response of the KSM44 changes with the pattern. In cardioid mode, the graph shows a peak of approximately 3 dB at 6 kHz and another smaller peak around 12 kHz. If used

▶ SHURE KSM44

MANUFACTURER: Shure. Inc., 222 Hartrey Avenue Evanston, IL 60202-3696. Tel: 847-866-2200. Web: **www.shure.com**.
SUMMARY: A natural-sounding large-diaphragm mic offering three polar patterns.
STRENGTHS: Versatile. Natural sound quality. Low self-noise. Smooth off-axis response.
WEAKNESSES: None to speak of.
PRICE: $1,340
EQ FREE LIT. #: 117

in the close field (approximately six inches from the diaphragm), there is a broad peak centered around 50 Hz due to the proximity effect, though in the far field (two feet or more) the low-end response is flat down to around 30 Hz. In the omni pattern, the frequency response is fairly flat except for an approximately 4-dB peak, again at 12 kHz. In the Figure-8 pattern, the graph shows a broad rise from about 2,500 Hz to a 4-dB peak at 7 kHz, where it falls abruptly to flat at about 9 kHz and then continues down to –5 dB at 20 kHz. The proximity effect in the Figure-8 pattern is more pronounced than in either cardioid or bi-directional patterns — up to 10 dB boost at 50 Hz. The polar response patterns are pretty much what you'd expect them to be — the higher the frequency, the more directional the response in every pattern. However, there doesn't appear to be any anomalous bumps in the polar response of the KSM44, which helps to keep the off-axis response fairly predictable.

The Sound

The first question asked with any microphone is "How does it sound?" In the case of the KSM44, the answer is, "Very natural and surprisingly good." I was a little surprised because the KSM44 sounded great in a couple of applications where I would normally not use a large-diaphragm mic. For the last four or five years, I've preferred to use small-diaphragm condenser microphones for most applications other than as a main vocal mic — for acoustic instruments, drum overheads, percussion, and for ambient miking. (Some readers may not be aware that the size of the diaphragm has very little bearing on the low-frequency response of a

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microphone; high-end measurement microphones with a flat response to below 5 Hz can have diaphragms as small as a quarter- or half-inch in diameter. Size isn't everything, though large-diaphragm condenser microphones are most commonly used for recording vocals and are therefore deemed to be "sexier" than small diaphragm mics.)

For my first test, I set up my typical two-microphone acoustic guitar placement: One mic pointing up toward the 12th fret and the other above the body of the guitar and pointing down toward the body. Each mic was placed 8-10 inches from the instrument, and set for cardioid. In this application, the KSM44's recorded the best and most natural-sounding acoustic guitar tracks I've gotten in recent memory. And when I tried these mics as drum overheads, I found the same thing. The overheads sounded like the drums did in the room — which can actually be a little disconcerting until you get used to it. I honestly didn't realize how good the Shures were in this application until the next session, when out of habit I swapped back to my usual overhead mics while miking up the drum kit.

The proximity effect (in the cardioid pattern) is present but not overbearing — it also sounds (here's that word again) natural, and the milder of the two available highpass filters allowed me a second opportunity to control the proximity effect if the application required. On kick drum (placing the KSM44 about a foot in front of the drum in a tunnel made of packing blankets — I couldn't bring myself to put someone else's condenser mic inside the kick) the low-end response of the mic was very nice; deep, but not muddy. And on an acoustic bass, there was enough air in the tone (with the mic placed about 18 inches away from and a little bit left of the bridge) to add a nice clarity to the sound of the bass.

The fairly smooth off-axis response of the KSM44 means that reflected sound (especially in confined areas such as iso booths) doesn't add a great amount of coloration to the sound of the room. I found that the KSM44's omni and bi-directional patterns are more coherent (and usable) than most microphones in the this price range.

A natural sound has its place, of course, but there are applications where it may not be the best choice. For example, to my ears the KSM44 was a bit too bright when placed above a fiddle, though it worked fine when placed a couple of feet in front of the player so that the mic wasn't picking up as much direct radiation from the top of the instrument. And if you wish to emphasize certain characteristics of an instrument — the

midrange honk of a dobro or the percussive attack of the pick on a mandolin, then the best choice may be to use a mic that emphasizes those things. That's not to say that the KSM44 sounded bad in these applications — it didn't sound "bad" anywhere I tried it. But when you're trying to emphasize a particular character associated with an instrument rather than the overall tone, an accurate and neutral microphone like the KSM44 can make it a rather difficult task to accomplish.

As a vocal mic, I had mixed success with the KSM44. It always worked well with group vocals, where the microphone was a few feet away from any singer, and it also sounded great on about 40% of the lead vocals I recorded with it. However, making this sort of judgment is definitely subjective, and is influenced by what mic you normally use in these applications. For example, if you're used to the sparkling high end of an AKG C12 or the warmth and low end of a Neumann U 47, then any mic without those qualities will seem to be lacking something. In addition to this subjective aspect to mic selection, the sound of the singer's voice will make a big difference. A singer with a thin voice would likely sound better on a microphone that adds warmth and fatness, while a singer without much in the way of upper frequencies may need a brighter mic to pull out some of the tones that are missing at the source. That same natural quality that I've mentioned above shows up when recording lead vocals; if the vocalist sounds great without any added hype from the microphone, they sound great when using the KSM44. Further on the plus side, those that didn't sound "great" with the KSM44 never sounded bad, they simply didn't sound bigger than life — whether it's the right mic to use in a particular application just depends on the sound you're going after.

Conclusions

In my opinion, the best thing about the Shure KSM44 is not that it sounds great on acoustic guitars and as overheads, but that it sounds pretty darned good on just about everything you put it on. While a well-equipped commercial facility may have a microphone locker equipped with the "perfect mic" for each and every application, most of us don't have that luxury. Instead, we must choose tools that provide the best results for the widest range of applications. With a list price of $1,340, the KSM44 isn't the cheapest large-diaphragm mic on the market today, but because it works so well in so many applications, it certainly is one of the best values. ∎

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Thanks.

Bill Thompson

President, Ashly Audio, Inc.

(and just for the record, we think the Redwood trees in the picture are the perfect metaphor for Ashly...)

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TC Electronic Triple•C Multiband compressor/limiter

TC's latest offering goes beyond standard compression into the multi-band zone

Multi-band dynamics processing has really taken off lately, and for good reason. Being able to separately process different frequency ranges in a signal allows for more effective control, with more natural sounding results. In many cases, multi-band compression offers tone control results that are superior to those provided by an equalizer — the multi-band compressor can react to changes in the level of various frequencies, and help to maintain a more consistent overall timbre.

And for more standard dynamics control applications, multi-band compressors are a godsend, especially with today's bass-heavy styles and high-impact mixes. A full-range compressor can have difficulty dealing with these kinds of signals — the heavy bass can compromise the compression results, for example. But with a multiband compressor, the low-frequency region can have its own compression, leaving the other bands available to independently process the remainder of the signal.

TC Electronic has been working on multi-band processors for some time; their ubiquitous Finalizer series, for example, offered some of the first hardware-based multi-band dynamics control. Now the company has released a new, three-band compressor, the Triple·C. But the Triple•C goes beyond just three-band compression, offering a number of useful extras. Both single and dual-mono/stereo versions of the Triple•C are available. For this review, I checked out the single-channel version

Ins and Outs

To start with, a huge feature of the Triple•C is the quality of its A/D and D/A converters in combination with its S/PDIF digital I/O. The Triple•C offers 24-bit converters with dithering to 20-, 16-, and 8-bit resolutions. I found its I/O was superior to the standard converters on many other digital devices aimed at the home recording and project studio markets. Among other examples, I compared the A/D converters of the Triple•C to those of the Akai DPS16 at 24-bit resolution and found the Triple•C to be better in all tests, particularly in the upper frequencies. When I repeated the tests at 16-bit resolution the difference favoring the Triple•C was even more pronounced.

For even more flexibility the Triple•C offers a sidechain input that allows you to trigger compression from external sources or to connect an external equalizer for frequency-dependent

▶ TC ELECTRONIC TRIPLE • C

MANUFACTURER: TC Electronic, Inc., 742-A Hampshire Road, Westlake Village, CA 91361. Tel: 805-373-1828. Web: **www.tcelectronic.com**.

SUMMARY: The Triple•C is a dynamics processor that can function as a multi-band compressor with definable high-, mid-, and low-frequency bands or as a traditional single-band compressor. It is suited for both tracking and mixing applications.

STRENGTHS: Excellent audio fidelity. Outstanding A/D and D/A converters. Superb metering. S/PDIF digital I/O. Very flexible. Easy to use. Externally controllable via MIDI. Usable factory presets. Ability to store user presets. DRG function simulates tube warmth. Way-cool Envelope Mode.

WEAKNESSES: Poor user's manual. Unable to set separate ratio, attack release, and threshold settings for the three frequency bands.

PRICE: Single channel, $699; stereo/dual-mono, $999

EQ FREE LIT. #: 118

compression for applications such as de-essing.

The Triple•C has separate LED-style meters in its LCD window for input and output levels, as well as for gain reduction levels for each of the three frequency bands. The input and output meters also have easily viewable LED clip indicators. I prefer being able to see the input, output, and gain reduction levels simultaneously and many compressors don't offer that feature; I was quite happy to have that capability in the Triple·C. Overall I felt the metering on the Triple•C was very good.

The MIDI implementation of the Triple•C further expands its flexibility. The unit has separate MIDI in, out, and thru connectors, which can be used for storing and recalling presets as well as for program (preset) and control changes.

User Interface

I really liked the user interface employed by the Triple·C. It takes a hybrid approach combining

old-school dedicated buttons/knobs with more modern soft buttons and knobs, and an LCD-based menu editing system. The result is an easy to use product that's flexible, powerful, and highly editable. I found the 50 factory presets and 100 additional user presets to be very useful. The user presets can be stored and reloaded via bulk MIDI dump. The factory presets are aimed at common applications and in many cases worked very well with only minimal tweaking. Primary parameters such as threshold, ratio, attack, release, lo-band spectral level, hi-band spectral level, makeup gain, and input level are adjusted using dedicated front-panel knobs, and it's easy to see the effects of what you're doing with aforementioned meters. Other, less-frequently used parameters are accessed using front-panel soft buttons or by stepping through menus on the LCD window.

A parameter wheel and a great display simplifies the editing process while maintaining the high degree of tweakability. The biggest disappointment of the Triple•C was the user's manual. It would have been helpful to have at least some documentation about the 50 factory presets, and in one case, a feature in the edit menu called "Comp Style," seemed to be missing completely from the 30-page manual. It also would be nice to have a little more information about Envelope Mode (see below), which is a somewhat unusual feature.

Compression

There are two basic compression modes on the Triple·C: full-range and multi-band. Full-range mode works like a standard compressor with the compression being triggered and applied uniformly across the entire frequency spectrum. Multi-band mode divides the signal into three user-definable frequency bands to which compression can be applied independently. The user-definable high and low crossover points can be chosen from 31 different frequencies between 20 Hz and 20 kHz, allowing significant flexibility. Multi-band mode also affords control over the output level of both the high- and low-frequency bands via front-panel knobs; this lets the Triple•C serve as a sort of dynamic broadband equalizer — a very powerful application.

The Triple•C is normally an RMS-based compressor that responds to average level. There's a front-panel switch in multiband mode that allows the Triple•C to become a peak-based compressor, which is often better for percussive material. Another

TRIPLE • C SPECS

DIGITAL I/O
Connectors ....RCA (coaxial)
Formats ....S/PDIF (24-bit)
Output Dither..........................HPF/TPDF dither 24-/20-/16- or 8-bit
Sample Rates ....44.1 and 48 kHz
Processing Delay ....0.1 ms @ 48 kHz (excluding optional 3 ms look-ahead delay)

ANALOG INPUTS
Connectors ....1/4-inch TRS balanced
Impedance. Bal/Unbal ....24 kohm/13 kohm
Max. Input level ....+24 dBu
A/D Conversion ....24-bit, 128x oversampling
Dynamic Range ....100 dB typical, 20 Hz-20 kHz
THD ....< 92 dB typically (0.0025%) @ 1 kHz
Frequency Response ....+0, −0.1 dB @ 48 kHz, 20 Hz-20 kHz
Crosstalk ....< −95 dB, 20 Hz-20 kHz

ANALOG OUTPUTS
Connectors ....1/4-inch TRS balanced
Impedance. Bal/Unbal ....40 ohms
Max. Output level..+20 dBu (balanced)
D/A Conversion ....24-bit, 128x oversampling
Dynamic Range ....104 dB typical, 20 Hz-20 kHz
THD..< −94 dB (0.002%) @ 1 kHz typical
Frequency Response ....+0, −0.5 dB @ 48 kHz, 20 Hz-20 kHz
Crosstalk ....< −95 dB, 20 Hz-20 kHz

CONTROL INTERFACE
MIDI..In/Out/Thru
Pedal ....1/4-inch TS phone jack

button on the front panel of the Triple•C enables a feature called SoftLim, which activates a limiter that triggers a few dB before clipping. SoftLim is subtle in most applications but will often inaudibly prevent nasty clipping on transients.

A feature called "Look Ahead" is available via a front-panel switch in multi-band mode and enables a three-millisecond delay. This brief delay allows the processor to analyze incoming material and process it more accurately. Compression can be triggered on the Triple•C by the incoming signal, by an external sidechain input, or by the sum of the incoming signal and the sidechain input. The Triple•C offers a direct out in conjunction with the sidechain input to allow the use of an external equalizer to emphasize key frequencies that you want the compressor to respond to. This allows for applications such as de-essing, etc.

The Triple•C has a user-adjustable parameter called DRG, which stands for Digital Radiance Generator. DRG adds second harmonic distortion to simulate tubes and add warmth to the signal. While I didn't feel

that it made the Triple•C sound like a vintage tube compressor, DRG did add a nice coloration to the signal in some cases.

Envelope Mode
Another nice extra in the Triple·C's bag of tricks is Envelope Mode — a feature I don't recall seeing in a compressor before. Envelope Mode uses the compressor to shape the envelope of an incoming sound by changing the level of the signal at the attack and release points. When in Envelope Mode, the Lo-Band and Hi-Band rotary knobs take on the function of Envelope Attack Gain and Envelope Release Gain respectively. This works as follows: When a signal crosses the user-defined threshold, it will be boosted or attenuated to the level defined by the Envelope Attack Gain in 0.1 ms. Following this, the Attack knob defines the time it takes for the signal to return to the threshold level. After the signal drops below the threshold, the Release knob defines the amount of time that the signal decays before being boosted by

Aardvark Direct Pro 24/96

Audio interface for Windows computers

The Aardvark Direct Pro 24/96 is a PCI audio card with breakout box plus custom control software that emulates a standard mixer. Minimum requirements are a Windows 95/98 PC with a 133 MHz or better CPU and at least 32 MB RAM; we're told that Windows 2000 and Macintosh drivers will be forthcoming soon. The breakout box offers four balanced analog inputs (thoughtfully provided on combo jacks that accept both XLR and 1/4-inch TRS or TS connectors) and six analog outputs (four balanced, on 1/4-inch connectors, and two unbalanced, on RCA connectors). In addition, there's a headphone jack and a MIDI input and output, allowing Direct Pro to act as your computer's MIDI interface as well. A phantom power switch applies +48V to all XLR inputs simultaneously (line inputs must be connected with 1/4-inch plugs), allowing you to use condenser as well as dynamic mics. S/PDIF in and out (on standard RCA jacks) are provided on the card itself. However, when using the S/PDIF inputs, only two analog inputs are available. For those of you who need expanded systems, up to four cards can be installed simultaneously in a single PC, though these have to be daisy-chained externally, connecting the S/PDIF out of one card to the S/PDIF in of the next one. It's too bad that Aardvark didn't at least provide connectors on the cards for use with internal ribbon jumper cabling. This would have been a much more elegant solution, and one that would have allowed the use of more than a single pair of S/PDIF I/O in a multiple card system.

Speaking of the PCI card, one of its most notable features is its heavy-duty shielding; in fact, all the circuitry is encased in what appears to be an epoxy covering. The shielding works — Direct Pro is far and away one of the quietest sound cards I've ever popped into my PC (and believe me, I've popped a lot of them).

The breakout box not only houses the A/D and D/A converters (which are 24-bit and capable of supporting sample rates as high as 96 kHz), but also four discrete mic preamps (not the inexpensive "preamp-on-a-chip" design used by many sound cards). The converters all seem to be of excellent quality. The preamps seem to be about equivalent to other preamps in this price range. They're fine — quite good, actually — but I didn't feel they had the definition of most of the outboard

Multichannel 24 Bit Digital PCI Bus Architecture Interface

▶ AARDVARK DIRECT PRO

MANUFACTURER: Aardvark, 202 East Washington #306, Ann Arbor, MI 48104. Tel: 734-665-8899. Web: **www.aardvark-pro.com**.

APPLICATION: Audio I/O and mixing inside your computer.

SUMMARY: An excellent, cost-effective way to interface your PC with your audio gear and at the same time enter the brave new world of 24/96 recording

STRENGTHS: Zero-latency monitoring with reverb, EQ, and compression. Effects can be recorded if desired. 24-bit/96 kHz converters. Works with all standard Windows software. Balanced analog and S/PDIF I/O. Extensive shielding. Unusually large amount of preamp audio gain (pre-A/D converter).

WEAKNESSES: No dedicated headphone volume control. Limited patchbay routings. 88.2 kHz sample rate not supported. Interconnection between multiple cards precludes use of more than a single pair of digital I/O. No gain reduction meters. No MIDI control of mixer software. Preset saving counter-intuitive. No factory presets provided.

PRICE: $699   **EQ FREE LIT#:** 119

(and more expensive) preamps in my arsenal.

The Direct Pro 24/96 Control Panel is the included mixer software. (See fig. 1) Designed to be used in conjunction with the Windows program of your choice (a copy of SEK'D Samplitude Project software is included in the bundle), it doesn't allow for either recording or playback of audio files. Instead, it emulates a hardware mixer, providing onscreen faders to control the levels of each of the ten available inputs: the four breakout box analog inputs (or the stereo S/PDIF input plus analog inputs 1 and 2) and three stereo WAV files (alas, there's no provision for independent control of six mono WAV files, since the fader pairs are ganged together). There's also a pair of ganged faders for the monitor output. Each of the four "live" input channels is accompanied by an input trim with, unusually, three independent gain stages. If you're using a line-level input, it can be set to provide up to 9 dB of boost or −14 dB of attenuation (allowing you to interface with both −10 and +4 equipment). If you're using a mic-level input, there's an unusually large range of gain available here, all the way up to +75 dB. More significantly, all this gain occurs in the analog domain, prior to the A/D converter,

thus facilitating optimum signal-to-noise ratio with maximum resolution.

Each of the four "live" input channels also offers a three-band equalizer and compressor. The high and low bands are fixed at 8 kHz and 220 Hz, respectively, while the mis-named "mid" band can actually be swept between 50 Hz and 15 kHz; each band can be boosted or cut by up to 12 dB. Each compressor offers threshold, ratio, attack, and release parameters; ratios can vary from 1:1 to 11:1, so it can't be used for limiting. Attack times range from 1 to 100 ms and release times range from 20 to 2,000 ms; unfortunately, there's no gain reduction metering. There are dedicated Bypass and Preset switches next to each equalizer and compressor, allowing you to A/B the signal and to save your own settings (no factory presets are provided). I do have a minor niggle about the way presets are saved. The dialog box has an "Add" button, but if you press it before typing in a name, a blank preset (with no name) is saved. Conversely, typing in a name and then pressing the Enter key (the standard way of saving things in Windows dialogs) doesn't save the preset — you need to type in the name first and then use your mouse to click on the Add button. At best, this is counterintuitive and clumsy; in a busy mix session, it's all too easy to lose your settings.

There's also a master reverb section, with three room sizes (Room, Hall, and Church) and three adjustments (Decay, Diffusion, and Brightness). Interestingly, while there are independent channel reverb send levels, there is no master reverb return control (presumably, it's always on, at unity gain). Reverb settings can also be stored as presets, independently of EQ and compressor presets.

Truth be told, none of the three effects is sonically all that great; chances are that you'll be able to generate better sounding effects from your software plug-ins. But what is exceptional is the fact that all three effects are created by the card's dedicated DSP chip instead of by the host computer's CPU, which means that there is no delay whatsoever — "zero latency," in the parlance — when using them. Essentially, Direct Pro has been designed to serve as a sophisticated monitoring system, allowing you to overdub to prerecorded tracks while adding selective EQ, compression, and/or reverb in your headphones, the same way an external mixing board with a dedicated foldback works.

Of course, if you want to, you can record these effects to the tracks you're laying down. Adding EQ and/or compression to live tracks is as simple as clicking on the "Rec FX" button beside each input fader. Recording reverb is a little less straightforward; you need to designate the virtual input pair 5/6 as your
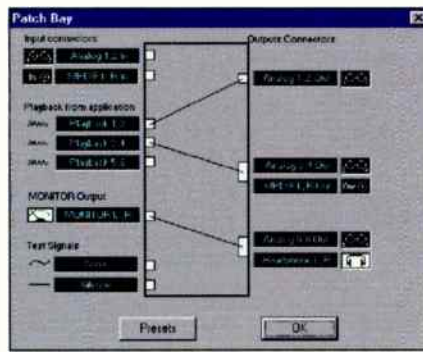
FIGURE 1

FIGURE 2

recording source in the host software. Adding the Direct Pro effects to a prerecorded WAV file, however, requires manually routing the track to a physical output and then using a cable to plug it into an input; this, of course, means that the signal needs to undergo two extra conversions (digital to analog, then back to digital again), which usually isn't a good thing to do. It's hard to view this as a limitation, however, since Direct Pro is really not designed to be used that way.

Each channel has a bar level meter (with optional peak hold) and mute and solo buttons, plus horizontal pan sliders for each of the four input channels and a pair of VU-style "master" monitor meters. The latter do exhibit some latency even though, as we've noted, there's no latency in the audio signal itself; no doubt this is due to the fact that the screen redrawing is left up to the host CPU and is clearly — and properly — low priority. There are also stereo link buttons for channels 1 and 3 (marrying them to channels 2 and 4, respectively) and handy Signal Present "LEDs" are available when the S/PDIF inputs are assigned to channels 3 and 4. Last but not least, a master Preset button allows you to save entire "snapshots" of all console fader, knob, and switch settings. There's also an "Advanced" dialog that allows you to set the PCI bus efficiency and configure the ASIO drivers so as to optimize performance in your particular computer. It's worth noting that none of the Control Panel

functions can be controlled remotely via MIDI; Direct Pro's MIDI I/O is strictly for porting data to and from the host software.

The Control Panel also provides a software patchbay, where all signal routings are set. (See fig. 2) Simple drag-and-drop operations allow you to connect inputs to outputs, and there's even a Tone generator, handy for troubleshooting (a Silence generator is used for disconnecting routings). Any input can be routed to up to three different outputs, but analog outputs 3 and 4 are "tied" to the S/PDIF outputs and analog outputs 5 and 6 (the RCA jacks) are tied to the headphone outputs, meaning that they can't be used independently. These limitations are slightly frustrating. It's true that, most of the time, you'll want to route the "monitor" (i.e., master) output of the control panel to the headphone jack, but I could see situations where you'd want to send a different signal to outputs 5 and 6.

Installation of the Direct Pro hardware and software was a snap in my Pentium II 450 system. I encountered no crashes, problems, or bugs using the Control Panel with various different host software packages, including the bundled SEK'D Samplitude Project, as well as Cakewalk Pro Audio, Sonic Foundry Sound Forge, and Steinberg WaveLab and Cubase VST. (The owner's manual provides "quick-start" setup guides for all of these products.) To operate correctly, the Control Panel must be manually set to the same sample rate being used by the software; options include Internal 32, 44.1, 48, or 96 kHz, or external S/PDIF. The latter must be used when bringing in signal via the S/PDIF inputs, which makes it somewhat surprising that this option isn't automatically selected when assigning the digital inputs to channels 3 and 4. When using the double-speed sample rate of 96 kHz (why no 88.2 kHz support?), the DSP functions of Direct Pro are disabled and no effects are available. Not every software package supports it, but if you haven't yet done any 96k recording, you may be pleasantly surprised at the extra "air" that it adds to the sound. The A/D converters in the breakout box are of sufficiently high quality to allow you to hear the difference. If you're doing work destined for release on DVD-Audio, you may want to begin entering this new territory. However, don't forget that there's a twin-whammy tradeoff of losing the onboard effects and generating much larger file sizes.

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World Radio History

the amount defined by the Envelope Release Gain knob. Envelope Mode can be used to increase apparent sustain in a signal, add dynamic contrast, or for creative/expressive control. The Envelope mode can be triggered in three different ways: 1) using the incoming signal; 2) using the external sidechain input; or 3) off the sum of the incoming signal and the external sidechain input.

The values of the Envelope Attack Gain and Envelope Release Gain can be controlled from the front panel or via MIDI continuous controllers. The real creative control comes from tweaking the knobs, riding a control pedal, or automating continuous controllers from a sequencer while processing a signal.

In Use

I used the Triple•C for a variety of applications including miked sources such as acoustic guitars, electric guitars, vocals, snare drum, bass drum, and miscellaneous percussion instruments. I also evaluated the Triple•C on bass guitar via a direct box and on a variety of sounds from my Kurzweil K2500S keyboard. I quickly discovered that the Triple•C offers a lot more than simple dynamics control. This doesn't mean the Triple•C doesn't work very well as a standard compressor — it does — but when used in multi-band mode it becomes an even more powerful dynamics control and timbre-shaping tool.

On kick drum, with the Triple•C in multi-band mode, I was able to use the low band to shape the "body" of the drum sound and the high band to control the amount of "click" when the beater hit the drum. In another session, I tracked some vocal takes using a Rode Classic tube microphone, an Oram MWS preamp/EQ, and the Triple·C. I found this to be an ideal tracking setup as I was able to tweak for the best timbral and dynamic content. In this case, I used the Oram EQ to tweak specific parts of the vocal timbre while using the Triple•C for broader shaping and dynamics control. I was able to get significant compression on the midrange of the vocal without losing the bottom end or pinching the top end. I really liked the Soft-Lim feature, as I didn't find any instances where it was inappropriate to use and many where it saved a take.

I played around with Envelope Mode and a controller pedal on all sorts of signals. The results ranged from making a track more realistic sounding to making it really bizarre. In one case, with a little practice, I was able to add some much-needed sustain to a bass guitar track. I also successfully added some dynamic fluctuation to a rather static synth bass track to make it more interesting. The really cool stuff happened when I MIDI-controlled the Triple•C from Digital Performer. Being able to automate parameters and edit them to perfection resulted in exactly the sound I wanted. This points out something about the Triple·C: It functions very well as a "plug 'n' play" compressor, but if you take the time to play with it and explore the many extras it offers, it can become a powerful expressive and creative tool.

Conclusions

I was thoroughly impressed with the Triple•C for both tracking and mixing applications. It offers impeccable audio fidelity and I found it to be a powerful tool for dynamic control and timbral shaping. At its list price, the Triple•C certainly provides an outstanding value when you consider its immense flexibility. Its balance of functionality with ease of use and extended capabilities should find it a welcome place in many home and professional studios.

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

World Radio History

Hotter tracks without
losing dynamics

Compression and the Art of Mastering

CRAIG ANDERTON

Dynamics processing (*e.g.*, compression, limiting, etc.) for mastering is a hot topic. Although some engineers overcompress, giving a lifeless, dynamically unsatisfying sound, the reality is that cutting CDs with hot levels is the norm. If your music shows up in a pile of demos and sounds weak in comparison, you're in trouble.

Although I usually advise going to a pro mastering engineer for a major release, in some cases (like submitting demos or posting material on the Web) you might not want to go through the expense. Therefore, it's worth knowing enough about mastering to be able to master your material for non-critical applications.

For dynamics processing, some people think that all they need to do is buy something like a TC Electronic Finalizer, dial up a preset, and kick back. But the proper use of dynamics control requires analyzing the signal itself to determine the most appropriate type of processing.

Revisiting Dynamic Range
Dynamic range represents the difference between the softest and loudest signal levels you can record. However, note the difference between *peak* and *average* signal level. For example, heavily distorted guitar has a very high average level, because the signal is clipped so that even "soft" passages end up being high-level. Conversely, a drum has a high peak level but low average level because of an initially high transient that decays quickly.

If you set the levels for a drum hit and shredded guitar chord so that their peaks both hit 0 (maximum level), the guitar will sound much louder because it has a higher average signal level.

The point of adding dynamics control during mastering is to raise the average signal level and produce a "hotter" signal, while keeping peak levels from distorting. However, there's a tradeoff: too high an average level kills dynamics, while too low an average level gives an anemic sound compared to music mastered at a higher average level.

Tools of the Trade
Let's look at various ways to do dynamics processing:

Limiting
Basics: This clamps all signals above a threshold to the specified threshold value (referenced to 0 dB). Everything below the threshold

remains unaffected.

Why you'd use it: This is most useful for audio having a low but consistent average level, and a few spiky peaks that hit 0. For example, suppose most of your peaks top out at –9 dB, but a few "rogue" spikes go to 0. If you set the limiter's threshold to –9 dB, it will clamp those
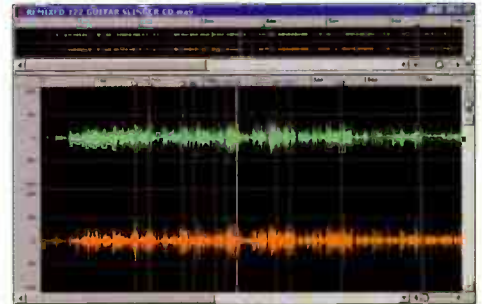
FIGURE 1: Using Steinberg's WaveLab to find the peaks in a tune.
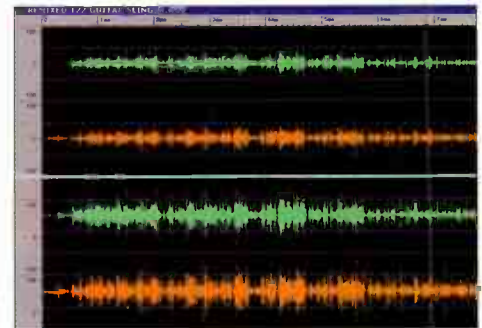
FIGURE 2: The uncompressed signal (above) is a good candidate for compression. Even though its peaks hit 0, note the relatively low average level. Setting a compression threshold of around 6 dB, with a fairly high ratio (10:1), produces the waveform shown below. The dynamics are still pretty much intact, but the average level is much higher.

rogue spikes to –9 dB. You have now recovered an additional 9 dB of headroom, which allows raising the gain or normalizing to a higher level.

Drawbacks: When used to clamp the occasional peak, limiters can give very unobtrusive dynamics control. However, attempting to use limiting to greatly increase average level can lead to a squeezed, unpleasant sound.

Compression
Basics: This processes all signals above a threshold so that output signals increase at a lower rate as the input signal increases. For example, with 2:1 compression, making the input

HOT CD-R COOL PRICE

CONCERNING NORMALIZATION

The caution about normalizing all tracks on a CD is valid, as there may be songs with varying average levels, and normalizing based on peak levels could make some songs appear drastically louder than others. However, these days my last step in mastering is to normalize all songs to 0.1 dB (this brings down the level just enough to make sure that a pressing plant doesn't interpret a string of 0 dB peaks as being "overs," thus causing it to reject the master). Then, I sequence the songs in Sonic Foundry's CD Architect, and listen to the tracks. If needed, I lower the level of songs that appear too loud compared to other songs.

If all the songs are okay, except one or two appear too soft, rather than lower the levels of the louder songs (I can't bring up the levels of the soft songs, because they're already normalized to their maximum possible levels), I'll go back to the soft songs and add 0.5 or 1 dB of loudness optimization. This isn't necessarily an ideal solution, but it gets the job done without causing too much disturbance to the time/money equation.

One other advantage of normalizing all cuts is if I need to submit an individual cut for a compilation CD or Web site, I needn't keep track of which ones were cut softer or louder.

signal 2 dB louder will only raise the output 1 dB. With 10:1 compression, raising the input signal by 2 dB will raise the output by 0.2 dB. Signals below the threshold usually remain unaffected.

Why you'd use it: This is ideal for making program material with a widely varying dynamic range more consistent. A good example would be mastering a live recording where a band does a soft *a capella* section, then blasts loose with a hot solo. Compression can appear to bring up the softer parts, while taming the louder sections. *Drawbacks:* Compressors are actually very complex to use, with many interacting parameters. (In addition to threshold and ratio, an attack control determines how long it takes for compression to kick in when a signal exceeds the threshold, and a decay control sets the time for the compressor to release after a signal goes under the threshold. There are also different types of compression curves and other variables.) As a result, just tweaking knobs without understanding what you're doing can do more harm than good.

Loudness Maximizers and Optimizers

Basics: These are related to compressors and limiters, but usually incorporate some sort of "smart" algorithm that maintains the illusion of dynamics better than traditional processors. Simply stated, they seem to squeeze the dynamic range variations into the top 10 dB or so of available headroom. That's enough to give a sense of dynamics, while greatly increasing the average level.

Why you'd use it: These boxes are ideal for dance tracks, because DJs don't want huge level variations between tracks. They're also great for mastering for the Web, because data compression algorithms (*e.g.*, MP3, RealAudio) seem happier processing data with a restricted dynamic range. As to standard CD releases, they're seen as a "one-stop" solution to getting a hotter level on CD. If used carefully and appropriately, they can do just that. *Drawbacks:* These processors are often misused to get hot levels at the expense of added distortion and dynamic range demolition. Also, they're not always the most appropriate solution for increasing average level.

The Right Tool For The Right Job

I like a natural sound, but because I do a lot of dance music, the mastering is expected to be really hot. However, each tune, each situation, is different. As a result, different program material works best with different "styles" of dynamics control.

Multi-Stage Dynamics Processing

Sometimes applying multiple instances of gentle processing works best. For example,

► WEBLINK

Keeping it cool online

Fanning/Banning the Flames

About six months ago, *EQ* started their on-line discussion forums. I don't think anyone was prepared for the deluge of responses from readers and interested parties that has been taking place. I've only been hosting for a few months, but have found it to be a primarily rewarding experience. Now, I'm certainly not the world's foremost authority on *anything,* but during my 43 years in the trenches, I've certainly experienced a lot of good and bad situations. And I'm happy to share the benefit of those experiences with people who are interested in learning something — and who are interested in sharing their own experience and knowledge. My favorite moments, in fact, have come from questions I haven't even had a chance to reply to that were quickly and competently answered by other participants. I think that's the true beauty of the forums — we're a community where everyone is sincerely reaching out and helping each other via the Internet.

However, occasionally, the other shoe drops — it's only human nature. Debates develop, tempers heat up, people say some amazingly nasty things to others, and things career out of control. Surprising? Hardly. But let's talk about it for a moment.

I've hosted a forum on AOL for about eleven years. It used to be open season on occasion. After all, there I was, online and unmasked, and everyone else involved had his or her "mask" on. Email gives people an enviable anonymity that can sometimes be used like a Klan sheet. Crosses can be burned on people's lawns and no one *really* knows who the culprits are. I have heated debates with rock writer Dave Marsh in print on occasion, but we both know who the other is. There's something inherently honest about that. But if somebody comes into a forum and anonymously starts flaming and heating things up, I think it's a counter-productive assault carried out under a pointed cyber-sheet. Everyone's entitled to their opinion — after all, this is America. But when the expression of that opinion gets nasty, becomes a personal attack, then it's time to take a step back and rethink. When it gets personal, *take it to private email.*

Recently, I hosted a Napster thread in my forum (big mistake). We all know from last month's column what my stance is on that

subject. Well, my feelings and those of the opposition can easily reach a fevered pitch on that particular subject. Somebody baited me, and, being passionate about what I believe, I took the bait and stooped down to their level. It got incredibly ugly and personal. I kicked myself for losing my temper and thought it best to erase the thread and get that venom out of the forum as soon as I could. I apologized to all on the forum for losing said temper, and the protagonist, who had stated that he was out of there, came right back and stirred the pot up again. I recalled what I used to do in my AOL forum when things like this happened: I asked everyone to just ignore the flamer, and when they complied, he left, bored. I'm gonna suggest the same thing the next time something like this happens on my *EQ* forum; there's too much good being done there to let flamers take it down.

There are plenty of places to vent and rant on the 'Net on just about *any* subject under the sun. I refer you ranters to The Velvet Rope (**www.velvetrope.com**), the music-biz insider's forum, where the sharks feed on each other regularly. Let's hope we don't come to that here. Where else are you going to find experts of the caliber of George Massenburg, Chris Stone, and Roger Nichols (to name but a few) who will correspond with you one-on-one *for free*?

To conclude, I'm certainly not advocating censorship. If you wanna rant, at least take the sheet off your head and let us know who you are and what qualifications you have for your opinion and heated stance. Then, when it's appropriate, take it private and allow a terrific community to exist without unnecessary intrusions. Let's be professional.

There's no need for a cyber-recapitulation of Altamont; hopefully we've learned something since then — peace, love, and the dissemination of helpful information for the betterment of the audio community. Okay, you can turn off the John Phillip Sousa CD now and put Massive Attack back on. See ya next month. ∎

AL KOOPER

Good craftsmanship
is good business

Als ik Kan

JIM BORDNER

Welcome to 2001 (the true millennium, for you hardcore Julian calendar fans out there). Over the holidays, I've been doing some reading, and one of the true-life characters I encountered has really captured my imagination. I'd like to introduce him to all of you as a sort of keynote for the New Year.

Gustav Stickley was a furniture maker and a leader of the Arts & Crafts movement. His original furniture is highly prized by collectors for its beauty, simplicity, and durability. The company that bears his name still strives to create pieces that exhibit these same values.

Stickley lived as he worked — simply. He had no lofty dreams about wealth, fame, or making a chair that was number one with a bullet on the Arts & Crafts Charts. Gustav Stickley was a *craftsman*. He stated his entire philosophy, in his native Dutch, on a label he placed on the underside of each finished piece. It read, "*Als ik Kan*," which means "the best I can." By adhering to this motto and making each piece the best he could on that particular day, Gustav Stickley became a master of both his craft and his business.

In any creative business, it's easy for people to succumb to complacency ("well, that's good enough for what they're paying me"), arrogance ("these clodhoppers don't deserve anything better"), or creative despair ("man, I'll never be as good as James Newton Howard"). It's especially tempting in a business such as music or recording, where the practitioners hold a specialized knowledge. But these attitudes are counterproductive, and therefore bad for business. You can't do your best work when you're comparing everything to your personal "Gods" or the truly *great* project you got to work on last month. The only way to approach a project, regardless of budget, time constraints, or the relative thick-headedness of a given client, is with Stickley's attitude. Always create simply the best you can.

You never know who is going to talk to who, how a project may come your way, or when today's no-budget gig will lead to next year's hooray-we're-going-to-Europe gig. Because of this, you want all the work you send out the door to be the best you can do. If you apply some of the principles of craftsmanship, your best will always lead to better.

A Craftsman is Always Learning
You can always get better. Stickley never stopped studying and experimenting, and neither

should you. Read a book on mic placement. Spend an afternoon doing nothing but goofing with EQ settings on a bass guitar track. Attend a seminar about something you think you already know. If you come away with even one or two pieces of solid information from the exercise, you've achieved something amazing — you've gotten better at your craft, a feat accomplished by very few individuals in this world today.

A Craftsman is Always in Service to the Customer
There's a buzzphrase in business circles these days — "exceeding customer expectations." Everybody is trying to do it, from AT&T to the

> THE ONLY WAY TO APPROACH A PROJECT, REGARDLESS OF BUDGET, TIME CONSTRAINTS, OR THE CLIENT, IS WITH STICKLEY'S ATTITUDE. ALWAYS CREATE THE BEST YOU CAN.

pet store on the corner. Whole books have been written on the subject, but it can be expressed in a nutshell. Your clients expect something for their money: an experience, a finished product, a good feeling, something. If you fail to meet their expectation, you've lost a customer. If you meet it, they'll be satisfied. But if you exceed it, you'll amaze and delight them, and win their loyalty.

They may be perfectly happy with a recording engineer who says, "Well, if you guys are okay with that guitar sound, then let's call it a day." But they'll never forget the one who says, "I'm going to hang around here and try a cool idea I have for that guitar hook…now, you guys go get some food…bring me a cheeseburger." A true craftsman doesn't stop just because the
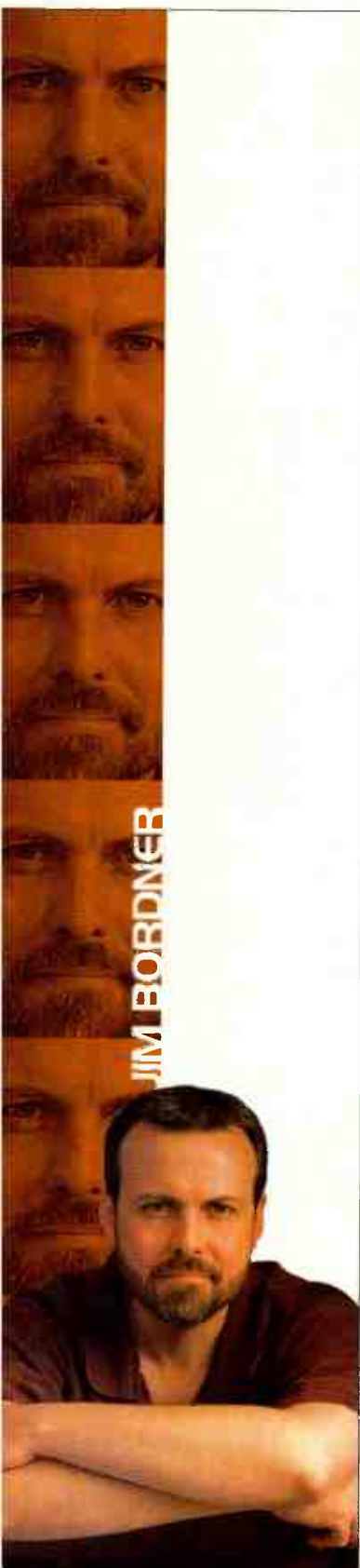
customer is happy. A craftsman doesn't stop until he or she is happy.

A Craftsman is Always Educating the Customer

We'd all love to work with clients and customers who trust our judgement, who solicit our opinions, and who are loyal to their relationship with us (and believe me, there's nothing *but* relationships in this business). You can help achieve all three at once by simply educating your customers. It's easy for us to guard our knowledge with language that sounds like a code to outsiders. Doctors do it, lawyers do it, pickpockets and grifters do it, and so do recording engineers. I've heard a hundred funny stories about engineers having duped their customers to avoid incorporating bonehead suggestions: playing back a mix from the two-track while pretending to raise and lower levels, silencing a client with meaningless double-talk, and

the like. I suspect that many of these stories are urban legend, but true or not, they exhibit an attitude that would be better left outside the studio door. Baffling a customer might make the engineer feel pretty damn smart in the short term, but what effect does it have on a customer?

Well, it makes them feel stupid. Nobody likes feeling stupid, and they don't like to go back to a place where they were made to feel stupid. Next time, they're going to talk to another engineer at another studio, and this person may take the time to explain, in simple terms, why he's doing things the way he does and why he feels that the customer's suggestions, while valid and taken seriously, aren't really the best approach. This guy will make them feel smart and involved in the process. And that's where they'll go the next time.

How many times have you heard someone say, "I have this great doctor," or "I know this wonderful mechanic?" What they're really saying is, "I know a guy who makes things clear to me and lets me make intelligent decisions based on the knowledge he shares with me." Share your expertise, translate it, bring your customers into the

process, and you'll build relationships that your competitors can't challenge.

A Craftsman Works for the Love of the Craft

This is absolutely the only good reason to do anything in this shockingly brief life. Gustav Stickley loved the whole process of making furniture — measuring, cutting wood, sanding and polishing, watching the grain emerge under his hands. You have to love every part of the process to be a craftsman. I would guess that the great recording engineers would tell you that even they get a charge out of the simple act of threading up a fresh reel of two-inch tape.

That's how being a craftsman makes your business more solid and more profitable. Here in this glittering New Year, don't hamper yourself with odious comparisons. Just ask yourself if this piece of work, the one right in front of you, deserves the *Als ik Kan* label. It's good craftmanship and good business.

Jim Bordner makes music, records audio, and studies the twisted anthropology of studio customers at Gravity Music.

From Here On Up It's All Downhill
(apologies to Walt Kelly)

Sometimes it feels like the insanity will never stop. In an epoch where you can buy a CD through your cell phone (soon your Palm handheld — maybe even your palm!) and prices for storage media drop through the floor on a daily basis, it feels as if the deeper we investigate Internet audio the more it appears the same people are running everything. The year 2000 has seen many venture-capital-backed indie companies plucked up by large businesses, and others burn through millions of dollars until there's nothing left, their workable business model dangling somewhere in the future. The Big Five (perhaps soon Four) labels continue their cozying up to the Internet industry, injecting their money (and their will). Bertelsmann's "loan" of $50 million to Napster for the development of a subscription model featuring BMG music is a perfect example.

This month we will depress and elate you. Recent Internet audio company downers will be balanced with displays of actual commonsense by the oft-times over-funded online music community. That sense of entitlement oozing from the pores of corporate "cybermusic" whores is finally washing off in reality's cold shower. It's the wake up call we've been anticipating.

First, let's talk about how the major labels have bitten the bullet. Streaming companies like Musicbank (**www.musicbank.com**) and Streamwaves (**www.streamwaves.com**) are finally able to buy licenses to legitimately stream major label catalogs (*i.e.*, the latest Mariah Carey and Sting releases can be legally listened to via Internet radio). EMI will license music to Streamwaves for their own subscription model (between $10 and $13 per month) set to launch at the beginning of 2001.

The above shows adherence to the "if you can't beat 'em, join 'em" business model. That and "duck and cover" are all the tools the techno- and bureaucrats seem to be using. It ain't easy sailing for Internet music companies right now. Tech stocks are way down and many companies are hemorrhaging cash or have simply tanked.

• Venerable Riffage (**www.riffage.com**), having already received large infusions of cash from BMG and AOL, is having a rough time. They need even more money and are considering selling themselves off. Who, other than an old-school, cash-heavy big label, is going to step up to the plate? Riffage recently spent one million dollars on a site redesign and another million buying the hundred-year-old Great American Music Hall in San Francisco. Reports from the frontline are: Web operations remain a far third in revenues, well behind 1500 Records (their acquired private indie label) and the live venue. The FezGuys never really cared for the domain name (c'mon, man: "riffage"…like…"Dude!") but wish them the best.

• Emusic (**www.emusic.com**) has been making overtures to Napster (**www.napster.com**) regarding illegal sharing of Emusic-copyrighted material. Rebuffed in their negotiation attempts, Emusic is now claiming Napster has "forced" them to report offenders. They'll use a 'bot to find 'em, an email to warn 'em, and 24 hours later demand Napster remove 'em if those files are still being shared. In spite (or because) of this frantic proactivity, Emusic's stock remains bottomed out.

• Online indie distribution company The Orchard ("A Place To Grow" — **www.theorchard.com**) has pruned staff and owes tens of thousands of dollars to employees and artists.

• Independent label Platinum Entertainment is selling all its assets and filing for bankruptcy — it was probably that horrid domain name for their Web site: (**www.heardon.com**).

• Once proud Artistdirect (**www.artistdirect.com**), valiantly attempting to staunch the bloodletting, has dumped 12 percent of its staff and more layoffs are expected. The company had been blissfully burning through $10 million a quarter. Of course, it still has $97 million in cash, so anything's possible. How about one last megalithic party?

• Punch-drunk but still standing, MP3.com (**www.mp3.com**) has settled with the last of the Big Five record labels. That's a total cave-in of about $170 million to mollify the plaintiffs and pay the lawyers. Of course, MP3.com probably still has a cool $100 million in the bank. Maybe *they'll* buy Riffage. Probably not, because labels that have already settled with MP3.com still have the right to come back and collect more damages. This raises the possibility of the defendant's dissolution into bankruptcy, which means, of course, that no one gets paid. And so MP3.com has suddenly added a retired judge who specializes in intellectual property rights cases to its corporate board! Smart thinking.

The online music flash-in-the-pan is in danger of becoming cold, curdled grease. Most everybody's finally stopped screaming "There's gold in them thar hills!" Of course, this is all good news. Realistic business models and realistic expectations can now take their rightful place. One potential downside here is cool indie sites being bought by profit-minded suits, branding kept, and brain trust fired. Once again the big guys can own the space by simply waiting to get a good deal to buy the intrepid trailblazers. Think Microsoft strategy: the Law of Attrition. But we're not bitter about it! "Always look on the bright side of life...."

Now, let's take some questions....

Questions!

Can you explain how, if Napster is free, no fees or nuttin', how does it generate any money? Who pays for the servers/electric bill? The guy who "invented" it sez he quit school to "run the company." Who pays his salary? How can it exist as a company with stocks etc, if it doesn't really exist? —Steve

Steve — You've hit upon the essence of this era of online music! Napster, the technology, is a small app that a college student developed in his spare time for the hell of it and released to anyone and everyone. It works without money or sales because everyone simply donates the disk space on their hard drives to share and share alike. As to Napster, the company, with payroll, etc., well, it's called "investment." People with money to burn offer some (for operational costs and frequent trips to music conferences) to a bunch of heads who are attempting to turn a free tool into a profitable one. Whether or not it'll work depends on an infinite number of variables including, to a large part, luck. If Napster, the company, runs out of money and they can't get any more, then employees will migrate to the next company and the investors will own what's left, including the name and the technology. A fire sale will ensue. Isn't the free market cool?

I need some advice. My dream is to own a record label; I've done a lot of songwriting and a little producing, but my two big questions are how do I get my foot in the door, and how do I keep my beautiful ship of artists from sinking? Would I need to start my own Web site? —John

John — You can't get your foot in the door without first putting your boots on. Do this by playing live as much as possible. You can't keep your artists' ship afloat if they aren't naturally buoyant. Let them sink and go find someone else who already knows how to float. You've got enough work to do just taking care of your end. Remember, the music comes first and then the Web site. Take advantage of free upload sites (IUMA, MP3.com, Riffage, etc.) to create simple but useful Web pages for your music, bio, and tour dates. Your local dialup ISP account probably includes some Web space. You can play with it to create a more personalized site for your band. Also remember to register your band's domain name.

When are we Mac users going to see a Mac version of Shoutcast? I saw a version for OS X, but in an attempt to download, it could not find the file. I'd like to see a pre-OS X version as I'm not ready to jump on the OS X bandwagon. —Sean

Sean — We feel your pain, but the only way to get some love is to ask the folks who develop Shoutcast when (and if) they plan to

I'll typically apply some light limiting — 1 to 3 dB — to tame the biggest peaks, then normalize up to 0 dB (see sidebar). That produces a much hotter track, but the dynamic range remains virtually intact. WaveLab has a useful feature that can analyze a signal, then place markers at the most prominent peaks (fig. 1). If there are only a dozen or so "rogue peaks" in a tune, limiting is a great way to make the tune hotter without losing dynamics.

Next, adding a few dB of loudness optimization will make the track really "pop." Be sparing, though. If the program indicates that it's possible to increase the overall level by 7 dB without distortion, I'll go for 3 or maybe 4 dB max. Between those dB, and the typically 2–3 dB recovered by limiting, I'll end up with a track that's 5–7 dB hotter than the original mix — that's a lot — but it still has dynamics.

That's usually where I like to leave a track. Some clients, though, want things even hotter. For the final *coup de grâce*, I'll add a slight amount of overall compression (*e.g.*, threshold of –2 dB, and a compression ratio of 3:1). This gives just enough of a snap to make them happy.

Compression: Fixing it in the Mix
Sometimes a mix will not turn out quite right — there may be too many level variations, or some drum hits might come on too strong. The obvious answer is to do another mix, but this is not an ideal world, and often time or budget constraints work against making every little tweak.

Here, compression can be a savior by producing a more uniform-sounding mix (fig. 2). Again, don't go overboard; setting a threshold around –3 to –6 dB and a ratio of 1.5:1 or 2:1 might be all that's needed. Once that's accomplished, a little loudness optimization can heat up the levels.

All You Need is Loud
Now suppose you've produced a mix where you used some compression on individual tracks, so there aren't any wild transients, and the mix is uniformly great...but you just want a hotter level. This is a case where all you might need is some loudness optimization. Dial in a few dB, and you're ready to go.

Well, that's enough for now. Just remember that, despite the hype, there is no "one size fits all" dynamics processing solution for mastering. Always choose your processing on a case-by-case basis for the optimum combination of hot signal and dynamic range.

Craig Anderton, creative director for MusicPlayer.com, is also the author of Multi-effects for Musicians (AMSCO). His song "What Are You Waiting For?" was recently remixed for a European compilation CD.

ROOM WITH A VU
▶ continued from page 31

in a musical and cohesive way. Synth-wise I have the most fun with my Nord Rack 2 — I think everyone ought to have at least one synth with buttons! I'm a big fan of the Lexicon Studio Core32. Although it hasn't been a huge marketing success, the unit has excellent features and sound."

PRODUCTION NOTES: Ostwald concludes: "When we built the studio I ran PVC pipes through the walls and ceiling to the house. There's wiring in the pipes for microphone, phones, video, and ethernet connections, so the studio is tied into the main house where I have my Yamaha C7 grand piano and a remote overdub room. With all the sequencing and programming going on, I always find it *very* refreshing to set up the band and record live — it's a blast! I can have live drums in the studio room, keys and bass in the control room, and a live percussionist, horn player, or vocalist in the remote room." ■

From Disk to Disc

Basically I think that two things need to happen before tape is dead. First, optical disc technology needs to improve by a couple orders of magnitude. There's a new technology called FEM recording that's based on photo-luminescence. This technology allows 100–200 gigabytes to be recorded on a single DVD-type disc. DVDs are less susceptible to damage from travel and storage than tape or hard disk. Second, we need updates in software to allow multiple time stamps of data so that once a region has been positioned in the desired location, a local time stamp (in addition to the original time stamp) will assure that a region that was accidentally moved can be repositioned with a purposeful click of the mouse.

Until optical technology catches up and spawns inexpensive DVD-based multitrack recorders, tape machines such as DA-88's and ADATs will remain the medium of choice for most project studios throughout the world. Tape still allows digital recording technology for a massive amount of album product that is released every year.

I get projects to mix from South America frequently. Twenty percent of the projects I receive are on DA-88 tapes, 65 percent are on ADAT tapes, and 10 percent are Pro Tools files on CD-ROM. None are sent on hard disk or reel-to-reel multitrack formats. In Caracas, there are many small project studios with one ADAT machine. Producers move around from studio to studio with their ADAT tapes. Recording a guitar here, a horn section there…and once in a while they have to go to a larger studio to make more slave tapes so they can continue to record overdubs in project studios that only have one or two ADATs to work with. These machines get lots of work, too. When I was in Caracas last year, it was very hard to find an ADAT machine with less than 8,000 head hours. The hour meters on most of the machines I saw were pegged at 9,999 hours. That is a lot of music.

Project Update

I have been upgrading my little studio, and have been working in other project studios in Florida.

I just finished mixing an album in a project studio in the Tampa area. The artist was Gumbi Ortiz, a percussionist. The producer was Dan O'Brien. I mixed on a MOTU 2408 setup with Digital Performer. Some of the mixing is done with the volume automation in Digital Performer, and then stems are sent out to a Panasonic DA-7 where they are combined and sent to an Alesis Masterlink. The studio has two isolation booths for overdubs when necessary.

I helped an artist/producer named Ariel Remos finish up his album. His Miami project studio is Pro Tools monitored through a Mackie analog console. Recording is done through external mic preamps directly into Pro Tools. The console is just for monitoring and all of the mixing is done in Pro Tools.

My studio has added a pair of Mackie HDR 24/96 recorders and a Sony DMX-R100 console. I have been dumping projects into Pro Tools from my six ADAT M-20 machines. I then mix in both Pro Tools and the Sony board. I use plug-ins in Pro Tools and use the volume automation for vocal rides and such. Each track is then fed to the Sony for EQ and reverb sends. From there the mix is printed on Alesis Masterlink.

Life is good, but new equipment makes it even better. ∎

PARTY IN PARADISE

All I did for this record was to play a tune and say, "How do we want to cut this?" We let it organically develop and then we recorded it. The smart thing we did this time was to rehearse for a week. We went to The Alley and rehearsed 18 tunes, two or three times each, and then had a couple of days to think about them. Then we went directly into the studio and when I said, "Let's do 'Penguin In A Palmtree,'" everybody knew what I was talking about, instead of working it out in the studio. It made a big difference, because I wanted to put a set together and go in and play it. We had a list of songs and we just went in order. We'll sequence the album later after we've had time to think about how the songs fit together.

I was impressed at how briskly things moved along when typically, over the years, sessions have gotten slower.

Nash: Well, people get lazy. When I was recording with The Hollies we cut our first album in an hour and a half. The entire record. We set up our instruments, then we did our two 45-minute sets. We played the set right through. We knew what we were doing, because we'd been doing it for a couple of years. We took the best of both sets and that was the first album.

Do you remember the studio?
Nash: Sure, Abbey Road.

Is it true that The Hollies was the only group that had more hits in England than the Beatles?
Nash: It's very possible, but you have to realize that The Beatles stopped in 1967. We kept rockin' right on through. We did have an enormous amount of hits — I think it was 18 Top Ten hits. We were very big, and we were a good, popular group. Not very profound, but who was then except for The Beatles and Dylan?

Do you have any thoughts for young people thinking about a career in music?
Nash: The only thing that kept me alive, that kept me moving forward, that kept me on the ball, is a love for what I do. I don't want to get heavy-handed, but there's a great responsibility as a musician to write true music, meaning that it comes from within your soul, something that you truly feel. I spent many years not doing that and I love those years, but regret them at the same time. I realize it was part of the growth process that brought me to the point where I am right now.

Well, I can't think of anything else to ask.
Nash: Good — let's go get some lunch.■

DARYL KELL

The best thing was the personnel involved. The director really understands music, so it was a pretty good collaboration between he and the composer [Edward Shearmur]. That makes it nice compared to projects where the director is unsure as to what he wants.

What gear do you normally use?
Most editors own their own gear since you can set it up so that it's the same every time, which you can't really do with rental gear. I basically use a Mac G4 with [Digidesign] Pro Tools and [Mark of the Unicorn] Digital Performer software. I use Digital Performer for scoring sessions for clicks and steamers and to play back any prerecorded audio tracks. I'm sure other sequencers do exactly the same thing and would work as well, but most of the composers I work with use Digital Performer, so it's easiest to just stay with it. I also have a Mackie LM3204 and a couple of Genelecs for my personal monitoring purposes. For picture I use a MiroMotion card for video capture. At this point I still receive a tape and digitize it, but that's already changing in that you can now receive just a QuickTime file. The tape step will be gone pretty soon.

My system gets moved around and plugged into whatever console is being used on the dub that I'm on. We now have the technology to not lose any generations from the initial recording of the orchestra right through mixing and dubbing. For *Charlie's Angels*, the orchestra was recorded onto a Euphonix R-1, then we came out of a Euphonix System 5 digitally into Pro Tools when it was mixed. Although there wasn't a digital console on the dub stage, theoretically you can stay digital from when it gets recorded up until when it reaches peoples' ears in the theaters. That would be the first time there would actually be an analog conversion.

What's the most difficult part of your job?
The time period leading up to the scoring sessions is the most hectic and needs the most organization. Once things are recorded and the project is in transition to the dubbing stage, then the biggest problem is trying to keep up with the picture changes. You try to keep the files of the scoring session up to date with the most current picture, but sometimes it's already been orchestrated when the changes come in. On a whole, organization and preparation are the hardest parts.

FEZ GUYS

produce a Mac version. Get your friends who also want a Mac version to email them. Show Shoutcast there's a demand. That will help them prioritize your request. By the way, we assume you're talking about the server component, as there are several MP3 players out there for the Mac that play Shoutcast streams (Audion, Macast, etc).

What's the technology used to auto-start sound on homepages? Is it simply Java applets plopped in? Will Javascript do the same? Or am I so far behind that I should just go pickup my Sigma acoustic bass and pass the time away? —Peter

Peter — There's lots of ways to start audio playing as soon as a fan brings up your Web site. Macromedia flash easily embeds MP3 audio in a page, and most browsers (certainly Netscape) can play WAV files with the EMBED tag, also. Javascript can be used to automatically pop up new windows with music in them, too. But wait...take a moment and ask yourself this question: Do people really *want* music to start playing when they arrive at a site? Here are the FezGuys's Top Five reasons to avoid embedding music on your home page:

1. Most people are already listening to their own music and when yours mixes in it's instant cacaphony.

2. It may require software that users don't have installed. This would result in a warning message — not a good first impression!

3. Many users are still connecting with dialup modems. Including music in your home page will take much longer to load — also not a good impression!

4. After clicking on one link, clicking "back" on the browser restarts that same audio clip over again from the *same place*, often delaying the time until your visitor can click on another link.

5. It's simply not expected and can be jarring.

Visiting your site should be easy, quick, and pleasant. We encourage you to incorporate all sorts of features into it, but try to place them where people know what to expect. Offer links to this information by using helpful messages along the lines of: "Click here to see a cool multimedia display of our music video!" or "Check out our music in our interactive jukebox!" It's an aesthetic choice only you can make. Choose wisely.

World Radio History

World Radio History

World Radio History

World Radio History

*Life is good, but new equipment
makes it even better*

2001, Space Odyssey

I have been waiting for the 24-hour *2001* channel. I figured out that in the year 2001 you could play the movie *2001* exactly 2001 times with some space for commercials. The commercials would undoubtedly be for the 24-hour *2010* channel to be launched in nine years to show Stanley Kubrick's sequel.

My mastering/mixing room has become a space odyssey of its own. I have no space for all of the stuff that I can't seem to get rid of. Some outboard gear has been gathering dust for five years or more, but just when I get ready to get rid of it, some problem arises that only this obsolete piece of gear will solve. This, of course, justifies hanging on to everything forever. I have come up with a temporary solution. I tell my wife, "This gear will help you with your demos, so I'll just leave it here in your room, okay?" This is working for now, but she is starting to ask questions about all of the gear piling up on her side of the house. I hope she doesn't read my column this month.

Hard Disks Are Free
Well, almost free. Considering that one year ago I paid $1,800 for 18-gigabyte SCSI hard disks, $199 is basically free. (I won't even tell you about the 45-megabyte drive I bought in 1983 for $5,995, or the five-megabyte drive that was $3,995 in 1980.) Eighty-gigabyte ATAPI drives are now $349. Seagate just announced a new 180-gigabyte drive due out soon. Samples are available now.

Hard disks are now cheaper than two-inch analog or reel-to-reel digital tape. Removable drive bays now allow you to plug in the disk drive, record your music, and, when you're done, unplug the drive and put it on the shelf. New project? Just plug in a new drive.

Many of the disk-based multitrack machines, such as the Mackie HDR 24/96, save their files in native DAW formats so that you can plug the drive into your DAW and read the audio directly. The Mackie also has an Ethernet connection for transfer of audio between the 24/96 and any networked computer.

I am waiting for Sony to produce a professional 48-track (or more) digital recorder that's hard disk- or optical disc-based. This milestone will surely mark the end of tape-based recording as far as I am concerned.

The worst thing about tape is that you can't slip tracks around to make them earlier or copy choruses to other choruses without a second machine. Also, the best thing about tape is that you can't slip tracks around to make them earlier. There's a syndrome I like to call "middle-finger mouse reflex" that affects most DAW users. As you're repositioning the mouse from one corner of the computer screen to another corner of the computer screen, your middle finger sometimes reflexively clicks the mouse button. If the mouse pointer was passing over an empty area of the screen, then no problem. You just pause for a second, shake your hand in the air, clench your fingers, and go on about

> **I AM WAITING FOR SONY TO PRODUCE A 48-TRACK DIGITAL RECORDER THAT'S HARD-DISK BASED. THIS MILESTONE WILL MARK THE END OF TAPE-BASED RECORDING.**

your business. If, on the other hand, the mouse pointer was passing over a region of audio, that region would be dragged from its original position by some arbitrary amount. If you noticed the finger twitch, then everything is fine...you just click "undo" and go back to work. If you didn't notice the errant click until after your next editing action, you are dead meat. Some region that you didn't notice was moved, and you don't know by how much. You can't reposition all of the regions in that area to their original timecode, because they were all moved anyway in the course of editing. I like to use tape as my "reference master" during a project. If I have a question about relative position between tracks, I can recapture the tracks from tape and compare their position. This method has saved my butt on more than a few occasions.
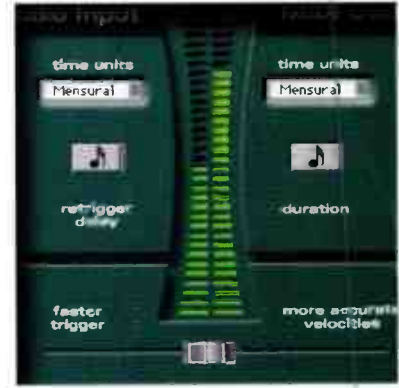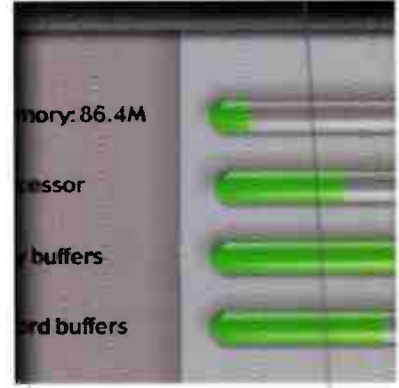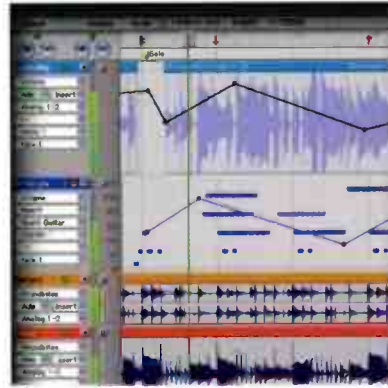
DP3
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