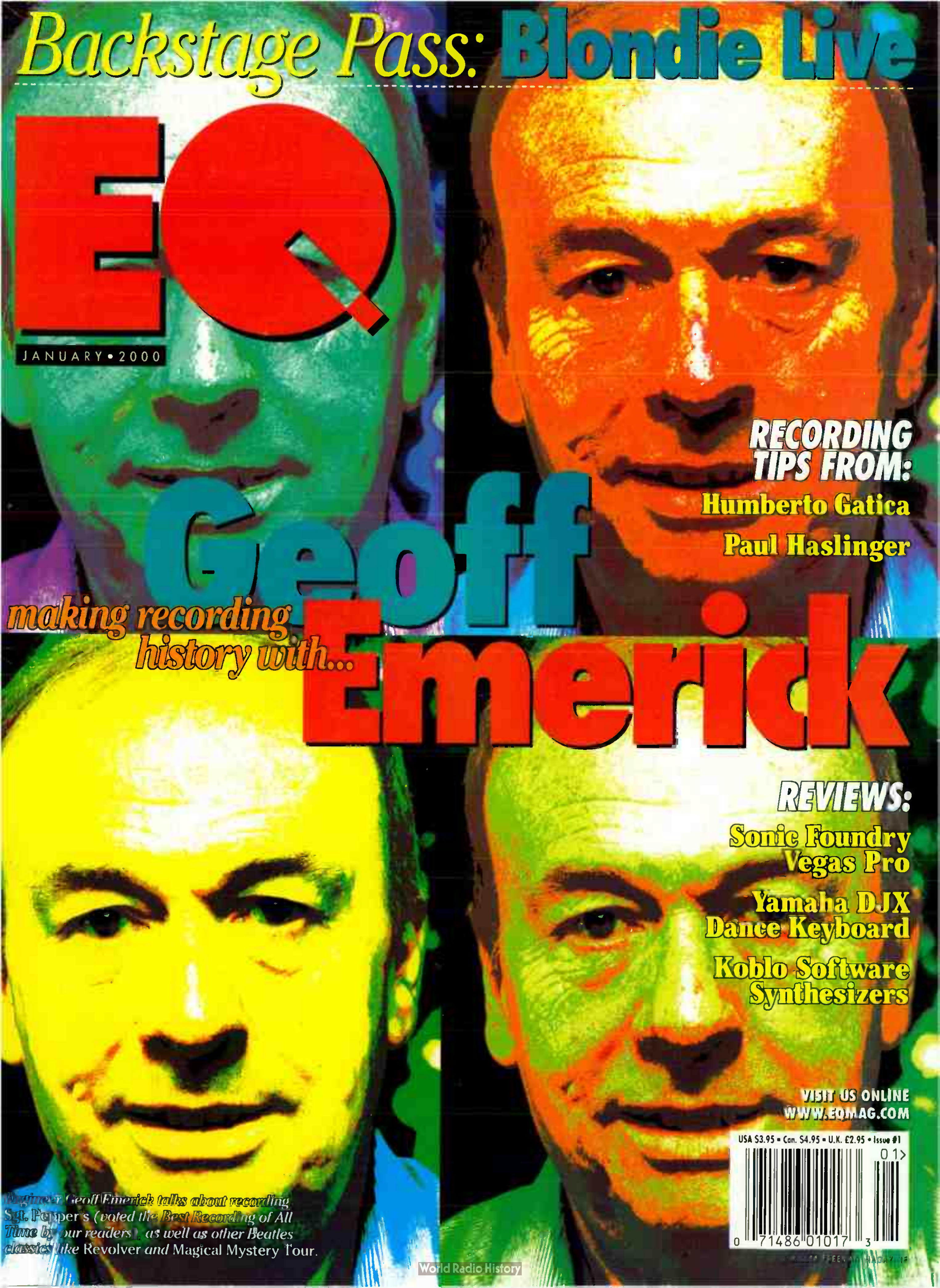

Backstage Pass: Blondie Live

EQ

JANUARY • 2000

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making recording history with...

Geoff Emerick

REVIEWS:

Sonic Foundry Vegas Pro

Yamaha DJX Dance Keyboard

Koblo Software Synthesizers

Engineer Geoff Emerick talks about recording Sgt. Pepper's (voted the Best Recording of All Time by our readers), as well as other Beatles classics like Revolver *and* Magical Mystery Tour.

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VOLUME 11, ISSUE 1
JANUARY 2000

ON THE COVER:
Geoff Emerick gets the Warhol treatment thanks to photographer Edward Colver and assistant art director Greg Gennaro.

FEATURES

TECHNIQUES/WORKSHOPS

EQ LIVE

DRIVE!

GROOVE

COLUMNS/DEPARTMENTS

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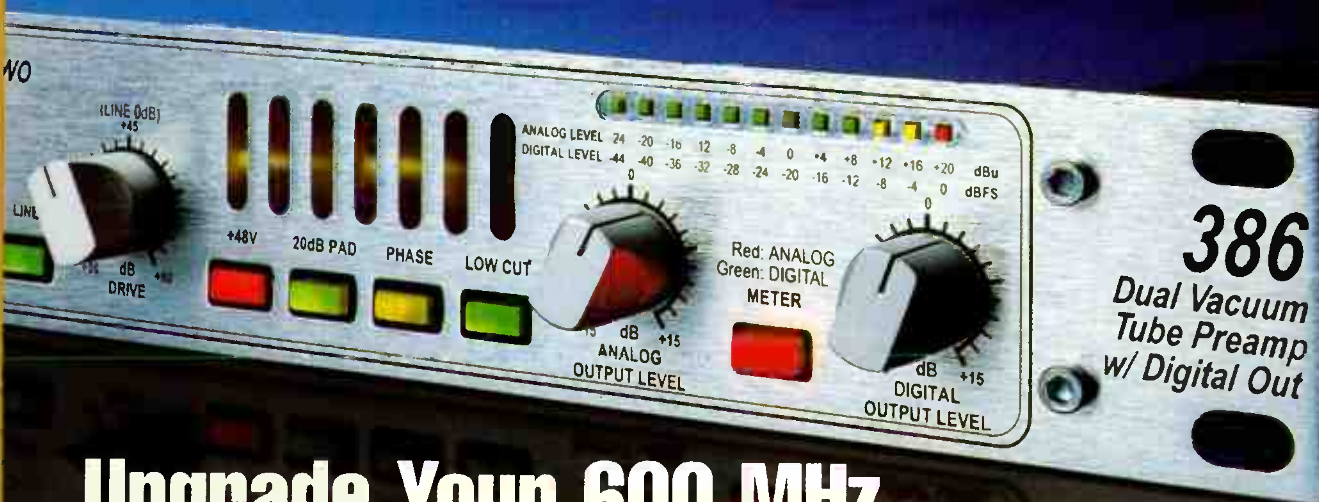
*Actual unit is 19" wide and will not fit in your computer drive bay.



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CIRCLE 50 ON FREE INFO CARD

NO FEAR

Regarding the "No Fear" editorial in the November issue: Thanks for a superlative article about a superlative person. I almost never write to magazines, but this article prompted me to as it really touched a nerve. Who was this wonderful being? What was her name? Maybe you can't give it out, but I would sure like to find out more about this courageous dancer, and I bet many of your readers would like to know, too. When we have all of our faculties, it's almost impossible to truly imagine what it might be like without one or more of them. Like you, I know that if I couldn't see, I'm sure I would play it safe and do something not too dangerous, like editing or mixing. But to be blind and partially deaf and go out on a limb as Miss Diva Divine did, requires a faith in oneself and, I would think, in a power greater than oneself. I acknowledge and applaud this incredible person.

I think this could lead to a profile on "challenged" people in our business. Everyone knows about the great and beloved Stevie Wonder, but there are many other unknown people like the girl you brought to light who deserve some recognition.

Thanks again for sharing such a wonderful story.

Gordon Nicol
Scotch Productions, Inc.
Dallas, TX

TOP RECORDINGS

While I'm sure you'll receive any number of opinions on your selections for the list of landmark recordings [December, '99] (it is a matter of opinion, right?), I couldn't resist adding mine!

I kept expecting to find the one artist I couldn't imagine being left out of any such list, and, to tell you the truth, I started off just kind of skimming the article while keeping an eye on both your list and Al Kooper's for a mention of one name, because I was curious which of his albums one or both lists would select, and what you'd say about it, and yet, to my amazement, I reached the end of both lists without seeing it!

Where, oh where, was Frank Zappa? Need I say more?

John Prusinski
S2N Media, Inc.
via Internet

OK, I know it's taste, and everyone would choose a different list. I just thought I'd add a couple thoughts to Al Kooper's list; although it is a really good list. My thoughts:

Top 100 and no Aretha?! Yikes. There has to be one of the first three albums she made for Atlantic here, and my choice would be *Lady Soul.* "Since You've Been Gone" rocks as hard as any song ever written. Throw in "Chain of Fools," "Good to Me As I Am to You," "Ain't No Way," "Groovin'," "People Get Ready," and, well, it has to be on the list.

Sticky Fingers for the Stones?! Come now! *Begger's Banquet* was their crowning achievement, and the last album Brian Jones *really* worked on.

Dylan's a little tougher. Yes, *Blonde on Blonde* was a great album, and it seems like it was truly done on the fly, which makes it quite an achievement. But *Bringing It All Back Home* was the groundbreaker. Great songs on the acoustic and the electric sides. I knew people who scratched the electric side up so it couldn't be played, kind of like slashing the Impressionists. This is the stuff that altered the Beatles way of looking at song and the rock world forever.

Now, here are three I think should be at least honorable mentions:

Gladys Knight and the Pips version of "Heard It Through the Grapevine" just kicks Marvin's version's ass. This rocks like Aretha's stuff.

Alone Together by Dave Mason (and the Shelter People without the name) has great songs, great arrangements, and great production. I put it way up there somewhere.

They Only Come Out At Night by Edgar Winter and produced by Rick Derringer has everything. Really amazing production! Really! And songs that range from hard-rockin' blues-based numbers like "We All Had a Real Good Time," to the country rock of "Round and Round," to a ballad like "Autumn," to the synthesizer funk of "Frankenstein." Listen to this album on a great stereo, real loud, and be really impressed!

Fred Towne
via Internet

SLOP SINK STUDIO

After reading the article "Joe Jackson's Coat Check Sessions" [October '99] with Steve Remote (I've worked with Steve in the early '90s through the band Psychic Orgy), I thought I'd mention a similar but somewhat backwater experience. I was hired to do a live multitrack recording of the hip psychobilly outfit Slick Pelt. The club was a place I had been to before called the Blue Comet in Philadelphia, so I had an idea of the layout. I loaded up my '79 Firebird with my two ADATs and a Mackie 1604VLZ and was on my way.

My plan was to run a 100-foot snake into the basement so I would have a nice quiet setting to work in. As it turns out, the basement was partially condemned (the building was collapsing or something). On to plan B: Find a new control room. The best I could do was a slop sink closet next to the girl's bathroom.

Time to rig it up. The cook lends me a cutting board that I put over the sink as a tabletop for the Mackie. Then, I clear a shelf of some toilet paper for the ADATs, which I hold up with a mop handle for safety. I use a combination of the direct, sub, and aux outs to the ADATs. I run the snake to the stage, facing out, and get ready for line check. I'm only using bass on 5–6, guitar on 7, vocals on 8–10, and crowd on 11–12. Since I have a few tracks open, I make a music submix on 13–14 and a vocal sub mix on 15–16. These "live" submixes really added something when we were mixing the tracks back at my studio.

I had a great time doing this. There are even some pictures of me with two women and a cowboy hat, but that's another story. It just goes to show how, with a little inventiveness, plenty of duct, and a lot of beer, anyplace can become the slop sink of your dreams.

Vin Cin
Electric Plant Studios
Brooklyn, NY

WATCH YOUR MOUTH

Dittos To Steven Durr's great letter [December '99]. In fact, keep that word, the other s-word, and the f-word out of the entire magazine! If you're quoting from an interview, use dashes or asterisks, like the rest of the publishing industry. I read the magazine holding my nose, and cover it with papers when I leave the room because I need the technical info from Eddie Ciletti, Mike Sokol, and others.

John Flenniken
Fort Worth, TX

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CIRCLE 2 ON FREE INFO CARD

WHAT'S THE BUZZ

Q *I am trying to record from a mic into my Dell computer mic input (SoundBlaster-compatible) with a Radio Shack dynamic mic. All I get is buzz. I have read that this may not be the correct type of mic. Is that the problem? Also, when I try to record from a telephone line using a Radio Shack recording controller, I get the same thing. Do I need a preamp?*

Bernard H. Doft, MD
via Internet

A Your basic problem is that a dynamic microphone puts out a very low level signal that needs to be amplified up to a level that your sound card can deal with. The cheap way is with a low-impedance to high-impedance audio transformer that trades off impedance for signal level (as all transformers do). You can get one at Radio Shack for about $15. Then it's simply a matter of getting the right adapters (1/4- to 1/8-inch) to get from the transformer into the sound card.

A better way to do this is with an external mic preamp (usually called a "pre"). This will have some sort of balanced XLR input that rejects hum picked up in the mic line, as well as an active amplifier to provide the needed voltage gain. Plus, it will usually include a meter of some sort and 48-volt phantom power (used for condenser mics). Other bells and whistles are things like phase-invert and an input pad for really hot signals. Many also include a tube to soften the transient peaks and add a little harmonic warmth...which can be very flattering. You can get something like an ART tube preamp for around $100, and top-grade mic pres go for as much as $5000 per channel (and more).

The best way to accomplish getting audio into a computer is with an external A/D converter and a preamp. This avoids all the nasty electrical noise inside a computer that makes analog signals very unhappy. I've used systems from Event, Aardvark, Frontier, and Yamaha with world-class results, but you can expect to pay from $300 on up for a basic 2-channel card, maybe $800 for a basic 8-channel card, and thousands for a professional setup such as Pro Tools. These always sound better than a cheap sound card that came with your computer, since it was probably optimized for game sounds, and never designed to be a serious recording tool.

As far as recording from a "recording controller," you should be getting sufficient level from it. Are you sure you have the proper recording device selected in the audio-controller window? In my Win 98 system, when you click on the speaker icon on the tool bar you'll get a window that allows you to select mic, line, CD, or WAV inputs, as well as their recording levels. Make sure you have the desired input "checked" and the gain turned up on that virtual "fader."

Hope this helps....

Mike Sokol
JMS Productions Inc.
jmsokol@intrepid.net

TO DI, OR NOT TO DI

Q *I do sound for my church, and we have the bass and the keyboards plugged into transformers to connect them to the snake. I don't seem to have problems with it, however, would I be better off connecting to DI boxes? What would be the difference in plugging an acoustic/electric guitar to a DI box as opposed to a transformer?*

Mike
via Internet

A Passive DI boxes are basically audio transformers with extra switches that allow you do tricks like lift a ground or attenuate a signal that may be too hot. So, if your transformer sounds OK, then going to a passive direct box won't improve the sound. But an active DI box uses an FET or tube to provide a high input impedance that will reduce loading effects on the instrument pickup. This isn't normally a problem for active pickups and output drivers in keyboards, but it can make a significant difference in the sound from, say, a P Bass or acoustic guitar with a passive contact pickup.

Usually loading from a passive pickup will result in a loss of highs and definition. So go borrow an active DI box and try it in place of your transformer. Depending on the pickup and the instrument type, it may or may not improve the sound. But usually the clarity improvement on a passive bass guitar can be quite noticeable, and an acoustic guitar with a good active DI can actually shimmer. Electric guitars with passive pickups are the best candidates for an active DI box, since loading them with anything less than a megohm can cause significant timbre changes and loss of level.

Mike Sokol
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PREAMP IMPEDANCE MATCHING

Q *I use a TASCAM MX-4 mic preamp for live recording because it has a stereo bus. I also have a TL Audio Indigo series mic preamp, which has four warm and spacious-sounding (at least to my ears) mic pres. The TL Audio mic pre has an output impedance of 47 ohms; the TASCAM MX-4 has an input impedance of 2 kohms.*

Is it advisable to use the TASCAM just as a line mixer connecting the outputs of the TL Audio pres to the inputs of the MX-4, or will I have some sonic degradation? I hear some slight difference, but I'm not sure in which direction this difference goes. Could you give some general rules about impedance matching? Do you have other suggestions how to get good microphone sound from more than two microphones on two channels?

Robert Jaroslawski
Freiburg, Germany

A I have worked for TASCAM for ten years, and yet, when I read your question, I was not familiar with the MX-4. After a search for information, I discovered why I was not aware of it: TASCAM did not distribute the MX-4 in the United States. I found an owner's manual and discovered what a useful tool it can be. The MX-4 is a great line mixer and/or mic preamplifier, but it's not doing justice to the TL Audio mic pre you are using.

Here is a general rule to follow regarding impedance matching: Impedance is the total opposition a part presents to the flow of signal. The outputs of circuits have an impedance rating, and so do inputs. What's best depends on the direction of signal flow. Generally, the output impedance should always be as low as possible. A circuit with a low output impedance will have no trouble driving a variety of inputs.

In other words, the TL Audio mic pre (47-ohm output) is not being used to its best sound quality potential running it into the MX-4 (2-kohm input). In theory, the TL Audio mic pre should be routed directly to the multitrack to take advantage of its great sound, but then you would be using four tracks instead of two. What I gathered from your question is that the TASCAM MX-4 is being used in place of a line-level mixer to sum four mic signals into two channels or tracks of a recorder. If it works for you and the sound is acceptable, do it. As much as the audio community likes to hold up rules to follow, this is all about art and accom-

plishing a task. If that's how you want to do it, and it doesn't blow anything up, go for it.

Karl Moet
TASCAM
www.tascam.com

A *TL Audio responds:* Generally speaking, for best audio signal transfer, a device should have a low output impedance and should drive into an input device with a much higher input impedance. Line outputs will normally have impedances of less than 100 ohms, while line input impedances will be much higher, e.g., 10 kohms or more.

However, the Indigo 2001 preamp can happily drive input loads as low as 2 kohms, the only difference being that there will be a very slight loss of level. You will need to be careful about setting gains, though. You may need to reduce the output gain of the Indigo preamp and/or reduce the input gain of the TASCAM unit to prevent overload occurring in the input stage of the latter.

Howard Jones
Technical Sales Manager
TL Audio Ltd.
www.tlaudio.co.uk

DRUM MIKING HELP

Q *I'm supposed to be the technician for a band, but I need a few tips to get me going.*

I can set up the equipment, but there's the problem with the drums: I don't really know how to mic a drum set. I have a 12-channel mixer in front of me, and the drums take up six of the channels.

Could you also provide a few tips about live performances in general? One problem I had was that the guitarists with their own amps would crank up the volume so high that I couldn't route them through the mixing board; hence, they drowned out the rest of the band. They say the amps give a better sound than a mixer. How do I control the overall balance of volume?

Gareth
via Internet

A Check out some of my columns I've written on the subject via my Web site (www.soundav.com). Some of those should help with "adjusting" the musicians' attitudes about turning up their guitar amps.

In terms of miking drums, if everything else on stage is at a reasonable volume, then you can do drums with a kick, snare, and overhead mic (three channels). This works especially well in a club setting with a tight stage where you can control the instrument volume. If, however, you want that big, isolated drum sound on a big, loud stage, then you either put a separate mic on each drum or put a trigger on each drum and use sampled drum sounds. For major concerts, I normally use ten channels (and mics) for drums: kick, top snare, bottom snare, hihat, rack 1, rack 2, floor 1, floor 2, overhead 1, and overhead 2. You can usually split the rack mics over two drums if you have four rack toms. If not, then it's 12 channels for drums. If you add in a few specials for chimes and gongs, you can easily eat up 16 channels. Furthermore, of course, you'll need to add noise gates to most of those channels, and a few compressors are good to have. That's how the big shows do it, and why their live drums sound so awesome.

Mike Sokol
JMS Productions, Inc.
jmsokol@intrepid.net

SURROUND MICS

Q *Do you have any experience with various microphones built to capture surround sound using a single unit? I have read about a few, but don't know where to check them out or buy them.*

Gary Reece
via Internet

A I've been experimenting with the Crown SASS-P, which is a binaural-style mic, along with the AMS Soundfield quad-capsule mic. Both of these can be decoded via external processors to derive true "surround" using two outputs from the Crown or four outputs from the AMS in B-Format mode. The Crown SASS-P can have rear surround "extracted" from the binaural signal. The AMS Soundfield, on the other hand, is a true "steerable" mic that can be selected in postproduction to act like an omni, cardioid, steerable shotgun, M-S stereo, or full surround mic from a "point" source capsule. Technically, it has to be the coolest mic I've ever played with.

Hector La Torre, from Fits & Starts Productions, and I will be adding a "Microphone Techniques for Surround" workshop to our regular "Surround for Project Studios" seminar. Stay tuned to my Web site for details as our West Coast Road Tour is now in the planning.

Mike Sokol
JMS Productions, Inc.
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CIRCLE 25 ON FREE INFO CARD

EQ NEWS
JANUARY • 2000

MusicPlayerNetwork to be Launched at NAMM

Miller Freeman PSN and GPI Music Group, the world's largest family of active musician magazines, has scheduled the launch of MusicPlayersNetwork.com — the definitive online resource for musicians — for the 2000 NAMM show in Los Angeles.

The super site will be the mothership for content from the editors of *Guitar Player*, *Keyboard*, *Bass Player*, *EQ*, and *GIG* magazines. A variety of exciting innovations are also planned by the site's online content editor Craig Anderton (well known for his landmark Studio, Sound & Stage Web community formerly on AOL). GPI music group editorial director Michael Molenda will oversee the project with Anderton. Angelo Biasi, Miller Freeman PSN e-business development director, is the site's director of sales & marketing.

MusicPlayersNetwork will include a mega search engine that encompasses all Network affiliates, as well as product reviews and updates, music lessons, daily industry news, downloads, community chats, product-specific forums, personalization options, and much more.

"Our company is making a great financial commitment to build the most active network of musician sites on the Web. This major collective effort between our core music magazines is an exciting venture for the online music community," states Paul Gallo, president of Miller Freeman PSN.

Adds GPI Music Group publisher Ed Sengstack: "Anybody can build a music Web site, but nobody has our content or our history of editorial integrity. This site will ultimately draw musicians from all over the world into a unique music community, and will have a tremendous impact on the future distribution of music information."

In addition to the core titles initially involved in the MusicPlayersNetwork, other Miller Freeman PSN properties — including *Gavin*, *Pro Sound News*, *Studio Sound*, *www.musicgearonline.com*, *EQ Buyer's Guide Online*, and *Surround Professional* — will contribute content to the site. MusicPlayersNetwork is also entertaining a limited number of Network affiliations and advertising commitments.

Take a Virtual Reality Tour of Hafler's Site

Hafler's new VR Studio (at www.hafler.com) is a collection of virtual-reality movies that support 360-degree views of its products that allow viewers to experience Hafler products in real time by seeing objects from all sides. Now sound contractors, engineers, producers, studio owners, and musicians shopping for new amplifiers and reference monitors can get a close-up view of all Hafler products.

"Hafler's VR Studio lets our customers get a close up view of amplifiers and reference monitors from the convenience of their home, office, or studio," explains Hafler Web wizard and technical writer Eric Russell. "Customers can also zoom-in on controls, connectors, and LEDs, as well as click on hot spots to see the inside circuitry and craftsmanship that goes into each and every Hafler product."

Jerry Cave, managing director for Hafler, says, "This is a great first step for dealers and potential customers to see what our products are all about. Not only can they see the necessary specs on a particular amp, active monitor, or subwoofer on our Web site, but now they can see how the components are actually laid out and visualize on a 3D scale how the gear looks from all sides."

For further information on the Hafler VR Studio, please visit www.hafler.com/vrstudio.

Multimedia Education Becomes More Financially Accessible

In a move that makes the multimedia and audio industries' hands-on training and education available to as many students as possible, SAE Institute, the world's largest network of audio and multimedia education and training centers, has announced it has teamed up with the Sallie May Financial Corporation to provide potential students with a flexible financial aid program. SAE students at the Nashville and New York City institutes are immediately eligible for aid through the program.

"Students who have always dreamed of a career in audio, film, video, and multimedia, now, for the first time, have the opportunity to realize that dream," says Marcel Gisel, SAE vice president of U.S. Operations. "This program is truly unique in that it exposes the greatest reservoir of talent to the very best education and training. This promises to further improve the quality of graduates entering the industry."

For more information, call SAE at 212-944-9121 or visit www.sae.edu.

Yorkville Sound Acquires A.R.T.

Yorkville Sound has announced the acquisition of Applied Research & Technology (A.R.T.). With the new ownership, A.R.T., manufacturer and distributor of electronic equipment for the recording and live music industry, will be able to invest in key areas of the business, in order to stimulate production and sales, as well as new product development. A.R.T. will continue operations at its existing Rochester, NY location.

"We could not have hoped for a better deal for our customers and employees," says Phil Betette, president of A.R.T. "We keep our location, our people, and most of our existing business ties."

For more information, call Yorkville Sound at 905-837-8550.

"Blinded" Winner has Clear Vision

After his online remix of Thomas Dolby's "She Blinded Me with Science" snared him an $8000 scholarship, Ex'pression Center for New Media student Ned Creed came away with a clear vision of his future. That vision places him squarely behind a computer audio workstation or mixing console.

Creed, a casual guitar player with no recording or mixing experience before his enrollment at Ex'pression, was part of a groundbreaking program and partnership between Beatnik, Inc., a leader in interactive audio technology, and Ex'pression Center for New Media, a multimillion-dollar, high-tech, multimedia and sound arts school located in Emeryville, CA.

To launch the new program, which was designed to advance the study of interactive Web music and sound in a series of new certification courses (dubbed "The Art and Science of Sonification"), Dolby offered an $8000 scholarship to the one student who could remix his hit song. The "Blinded" contestants created their remixes online using the Beatnik Player interactive Web musical tool, a standard browser, and song elements that were posted on the Yahoo! Digital Web site.

Open to all Ex'pression students, the contest was held during the MusicBiz 2005 conference, a music futurists event that was hosted by the Ex'pression Center. Scholarship-winner Creed topped the list of 13 prize recipients. Second-prize winners received $100 gift certificates from Tower Records, while third-prize winners received Beatnik berets and T-shirts.

For more details, call the Ex'pression Center at 510-654-2934 or visit www.xnewmedia.com. Call Beatnik, Inc. at 650-295-2328 or visit www.beatnik.com.

108th Annual AES Update

Paris, France will be graced by the first Audio Engineering Society Convention of the new Millennium. During February 19–22, countless audio professionals from around the world will attend the Convention. AES executive director Roger Furness reports that a stellar combination of cutting-edge technologies, papers, workshops, unique special events, and exhibitors will converge to help guide attendees through the industry's constantly changing terrain of trends and innovations.

The Committee has developed 19 Paper Sessions featuring over 100 individual papers, covering subjects ranging from "Active Control of Noise by Wave Field Synthesis" to "Audio Watermarking of MPEG-2 AAC Bit Streams," "Real Time Streaming of Multi-channel Audio Data through the Internet," and "The Largest Music Studio in Russia."

Fifteen challenging workshops have been designed to provide attendees with opportunities for in-depth study of established and emerging technologies and to address issues of serious consequence to the industry. Some of these sessions include: "Multichannel Sound in the Cinema," MPEG-4 Version 2 Audio — What is it About?," and "Audio On-Line: Should Networks Handle Audio?"

Other planned happenings for the show include tours of famous French studios, excursions to notable performance venues, and much more.

For more information and a constantly updated conference program, or to register online for the Convention, surf to the AES Web site at www.aes.org.

Since we introduced the MX-2424 hard disk recorder, there has been a lot of speculation about its price (which is so low it seems too good to be true).

So we get questions. Like...

"24 tracks is an upgrade?" (No, it's 24 tracks right out of the box.)

"24-bits is an upgrade?" (No, all the bits are there too.)

"Do I have to pay extra for inputs and outputs?" (No. At $3,999 estimated street price* you get a full set of 24 TDIF-1 or ADAT* optical digital inputs and outputs — plus an assignable stereo AES/ EBU - S/PDIF pair. For a little more you can get 24 channels of AES/EBU digital I/O, or analog — or both digital and analog!)

"Does it need an external computer?" (No. The MX-2424s front panel has a full set of professional transport, editing, and track assignment controls, including a shuttle/ scrub knob. So you don't have to have a computer to run it. But — if you happen to own a Mac or a PC, you can take advantage of the digital audio editing and control software that comes standard with each MX-2424 to do even more. Your choice.)

"Before I start recording do I need to buy a monitor, a keyboard, or a hard drive? Or anything else?" (No. Nyet. Nope. Not at all. Just hook up power and start recording.)

So let's make this as plain as we can: The MX-2424 is an amazing, full-featured professional 24-track digital recorder. And there's never been anything like it at this size or price.

Its sonic performance is outstanding. Lots of companies claim 24-bit 48k performance, but only the MX-2424 is part of TASCAM's M Series family of multitracks — the products chosen for their sonic performance by such discriminating facilities as Skywalker Sound, Universal Studios, and 20th Century Fox.

$3,999 ESP*.

Superior reliability is guaranteed. The MX-2424 was designed from the bottom up to be a great recorder, and nothing but a great recorder. Its processors and circuitry are fully optimized for audio - not video games, spreadsheet software, or surfing the web. And isn't that absolute focus and rock solid performance exactly what your music deserves? Over the last three decades we've designed and built literally millions of professional recorders and recording systems; the MX-2424 is the culmination of everything we've learned.

So easy to operate, you could do it blindfolded. Of course that way you'd miss the great light show from the 24 tracks of level metering and channel status displays... but the real point here is simplicity. When you want the MX-2424 to start recording, just reach over and press REC + PLAY (just like a traditional tape recorder). In a fast-paced production environment, you can record to hard drives that mount into standard Kingston® carriers and plug into the front panel drive bay. Just pop in a new drive at the start of each session. It doesn't get any simpler than that.

It's a complete professional hard disk multitrack in a portable, affordable, rack-mount box. You can plug it in, turn it on, and start recording.

✔ Back panel ports include Fast/Wide SCSI, ethernet, MIDI, RC-2424 remote, and TL-BUS!

Extend your reach — Want a remote control? Get the one that's made to take advantage of the power in your MX-2424. The RC-2424 remote is a powerful, professional multi-machine controller with all of the MX-2424's front panel features, plus macros and more.

Really.

The power to meet your needs. A standalone MX-2424 is an incredibly powerful unit, with enough internal hard disk capacity to hold about 45 minutes of 24-bit 24-track audio. The MX-2424's Fast/Wide SCSI port lets you connect up to 15 external drives and record directly to all of them. And if you need more than 24 simultaneous tracks, just add additional MX-2424's. Up to 32 MX-2424's can be locked together in *sample accurate sync* to act as a single recorder.

Professional recorders need to interface with increasingly complex systems.

✔ It provides video and time code lock capabilities as standard features, making it easy to integrate with external workstations.

✔ It resolves to AES/EBU, S/PDIF, word clock, TDIF-1, ADAT optical, SMPTE Time Code (LTC), and video, and chases MIDI Time Code.

✔ Available Input/Output modules include TDIF-1, AES/EBU, ADAT optical, and analog.

MX-2424 shipments are about to start, and there is already a waiting list. To get yours sooner instead of later, contact your authorized TASCAM dealer!

So... what's this Estimated Street Price?* **Instead of quoting you some meaningless "List Price," ESP is what we expect typical U.S. customers to actually pay for an item. It gives you a better way to compare value when you shop.

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EQ PRODUCT VIEWS

FOR REAL

Kind of Loud Technologies' RealVerb 5.1 multichannel reverberation plug-in for Digidesign's Pro Tools uses auralization technology to model physical spaces. RealVerb 5.1 can morph between room shapes and textures, giving users creative control over their sound. The RealVerb 5.1 also has the ability to map reverberation spatially for surround mixing. RealVerb 5.1 allows the user to independently place the direct path, early reflections, and late field reverberation in the soundfield. For more information, contact Kind of Loud Technologies at 831-466-3737 or visit their Web site at www.kindofloud.com. Circle EQ free lit #105.

GREATER REVERBERATOR

Sony's new DRE-S777 sampling digital reverberator is a digital effects processor that re-creates the "real" ambience reverberation of various sound environments. The DRE-S777 uses processing that allows audio signals to be combined with sampled data taken from real reverberant sound spaces, collected by Sony engineers who have sampled data from some of the world's most highly regarded halls, studios, and other acoustic environments, plus a classic analog plate reverb. The reverb accepts input signals with 24-bit precision and at sampling rates up to 96 kHz. Coupled with multichannel surround sound capabilities, this audio resolution extends the applications for the DRE-S777 to cover a range of modern production tasks. The DRE-S777 offers a multichannel output, and, with the optional DABK-S703 DSP board, allows five-channel surround sound effects to be delivered. For more information, call Sony at 800-686-SONY or visit www.sony.com/proaudio. Circle EQ free lit #106.

AMP CAMP

QSC's new RMX Series of amplifiers includes a three-model line offering the popular power points of 830, 1400, and 2400 watts in a low-profile, 2-RU chassis. The RMX 850, RMX 1450, and RMX 2450 feature high-current torodial transformers for solid 2-ohm performance and low noise. The amplifiers include low-frequency filters to boost system response, protect speakers, and ensure low-end tightness by more precisely matching the amplifier's range to the loudspeakers. Each channel can be independently adjusted — 50 Hz for most compact full-range systems, 30 Hz for subwoofers and large full-range systems, and full-range for studio monitoring. Independent user-defeatable clip limiters are also included to reduce distortion without sacrificing performance. Other features include: front-mounted gain controls for easy access; signal and clip LED indicators to monitor performance; binding posts and Neutrik Speakon outputs; independent DC and thermal overload protection on each channel to automatically protect speakers and amplifier; and a continuously variable speed fan with back-to-front air flow to keep amp and racks cool. For more information, call QSC at 800-854-4079, fax them at 714-754-6174, or visit www.qscaudio.com. Circle EQ free lit #107.

EQ PRODUCT VIEWS

START YOUR ENGINE

Yamaha has introduced a programmable DSP engine, the DME32 digital mixing engine. The DME32 is a 32-input, 32-output processor that is expandable to 128 inputs and 128 outputs with 32 cascade busses. High sonic fidelity is maintained with 32-bit resolution and operation at a 48 kHz sampling rate. Parameter adjustments, scene changes, and other functions can be accessed from the front panel of the 3U rack-space unit. With four MY card slots, a wide assortment of digital and analog I/O is available. The DME32 can be expanded to 32 inputs and 32 outputs either by adding remote I/O components — the AD824 (input x 8) or DA824 (output x 8) — or by utilizing Yamaha's MY8/MY4 series mini-YGDAI interface cards and plugging them into the unit's four mini-YGDAI card slots. For more information, call Yamaha at 714-522-9011 or visit www.yamaha.com/proaudio/. Circle EQ free lit #110.

Y6K-COMPLIANT

TC Electronic introduces the System 6000 multichannel audio processing system. It is targeted specifically towards music, film/post-production, broadcast, and mastering applications in surround environments. System 6000 algorithms include the industry standard VSS reverb technology in a newly developed multichannel version, called VSS-5.1. VSS-5.1 is a multi-source input to multichannel output space simulator that incorporates positioning generators and five totally uncorrelated reverb diffused fields. Multichannel, multiband dynamics processing is available for System 6000. Four-band processing of multichannel audio can be realized; including expander, compressor, and final sample accurate brickwall limiter. Sample rates up to 96 kHz are covered. For mono or stereo signals, the MD2 algorithm from the M5000 has been improved to MD3, incorporating dual mono modes, sample-accurate brickwall limiter, and 96 kHz operation. The TC ICON remote is the System 6000 remote user-interface controller. The System 6000's retail price start at $8000. For more information, call TC Electronic at 805-373-1828 or visit www.tcelectronic.com. Circle EQ free lit #108.

MASSIVE PASSIVE STEREO TUBE EQ

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Perhaps, but would photographs of our Variable Mu or VOXBOX have created their successes alone?

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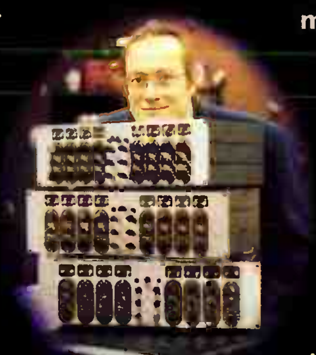
Engineers who have already gotten hold of the MASSIVE PASSIVE have told us: "Why does it make everything sound so much better?", "It's organic and orgasmic.", "It's a f%#king powerhouse.", "It's unlike <u>any</u> other EQ.", "This is IT. The sound I've always dreamt of but couldn't ever get until now."

GOT THE PICTURE?



Craig 'HUTCH' Hutchison designed these monsters... The **MASSIVE PASSIVE** is a two channel, four band equalizer, with additional high pass and low pass filters. "Passive" refers to the tone shaping part of this clever new EQ design not using any active circuitry. Only metal film resistors, film capacitors and hand-wound inductors sculpt the sound, kinda like a Pultec EQ on hyper-steroids. Super-beefy, hugely-high-headroom Manley all-tube make-up gain amplifiers deliver your tunes into the next realm. You'll need to experience this.

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Great Egg-spectations

Engineer/artist Meg Lee Chin makes attention-getting mixes from her London studio/bedroom

STUDIO NAME: The Egg Studio (www.yolk.demon.co.uk/eggstudio.html)
LOCATION: Soho, London
KEY CREW: Meg Lee Chin, owner
CREDITS: Ten Benson: "City Hoppers," "Dead Celebrities," and "Transport Overseas." Pigface: album track "Nutopia" from *A New High in Low.* Ali Zapatak: "Angel Station." Crunch: *Bootleg,* released in the former USSR. Invisible Records: "Scarecrow" album track from *Wish You Were Queer* Ministry cover album. Meg Lee Chin: *Piece and Love,* Meg's debut solo album
CONSOLE: Yamaha ProMix 01 (18 channel automated mixing, built in effects (Yamaha SPX990 [2], Yamaha Compressors [3])
MONITORS: Tannoy PBM 6.5 monitors.
COMPUTER: Pentium II 333 MHz 128 SDRAM; Barracuda SCSI drives [2]; 2.1 GB each Ricoh 6200s CD-ROM writer/rewriter; TEAC 32x CD-ROM; Event Gina PCI audio card 2-in, 8-out analog plus S/PDIF in and out (4-in and 10-out); Jaz Drive
SOFTWARE: Cubase Audio VST; ACID; Wavelab; Sound Forge; Rubber Duck
OUTBAORD GEAR: DigiTech Valve FX; TLA Valve preamplifier and compressor
SAMPLER: Peavey DPM SP Sampler (16 MB RAM)
MICROPHONES: Langevin, AKG C1000s, AKG 451
MIDI: Kawai MIDI controller
RECORDERS: Sharp MiniDisc; Definitely no DAT machine
STUDIO NOTES: Meg Lee Chin states: I started with a Portastudio™ and built it up from there. I've been a hobbyist for years. It began getting serious when I got my Atari Falcon about five years ago. I built my PC about two years ago. Egg Studio is in my bed/livingroom.

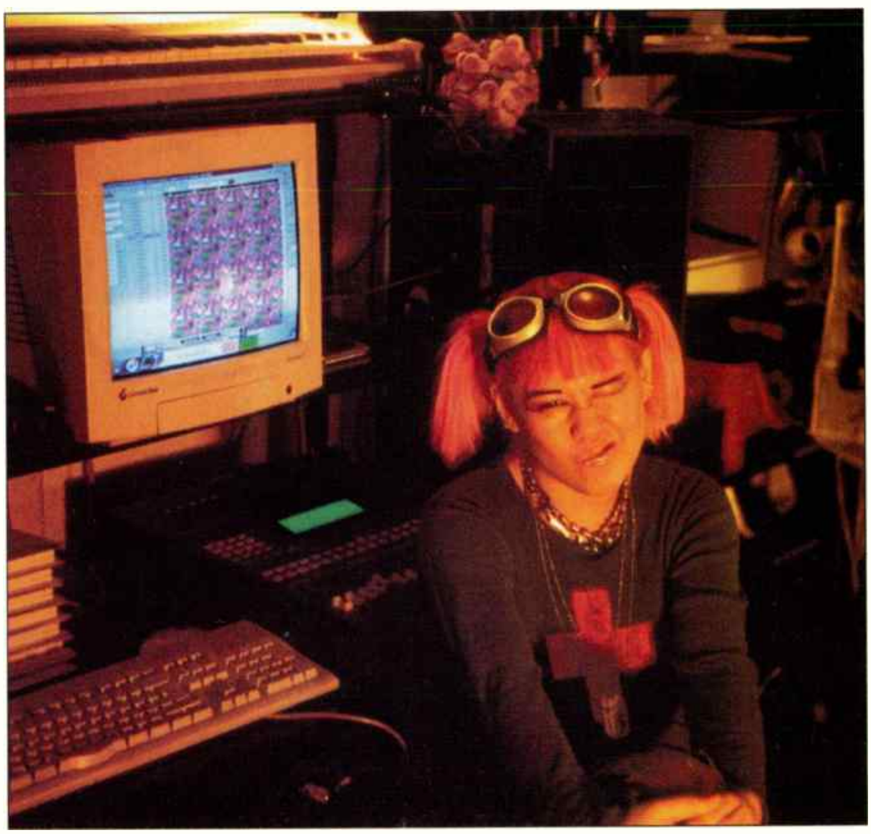
Egg Studio has been built piece by piece over a long, long period of time. It may not look like much, but it's mine and I've done it myself. But just wait till I get my paws on some *real* equipment! (*Drool!*)

CARREER NOTES: Chin continues: I took a sound engineering class at San Francisco State University. I was the only girl. The recording studio had yellow walls and fluorescent lights. I kept falling asleep. Whenever we had projects, some guy would always come over and tell me I was doing it wrong. All the guys acted so authoritative and serious and because I was always sleeping during lectures, I assumed they were right and that I was the dumbest in the class.

When it came time for final exams, we had to book time in the studio and mix a track on our own. As it transpired, the professer was playing back the results and said there was one track that stood out as being noticeably the best of the lot. He played it and it was quite obviously the winner. He looked at the box and his face looked puzzled as if there had been a mistake. Finally, there was no avoiding it. "That track was mixed by Meg." The mouths of everyone in the class fell open." I nearly fell off my chair. This was not difficult, seeing, as beforehand, I had been half asleep!

I had no choice but to learn sound engineering. Except for my brief course, I am self-taught. It was the only way I could ever get to do anything other than be the dolly-bird singer. Guys in studios don't listen to me at all. It sounds a boring feminist cliche, but it's true. Guys in bands just want me to sing husky, suck a lollipop, and flash my knickers. I've always looked kind of young and babyish. I could tell you some hair-raising stories that would make your blood curdle on how I've been underestimated, but I don't want to bum you out! Honestly, it's a never-ending saga, and it never changes. [It's] nobody's fault — that's just the way the world is. People *do* judge a book by its cover, and that applies to both men *and* women.

Consequently, my situation has made me a popular engineer among women. I'm not intimidating in the studio and I let them have whatever they want. I know what it's like. EQ

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Hear **Here**

What may look like a living room is actually an innovatively acoustically tuned listening room

STUDIO NAME: 1130
LOCATION: Chicago, IL
KEY CREW: Harry Witz
CREDITS: Owner/engineer Harry Witz has mixed front-of-house and engineered recordings for a multitude of acts such as Kansas, KISS, Heart, REO Speedwagon, Triumph, and Cheap Trick.
PROJECTS RECORDED: Cheap Trick *Music For Hangovers* (5.1 remix for DVD to be released in February 2000); Smashing Pumpkins; Tori Amos; 311; Poi Dog Pondering
MONITORING EQUIPMENT: M&K MPS-2510 powered monitors [5], powered subwoofer, and LFE Management System; Genelec 1031 powered monitors [2]; Studio Comm VCA Volume Control System
COMPUTERS & SOFTWARE: Apple PowerMac 9600/200 MHz with 256 MB RAM; ViewSonic 15-inch flat display; APS Technologies 4mm DDS3 DAT backup; Yamaha CD Burner; Glyph Tripp system w/two 9 GB hard drives; 8mm AIT DAT backup and CD burner; Glyph Dual 9 GB hard drive; multisync color monitors [2]; Adaptec SCSI Bus accelerator; Digidesign Pro Tools
RECORDERS: TASCAM 122MkIII cassette deck and DA-45HR 24 bit DAT recorder
DIGITAL RECORDING EQUIPMENT: Digidesign Pro Tools Mix Plus and Pro Tools Mix Farm Cards [5]; Mackie HUI control surface
OUTBOARD GEAR: Allan Smart Stereo Compressor; TASCAM IF-AE8 TDIF to AES/EBU converter; Summit TL-100 tube compressors [2]; Spatializer; Lexicon DC-1/ AC3 DTS; Digidesign 888/24 I/O; Apogee AD8000; Lexicon PCM91 reverb; 96 Point TT patchbay
MIDI EQUIPMENT: Opcode Studio 64x interface
POWER CONDITIONING AND BACKUP: APS UPS650 uninterruptible power supply
VIDEO EQUIPMENT: Pioneer DVD 909 and 56-inch wide screen HDTV; Mitsubishi U-80 S-VHS; Sony PVW-2800 Betacam SP deck
STUDIO NOTES: According to Harry Witz, the construction of his new room was precipitated from necessity: "I needed to mix in 5.1, and I couldn't find anyplace equipped to do it here in the Chicago area. So I built 1130. It's a comfortable, 24-x18-x9-foot room with the look and feel of a living room. I can bring in a band for a listening session and we can all be very comfortable. Even though it looks like a just living room, we went through a lot of effort to acoustically tune the room. In addition to pouring concrete on the floor to dampen resonance, we also tuned the room using a couple of interesting techniques. A set of 16-foot wide wooden blinds is located at the back of the room. They are backed with acoustic foam for dampening purposes, so, by opening the blinds to varying degrees, we're able to adjust the reflectivity of that part of the room. Also, there are several doors that lead into the room from small closets. These doors have vents that lead to tuned ports, allowing them to act as bass traps. Our end result is an incredible reference environment where we can really crank up the mix and not worry about bass nodes interfering with the mix process."
PRODUCTION NOTES: Witz also owns Top Note Studio, which is tied via 48-channel, 3-way recording split to the Metro Club in Chicago, IL. Using this arrangement, Witz can record Metro shows live to the TASCAM DA-88's installed in Top Note. The tapes are later transferred into Pro Tools for 5.1 mixing at 1130. Harry is currently in the process of mixing the Cheap Trick 25th Anniversary Concert in 5.1, and has recently recorded and mixed (stereo) live sets by Smashing Pumpkins, 311, Poi Dog Pondering, and Tori Amos. Harry notes that sometimes, "mixing audio can be easier when you can see what was happening on stage at the time, so a lot of times I'll run a Betacam tape from the live performance on the high-def TV here. It helps me focus my mix and also gives me an idea of where (or which musician) a sound may have come from." EQ

Astatic JT-30

Go wide with this harmonica fave

MICROPHONE NAME: Astatic JT-30
FROM THE COLLECTION OF: Joe Manzella, Audio Arts, Colonia, NJ
YEAR OF MANUFACTURE: ca. 1937
PRICE WHEN NEW: $16.95 (catalog price circa 1950)
TYPE OF MIC: Crystal transducer
POLAR PATTERN: Nondirectional
FREQUENCY RESPONSE: "Wide range" (See notes.)
OUTPUT LEVEL: –52.0 dB
OUTPUT CONNECTOR: Concentric two-conductor
DIMENSIONS: 2-5/16 diameter x 3-1/16 deep (inches)
MIC NOTES: The Astatic JT-30 was part of Astatic's JT-Series of crystal- and ceramic-transducer microphones, which also included the JT-40 (crystal), and JT-30C and JT-40C (ceramic) models.

According to an Astatic technical bulletin from the time period: "The terms 'voice range' and 'wide range' refer to the frequency response characteristics of Astatic microphones. 'Wide range' microphones have a flat frequency characteristic that makes them suitable for voice and music, and makes reproduction sound quite natural. 'Voice range,' on the other hand, emphasizes the frequencies above 500 cycles per second, which greatly increases the intelligibility of speech, although reproduction is not quite so natural as with wide range. The increase of high-frequency response with respect to low frequencies makes possible greater effective output in terms of intelligible speech. This means that a public address amplifier or radio transmitter of a given power rating will deliver greater useful output on speech when used with the voice range model mic."

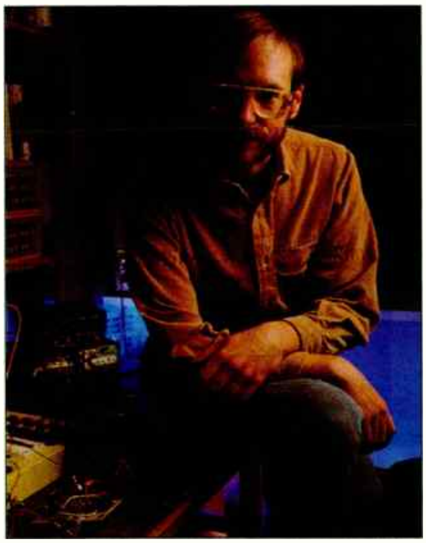
While the JT-30 was considered to be a "wide range" microphone, the JT-40 was classified as a "voice range" microphone; the two models shared the same external case.

USER TIPS: Due to its compact size and shape, the JT-30 has been and continues to be a very popular microphone for handheld use by harmonica players. Generally, the player forms a cup with their two hands around the JT-30. The harmonica is positioned across the front of the microphone, and held in place by the thumbs. EQ

PHOTO BY EDWARD COLVER

The Mixing Protocol

How to sonically get your mix ready to be mastered



BY CRAIG ANDERTON

As we all know, mixing is very easy: you simply move the faders around until everything sounds wonderful. Well, at least that's the theory; in practice, multiple complications enter the process.

Although I've seen many articles devoted to specific aspects of mixing, such as adding reverb, altering EQ, and the like, I haven't seen much that addresses the process itself. One of the main questions I receive goes something along the lines of, "How can I get my music to sound professional?" So let's look at one way to approach doing a mix.

THE ACOUSTICS

Before even thinking about mixing, assess your listening situation. Most home studios are not acoustically treated, which just about guarantees response anomalies. For example, if you're not aware of some low-end cancellation that weakens the bass response, you'll mix the bass too high. When you play the mix back over a system with a more accurate frequency response, the bass will be overly prominent. More ominously, if you have hearing problems (such as high-frequency loss), you'll mix to compensate for those problems. When I hear mixes with really sizzling high ends, I sometimes wonder whether it was mixed by someone whose high-frequency hearing was blown out by mixing too many live gigs without suitable ear protection.

There's not much you can do about hearing-related problems, other than test your hearing to discover any problems and compensate for them. In terms of acoustics, though, many smaller studios prefer using nearfield monitors because, in theory, these take acoustics out of the equation. When your ears are close enough to the speakers (typically 3 feet or so), room reflections become a negligible factor in the overall sound. Of course, not all nearfield monitors are the same, as a quick listening test of different models will confirm. You need to "learn" your speakers, and the best way to do this is to listen to some material you know really well over a truly accurate set of speakers at someplace like a professional mastering studio. Listen carefully for any variances with respect to the sound in your studio. And remember that some nearfield monitors simply cannot reproduce sounds in the 40–60 Hz range, so if you have a lot going on down there, having an accurate reality test becomes even more critical.

There's always the old tip about A/B'ing your mix with that of a well-mixed CD and aiming for the same kind of spectral balance as the reference CD. If you do this, though, level-matching between the two sources is crucial. If one source is slightly louder, you'll hear more bass and treble because of the Fletcher-Munson effect (i.e., the ear's high and low-end response drops off at lower volume levels). Match peak rather than average levels, as that will also indicate whether you have too little or too much dynamics processing. If your mix reaches the same peak levels but sounds much softer than the reference CD, you could probably use a little compression or limiting to bring up the level. Conversely, if your CD sounds much louder, you're probably using too much compression. Yes, ultra-loud CDs are kind of a kick, but the lack of dynamics wears out its welcome after a while.

TOO LOUD OR NOT TOO LOUD?

One of the biggest mistakes made by inexperienced engineers is to mix at really high levels. Because of the ear's physical construction and logarithmic response, it's harder to judge fine level differences at high volumes. Here are the steps I use in setting up a mix:

1. Listen at moderate volume through headphones to catch any ticks, pops, glitches, throat-clearings, etc. that may not be as noticeable over speakers. This often involves listening to each track through headphones and correcting any small problems in a digital audio editing program.

2. Listen through headphones at a barely audible level. I mean really, really quiet levels — just above the threshold of hearing. At these extremely low levels, gross level mismatches become super-obvious. Make any level adjustments needed to iron out these differences. However, do not make any decisions about bass, as the ear's response is not accurate at these super-low levels. The one exception is if the bass really blasts through, because then you know it's way too high.

Generally, I start listening midway through a tune, where the levels are more or less established. This gives a more representative idea of the average levels. Then, when you return to the beginning, which will often be quieter, you'll be able to assess it in the context of the overall tune.

3. After getting the levels more or less balanced at low levels, switch over to speakers at a barely audible level. The overall mix should be similar to what you heard on headphones. If there are variations, split the difference — for example, if the vocals seem just right on 'phones but somewhat loud over speakers, back off the volume just a tad so that vocals seem just a little under on 'phones, and just a little over on speakers.

When changing levels, a mistake beginners often make is to do fairly major changes, like boosting or lowering something 3 dB or so. Three dB is actually a really major level change, and, by making big changes, you'll upset the balance and need to change other channels to

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compensate. Even a 0.1 dB level variation for a channel makes a subtle, but noticeable, difference in the overall mix. Make very small changes as you listen; if you think something needs to come up, bring it up a tiny amount, then listen for a while to "settle in" to the new levels before you decide whether to bring it up any more or not.

4. Now turn up the speaker volume nice and loud. Your ear's response is flattest at higher levels, so this is the time to pay particular attention to the bass and treble regions. Make sure your bass is strong without being overpowering, and your high end bright and balanced, without being screechy.

5. At this point, the levels are pretty much set. I then play back at a moderate, comfortable listening level, and run through a tune several times, listening for a different aspect of the mix each time. For example, on one pass I might listen for any clicks or other glitches that need to be excised. On another, I might concentrate on just the EQ, and so on. If you try to listen to too many things at once, you may miss one problem while you're listening for a different problem.

6. Next, I play the tune as background music while I tidy up, do routine maintenance, or whatever. Placing it in the background, while you're not paying as much attention, allows anything really strange to jump out at you. For example, suppose you mixed an instrument a little too low. When you're concentrating on listening, you'll hear the instrument because you're playing close attention, so you might not realize it needs to come up a bit. When played as background music, the instrument will tend to drop out of the mix, so you'll know to raise it up a bit.

Similarly, many musicians like to play tunes while driving. My theory is that the road noise provides enough ambient level to mask low-level sounds, causing them to drop out of the mix, and thereby alerting you that you might want to make them louder. I've noticed the same phenomenon when I turn on the air conditioning while mixing; it makes enough noise that it will mask sounds that are too low. I'll often go back and raise the level of softer instruments somewhat so that they can be heard over the ambient noise.

7. The final test is to burn a CD and play it over a variety of systems, in a variety of listening environments. This shows whether the mix is "transportable," i.e., whether it can play over different systems and still survive more or less intact. If, for instance, on seven out of ten systems the bass sounds too low, then bring it up just a tad in the mix (ah, the beauty of automated mixes — it's so easy to go back and make these kinds of tweaks!).

Once the mix passes the transportability test, you're done — except, of course, for the mastering. But that's a whole other can of worms, which we'll open some other time.

Craig Anderton is the author of the classic text Home Recording for Musicians *(available from www.musicbooksplus.com), and has played on, mixed, or produced 15 major-label releases. His latest CD,* Sexy World, *is also available in Acid loop library format. Send e-mail to Craig at www.craiganderton.com.*

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CIRCLE 73 ON FREE INFO CARD

Resolute Resolutions


PHOTO BY JONATHAN POSTAL

Get ready for a whole new Al in 2000 (well, maybe not a whole new Al, but definitely a partially new Al…)

BY AL KOOPER

Well, if you're reading this, we're still here and we have to deal with that. Towards that end, I've made Ten Millennium Resolutions.

1. I will take advantage of technology and resize my project studio. I don't need my gigantic TASCAM 3500 console anymore. It takes up too much space. However, I'm never gonna get back the $7K it cost. But friends say, "You got great usage out of it — that's worth the money." *I don't see any of them offering to buy it.*

I'll be going digital with the console, natch, and I'll have to wade through all the possibilities and hope I make the right choice. Same with the storage devices. I've had a DA88 and two 38's for a few years now, and have not had a lick of trouble with any of them. But I must resize in 2000, so I'll look into one of those all-in-one 24 tracks-on-a-hard-drive-candies. Wish me luck.

2. I will replace my 5-year-old Apple 7500 computer with a G4 400 MHz with a DVD-RAM drive and a 13 GB hard drive.

They must plan the obsolescence at Apple, cause my computer *and* my Apple Monitor are both in assisted living mode as we speak. So they got me now. When I do break down and get the G4, none of my SCSI peripherals will work unless I buy cabling that costs more than the peripherals. I will become a USB boy involuntarily. Then I need another hard drive if I wanna go FireWire. What a world! Of course, my favorite old shareware games will cause my new G4 to lose it's new-fangled mind and crash, and I'll be forced into making John Madden rich like the rest of the world. Well, at least I can watch *Dr. Strangelove* in my office with amazing clarity anytime I want.

3. If my royalties are kind to me, I will buy my own personal Phonolog. Most of you are going, "What in hell is he talking about? A phonograph that looks like a log?"

Nope. It's an amazing research tool that had its heyday in the '50s and '60s. A ridiculously *huge* book that chronicled every song and artist that was ever released. *The pedestal the book sits on is $300 just by itself.* You would see these books in record stores behind the counters. Sort of the anal-retentive *Schwann Catalogue.* Well, I wanna be able to hear a song and look it up and see who is singing. No senior moments in the Kooper household anymore. So, in the year 2000, hopefully, there will be a Phonolog in the house.

> *Resolution #5:*
>
> *I will become twice the Gear Sl*t Roger Nichols is, but find a nicer phrase for it.*
>
> *Resolution #10:*
>
> *I will do my level best to inform and amuse my loyal, wonderful EQ readers no matter what the letters to the editor say.*

4. I will sell CDs and T-shirts from my Web site. I know I will never venture into the valley of the music business again, so I better come to grips with getting my "new" music out there. I prefer the CD with the booklet to the MP3 thang. So, in the year 2000, this will happen. Then, hopefully, I can unload the myriad shirts I have left over from my 50th birthday party-recording-session-subsequent-album-that-went-out-of-print-in-1999.

5. I will become twice the Gear Sl*t Roger Nichols is, but find a nicer phrase for it.

6. In 2000, I will endeavor to not offend as many people as I did in 1999, but still more than I will in 2001.

7. If "the '70s rock star look" is what is setting the jewelry world on fire in 2000, then it might behoove me to design some jewelry this year.

8. I will kill my PlayStation.

9. I will *not* produce the *Kenny G plays Coltrane* CD or the *Kid Rock Does Hendrix* CD, but I will consider an offer to do the Kenny G/ Kid Rock duet CD.

10. I will do my level best to inform and amuse my loyal, wonderful *EQ* readers no matter *what* the letters to the editor say.

EQ

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CIRCLE 61 ON FREE INFO CARD

Bob Mothersbaugh

PHOTO BY ED FREEMAN

A talk with Bob #1 of Devo

BY MR. BONZAI

Bonzai: I understand you're doing a spectacular campaign for McDonald's — how's it going?
Mothersbaugh: Don't ask. Don't tell. [*Laughs.*] I'm nine days past the deadline.
What are you going to do?
Keep sluggin' it out. It's a 45-minute cartoon they sell at the stores. We've done three — this is number four.
In your own studio here, what are your main recording tools?
Mackie console, 32-channel, 8-bus, with three sidecars. I usually record to Pro Tools, and I have some DA-88's, but they are a couple of milliseconds slower than Pro Tools.
What are these guitars you have here?
Here's a reissue 1952 Telecaster, a 1980 Guild X1-70 hollow body, a reissue Rickenbacker George Harrison model, and my favorite here is a 1964 Gibson SG — really lightweight with a really thin neck — you don't have to work hard to play it.
Amplifiers?
I've got a Fender Tweed Deluxe, totally refurbished; a 1956 Rickenbacker; a '59 Fender Champ, and a Music Man 50-RD — great sound and it has a line out so I can go directly into the board and just use the tubes with overdrive.
If you were a musical instrument, which would you be?
Jimi Hendrix's guitar at Monterey Pop — but I don't like the way it would end up. Maybe the drums on Bob Dylan's "Like a Rolling Stone" — I'd like to be the tom-tom that gets hit to start the song. Does Madonna play an instrument? 'Cause I'd be any instrument she played.
I think she plays the organ. By the way, what's wrong with the music industry?
What's right with it? I think what's wrong has always been wrong, but it's just bigger now, more obvious. It's the lawyers and the bean counters who are running things. I don't want to say anything too bad, because I have a lot of friends who work in the music business...
What music would you like played at your funeral?
Just about any good sitcom theme song from the '60s: *The Andy Griffith Show, Mary Tyler Moore, Jetsons, Flintstones, My Three Sons.* That stuff is just etched into my brain.

Suspect: Bob Mothersbaugh

Ancestry: German?

Occupation: Guitar slinger, composer.

Birthplace: Akron, Ohio. "Nice place to re-tire."

Residence: Los Angeles

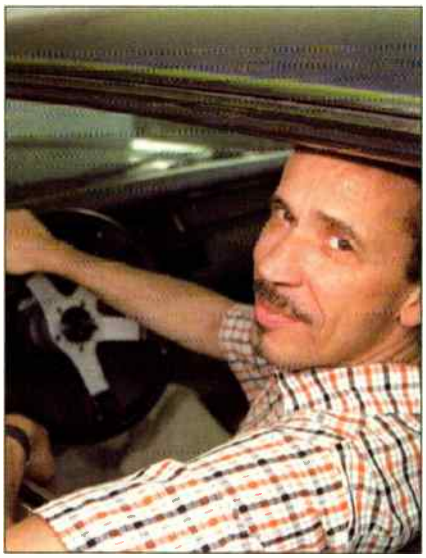
Vehicle: 1969 Dodge Coronet 440 with 383-cubic-inch engine.

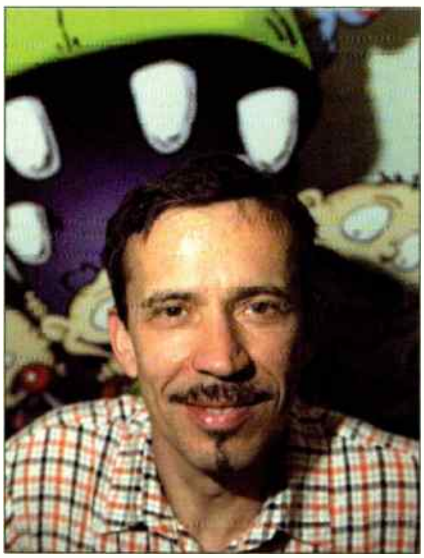
Diet: Often seen at Bosco's House of Steak and Donuts.

Identifying Marks: "My brother...and I have a scar on my upper right arm from a Rolling Stones concert in 1972. I fell on a fence trying to sneak in."

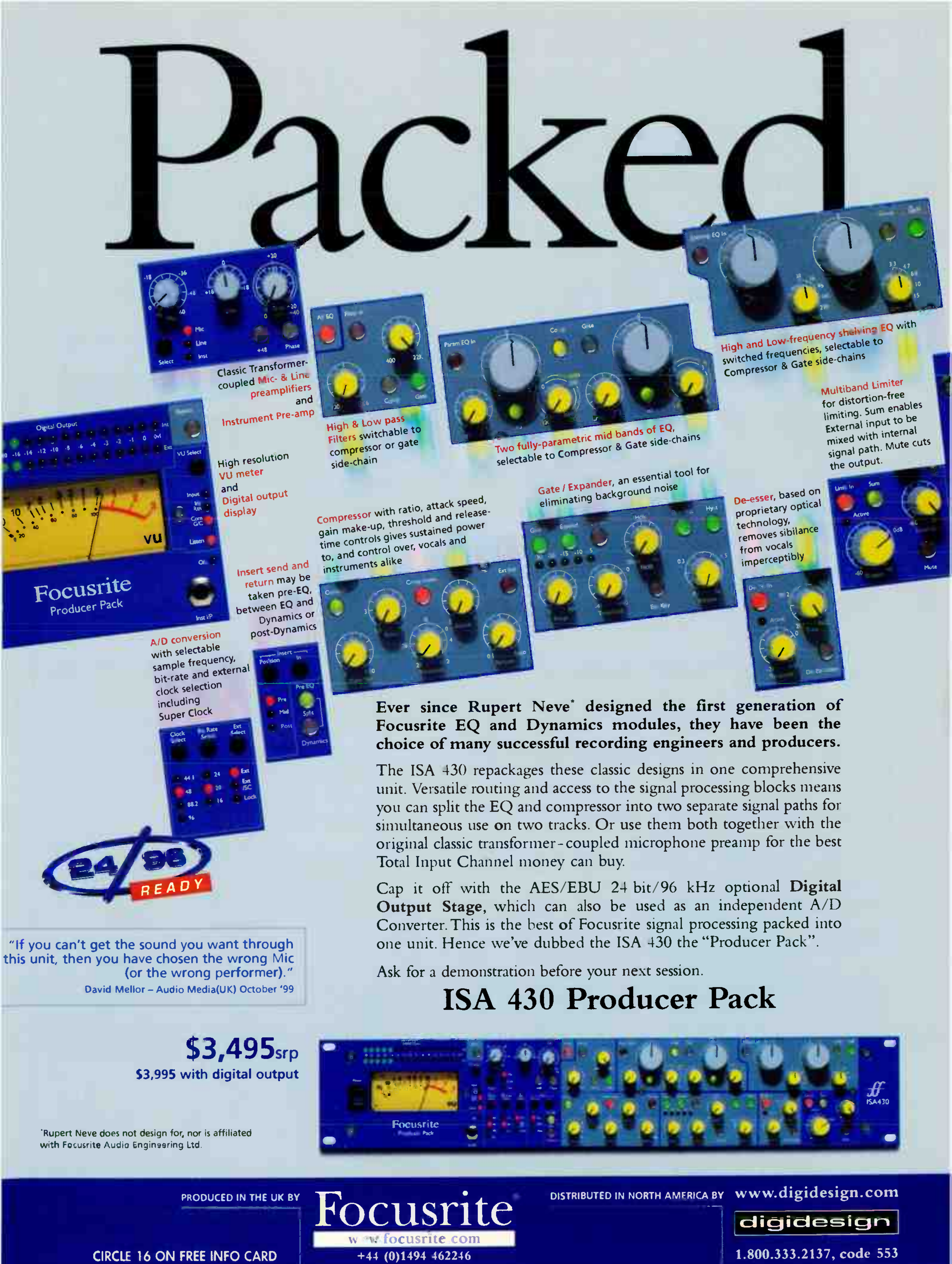
Pet Peeves: "None. It's a beautiful world we live in."

Credits: Co-founder of Devo, scored four seasons of *Rugrats* TV show; the film *200 Cigarettes*, and numerous commercials, video games, and animation projects.





BOB MOTHERSBAUGH PHOTOS BY MR. BONZAI



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CIRCLE 16 ON FREE INFO CARD

What is the first music you remember hearing around the Mothersbaugh house?
Well, the General — "General Boy" — who sometimes filled in as our father, liked Dixieland Jazz, Calypso, Spike Jones, Bob & Ray — all in equal doses. He wasn't shy about putting it on his hi-fi stereophonic set and turning the volume way up.
What did you learn from Mark?
Hard to say. I'm sure I've learned a lot from my older brother, but there are things I wouldn't tell ya about what he taught me as a kid. He can look at you with a straight face and you don't know if he's telling the truth, or lying, or being funny, or making fun of you. I can't do that stuff very well. I remember we used to go down to the basement where all the dirty clothes were piled up. We had a little 45 rpm record player and we'd lay on the pile of clothes and he'd make me listen to The Kinks.
Who were your musical heroes when you were getting started?
Before I could play guitar very well I used to practice in the basement, swinging my arm like Pete Townsend. I loved Hendrix, but I never tried to play his stuff. He was amazing. I liked Keith Richards, Jeff Beck, and Chuck Berry. I had a job one summer where I had to

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CIRCLE 35 ON FREE INFO CARD

demolish a building and I had a Chuck Berry double LP that I would put on at eight in the morning and then walk around with a sledgehammer beating out the walls and ceilings.

Who do you respect and admire today?

Anybody who made it through the '80s and is still in a band playing, or can still walk on two legs and put together a complete sentence.

How would you like to be remembered in history?

Bob was a real stand-up guy, and had a couple of nice guitars and a nice car.

Do you know any interesting business tricks?

Nope.

You mean you just keep getting screwed over and over again?

Well, I seem to be a very lucky guy. Most of the time I come out on top, and I don't know how or why. I'm an honest person — which is kinda bad in business. Just lucky, I guess.

What was your most ridiculous experience on tour?

Actually, one happened before we were touring. We were playing at The Crypt in Akron, back in 1976. Somebody in the band found these giant safety pins, like the kind girls wore on their plaid skirts. We thought we could use 'em with towels and look like we were wearing diapers on stage. We went out and played two songs and the novelty wore off and we had to stand up there with these stupid diapers on for another hour and a half in front of all 30 of our friends.

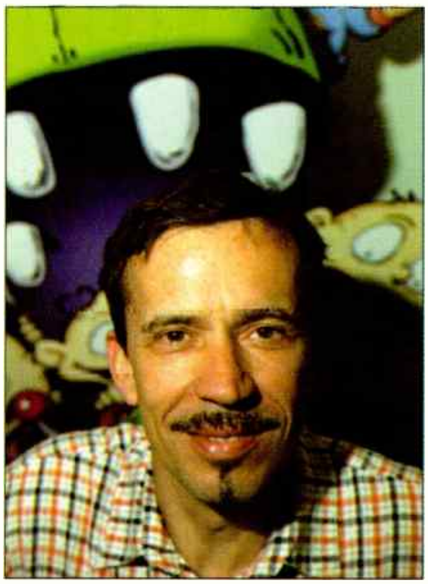
And, later, we played the Knebforth Festival in England, 1978, and we didn't have any roadies. We set up our equipment, ran backstage and changed into our yellow suits, and came out and played — but the power we plugged into was 110 volts and we had 220-volt equipment, so all the amps sounded like transistor radios. That's as loud as they would get. We had to play in front of a couple of hundred thousand people and all you could hear was the vocals and the drums. There was no stopping to fix things. [*Laughs.*] We were sandwiched in between Molly Hatchet and The Atlantic Rhythm Section, so it was the wrong crowd to begin with. They hated us. Luckily, we were on a stage about 75 feet high so the stuff that people threw at us couldn't hit us.

What makes a great producer?

Someone who can talk to the client and comfort them when they are freaking out and getting burned out.

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

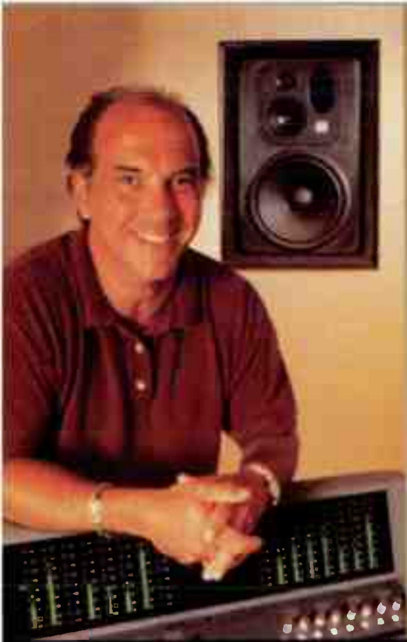
[*Laughs.*] Have an uncle or a dad that owns a record company. Save up a lot of money for a good lawyer. And — maybe — learn how to play an instrument. EQ



LSR. Profiles

The world's most noted recording professionals discuss the world's most advanced monitoring systems.

NO.1: New York / Los Angeles



Neil Karsh is the Vice President of Audio Services for New York Media Group. Recently, Karsh selected LSR monitoring systems for two of his Manhattan facilities, *Lower East Side* and *East Side Audio*.

"We've installed the first of our LSR 5.1 surround systems at East Side Audio and it's a great addition. The sound is extremely clear and is enjoyed by our mixers and our clients. Everyone is very pleased with the result."

New York

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Los Angeles

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CIRCLE 12 ON FREE INFO CARD

Vocal Virtuoso

Producer/engineer Humberto Gatica has done it all, and shares his experiences with *EQ*

BY HOWARD MASSEY

OK, trivia fans, try this one on for size: Who was the engineer for the most complex recording session of the 20th century? Hint: The session in question was 1985's "We Are The World," produced by Quincy Jones and featuring Michael Jackson, Lionel Richie, Bruce Springsteen, and what seemed to be a cast of thousands. The answer, of course, is the name at the top of this page — the supremely talented Humberto Gatica.

These days, Gatica is best known as Celine Dion's producer, but he has enjoyed a long and illustrious career spanning more than three decades. Gatica has earned a reputation as one of L.A.'s leading specialists in vocal recording. In addition to Celine, he has worked with many other equally demanding artists, including Barbra Streisand, Gloria Estefan, Julio Iglesias, Tina Turner, Marilyn McCoo, Bette Midler, Taylor Dayne, R Kelly, and Cher (Gatica mixed her 1999 hit, "Believe"). As impressive as that list is, it's really only the tip of the iceberg considering the wealth of material he engineers and produces for the Latin market (yes, he's worked with Ricky Martin) and his movie soundtrack and production work, which puts him with everyone from Chicago to Kenny G and from Johnny Mathis to Coolio.

With such a track record, you might expect the man to be somewhat unapproachable. However, nothing could be further from the truth. Warm, gracious, and unassuming, it's easy to see why he's so popular among the staff at L.A.'s Record Plant ("my home away from home," as he describes it). During a break between sessions, he talked with us about his philosophy towards engineering and production, as well as sharing some inside tips about the sublime "Celine Dion sound."

EQ: I've only interviewed one other producer who had more experience than you, and that was George Martin. How would you summarize the reasons for your longevity and success?

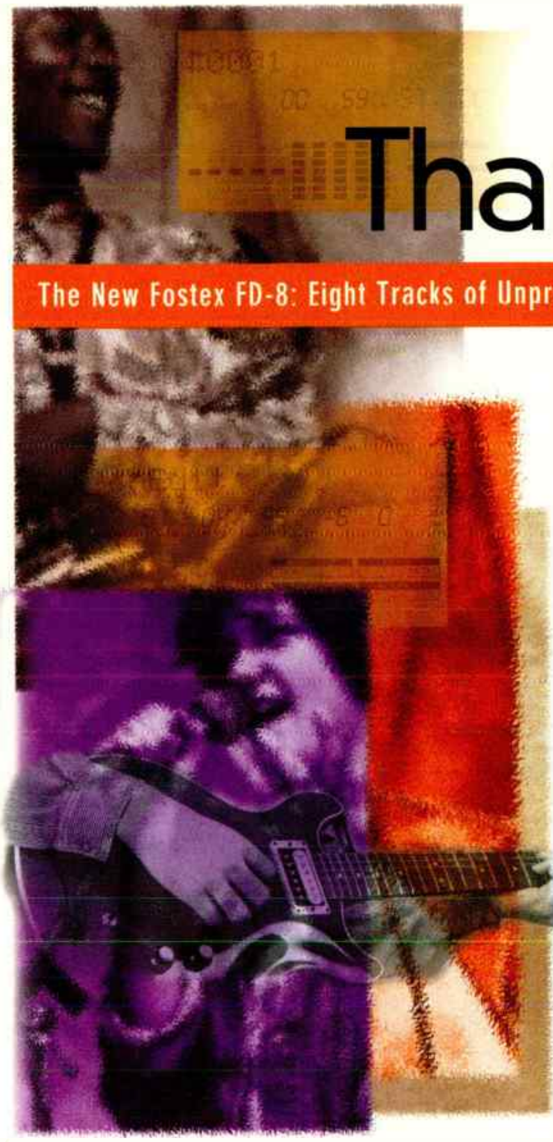
I'm just a pure and artistic person who knows how to use the tools that are available, and I apply and make them work to create what I think my sound is. I have gained the respect and the trust of many artists, and there is something that got me to work with them. Hey, I've been in the business for three generations, and that is what's keeping me busy!

I'd like you to comment on the track ("Here, There and Everywhere") that Celine recorded for George's 1998 solo album [*In My Life*, MCA]. It's a beautiful production, but it's qualitatively different from the tracks you've done with her.

Sir George Martin is the nicest human being in the business that I have ever met. He's a legend, so just being next to him was an unbelievable experience. I recorded the vocal for "Here, There and Everywhere," collaborating with him. He allowed me to do what

PHOTO BY DAVID GOGGIN

Way More Than Four More



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I do, and that was an amazing experience. So, it's interesting that you say it's different...

Maybe it's because the mix is different...

The mix *is* different. I was asked to do the mix, but time did not permit me. The approach they took toward Celine's vocal was that they interpreted it literally, which is somewhat different from what we usually do and the way she usually likes to hear it. Not that it was wrong; it was just a bit different.

What were the specific differences?

The way he approached the reverb on her. She's an artist who has a certain 'verb sound, where it's not swimming in 'verb, where the definition of her is right in your face, but then there's a cool trail that notes sort of sustain to.

Do you achieve that with a delay?

No. I use an AMS RMX16, which I think is the best 'verb for vocals. There are certain programs I use, and certain ways I apply them — sometimes post[fader], sometimes pre — depending on what I'm looking for. Sometimes I route the pre[fader signal] through a delay device, and then I combine the direct signal and the pre signal to create a certain effect that is triggered according to how loud or soft the artist is singing.

I think that Mr. Martin used a very short, dry, straight-ahead [reverb]. I don't think it was crafted the way we craft the vocal in reference to how she performs — all those little things that pay at the end of the day. Not that it's wrong, it's just different. And you were able to hear the difference. If you compare it to, say, the "Because You Love Me" vocal, you'll see it's a tremendous difference. You know, it's like any artist. Through the years, for instance, Phil Collins had a certain sound that was based on this cool, roomy sort of Harmonizer™ kind of placement in the mix, and it worked for him. If you took it out and instead put in compression and a straight echo, you'd say, whoa, that's not him, what's wrong with the sound?

What approach do you take when recording Celine's vocals?

Celine is an artist who has one of the greatest voices in the business. Through the years, we have developed a particular so-called Celine Dion sound. Sometimes that sound runs completely through a song; other times we have to accommodate it because it's a different type of track or a different type of song. Overall, it's a consistent sound. I use a Telefunken 251 microphone that has been modified by a very, very talented young man named Stephen Paul. He helped me locate this microphone and proceeded to modify it. I route the 251 to a custom mic preamp, one that I have used for the past 10 years. It was custom-built by a friend of mine, a very intelligent, very gifted sound engineer. I think the first time I ever used that mic pre was when I recorded "The Way of the World," which was a big challenge for me because we had 12 individuals and 12 different timbres and qualities. However, this particular mic pre allowed me to be consistent. I have never been able to use anything else, unless there was an emergency. I don't think there's anything in the market right now that is, in my opinion, able to surpass the quality of this mic pre.

> *"The whole idea of a great producer is to create a performance that's believable, and the engineer's job is to capitalize on that and put down on tape the best quality possible."*

I then route the signal through a GML equalizer, which I think is the most musical equalizer that anybody can have access to. Then I use an old 160 dbx limiter/compressor and, in terms of 'verb, an AMS RMX16 with an ambience program. That's the basic aspect of Celine Dion's sound, and when you add in an amazing voice, it all comes together.

We work very hard on vocals; she's very demanding in terms of what she likes to hear when she sings because that allows her to project. Doing vocals with her is a process, a big production — but it works; the results are great. The sound you hear is established at the recording, not in the mixing. I'm a guy who believes in commitment. If you like something, you sort of think a little bit toward the future where it's going to end up [in the mix]. What's going to be in front of this voice, what's going to be against it, that sort of thing. You make a decision and commit to it, and it's done. Come time to do the mix, it's already there.

In terms of the headphone mix, does Celine tend to go for less of herself in the cans so that she will project more?

The way she describes it to me is, "I want to be able to hear when I'm singing very soft and when I'm singing very loud." So what I give to her on the 'phones is a sound — it's not a mix — a sound that allows her to really hear every little thing that she's doing, and she likes to hear it very loud.

Do you strap a limiter across the headphones to protect her ears?

No. I never put a limiter on the 'phones, because, when she opens up, she will be oversinging, she'll be trying to fight the limiter. But I put another GML across the headphone mix and I create an EQ curve just for the phones. I don't care how good the headphones are, at the end of the day they are these little things that are just blowing volume up your ears. I use Sony headphones — I'm not sure of the model number — because they don't give me problems with feedback, even at close proximity.

You have to understand that this mic I use [the Telefunken 251] is so hot, sometimes I tell Celine, 'I can even hear your thoughts!' But even when she opens up, instead of the mic thinning her out, it just responds incredibly. I do want to make sure that the headphones are being driven by very powerful set of amplifiers so they don't clip. Celine can hear the slightest little increment of compression or overloading, even in the midst of a loud playback where you think, "Gee, how is she ever able to hear this?" Therefore, the headphone mix is clean; her voice is very defined in the phones, and the music takes a very second place.

Through the whole process of recording her voice, I'm constantly creating a kind of mini-mix; sometimes I have to hold her back just a little bit so she doesn't go over the top. We have an amazing rapport; there are a lot of visual things, a lot of signs and signals that she's giving me between takes or between lines. Or she'll just look at me and, without even saying anything, I'll know she's telling me to do something that might help. It's just a very trusting sort of thing that we have.

[Barbra] Streisand is a bit different. Streisand's one basic comment is: "I'm happy." That's it. Celine says, "I'm happy, but maybe on this line you're going to help me here, right?" Therefore, I might do something like change around the mix so she can hear more definition.

When you're recording Celine's vocals, are you monitoring the headphone mix in the control room?

She hears exactly what I'm hearing, but I don't monitor the EQ'd mix. Occasionally, after the sound is established, I'll put the 'phones on and I'll evaluate where she's at; what exactly is she hearing? And then sort of relate and make my differences so, when I listen to the big monitors, I know exactly where she's at. It works, and that is all part of her sound. I try to establish this from day one, so when it comes to mixing and the final record, that's it.

Celine is powerful; she has a lot of range, and a tremendous amount of energy. She's a person who is very mindful of what is done, and she definitely knows how to work the mic!

I guess what you're saying is: find the signal chain that works for you.

Exactly. Find what works best. It's funny, because the other day I was going through some tapes. I was looking for a specific sound in my library, and I ran into a DAT that was recorded during the "We Are The World" sessions. This tape had vocals only, from one of the passes — we did five passes that night — and it was completely un-EQ'd and unprocessed. It just had a good balance between one artist and the next. I was surprised because the quality and presence of the vocals — every one of those individuals — was unbelievable.

At the end of the day, in any kind of music, it's about how the artist feels, about how good you make the artist sound. You can have an amazing-sounding track, you can have an amazing balance, but if the quality and the presence and the placement of the vocals is not right — if the artist is not comfortable — you're going to end up with a record that has something wrong with it. Learn through the years where to place the vocal and then you can process it, you can effect it, any way you want.

Mutt Lange is an exceptional producer. On his records, from AC/DC to Shania Twain, listen to the quality of the vocals. Although he produces all kinds of different music, the vocals are always amazing — the quality, the presence, where they are placed, how they are recorded. I think behind him there is a great engineer who has worked with him through the years.

Start with a great, substantial, healthy foundation of rhythm, add a great-sounding vocal, and establish a great relationship between the two. When I begin my mixes, I start by establishing the rhythm, regardless of the kind of music. Then I add the vocal. If I am mixing and I come to the point where I feel comfortable and have established the relationship that makes it fall in place, it will be painless. The artist comes in or the producer comes in, and it's an amazing feeling when you know the vocal is right; at that point, you can always lift things up or take things out. Many times, though, certain panning or equalization can hinder that definition and clarity in the vocal, even if it's recorded correctly.

Another important thing to understand is that it's OK to take things down in a disciplined way; for example, trying to clear the air space that's been invaded by certain instruments. I want to hear things in my face sometimes, yes, but I also want to hear the vocal! Moreover, when I'm playing it loud, in the parts where the vocalist is also singing loud, I don't want it to be piercing my ear, either. Sometimes I hear CDs where all of the sudden the music builds and the artist begins to get thin and small, and all you hear is this frequency that just sticks out, even though it's not really having any impact. It's weird, it truly is.

So you're saying that you need to maintain emotional intensity, whether the music's going up or going down.

You've got to maintain it. Pretend you're singing, and you're there behind the mic, killing yourself, right? You're trying to deliver a performance that people will believe in, although they're not watching you moving or doing anything. Now I can hinder that very easily in the mix, even if I record you incredibly. To me, a great mix is when you have accomplished a tremendous feeling about the music being played, a tremendous feedback of whatever you're getting from the artist, whomever is motivating your emotion. The whole idea of a great producer is to create a performance that's believable, and the engineer's job is to capitalize on that and put down on tape the best quality possible. When the quality is good, the emotions come out.

You have to listen to the individual. Ask yourself, am I creating a tense feeling when they are singing by over-compressing? Sometimes it's kind of cool, but other times, at the end of the day, when you put everything around [it], it's not cool; all of sudden there's a sound, but the emotion goes down to nothing.

It's important to use a good mic. The

Telefunken is one of my all-time favorite mics; it lets me accomplish what I like to hear. In addition, don't be afraid to use equalization, there's nothing wrong with that. A lot of people say everything should be flat. Well, I've heard a lot of things that are flat, natural, but still sound bad. Equipment has been designed to be used in moderation.

Do you ever find yourself equalizing a vocal differently during the course of a song? Actually, tweaking settings during the choruses and the verses?

I've developed a consistent format of equalization that I use on most artists I work with. If I have to record an artist where I don't have access to certain equipment, like a GML equalizer, maybe then I would just do it flat. But I just use my experience, and whatever my ears tell me — my ears are the only tool that I completely trust.

How about in the mixing stage?

In the mixing stage, there are times where I may have to thin the vocal out because of the nature of the singing or the key in which the person is singing. Sometimes I mult the vocal and have the verse processed one way and the chorus another way; sometimes I use the 'verb aspect to change it a little bit. And that's great; if you have access to do that, it's wonderful. I don't do that all the time, but, with some artists, I do it maybe 60 percent of the time. Many times when Celine opens up in the chorus, I need to warm the sound up a bit, so I process it a little differently; I take away some air and I may add it back perhaps on the verses.

However, you're emphasizing the importance of using good equalization circuitry.

It's very important. You want to make sure your signal's clean and that you're not damaging the signal chain in any shape or way. Sometimes processors are not correctly aligned, and when you use an outboard compressor and equalizer along with whatever console you are using, they may not like each other. I usually don't use console equalizers; my whole vocal thing is done from outside equipment.

I used an SSL 9000 to mix five songs on the new Barbra Streisand album. In a few places, she said to me, "I can hear the compilations of performance, and that bothers me sometimes." I can totally respect that in an artist; she can hear the different emotional stages when we come in and comp the vocal.

Sure, you can't be in the same mood every time...

Exactly. And she can tell better than anyone, so she says, "Well, I was a little soft here, I was a bit angry there, and it bothers me..." To the average ear, no one would hear it, but I can really relate to her, so it was easy for me. But instead of doing a level thing [to compensate], I used EQ to take away some frequencies, and the SSL allows you to program the EQ and pop it in and out, and save a lot of anxiety.

How long do you typically spend on a mix?

In 90 percent of the mixes that I do — whether it's a 24-track, 48-track, or two 48-tracks linked together — I will build a mix in eight hours. Sometimes, though, maybe you are a little fatigued, and, if you're smart, you say, "Let's just come back tomorrow." And tomorrow, in one or two good clean hours, I can kill this. That's my theory. After the mix is done, my assistant and I dissect it from many different points of view.

For instance, right now I'm mixing Celine Dion's *Greatest Hits*. We're doing about eight or nine songs, and then, out of the nine, we'll select the best six. On this project, I'm working with many different producers — David Foster, Rick Wake, Mutt Lange, Max Martin (the guy who does the Backstreet Boys). My job right now is getting all the producers to where they are 100 percent

satisfied and happy. In their hearts and in their minds, they know that when it comes to the vocal — and, of course, they have an opinion, they can say anything they like — they know that department is taken care of by Celine and myself, in that order. So usually, they're very happy about where the vocal is. When it's all done, that's when I play all the mixes for her. She's very sensitive about the emotion that she puts down — meaning every little breath, every little thing in between breaths, every little pick-up of a line where sometimes you have to help matters. For example, she'll sing, "...the way I love you." I exaggerate it; I make sure that "L" is so divine that it really delivers what she meant to say. She's very into that; that's what she likes about working with us.

Do you take a different approach when producing music for the Latin market?

No, I just apply what I know. The language itself...sometimes that permits you to do things a little differently, but it's very limited; you have very little room to move.

I produce a girl from South America named Myriam Hernandez, a very successful artist — one of the greatest female singers in that part of the world. I consider her the Celine Dion of Latino music, and I use the same sort of tricks and approach in terms of recording her vocals. However, the Latino market, especially now, is so hip. And let me tell you, there are a lot of great-sounding Latino records recorded by Latino engineers, produced by Latinos.

So, technically, there's no qualitative difference.

No. Though sometimes you'll notice a bit of a difference in the way you equalize things, because there's a little more rhythm in the Spanish language; extra vowels, you know. They're different, so it's a different sound, and therefore certain EQ doesn't work. Same thing in French. Celine has done two French records and I have to tell myself to EQ her differently when she sings in French. A million things change because of the nature of the language; 1 dB in a certain frequency can make all the difference in the world. Certain languages do require certain crafts.

Do you have any general advice for someone just starting out in this field, someone who aspires to be the next Humberto Gatica?

Just be yourself, don't hold back, and be open-minded to both criticism and advice. At the end of the day, you can only teach an individual so much; at the end of the day, it's what you think sounds good and what feels good to you. It might be in line with other people who will want to work with you because they love the way you hear things. On the other hand, it might work against you because they don't like what you do.

You're going to make mistakes. The important thing is to learn from them, to tell yourself, "Oh my God, that sounds horrible, I will never do that again." Then, when you find yourself in the same position, you have to remember and to think, "No, I'm not going to do what I did before." You learn because that's how it is; you've got to make mistakes. You correct yourself and then you balance it so the good overrules the bad. In addition, you go on from there.

Howard Massey heads up On The Right Wavelength, a MIDI consulting company, as well as Workaday World Productions, a full-featured production studio. His upcoming book, Conversations With Record Producers, *is due to be published in fall 2000.*

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Console Patching Tricks

Welcome to my wiring nightmare...

BY MIKE SOKOL

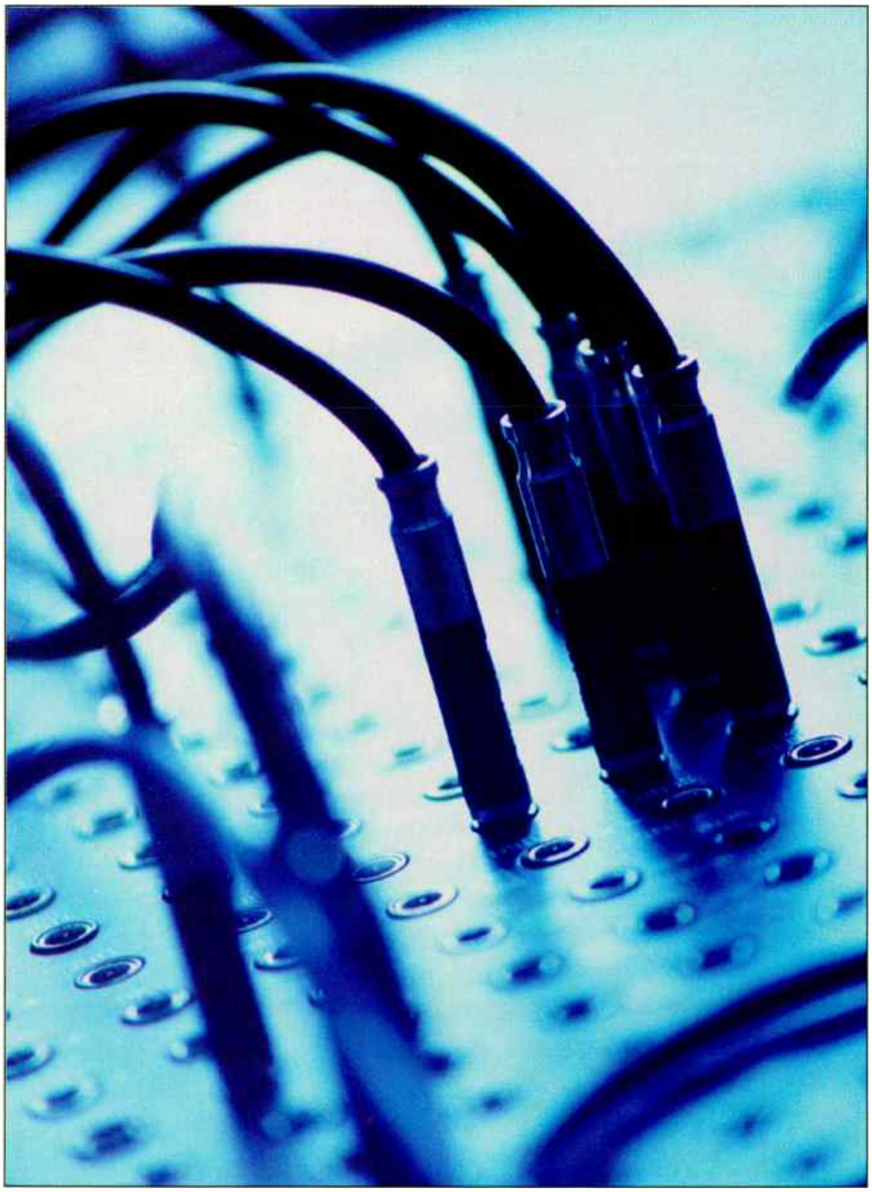
The lowly patch cable is severely underrated. It's a simple thing, nothing more or less than 6 feet of wire with 1/4-inch plugs. But its uses are endless. With a handful of them you can reroute audio signals in ways God (and the manufacturers) never intended, allowing you to use ordinary studio gear to perform feats of daring that will scare the masses and turn your contemporaries green with envy. For instance, while mixing consoles are hard-patched internally for specific functions, most have inserts and break points that will allow you to reroute signals in non-standard ways that will allow you to perform some incredibly useful, and sometimes scary, functions. Following are several examples:

INTERNAL DISTORTION

No, it's not the name of some new-age group. But you may have a recorded track that needs a little dirtying up to give it some life or make it stand out in the mix. There are tons of signal processors on the market that will do everything but sit up and dance, but sometimes you just need a little bipolar-junction distortion. Here's how: Use a patch cable to route the output signal from one channel strip into the input of another unused channel strip. Most medium-to-large analog consoles have a direct-out jack on each channel, which will output a postfader and EQ signal. Simply use a 1/4-inch patch cable to go from it into the 1/4-inch input jack on another channel. Then you can overdrive the second input with the signal from the first channel. You can get all sorts of interesting effects this way, from just a hint of "buzz" all the way to "robotz in heat."

And don't be afraid of using the equalization on both channel strips. Pre-distortion (the first channel strip) equalization works differently than post-distortion (the second channel strip) equalization. I find that you'll need to roll-off some the high frequencies from the second strip to keep from overwhelming the mix with too many distortion products. Dumping the bass from the first strip usually improves the "quality" of the distortion since it's not being modulated with bass frequencies. Remember that there will be a lot of signal level coming into the return strip; be careful with the fader so you don't do damage to your monitors. I've used this effect for voices, bass guitar, and electric guitar with very interesting results. And it doesn't cost a dime.

While you're looking for distortion tricks, don't be afraid to patch in some of your old or new "retro" gear. Vox Cry-Baby wah-wah pedals are great on background vocals, and nothing sounds quite like a Roland Space Echo that's being overdriven on guitar. You can even patch out from your sonically stunning high-end console into some old junk mixer you've been saving for a few decades too long. It's a great way to do unique processing on a track to



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make it stand out from all the other "proper" tracks in a mix.

DOUBLE DIPPING ELECTRIC GUITARS

Here's another patch trick often overlooked. Most of us will record both direct and speaker mic signals from a bass guitar. This gives you both a very dry punchy sound from the DI track along with nice rounded room tones from the mic in front of the speaker. But, if you've got an extra free tape track available while recording your next electric project, try the same thing for the electric guitar. Here's how and why: Just use an active DI box on the signal from the guitar before it plugs into the guitar amp. It should be an active DI so that it has a high enough input impedance so as not to mess with the signal level and timbre from the passive pickups in electric guitars. Passive boxes will mess with the actual sound coming from the guitar, and should be avoided in this application. Any good active DI box from companies like Countryman Associates or BSS will do the trick. It doesn't need to have a speaker emulator, since you're not trying to use it in place of the speaker, you want the full range signal of the guitar before it gets to the amplifier. So be careful not to overdrive this clean DI signal on tape...you *want* to capture the true sonic signature of an undistorted electric guitar.

Now, after all the tracking is done and you're ready to mix, patch the output the DI electric guitar track out to a guitar amplifier stack in the next room (or vocal booth). Use your favorite amp and mic combination, bring the mic signal back in just as you would an outboard effects device, and get ready for some fun. Now, when you play the clean DI track back, you can actually experiment with the overdrive and tone control of the guitar amplifier without pestering the guitar player to do multiple takes with more or less grunge for effect. This could be recorded onto another free tape track or used only during mixdown.

If you're really in an experimental mood, patch in that old Dan Electro or Silvertone guitar amp and add it into the mix along with Marshall stack that was previously recorded on the "mic" track from the session.

EMERGENCY EQUALIZERS

In addition to studio work, I also do a lot of sound reinforcement gigs. And, as Murphy will attest to, anything that can go wrong, will go wrong (and at the worst possible time). So here's a quick patch trick that could bail you out the next time you need just one more equalizer channel for a monitor wedge, or you've had a major electrical jolt take out some EQ channels right before the show with no time to get extra gear.

While it's highly desirable to have a dedicated 31-band equalizer for every monitor mix, anything is better than nothing. If you could get a parametric equalizer patched into the monitor send, you could finish the set without having the lead singer drop-kick the floor wedge over to monitor world. While in just such a situation a few years ago, I recognized that there was all the equalization I needed right on some

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Worldly Music

A behind the scenes look at *Score,* Paul Haslinger's eagerly awaited third studio album excursion as Haslinger.

BY JONATHAN MILLER

It has often been said that music is a universal language. However, sometimes it appears incapable of even traversing our continents, which seems strange considering that there can be no language barrier with instrumental music. For his part, Paul Haslinger has long been opining that contemporary music should consist of elements from all around the world, a conviction that has made him a fanatical sound collector and archivist who spends weeks and months tweaking, learning and modifying sounds.

Score perfectly encapsulates Haslinger's unquestionable pedigree, both technically and philosophically. Yet, with its myriad of influences and styles, the album appears worryingly at odds with a commercial music industry seemingly obsessed with concise categorization. And marketing is perhaps an appropriate springboard from which to dive into a dialog with the man himself — after all, consolidating pushing the envelope of contemporary music-making with corporate considerations is no easy task, right?

A former five-year veteran of legendary German electronic trailblazers Tangerine Dream, Haslinger evidently knows the score: "Truth be told, I just follow my gut feeling in almost everything I do. Sure, I'm obsessed with quality in each and every aspect, but my general direction is not calculated but rather 'felt'. I've always seen music as a language, fortunate in its lack of grammar; a language that every person around the planet relates to in one way or another, so it's quite natural for me to choose my vocabulary from as wide a range as possible."

As for commercial constraints, Haslinger remains unfettered with a refreshingly healthy sense of humor to boot: "The music of the future will be less predetermined by regional and stylistic barriers," he predicts. "It will be global in every aspect — not just pop, by the way; I think this will happen to any kind of music, no matter how simple or complicated its structure might be. On the other hand — let's face it, there will always be boring stuff, and people who sell and buy it. It goes well with the corporate culture out there, and it serves as nicely prepared plastic entertainment. Look back in time, though, and you'll find that even in Mozart's era the same principles were on display — promotion of the mediocre; neglect of the original; QED; nothing that biogenetics couldn't fix in the next 200 years!"

HASLINGER GOES TO HOLLYWOOD

Upon arriving in this movie maker's Mecca, Haslinger wasted little time in establishing what he terms "a music and media laboratory" — inevitably a state-of-the-art recording facility where one might expect a forward-thinking artist of Haslinger's caliber and background to inhabit. Originally sited in a loft-sized in-

SOME ASSEMBLY REQUIRED: Paul Haslinger sits in his well-equipped studio, named The Assembly Room.

dustrial building in West Hollywood, The Assembly Room is designed to accommodate a wide range of all-in-one-stop production, with a general focus on cutting-edge music-to-picture projects and new media work. More recently, Haslinger moved the studio out of town to a more idyllic, though no less hectic, locale — at his home in nearby Woodland Hills.

Numerous multimedia and new media commissions soon followed; in the expanding interactive multimedia scene, Haslinger was appointed music director for the prestigious SIGGRAPH Convention where he also composed music for the opening of the Electronic Theater show in 1995. Here he worked closely with leading computer graphic designer Frank Foster, who, in return, later produced and directed a stunning promotional video for the title track of Haslinger's second solo effort, *World Without Rules*, including computer graphics developed by Sony Pictures Imageworks. This went on to become one of the first pieces aired on AMP, a new show dedicated to cutting-edge music that MTV was launching at the time.

MULTI-TASKER: The total recall feature in Haslinger's two Yamaha 02R digital consoles allows him to work on several projects at once.

With a filmography stretching back to 1986 — completing several feature film soundtracks during his Tangerine Dream tenure, could it be that Haslinger's "renewed interest" in mainstream music-to-picture scoring is in any way related to the high-profile projects now open to him as Hollywood hotshot Graeme Revell's right-hand man? If we are to understand Haslinger's current work, clearly this period of musical reawakening is worthy of further discourse. The composer kindly obliges: "I was introduced to Graeme Revell through my friend Brian Williams [a.k.a. Lustmord] who works with him as a sound designer," he reveals. "Graeme, at the time, was looking for someone to help him on the programming side, which gets used as a term in Hollywood for a wide variety of activities, including sequencing, arranging orchestration, and general music realization. Every project is different, and every constellation changes according to style, orchestra size, and time restrictions. In Graeme's case, I'm fortunate to work with an established crew of music editors, orchestrators, contractors, and engineers who are all top notch in what they do. A big score, then, becomes this team effort; a very complex process — talk about recording a 95-piece orchestra over 100 tracks of synth playback, mixed with duduk and vocal solo — which requires everyone to work with everyone else in 'perfect sync.' It's in this 'scene' that I learned much more about collaboration than I did in any of my previous engagements."

I THINK THEREFORE I AMBIENT

Haslinger's "music and media laboratory" continues to be stretched to its technical limits, which was, of course, always the intention. Given the time-consuming nature of recordings like *Score*, The Assembly Room unsurprisingly "went total recall" in 1997 with the installation of two Yamaha 02R digital mixing consoles to facilitate simultaneously working on more than one project: "More samplers, more patchbays, and more total recall," best summates Haslinger's present-day needs. Haslinger proceeds: "I've found that I prefer to work with everything ready to go at my fingertips — hence the one-studio-does-it-all-model."

Not that Haslinger is about to be blinded by science, mind you: "Conceptualization and ideas are more important than realization," he says. "In other words, going for a walk and developing an approach can sometimes be more fruitful than hanging around the studio 24 hours a day. And, lastly, more and more of my work happens in other facilities: I just went up to Vancouver to record a string section for a new project with Graeme called *Idle Hands*. Before that, we had a couple of studio sessions for *The Siege* with Djivan Gasparian and my old friend Nona Hendryx — all of which happened in different locations and studios all over the West Coast. Actually, these days, lots of my inspiration for my music-only projects comes by way of project situations my film work puts me through."

These "music-only" projects have many faces — with *Score* (1999), *World Without Rules* (1996), and *Future Primitive* (1994) representing the compositional front-line, with more abstract sideline projects in close support: Haslinger has twice guested on outlandish albums by renowned French electro-acoustic ensemble Lightwave, the complex sonic tapestries of which he later took a step further with 1997's *Hidden*, the sole release to date under his Coma Vi®us pseudonym. Another is on the cards: "The new Coma Vi®us project is a conceptual album based on mathematics, numerology, and the Kabbala," reveals Haslinger. One to file under Easy Listening, obviously!

Of course, different projects demand different solutions: "Working practices differ a lot from the solo releases," Haslinger maintains. "With Coma Vi®us, the aim and intention is to challenge, to experiment, and to push the envelope of current music-making. Compared to this, the releases as Haslinger are pure pop — music for the car stereo! Therefore, techniques employed on Coma Vi®us are usually more complex and, ultimately, more time-consuming."

HOME IS WHERE THE ART IS

In revisiting themes linking *Score* and Haslinger's music-to-picture interests — likewise evident in the new album's pleasing artwork pastiche, this apparent homogenization was not entirely intentional from the outset, however: "The album's working title was *Hardboiled Wonderland*, inspired by a book written by Haruki Murakami," explains Haslinger. "After *World Without Rules*, I wanted to take the next step in the way I use and work with ethnic elements — hence a trip to Japan; both to learn and absorb, as well as to record and sample. The next phase started with my rediscovery of Les Baxter and Eumir Deodato, an experience that produced much of the lounge aspects of *Score*. Lastly, I stripped down from about 30 pieces in total to around 15 in the final group. 'Essentialization' is the term I use for this process — basically, finding and settling on the focus for the whole album, having compiled all the pieces. I did, and decided that the original title wouldn't work anymore, since most of the faster and edgier pieces were gone. Rather, I was looking for something fairly simple, a one-word title that had a multiplicity of meanings. RGB then suggested *Score*, and, even though I had my concerns about the connection to film scoring being too blatant, I finally agreed — mostly because I couldn't come up with anything better myself. I figured it's certainly 'neutral' enough to let the music speak for itself. Personally, I was much more concerned with artwork issues, which, in my view, has a stronger effect than the title."

SAMPLING CONUNDRUMS

Sampling, paradoxically, continues to play a major role in the Haslinger *modus operandi*. In the case of *Score*, he has taken the unusual step of listing his commercially available sample CD resources — even going as far as crediting some of their creators with Sound Design. Quizzed as to whether this was a conscious move against the grain on his part — litigation aside, it could be argued most artists actively disguise their sources, perhaps fearing ridicule — Haslinger is keen to downplay his actions: "I just like to give credit where credit's due; no special program attached," he counters. "Sampling, for me, is a technique. You're either good at it, or bad; it's that simple. As for the whole copyright issue, it's a complicated one, but common sense should prevail and tell us that it's fairly obvious, on a case-by-case basis, what's blatant theft and what's a creative and artistic transformation. In my own work, I've tried to stay with the second group. Samplers today are wonderfully powerful machines; each equipped with a specific sound — individual-sounding filters, modulators, and processing tricks, etc. They continue to be a central part of my musical universe."

Indeed they do. With several such units at his immediate disposal, Haslinger once cited the Roland S760 as his "absolute workhorse" in this department, attracted, in part, by its facilitating feeding an output into his Apple Macintosh computer's audio/video input, permitting sounds to be easily edited and re-arranged on-the-fly. With technology's inexorable march onwards, this instrument is now beginning to look a little long in the tooth when pitted against the latest offerings from rival manufacturers like EMU and Akai. Though remaining fond of his time-served Roland sampling hardware, Haslinger has nevertheless opted to jump ship: "I think the Roland format is here to stay," he remarks; "I just heard from a friend in Japan that the '700-series samplers are beginning to be hot commodities on the used gear market. People seem to be realizing that those machines actually sound pretty good, and Roland hasn't produced them in a while, so demand will go up."

Whilst it's not quite a case of out with the old; in with the new, Haslinger's self-defined sampling "workhorse" status now rests with EMU — several E4's, in fact; "...all maxed out to 128 MB RAM, mainly because I can use them on one SCSI line with the Mac and send stuff to and from the sampler for editing purposes.

"Actually, I could make a lot of music just using a G3 and an E4; it's a very powerful combination," enthuses Haslinger. "The new E4 Ultra looks very good, too — love the processing speed! Again, the main attraction for me though is sound quality itself; that's why I'm still not quite sure about the new Akais. To my ears, their samplers always sounded too much like DAT machines — thin and crisp. Now, my friends in Japan claim that this is because they have a more linear frequency response, and that Roland and EMU machines actually enhance — or falsify — the sound on playback. But hey, there was this song that went something like, 'If loving you is wrong; I don't wanna be right...'!"

FRONTLINE ASSEMBLY

Haslinger is on record as saying, "The most interesting point of making music for me today is how you put it together. You can look at it historically and philosophically and say, 'Well, the Latin root of *componere* — or composition — means to put together.' " In this context, The Assembly Room is a fitting epithet for Haslinger's recording environment, indicative of his ongoing interest in how music is assembled and the intensive research regularly undertaken there.

"I've always been chasing the dream of the perfect studio, or at least what is the perfect studio for me," says its creator. "Finally, after years and years of going through hundreds of studio configurations, I found one formula that really works for me at The Assembly Room, starting in West Hollywood. What I'm doing today is developing a little more; pushing it to be smaller and more powerful, but essentially it's the same studio concept since that time."

And that concept is? "I can summarize the concept in one word; *real-time*," explains Haslinger. "All I do revolves around that — the main difference to how composers used to work and it's here to be taken advantage of. Another big issue is nonlinearity. Again, there's no tradition for the fact that we can break music down into particles and mess with them. It's new; it requires different sensibilities, and it challenges the notion of 'original musician' as we know it. The Assembly Room then is simply a studio-turned-instrument; a tool that I use to do what I do, and which I 'tune and maintain' to fit particular needs of ongoing projects. One, of course, could call it a bunch of computers and some storage, but, then again, that may just be a little too insightful!"

Joking aside, there's no disputing that The Assembly Room is heavily reliant on technology — without which, Haslinger's penchant for re-assembling those musical "particles: into new forms would be nigh on impossible; unbearably time-consuming at the very least. As Haslinger once said: "I try to keep every unit in the setup on a computer screen or under direct hand control so I can, at every point in the process, tweak, change, and re-sample sounds. Over time, I've developed a whole new way of working that involves a huge amount of data processing. I'm trying to use technology in ways that enhances and helps me with that processing, rather than taking up my time troubleshooting or learning new systems."

Quite how this revelation sits with Paul Haslinger, circa 1999, remains somewhat tainted with ambiguity: "I wish I could do without the 'huge amounts of data to process,' " sighs Haslinger; "but, let's face it, on a very general scale — not just in music — we're facing the same dilemma: there's simply too much stuff out there to be absorbed in a lifetime for each and every one of us. To make an informed choice, one again needs to gather information on what the hell one's selecting from — pretty general, but very applicable in music, where, in order to dismiss a possibility, you should at least know what that possibility is!"

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TO KEY OR NOT TO KEY

Returning, briefly, to the performance arena, Haslinger is, without doubt, a technically gifted keyboardist. Yet, back in 1996, he appeared to be actively distancing himself from his roots. In re-appraising *Future Primitive*, for example, he commented, "As keyboard players, we have a problem; playing solos: it's either sub-Jan Hammer standard, or simply something that's not very interesting, so with *World Without Rules* I initially said, 'Okay; no solos, but if there is, they're not going to be keyboard solos...' " — hence contributions from inventive players like trumpeter Mark Isham and Haslinger's subsequent enthusiasm for the expressive properties of Yamaha's fledgling acoustic modeling technology in the VL1: "That, for me, completely differentiates *World Without Rules* from its predecessor," said Haslinger, "because there's really nothing in the traditional synthesizer sense on there anymore. It's something I'm proud of, because, ultimately, as keyboardists, that's where we're going. It doesn't matter if we're chasing a real orchestra or this kind of global sound; if we can make it sound like it's not done by a keyboard player then it's a good thing.'

Strong words indeed, though on the strength of *Score*, Haslinger remains true to his then-newfound philosophies: "Now the real chase is for something new. The emulation of other instruments — traditional Western or ethnic — are just facilitation devices; a trick to get your mind off existing patterns. The biggest goal and challenge is to transform and re-invent far enough, so that the end result speaks in a voice of its own. Again, I feel most of this cannot be planned, but has to come by intuition and a developed sense of sound."

And Haslinger's sense of sound is well and truly developed — to bursting point, in the case of his awesome sample library: "I could easily spend the next ten years just working on modifications of sounds I've already got in a raw state," he declares.

That one of Haslinger's instrument racks within The Assembly Room contains a selection of what might be considered outmoded gear — EMU Procussion and Morpheus; Waldorf MicroWave; Yamaha TG77; Roland MKS20, et. al. — is perhaps indicative of his enthusiasm no longer extending into the world of synthesis: "Well, you're missing some parts," Haslinger playfully retorts, before admitting, "Modules have indeed become very boring to me. I just bought my first synth in a long time, a Roland JP8080, which I'm using as a quick edit synth workhorse — always liked the Roland Jupiter8/MKS80 tradition of sound. At the same time, all those older modules you see I keep around for good reason: I know them very well; they do specific things, as opposed to the 'all-in-one-box' approach, which I hate, and they have some signature sounds I couldn't match with sampled versions."


ON THE ROAD AGAIN: Haslinger on stage.

Yet sampling is likely to remain at the forefront of Haslinger's manifesto for the foreseeable future: "The majority of my technical attention has been dedicated to samplers and audio processing," he clarifies. "That's where the action is for me; I prefer to start with a blank slate rather than a predetermined oscillation source."

VIRTUAL INSANITY

Given Haslinger's earlier interaction-oriented commentary regarding keeping every sound source in his setup instantly accessible via a computer screen or under direct hand control, his feelings toward both the virtual analog synthesizer phenomenon and the more recent proliferation of so-called "soft" synths, drum machines, and samplers remain stand-offish: "I love the concept," he says, "but, again, it's the *sound* that matters — as of yet, I haven't heard too much that's either particularly good or original in that respect."

But, according to Haslinger, "Emulating modular synths should not be the final goal; just a place to start from. New tools should be used for new stuff — that's how truly old-fashioned I am! It's very early in the game though, so let's see where Clavier, CreamWare, Propellerheads, and all the others can take it in the next few years. Concepts like Steinberg's Rewire could lead to some significant enhancements and to better integration."

Always an informative source of crystal ball gazing, many industry pundits predict the entire contents of the modern recording studio disappearing inside the virtual world of the desktop. As a musician active in several disciplines with numerous inside contacts built up over the years, Haslinger is likewise better placed than most to speculate, but remains unconvinced: "It will be the best of both worlds; just as there are still analog machines, turntables, and tube mics, there will be parts of our creative process that we don't *want* to be computer-controllable."

A QUESTION OF ZEN

Haslinger's longevity in a disposable age speaks volumes; not one for resting on his laurels, no doubt his singular musical vision will continue to be vindicated for many years to come. Though his Web site (www.haslinger.com) makes mention of several forthcoming collaboration and remix projects, Haslinger is sensibly playing them close to his chest: "All I can say is that I'm planning to produce a project with a singer in the near future," he discloses, tantalizingly. "I've started preliminary recordings with several people to test the waters. I'm a big fan of the production style of people like Tim Simenon, Marius De Vries, and Mark Bell; I'll try to take their work as a starting point for my own endeavors."

More concrete is Haslinger's online promise of "a whole lot of film music," including an enticing commission to score a documentary about the history of computer graphics: "The story is a fascinating one," he muses. "Computer graphics really came to the foreground of public perception and acceptance with films like *Terminator 2* and *Jurassic Park*, but the story began *long* before that; this is what the film is trying to tell. Musically, it will be a journey through time, reflecting on the victories and downfalls along the way, as well as the spirit of the people working to make a mathematical tool become an instrument for creative use."

Meanwhile, back in more familiar surroundings: "With Graeme Revell, I'm currently working on a sci-fi thriller called *Pitch Black*, Set on a distant planet and shot in the Australian desert, it follows a thread of film-making which started with the *Mad Max* series — desert sci-fi adventure; the strange, the brave, and the ugly."

Lastly, there is also a new TV project called *Eli's Theory* developed by Imagine — director Ron Howard's production company — for Warner Brothers: "They wanted unconventional, reflective, emotional music," imparts Haslinger. "It's about a father-son relationship; the kid's a mathematical genius and the father is raising him by himself. It seems to be one of those examples where people are trying something different for a change, and I'm glad to be part of it."

Jonathan Miller is a British freelance writer living in... well, England. He specializes in the "ancient art" of the high-tech music interview, and greatly enjoyed trading e-mail with Paul Haslinger. Jonathan can be reached at jonathan.millermusicmedia@virgin.net.

Revolutionary Recording

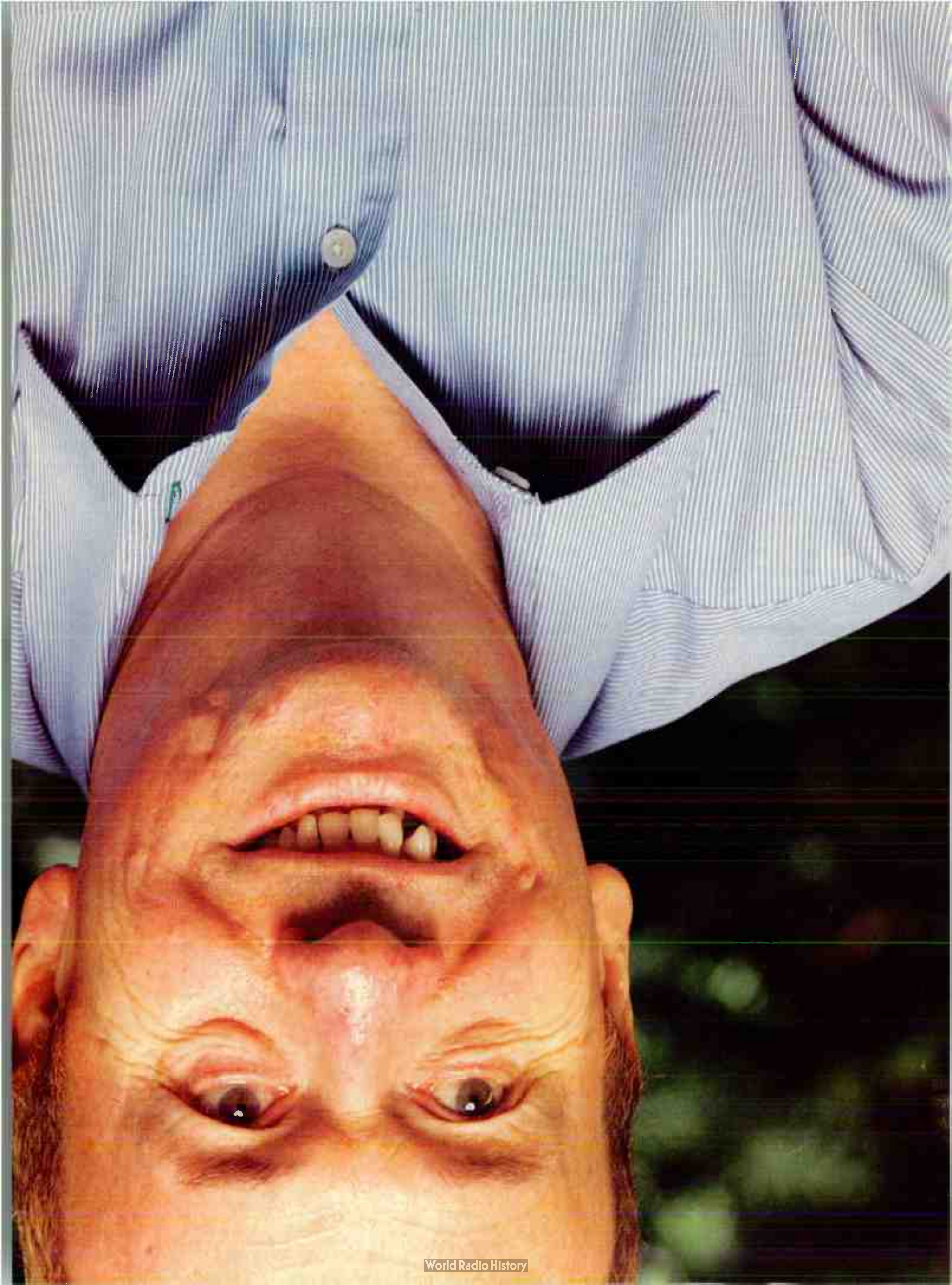
Geoff Emerick takes EQ behind the scenes for the making of history's most memorable sounds

By Howard Massey

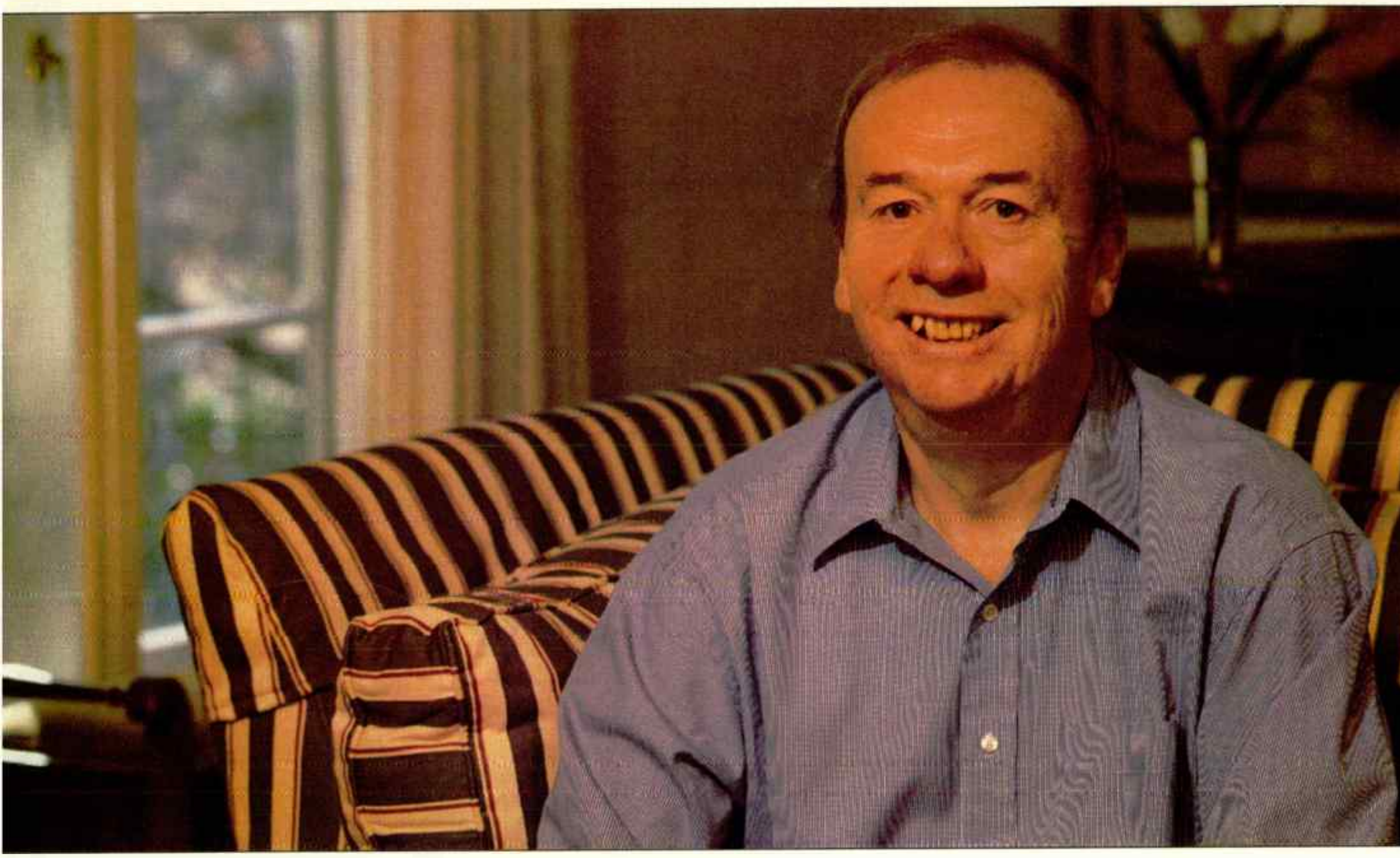
Most Beatles scholars agree that George Martin provided the musical polish that made complete the towering, raw talent of John Lennon and Paul McCartney. But perhaps the secret weapon in the group's legacy of recorded music was engineer Geoff Emerick.

As the man behind some of the most innovative sounds ever put down on tape, Emerick was the inventor of many recording techniques still in use today. Joining Abbey Road studios as a mere teenager, he quickly developed a reputation as a maverick — as someone who was willing to experiment, unafraid to challenge the staid ways of the past. Incredibly, he was promoted to full engineer a few months shy of his 20th birthday, and, shortly thereafter, was awarded the coveted job of becoming The Beatles full-time engineer, replacing Norman Smith.

Emerick was behind the board for The Beatles's most adventurous forays, including the legendary *Revolver*, *Sgt. Pepper*, and *Magical Mystery Tour* albums, as well as the "Paperback Writer"/"Rain" single (which premiered McCartney's new bold bass sound and was also the group's first experiment with backwards tapes), the live worldwide TV broadcast of "All You Need Is Love" (the first time such a feat was ever attempted), and what was perhaps the greatest double-sided single of all time: "Penny Lane"/"Strawberry Fields Forever." In the face of growing tensions within the band, Emerick departed halfway through the recording of the *White Album*, only to return for what many consider The Beatles's crowning glory — *Abbey Road*.



Revolutionary Recording



PHOTOS BY EDWARD COLVER

In the 30 years since the dissolution of the group, Emerick has forged an active career as an engineer/producer, working with Paul McCartney on many of his solo efforts (including *Band On The Run* and the new *Run Devil Run*), as well as with other major artists such as Elvis Costello, Badfinger, John McLaughlin, Robin Trower, America, Jeff Beck, Split Enz, and Ultravox. Mild-mannered, unassuming, and soft-spoken, not only is Emerick one of the most respected, experienced, and knowledgeable engineers on the planet, he's also one of the nicest people in the business.

EQ: You assisted for Norman Smith on some early Beatles sessions. Did he share his recording techniques with you, or was he secretive?

Geoff Emerick: There were no secrets because there was nothing to be secretive about. The main concept that Norman introduced to me was that it's got to happen in the studio, above all. He'd say, "You can open one mic and you'll know whether you've got a hit record or not." So it was more of an artistic slant than a technical slant. I was never that technically minded, but I had sounds in my head. I've always described the job as painting a picture with sounds; I think of microphones as lenses. Engineering is such a wrong term for music mixing, really.

For instance, when you're mixing, you may see strings as a silver shimmer, something else as a golden, something else as dark, something as green, something as red. That's the way I hear things.

That's an interesting point: Laymen tend to think of sounds in visual terms. A technical person would hear something and think, that sound's got lots of 150 Hz; a nontechnical person would think, that sound's dark.

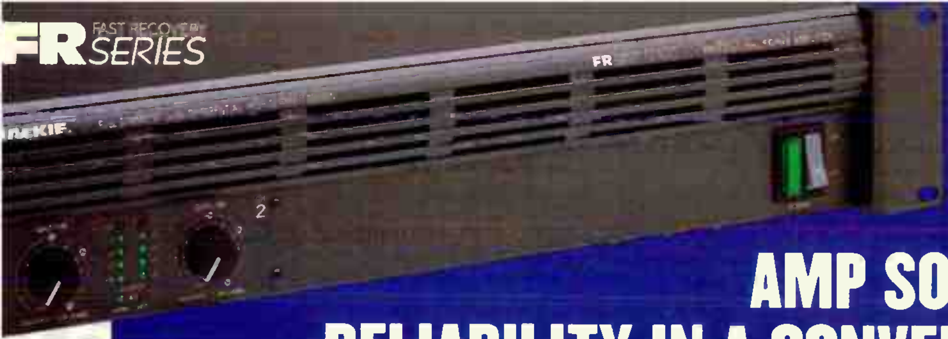
That's right. When I was younger, I had this thing with the maintenance people at Abbey Road. I'd do something and they'd say, "You can't do that because..." They'd go on about so many nanowebers of this and so many nanowebers of that, and I wouldn't even know what they were talking about! All I'd know is that I'd hear something and it would sound right to me.

So the training you received was just a matter of sitting back and observing?

Just observing. There was another good engineer at Abbey Road named Stuart Eltham, and I learned a lot from Stuart. But there was no actual training; it was up to you to pick up what you could. You can't really train anyone, anyway — it's like trying to train someone to paint a picture. If you can't paint, you can't paint.

How did you feel during that very first session, recording "Tomorrow Never Knows"?

Terrible! I had already been promoted to engineer, but I hadn't been asked to do The Beatles; I was doing other acts. I remember going up to the studio manager's office and them saying, "Do you want to do The Beatles?" They were explaining why, that Norman was leaving, that you're so young, blah, blah, blah. George [Martin] was there and a few other people. I was playing an eenie, meenie, minie, moe game in my mind, thinking, no, I can't do this, but if it stops on "yes," I'll just say, "Yes."



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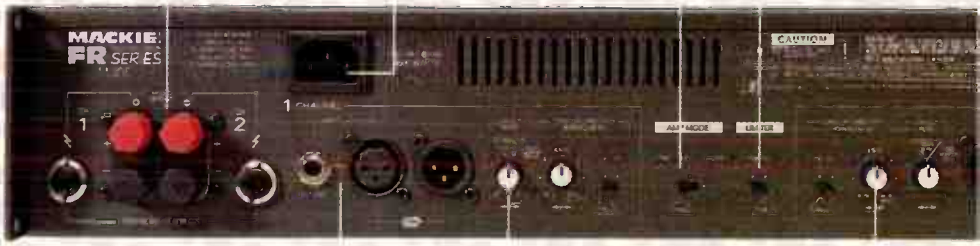

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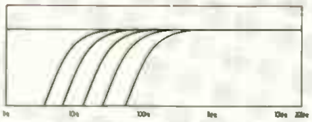

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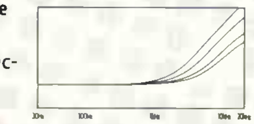

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[1] Music & Sound Retailer Magazine Survey February 1999

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What did I have to lose, really?

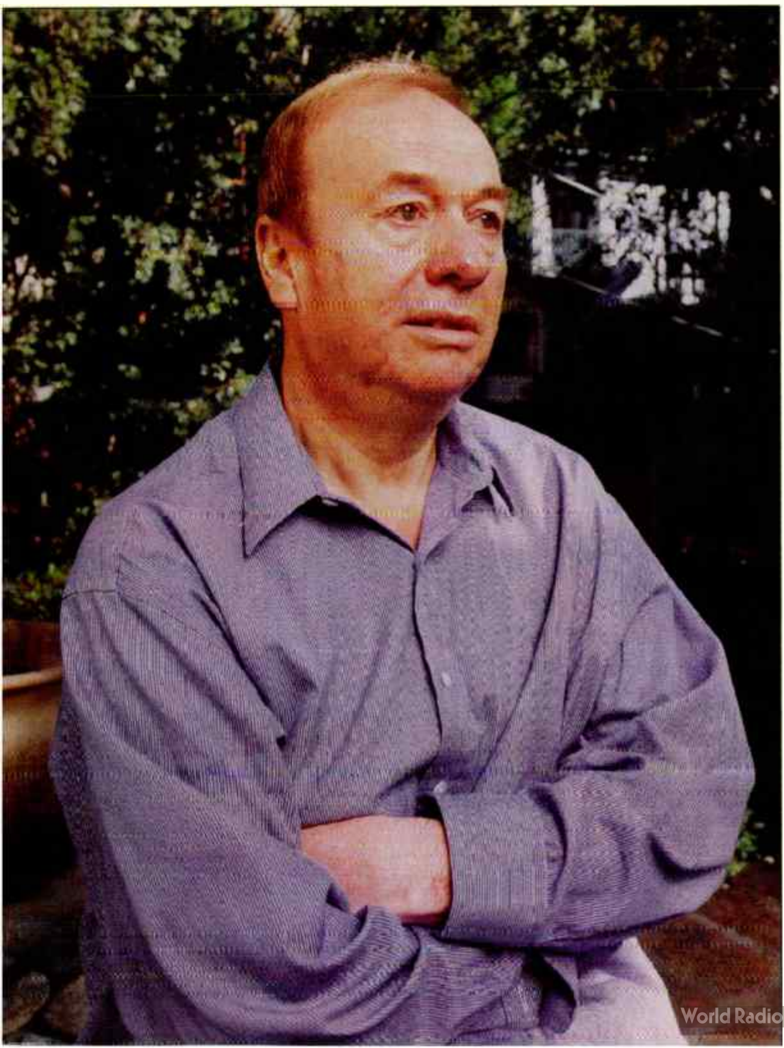
Did The Beatles accept you in your new role right away or was there a period of initiation?

They accepted me, more or less. I think Paul was OK; it was a little funny with the others. But it was alright after a week or so. Being young and somewhat innovative in my outlook, I guess I could really relate to George Martin. Although Norman would try things, there were certain things George would ask other engineers to do and they would say, "No, it's impossible, you can't do that." At least, together, we'd try things, so that opened up a whole new world; we could do anything he wanted because I was willing to do it.

Every day we'd try something new. We were listening to American records, and the bass was so good. I remember thinking, "How the hell did they do that?" We wanted to get more clarity into Paul's bass sound, and we tried everything, even to the extent of using a loudspeaker as a microphone! The theory was, if you can push signal out of a loudspeaker, you can use it to receive signal, and it should push it out with a bigger sound. I believe we used that on two tracks on *Revolver*, but I can't remember which ones.

What was the secret behind those incredibly rich, creamy bass sounds on *Sgt. Pepper*?

Even though we were working in 4-track — sometimes doing one or two reductions onto another machine — by the time we got halfway through *Revolver*, we always opened up one track that was purely for bass. Before, we were cutting the bass and drums together and then, when we'd mix to the second 4-track, we'd combine the bass and drums onto one track. The thinking was that if we wanted more bass later, we could turn up the bass EQ, and if we wanted a bit more drums, we could turn up the treble EQ.



On *Pepper*, I was bringing his bass amp out of the baffles because it was being done as an overdub. We'd do it late at night when everybody else had gone and we'd spend quite a bit of time on it. I've actually seen Paul's fingers bleeding from pulling out the notes, getting a note to speak properly. I used a tube C12 mic and put it about 6 or 8 feet away from the speaker cabinet and put it on figure-of-eight [polar pattern]; I also used an Altech compressor, though I don't remember the model number. And that was it, basically.

So you didn't take a DI signal from the bass?

Never, ever. I've never DI'd Paul's bass.

Do you remember what his bass amp was?

It was probably a Vox.

Were there a lot of bass punch-ins?

Oh, no. It was a performance. On those 4-track machines, you couldn't do quick drop-ins; bass drop-ins were especially hard to do.

Although he changed to the Rickenbacker at around that time, I don't think it was so much the bass guitar he used, I think it was having the mic on figure-of-eight. With the studio empty, you could actually hear a little bit of the ambience of the room around the bass, which seemed to help. At one time, on *Revolver*, I was trying to get extra definition on the bass, so I used to put a little chamber [reverb] on it — just a touch. Paul always detected it and hated it, so nine times out of ten I ended up taking it off; occasionally, it got through. But that gave the sound a certain roundness and put it in its own space, and that's what I was looking for. We had no electronic gimmick boxes — everything was organic, if that's the right word. There were no phasers or flangers; there was only so much we could do.

The sound you crafted transformed the bass from a supporting rhythm instrument into a lead instrument.

It surely did, unbelievably so.

It's a compressed sound, but it's not overcompressed, not squashed.

That was the Altech compressor. The Fairchild couldn't take the bass signal because the attack time was too fast.

The other thing I used to do when I was mixing — and Norman Smith taught me this — was that the last instrument [that I brought in] was the bass. So, at least through *Pepper*, everything was mixed without hearing the bass. I used to bring everything to –2 on the VU meter and then bring the bass in and make it go to 0, so it meant the bass was 2 dB louder than anything on the record, it was way out in front; the loudest thing on the record.

On "Rain," I may have compressed the bass two or even three times, just to give it no dynamic range whatsoever and get it way out front.

Would that recompression happen during the mix?

No, it would have been done during recording. The song would have been overall compressed during mixing because we were going to vinyl, so there were problems in mastering albums. It also had to be well in-phase because it helped make the end product sound good on vinyl.

When you were recording McCartney's bass, did you typically use EQ on the board?

The EQ on the board was just treble and bass, so I had very little control in the control room; I can't even remember what frequencies they were. I can remember it taking up to an hour to get a bass sound, doing most of the EQ from the guitar and the amps. Most of the sound originated in the studio.

Let's talk a little bit about the approach you took to drum sounds and drum miking.

Norman used to sometimes use the Telefunken 4038 ribbon mic, which is a fantastic mic. If the kit's well-tuned, you just have to put in some top EQ and you get the most unbelievable drum sound. But I was looking for new drum sounds, so I started close-miking all the toms and also putting a mic underneath the snare as well as a top mic. The top mic normally was an AKG D19C, which they've stopped making; it was a very cheap mic, made for talkback on the desks, but, for some reason, it sounded good on drums. We used to use a 56 or 54 condenser underneath, though often the capsules would go. At some point, we started taking the bottom skin off the toms and put a D19C underneath each of the toms, just to get a bit of tone. That was pretty much it — underneath and overheads on the toms [both D19C's], left and right overheads [Telefunken 4038's] for the cymbals, over and under the snare, hihat, and bass drum [B20].

So you had about a dozen mics or so on the kit.

By the time we got to the recording of *Abbey Road*, yes. You've got to remember also that all the equipment in the control room was tube, not transistor. The first transistorized equipment came in halfway through the *White Album*. When I went back and did the *Abbey Road* album, that was the first project that was done through the transistorized desk, which gave us more inputs.

Did you compress the snare drum?

We used to compress *all* the drums — including the snare drum — through a single Fairchild, at least up to the recording of *Abbey Road*. On those early transistorized desks, there was a good-quality compressor and limiter on every fader, which was a luxury. I mixed the *Anthologies* on that board — it was as near as I could get to the original sound, and I didn't want to color those tapes at all.

Having the tubes was a big part of it. I remember when I started the *Abbey Road* album [that] I couldn't get the same bass drum or snare drum sound that I'd gotten on the other records. The transistors wouldn't let the low-end distortion go through — everything was clipped.

Did Ringo tune his own kit?

Yes, and he always used to have a half-empty soft packet of Chesterfield cigarettes on the top of the snare, which gave it a certain sound. I think it was Chesterfields — you could put a soft packet of Peter Stuyvesants on there and it wouldn't sound the same! [*Laughs.*] No, I'm sure it was Chesterfields — the Stuyvesants never sounded the same. It just shows you how sensitive all that stuff is. We used to use tea towels on top of the drums, too; just mopping-up cloths.

Did the kit get moved into different areas of the studio for different tracks?

No, it was always in a locked position. A lot of *Revolver* was done in the smallest of the Abbey Road rooms, so there weren't a lot of options for drum placement.

What prompted you to move on from the basic three-mic setup to start experimenting with additional mics on the drums?

Everyone had always just put those mics there, but, for me, it was a question of putting your ear near instruments and hearing what they really sounded like. Plus, the request on certain tracks was, "I know that I'm playing piano, but I don't want it to sound like a piano," or, "I don't want this guitar overdub to sound like a guitar." So it ended up with me wandering around everything and putting my ear in different places and wondering what would it sound like if the mic were a 1/4-inch away from the skin? Or if the cymbal were a jack socket in your throat." He couldn't grasp that, he didn't even know what it really meant.

They weren't interested in technicalities, they were artists. Paul, even today, is not really interested in technical things. You start talking technicalities to any artist and they don't want to know.

According to [producer] Jack Douglas, Lennon always complained that the ADT at the Record Plant didn't sound as good as it did at Abbey Road.

No, it wouldn't, and I'll tell you why. Because it was tube equipment, we used a sweep oscillator, going from 20 cycles up to about 80 cycles. That changed the voltage of the rectifiers that went into the motor of the machine; it was a huge piece of equipment. But it also meant that you could actually play the guitar note with the varispeed, following the string bends. I used to play the varispeed to Lennon's voice to get that ADT sound, and you can't do that with the vernier scale knob you'd find on more modern varispeeds. Also, each tape track was 1/4-inch wide, so you could get absolutely perfect phasing. Everything had to be perfectly lined up to get that sound, but it lends itself to being a musical effect.

What was the standard Lennon vocal slapback echo?

It was just a straight tape echo at the ordinary 15 ips or 7-1/2 ips speeds. But don't forget that the length of the echo depends upon the gap between the record and replay heads, which is different on every machine. We used big EMI machines, so there was a considerably bigger gap than on today's smaller machines, hence a longer echo.

Jack Douglas has also talked about Lennon "working" the mic, singing certain words to the back or sides of the mic and flashing his esses by waving his hand.

That was a technique that they developed over the years. When artists used to come in for their first artist test — and Lennon must have gone through it — they were given certain instructions. For example, they were told to pronounce the letter "p" as a "b" so you don't get the pop, and they were taught to lean back on loud phrases. Flashing the esses, I'm not so sure about that. Paul's got a good mic technique as well.

What vocal mics did you use?

Mostly [Neumann] U 47's and U 67's. I'm sure we also used other mics, because there was this thing about, "Oh, it's the same old vocal sound." I remember, on one of John's vocals, getting a glass milk bottle and getting a very thin condenser mic and putting the mic in a plastic bag inside the bottle, then filling the bottle with water. The condenser mic, of course, had power within the capsule, so if water had gotten in, God knows what could have happened — the whole thing could have ended in death and disaster. But John sang to the milk bottle, so it was actually picking up the vibration of the glass, through the water, into the mic. I can't remember which song that was on, but it was one of the songs on *Sgt.*

Pepper. We'd do anything to be different.

Were vocals typically routed through the Fairchild?

Always. Just the sound of the amplifier, even if you didn't do any limiting, it just added a certain presence. We used it on guitars, too.

The lead vocal in "I Am The Walrus " sounds like it was sung through a fuzz pedal. Were there fuzz pedals available then?

Yeah, there were, but that was a bit of overload, as well as the way John was singing it. Those mic amps were like their own fuzz pedal — it wasn't a nasty distortion, it kind of helped the signal.

Did you come up with any specific mixing techniques during your years with The Beatles?

Well, everything was monitored in mono. If you're working in stereo and you have two electric guitars, one left and one right, it's very easy to pick them out. But if they're both coming out of the same loudspeaker, they each need their own character, so you have to spend a lot of time getting the sound right, placing each one individually in its own space. You wouldn't do all that extra work if you were mixing in stereo — that's the easy way out. I'd be filtering out various frequencies so there was no interference. For example, the only bass content would be coming from the bass guitar. That's why the sound's got so much definition, because there's nothing interfering with it.

So even the kick drum doesn't have a lot of low end; the bass guitar is actually below it.

That's right.

Did you have that in mind when you were recording the kick drum and bass guitar?

Not really. It was because, in the final mix, everything was coming from one single sound source. When we did give ourselves a treat and monitored in stereo, it really sounded fantastic because of all the work that had gone into the individual sounds.

When mixing, what sort of things did you do to differentiate Lennon's guitar from Harrison's?

It was just basically EQ and maybe a slight bit of one echo chamber on one guitar and a different chamber on the other. We very rarely used an EMT plate; we almost always used live chambers instead.

George Martin has said that *Sgt. Pepper* wouldn't have been as good had it been recorded in 24-track, that necessity had been the mother of invention.

I agree. We were put on the spot, and that was the sound you made at the moment; you had to put the right echo on, the right EQ, the vocal had to be right. It made things easier in a way, because otherwise there are too many variables, and what's the point? Where do you go? To me, that's why there's no great product today. There's good product, but nothing great.

Do you ever think there will be another Beatles, another artist that will so completely dominate the music scene?

No, because of the way record companies function now; they're purely a money-making machine. When I started, there was more of a focus on artistic considerations. Sometimes it didn't matter if the record made money, as long as the artistic aspect was put out to the public.

How do you see the role of producer as differing from that of the engineer?

The primary role of the producer is to pull the most talent out of the artist, to inspire the very best performance, even to the point of frustration if necessary.

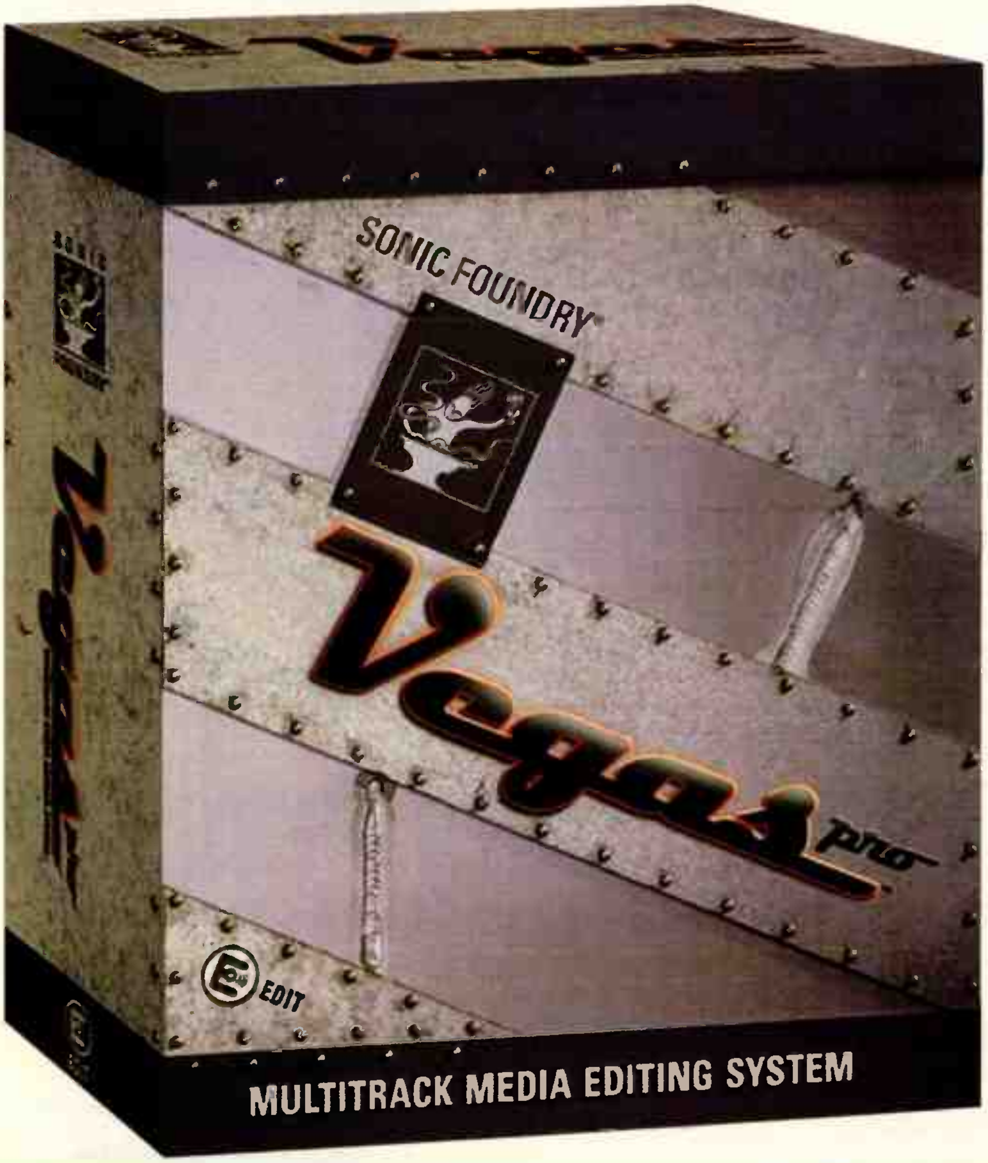
If you had to pick one single track that you were most proud of — in terms of your role as engineer — which would it be?

[*Pauses.*] I don't know. That's a tough one because there are so many. Probably "Tomorrow Never Knows." Or maybe "All You Need Is Love," because it was such a nightmare, doing it live and via satellite. Or "Eleanor Rigby," because of the string sound. I was also very proud of the work I did with John McLaughlin (*Apocalypse*, 1974).

This interview is excerpted from Howard Massey's upcoming book Conversations With Record Producers, *which will be published by Miller-Freeman Books in fall 2000.*

SONIC FOUNDRY

Veg



An intuitive new interface worthy of the city it takes its name from highlights Sonic Foundry's latest multimedia maker

Select the Record button: a track is created; input level indicators begin to jump; a waveform display blossoms in the track window, forming a trail behind the cursor; and time is counted out on a large numeric display. Within seconds of starting Vegas Pro, you can be laying down tracks. This immediacy is at the heart of the shiny new, multichannel, audio production tool from Sonic Foundry.

Without the inertia created by an existing (multichannel) user base, Sonic Foundry was free to develop new user interface concepts that form the core of Vegas Pro. Their success in building a fast and effective 2-channel audio editor, Sound Forge, has proven they know how to code for audio performance.

So Vegas Pro offers lots of technical capabilities such as: 24-bit/96 kHz audio tracks, which can also include regions in other bit depths, sample rates, and file formats; automatic crossfades; effects inserts on each channel; DirectX plug-in support on each bus; support for dual processors, dual displays, and multiple I/O cards; video timeline and preview displays; and audio (and video) scrubbing. There are many other features, including file formats and metadata specifically for producing streaming multimedia content for the Web.

The result is a powerful production tool for audio, including audio-for-video (in AVI, MOV, and MPEG formats). Hardware may become your limitation, as the software will support up to 32 assignable DirectX effects and 26 auxiliary outputs. Inevitably, bottlenecks within the PC, such as HD throughput (even with DMA), available processing power, and your audio I/O card's capabilities will ultimately limit the total number of tracks (unlimited in software) that can be simultaneously recorded and played back on any computer-based recording system.

It is, however, the fresh, clean design of the user interface that really sets apart this software. The program offers some innovative, yet practical, new approaches to managing the display of tracks, effects, audio clip libraries, and output busses. The most obvious distinction is the lack of separation between the tracking and the mixing aspects of the interface. The most pervasive difference is the use of 32-bit Windows conventions (as opposed to fixing Windows 3.1 or Mac OS legacies) in a way that makes sense to the process of developing complex audio tracks. Tracks behave in a "Windows" fashion, including the familiar Minimize, Maximize, and Restore buttons for managing the individual track views. The views of output busses, plug-in effects libraries, audio file libraries, the edit decision list, video preview, and audio trimmer window function as floating or dockable windows. These sub-windows include the typical "Close Box" and dock neatly below the Track View

vegas pro™

MULTITRACK AUDIO EDITOR

BY WADE McGREGOR

EQ LAB REPORT

MANUFACTURER: Sonic Foundry, 754 Williamson Street, Madison, WI 53703. Tel: 608-256-3133. Web: www.sonicfoundry.com.

APPLICATIONS: Multichannel audio production for multimedia applications, including the Web.

SUMMARY: A fresh new interface that provides efficient audio production for everything from low bandwidth Web audio to 24-bit/96 kHz DVD audio productions.

STRENGTHS: Uncluttered on-screen interface; handles multiple file formats within individual tracks; very fast displays; good track processing.

WEAKNESSES: Trimmer window functions are weak; doesn't support external MIDI hardware controllers; relies on Windows drivers for audio I/O features; doesn't directly address available DSP for mixing or processing.

PRICE: $699 **EQ FREE LIT. #:** 101

SONIC FOUNDRY

Vegas pro

MULTITRACK AUDIO EDITOR

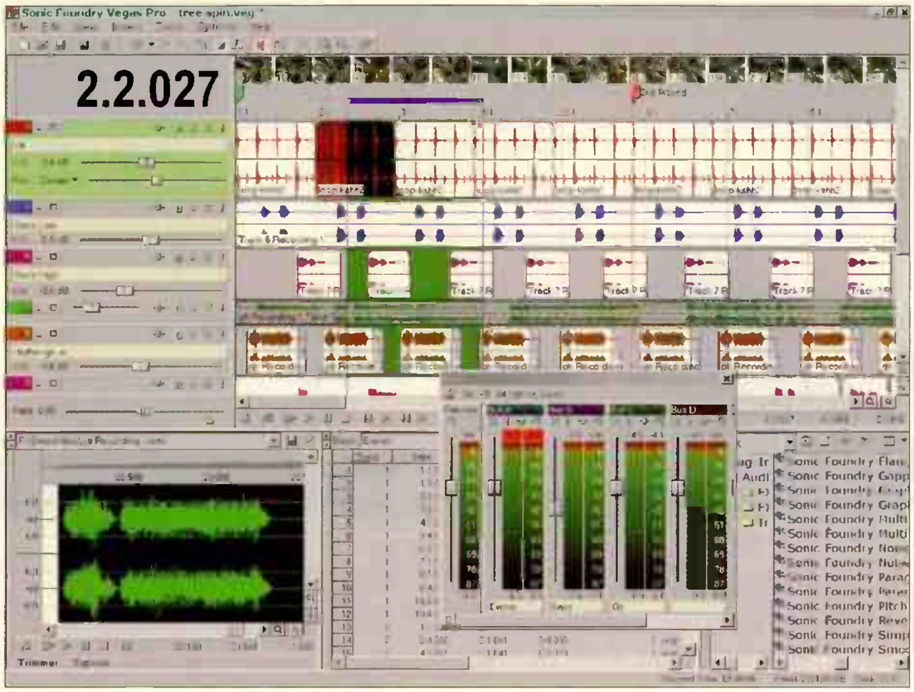


FIGURE 1: The Vegas Pro user interface places track-related controls in line with each track and offers a Mixer window for the output busses. Video is shown along a timeline (above the track waveforms) and all the various dockable windows can be arranged on a single screen.

window. Track functions, such as fade in/out; region (audio clips within a track) level; region boundary selection; and region location are all determined by the mouse pointer location within each track's regions. You needn't waste time mousing to and from the tool bar. Dragging a selected region of audio to another track can automatically create a crossfade, if an overlap over an existing audio region occurs.

Simple icons have been used to indicate the common functions of tracks and busses (see fig. 1). The overall volume of each track is controlled by a horizontal fader (in the Track View), and does not appear in the virtual mixer that is all too common in other applications of this type. This clarity in approach can only be made in the first iteration of a program. Thereafter, this form of innovation would be an unwelcome surprise to existing users. Current Sound Forge and Acid users (the SF product, not necessarily the pharmaceutical...) will find immediate familiarity with many Vegas Pro functions. However, there is a lack of slavish borrowing from these successful applications just to make them seem more similar.

Patching busses to (multichannel) outputs can be setup as a default, but is readily available on-screen (see fig. 2). Unfortunately, I found that the Yamaha DSP factory had to be reset using the Yamaha Patch application on first use of Vegas Pro (and after updates). Otherwise, audio assigned to a single AX44 output would be sent to all of the outputs. Selecting 24-bit as the default file format forced me to disable the SoundBlaster AWE64 Gold audio card (typically used for solo/preview on my system), as it didn't support sample depth beyond 16-bit. Thankfully, the program allows the user to determine whether projects generate new audio files (for security) or use existing files (to save HD space).

Laying tracks, dividing them into segments, slipping them around, and dropping other audio files into the project is all very simple and straightforward. All edits are nondestructive, never touching the original audio files. I would prefer the key-

FIGURE 2: Output routing is selected by selecting the (blue) routing icon, which brings up a list of available hardware outputs.

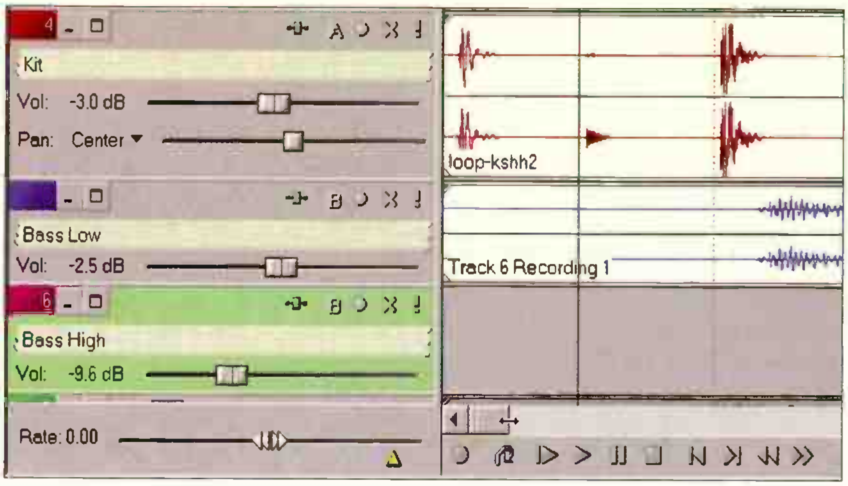


FIGURE 3: The horizontal slider for the Track View window can be dragged (double arrow cursor) by either end to quickly zoom in/out of the track waveform display.

board shortcut for Record to be a single button (currently CNTL+R), though, because sometimes my hands are nearly full. The bulk of the shortcuts, however, are familiar (such as space bar = stop/start) and expedient. A wonderful innovation is the integration of the horizontal slider (Track View window) with the zoom function (see fig. 3). The scrub function works well, although the speed steps incrementally rather than smoothly changing speed like an analog multitrack.

Selected regions of audio within a track can be displayed in a Trimmer window for closer examination, even down to the sample level. As this view offers almost no

continued on page 124

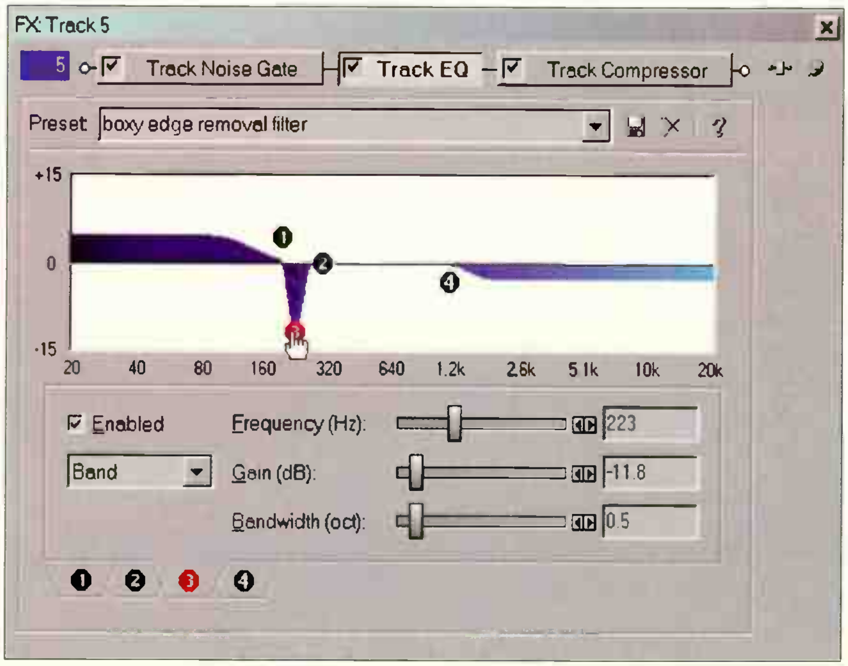
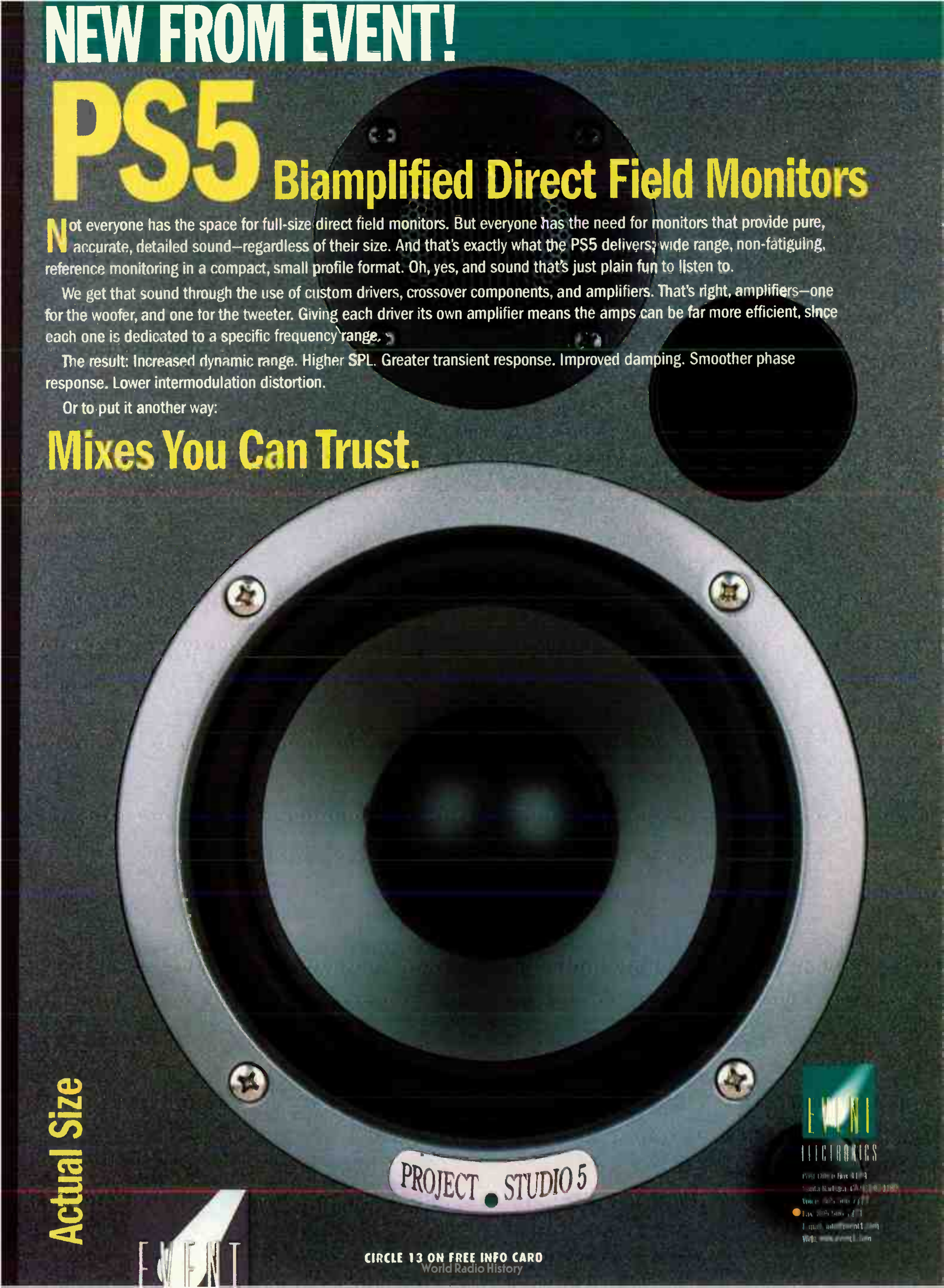


FIGURE 4: A chain of up to four dedicated effects (Dither not shown) can be inserted into individual tracks. The 4-band parametric EQ (shown selected) is comprehensive, yet simple to use.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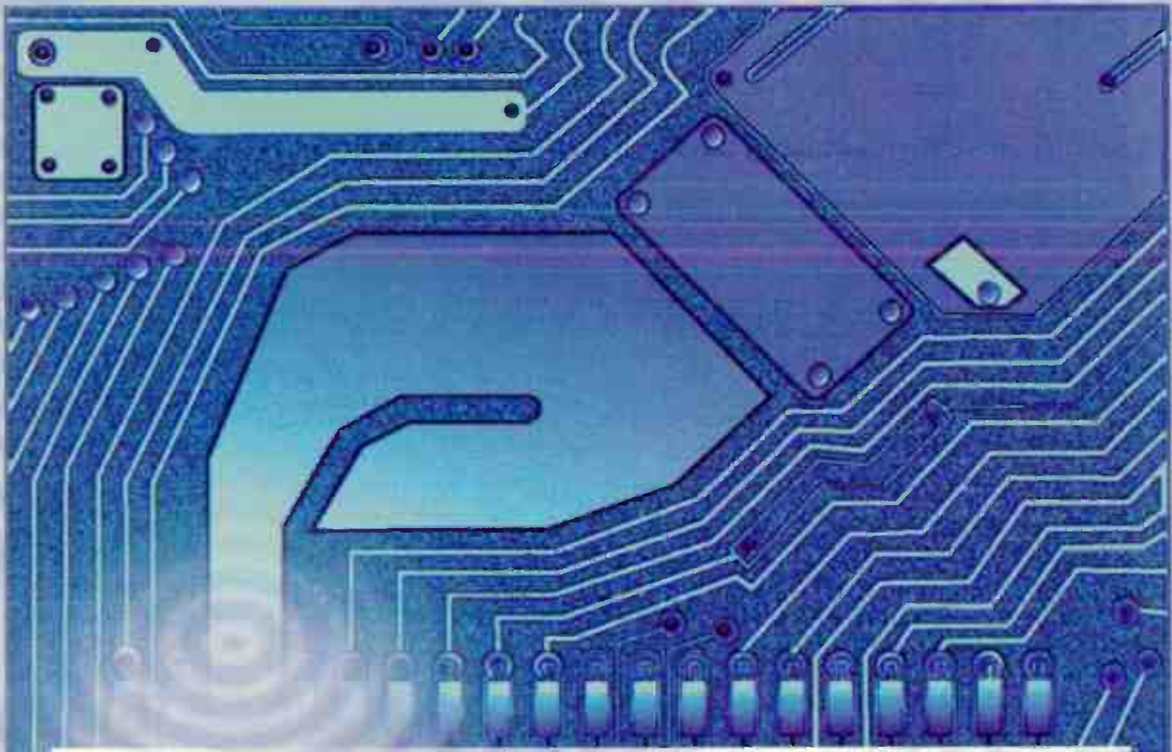

Cashing In The (Audio) Chips In 2000

Driven by ever greater demand for resolution in digital audio and the onslaught of digital delivery mechanisms, chips, from DSP to semiconductors to DACs, are among the hottest commodities in the pro audio business. This Audio Foundations special advertiser section provides information on some critical developments that will affect everyone in the industry for years to come.

Adding to the pressure for higher performance in digital audio is the impending release of DVD-Audio which is beginning to spark demand for high-resolution digital mixes. Further, within the disparate audio disciplines of film, television and music it is universally recognized that improved resolution throughout the production process translates to the final product regardless of the resolution of the release format. For these reasons, equipment capable of supporting not only high quality A/D and D/A conversion as well as 24-bit I/O but also double (88.2 or 96k) sampling will become the standard for digital studios.

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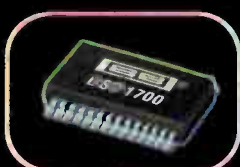

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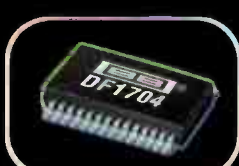

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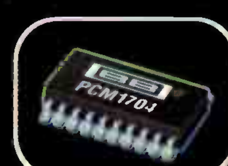
PCM1704



- Sampling Frequency (fs): 768kHz max
  (192kHz x 4 or 96kHz x 8)
- Input Audio Data Word: 20-, 24-Bit
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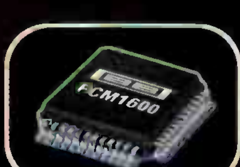

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- SNR: 105dB
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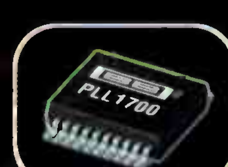

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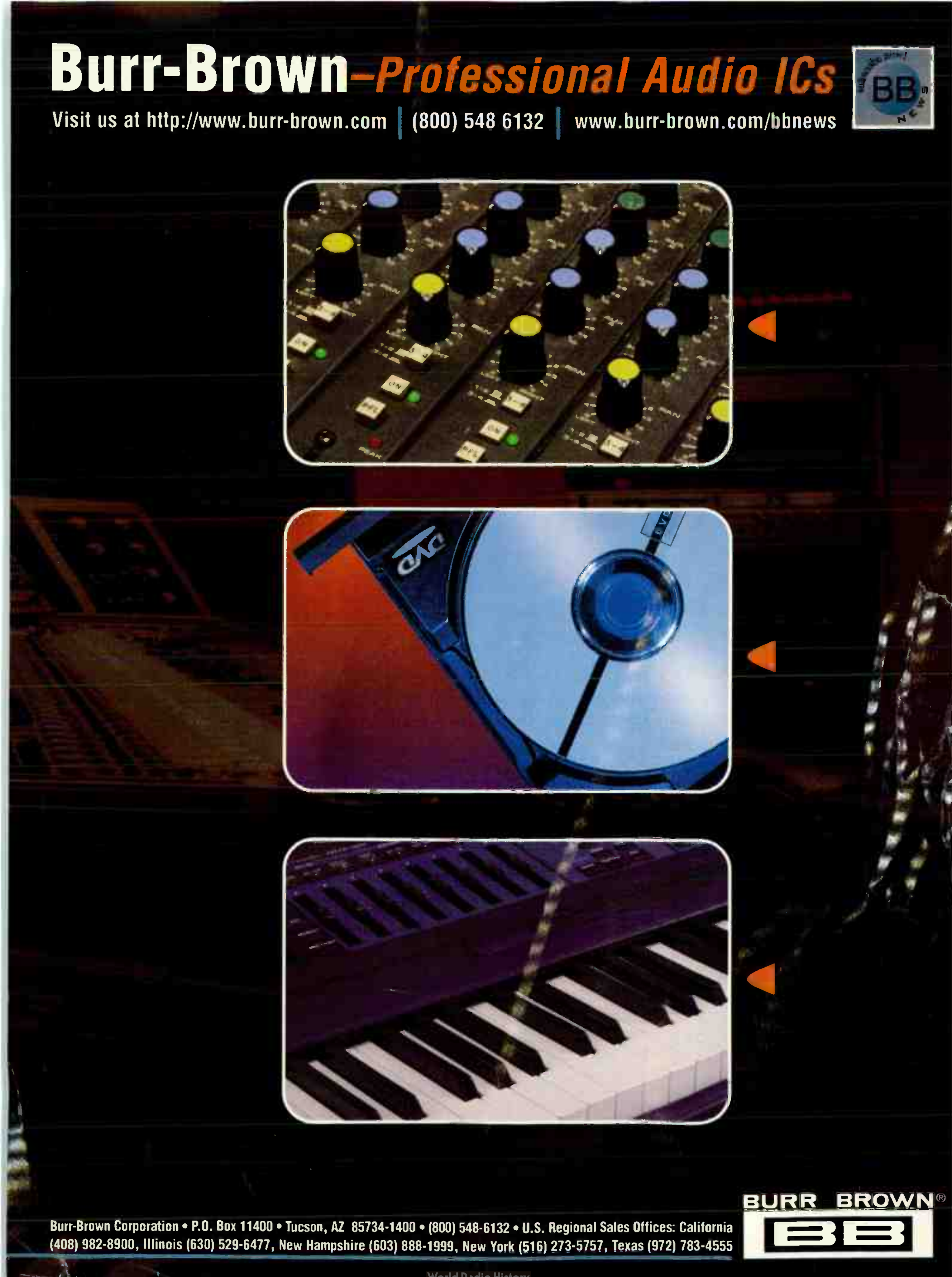
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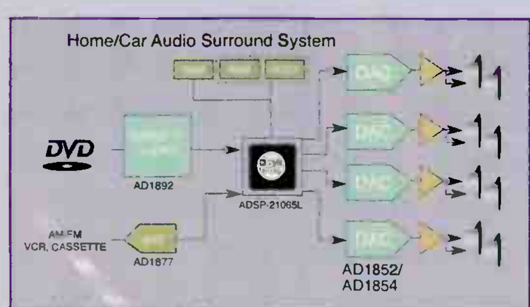
ANALOG DEVICES

AD1854—Stereo, 96 kHz, Multibit ΣΔ DAC

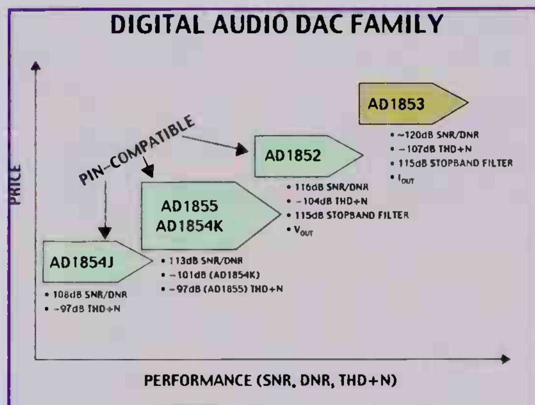
The AD1854 offers professional audio performance at a consumer price. The AD1854 is the second generation of the award-winning AD1855 (ISSCC 1998 paper of the year). It enables consumer electronics manufacturers to offer theater sound in their home theater products such as DVD players and A/V receivers. It also allows car audio equipment makers to offer "home theater sound" on the road.

The AD1854 uses ADI's exclusive multibit ΣΔ modulator with "perfect differential linearity" for reduced idle tones. It also features ADI's patented data directed scrambling to minimize sensitivity to jitter. The AD1854 achieves up to 113 dB dynamic range and signal-to-noise ratio without muting. It is available in two grades: The AD1854KRS offers 113 dB dynamic range (A-weighted) and -101 dB THD+N; the lower-cost AD1854JRS offers 108 dB (A-weighted) dynamic range and -97 dB THD+N. The AD1854 supports 24 bits and up to a 96 kHz sample rate. It also includes a click-less on-chip volume control.



AD1852—Stereo, 24-Bit, 192 kHz, Multibit ΣΔ DAC

The AD1852 supports the 192 kHz sample rate now included in the DVD-audio specification. It is a lower-cost version of the AD1853, which was the first complete audio DAC to support the 192 kHz sample rate. The AD1852 is fully compatible with sample rates from 32 kHz up to and including 192 kHz. It also achieves 116 dB dynamic range, signal-to-noise ratio without muting, and -104 dB THD+N. The AD1852 also features a superior digital filter with 115 dB stopband-attenuation. This means the AD1852 (and AD1853) offers the lowest out-of-band noise on the market today! The AD1852 uses ADI's exclusive multibit ΣΔ modulator with "perfect differential linearity" for reduced idle tones. It also features ADI's patented data-directed scrambling to minimize sensitivity to jitter. The AD1852 also includes a click-less on-chip volume control. The AD1852 is targeted to the performance requirements of high-end DVD players, DVD-audio players, CDs, A/V processors, mixing consoles and digital audio effects processors.



APPLICATIONS
- DVD and DVD-A players
- audio video receivers
- automotive audio systems
- mixing consoles

AD1852	Price $5.50
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AD1854J	Price $4.00
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at any sample rate

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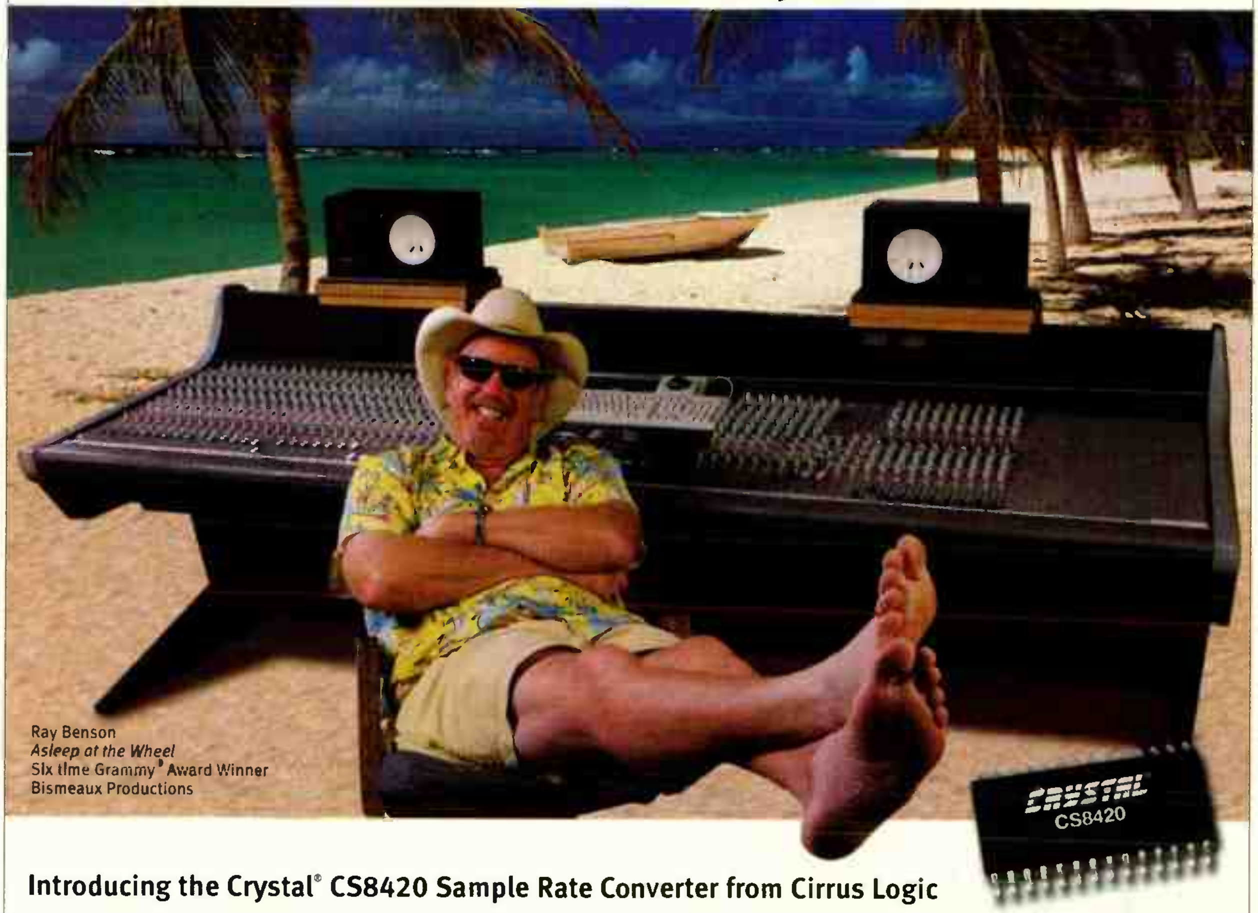
Contact: Fred Valenzuela, Crystal Audio Products Division at Cirrus Logic, 4210 South Industrial Drive, Austin, TX 78760, 512-442-7555, Web:www.cirrus.com.

Don't sweat the sample rates



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Designed for Your Needs

The CS8420 is a sample rate converter with excellent analog performance characteristics. Here are several good reasons why you can use it with confidence in your professional audio application.

- Complete AES3, SPDIF transceiver provides an integrated solution that eliminates separate transmitter and receiver chips
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To learn more about the Crystal® CS8420, kick up your feet and visit our website at www.cirrus.com or call (800) 888-5016 for more information.

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Audio Foundations

THAT Corporation 10 YEAR ANNIVERSARY 1989 1999

Do VCAs Really Color Sound?

While it is undoubtedly true that VCAs, like any electronic device, have some subtle sonic quality, they are far more transparent than is often thought. The "sound" of THAT Corporation's VCAs, in particular, is overwhelmingly the result of the external circuitry supporting the VCA. This is especially true of our latest designs, the THAT 2180 and 2181, perhaps the most sonically neutral VCAs available. Compelling evidence for this conclusion comes directly from the marketplace, where we regularly read reviews or hear from end-users that such-and-such compressor/limiter sounds great, but so-and-so sounds dreadful. All too often, the difference is cited as the (supposedly different) VCAs used in each product, when, in fact, the VCA is exactly the same. It is the external circuitry making the difference in the sound! We have designed or tested hundreds of VCA-based circuits, from feedback compressors to automatic gain controls, from noise gates to noise reduction, and if there is one comment that can be uniformly stated about the sound of these various devices, it is that the sonic character is overwhelmingly determined by the application circuit, particularly in the gain-control or "side-chain" processing.

This should not be too surprising, since it is in the side-chain that the VCA's dynamic response is determined, and where unwanted noise components may be introduced. As a result, subtle design decisions strongly affect the sound that is ultimately delivered. The type of level detector used, the amount and quality of pre-detector filtering, the design of threshold circuitry, even the board layout itself — all will have a contributory or determining influence on the final sound.

Other gain control components, like vacuum tubes and optical devices, may have their own sonic signatures. The same cannot be said of high-performance VCAs. Their transparency, for better or worse, puts the designer in complete control of the sound.

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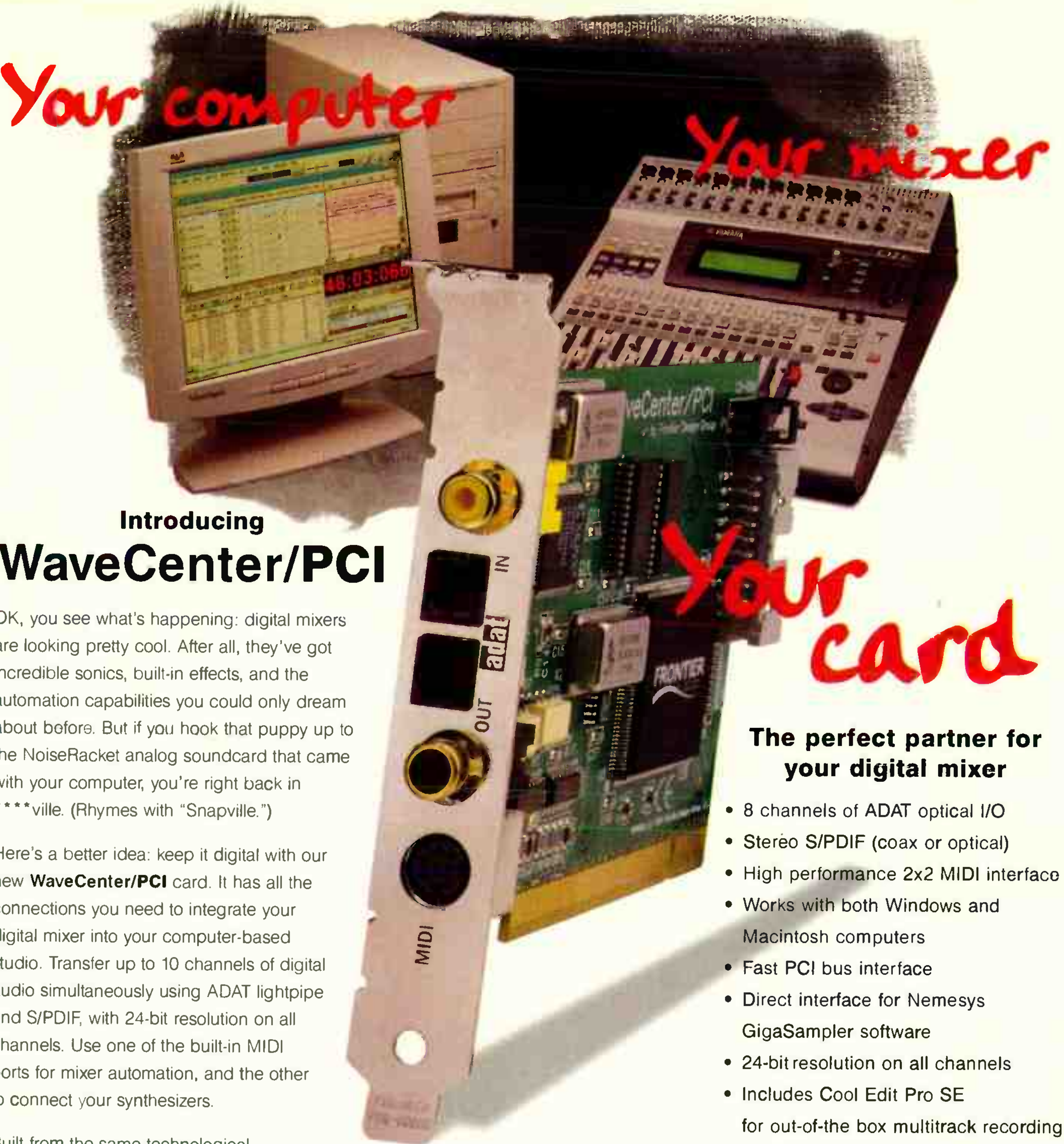

Introducing
WaveCenter/PCI

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

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

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EQ Live

Keyboardist Jimmy Destri takes to stages once again with his band, Blondie

PHOTOS BY TONY DI LORENZO

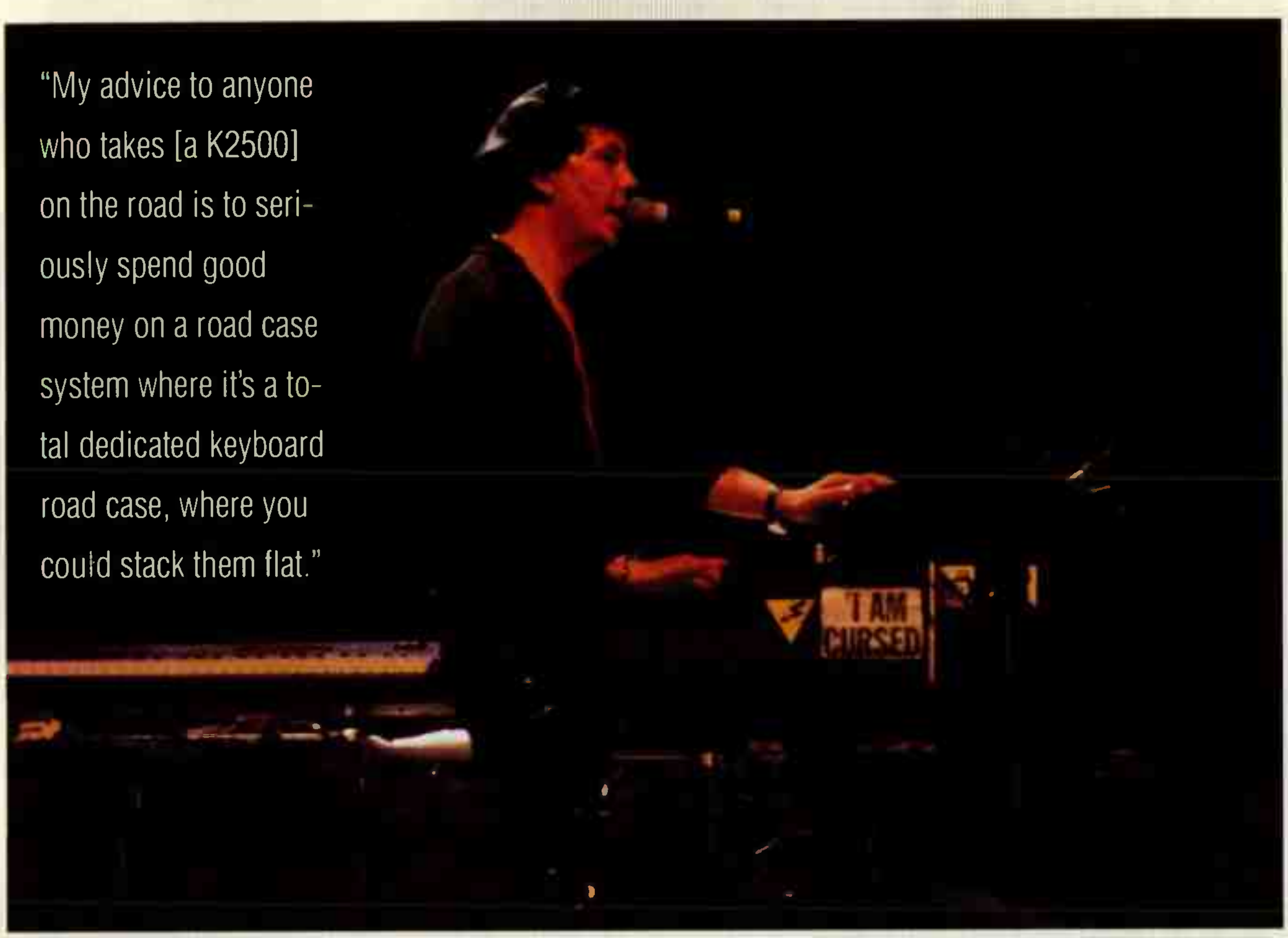
DESTRI RIDES AGAIN

BY TONY DI LORENZO

There was a time when our airwaves were dominated by sultry, sexy vocals, tightly woven keyboards, and guitar textures driven by some seriously "in-the-pocket" drums. Songs that had what every songwriter prays for: hooks! Hooks that stayed with you, that had you humming, whistling, and singing along well after the songs were over. Songs like "Atomic," "Call Me," "Dreaming," "Heart Of Glass," and "One Way Or Another" — these songs were definitely part of the soundtrack to our lives, as played by Clem Burke, Jimmy Destri, Debbie Harry, and Chris Stein. You know them collectively as Blondie, and they're back!

Whether Jimmy Destri is laying low to support Debbie's vocal or driving an organ chord right through your speakers, he knows what parts to play and where to play them. Take the Destri-penned hit "Maria," a shinning example of his strength as a songwriter. EQ recently had the opportunity for a sit-down with Destri to discuss his work with Blondie on their current tour, as well as in the studio for the making of their most recent album *No Exit*.

EQ: What synths were used for the



making of *No Exit*?

Jimmy Destri: The Kurzweil K2500x and K2500sx. This is a fully blown one with an internal hard drive and lots of memory. It's really a wonderful machine. It costs so much less than the maxed-out model, but you get just as much. You don't get the brushed chrome exterior. I was warned that the manual would drive me crazy. It turned out that I still haven't gotten through the manual, but, at a certain point in Kurzweil-eezz when you're going through the thing and using it, a light bulb comes over your head and you say, "Ahh, this is the method you're using. This is how they distorted Ray's vision." [Jim is talking about Raymond Kurzweil, designer of the original K250.]

I used to own a Synclavier II, and, of course, it was an old 16-bit machine with FM synthesis. It didn't have MIDI, but the great thing about the Synclavier was that the FM synthesis strings, horns, and a lot of their other sounds were so bizarre, so wild, fat, and thick — that it really had a character of its own. I haven't seen a keyboard match that until this Kurzweil series came out.

So the K2500 is like a Synclavier for the '90s and beyond...

Yeah, I really think so. What the K2000 lacked in memory it made up for it in the extent of its sounds and factory programs, which were just brilliant. It's a point where you could just leap off. This is even better (the K2500) because, besides having some of the same factory programs (and more), this has more memory, the ability to save songs, a mixer, a full sequencer, and total MIDI control.

Are you using the same rig live as for the record?

Yes. My live rig and my studio rig are the same. I didn't use any extraneous keyboards on the album. I didn't even use a regular grand piano — which I've used on every Blondie album — because I like the Kurzweil piano so much.

Wherever there is piano on the record, it's really the K2500?

Yes, and that was one of the reasons for getting the 88-note keyboard. When you dial up the piano sound, you really have a piano in front of you. If you're playing classical piano or if you're Elton John or someone who relies heavily on real piano when they're making a record, then you'll hear a difference. You don't get the wood resonating like when you mic up a piano. It's squashed down into the digital sample. But, for Blondie music, I don't think it matters at all. It was just so easy just to dial the sounds up and use them.

It does have it's flaky moments and it's not a real road machine. I've been using it on the road, and we had to buy an extra one — a rack module — just as a back up. Every now and then we need to send one out to be fixed. My advice to anyone who takes one of those on the road is to seriously spend good money on a road case system where it's a total dedicated keyboard road case, where you could stack them flat. When you put them on their back on a truck, they get shaken up and most of the brains and boards get jarred.

What other keyboards did you use for "No Exit?"

I used my Hammond XB-5, which I love. I'm very chagrined because they've stopped making them. It's a wonderful keyboard. It's a Hammond organ with a full array of programmable sounds. The XB-5 has presets beside the drawbars on each keyboard, both upper and lower. I've EQ'd number five to sound just like my old Farfisas — really tinny, bright, and sharp on the low end.

I know you have a vintage collection. Did you use any of your vintage gear for this CD or tour?

No. I had the Polymoogs and I had a MiniMoog, but that sort of fried. I used the Roland JP8000 to approximate a lot of those sounds (vintage analog). It's pretty good. I've tried the Nord and some other stuff, but the JP8000 was really the best of the lot as far as being user-friendly.

So you're using the JP8000 to replace all your old analog synths.

Yeah, I like to use it in a non-programmable form, like an old analog synth.

You mean actually playing the knobs, opening and closing filters manually?

The great thing about the JP8000 is that, when you recall a program number, it goes back to the way the sound was set before you started playing with the knobs. I'm using the JP8000 for the sweeps in "Heart Of Glass" on my right hand and the sequenced "Duga Duga Duga" from the Kurzweil. The JP8000 is MIDI'd to the Kurzweil, so I just trigger the K2500 to do the sequence and it uses the bottom half of the JP8000. The sweeps are on top and the bottom is the bass.

So, on "Heart Of Glass," you're getting a little added bass boost from the JP8000?

Right. For that "Duga Duga Duga." All those sequenced lines are called "Duga" in Blondie-land because of Giorgio Moroder. When we were recording "Call Me," he came up to me and asked "Can you play Duga Duga Duga?"

And now every repetitious 16th note pattern is a "Duga Duga Duga."

Yeah.

Are you making use of the sampling capabilities in the K2500?

We did a few samples for the album on a song called "Dig Up The Conjo." We really make use of the sampling for live performances. My sister Donna sings all the backing vocals on the song "Maria" in a choral effect. I had our record company send me her tracks solo'd. These were the digital tracks off the RADAR. [*No Exit* was recorded on the Otari RADAR system —Ed.] Rather than sampling the whole phrase, I sampled each note so I could play it in time. I have my sister sampled for the backing vocals in that song and I mix it with the Kurzweil choir program, so it really works well. A lot of samples work well when mixed with their factory programs.

Did you sample any of your vintage synths or are you using a sample library?

No. I was going to, but we found that we approximated the sounds pretty much the same. I do miss some of the sample-and-hold effects from the Polymoog. I just let you hear a track I'm working on and I'm using a sample-and-hold effect that I made up myself, but I haven't really gotten into the V.A.S.T. synthesis capabilities of the Kurzweil. Anything with a data wheel and a numerical keypad scares me. It's not like an analog synth. Analog is very intuitive.

"All the sequencing is coming from the K2500. I could have run a sequencer from a laptop, but that takes away from the spontaneity of the show."

I found one program that could be used like a sample-and-hold type sound. Having to flip through all those parameters is hell. It's not like reaching for a knob [as on an analog synth]. You have to find what you're looking for while staring at this little blue screen forever. It's very digital, but it's great because I'll always have it once I save it.

Any problems keeping this rig alive and well on the road?

Oh yeah. Like I said earlier, we started off using regular flight cases, and — although they're good — when you're going from studio gig to studio gig, or doing a few gigs, when you're on a major tour and the synths are going on container ships, air freight, and trucks, then it's much better to have a flat storage case that takes a series of keyboards — one huge case that your tech can deal with and unload all the keyboards from. So we've got a flat storage case just for the K2500's. The Hammond is built really well. It hasn't really broken down. The only problem with the Hammond is that the woodwork has gotten a bit scratched up.

Tell me about your stage amplification for the keyboards. Are you using a submixer for the keys and then sending a feed to the house mixer?

We use the Yamaha 01v mixer and we have one as a back up. We like it because it's automated. When you change songs, the faders fly. It remembers the MIDI patches, volume, etc.

What's driving the program changes for the mixer?

It's all being controlled by the K2500. What we do is put each song from the set list into a bank of setups. Each song contains four or five different setups split across the keys. Some of those setups contain sequences. Peter Danilowicz (my keyboard tech) usually has the sequences. My setups contain different sounds. Let's say Peter changes his song from 201, which may be "Dreaming," to 202, which is "Hanging On The Telephone." "Dreaming" may not have any sequences, but it tells the Yamaha 01V to recall all the levels and necessary settings for "Dreaming."

So the whole show is being controlled by the K2500.

Yes. A lot of the show is being controlled by the K2500 because that's what [drummer] Clem [Burke] listens to. It's basically the padding and the orchestration (if you want to call it that), for the whole band. What we hear on stage is through the monitors. We have the Leslie off-stage. The house gets a stereo mix from my keyboards, a separate mix from the organ, a mono mix from Peter's K2500, and another separate mix from my Roland JP8000. We were using JBL powered EON-10's, but now we're using personal monitors.

You don't have a speaker set up on stage?

No. As a matter of fact, if you get too close to the stage, you start to hear mostly drums and vocal coming from the floor wedges. We solved that problem by putting out little PA speakers that face the audience and put out keyboard sounds. Most of my stuff comes out through the PA without any stage ambience. It comes from my mixer right to the PA.

So you're going from the Yamaha 01v to the house mixer and you're controlling EQ and volume from the 01v.

Yes. Our soundman, Vincent Kowalski, keeps it flat. Sometimes a synth may jump out a little too loud because I've been working on a program and I didn't bring that keyboard's volume back. When this happens, Vinny just brings that synth back in the mix. It's different in every house. He usually has to boost the keyboards in open-air gigs because they get lost in the wind a bit.

It sounds like doing it this way makes Vinny's job a little easier.

Yes, it's much easier.

Are you using any computer-based sequencing at all?

No. All the sequencing is coming from the K2500. I could have run a sequencer from a laptop, but that takes away from the spontaneity of the show. Then everyone has to follow that computer. If something goes wrong with the computer, then you can't play that night because everyone is so used to following the computer arrangement.

It's like you're doing a track date.

Right. If my K2500's blow up, we can still play a Blondie gig, I'll just play organ. The show should never depend on machinery. It should always depend on people.

What kind of exercises do you do to keep your hands in shape?

Just some simple scales. I accept the fact that I have a different style and I use what I can. I don't consider myself to be an adept piano or organ player. I just use it as a tool to get the song done.

We know you're touring heavily to support *No Exit*. After the tour, will you be doing another album?

Yes, we're really looking forward to that. We'll probably start in the spring or maybe earlier like February or March. Maybe start recording in April and get it out by Christmas. We'd like to put the album to sleep and then sit with it for a while rather than rushing it out. I would really like to record a track in between the two albums.

I'd like to see [guitarist] Chris [Stein]come up with something that's so very Chris. Let him point the next direction because he really is the visionary. I'm too confused. I could follow, but I'm too confused to lead!

Tony Di Lorenzo may be reached via e-mail at thefront@interport.net.

EQ Live

ROAD GEAR

EQUALIZED

Two dual-channel models initially comprise the new dbx 12 Series equalizer line — the 2U 1215 (featuring 15 2/3-octave bands per side) and the 3U 1231 (featuring 31 1/3-octave bands per side). Both

the dbx 1215 and 1231 provide standard features like dual channels, ISO frequency centers, ±12 dB input gain range, and switchable 18 dB/octave 40 Hz Bessel low-cut filters. dbx's engineers also added 45 mm faders; selectable ±6 dB or ±15 dB boost/cut range; a choice of XLR, barrier strip, and 1/4-inch connectors; balanced inputs and outputs; chassis/signal ground lift capabilities; and an internal power supply transformer. For more information, call dbx Professional Products at 801-568-7660 or visit www.dbxpro.com. Circle EQ free lit. #116.

DOUBLE ELIMINATION

Electro-Voice has introduced the Eliminator Double two-way stage system. The Eliminator features EV's exclusive Ring-Mode Decoupling (RMD) found in the EV X-Array Touring System. The two-way speaker system delivers an on-axis frequency response of 50 Hz to 20 kHz, with a long-term power handling rating of 300 watts. The system uses the EV DL15BFH 15-inch speaker to handle bass, while midrange and high-end duties fall to the EV DH2010A compression driver and High-Q 60 x 40-degree constant-directivity horn. The system also offers a companion 18-inch subwoofer — the Eliminator sub — to extend the low-frequency response. For more information, call Electro-Voice at 616-695-6831 or visit www.electrovoice.com. Circle EQ free lit. #117.

VERSATILE CONSOLE

The ML5000, Allen & Heath's new flagship live console, recently made its U.S. debut. The console is a VCA-equipped, 16 aux, eight audio group mixer with eight mute groups, matrix, and enhanced LCR and IEM facilities, providing big-league sound reinforcement capability in a compact package. Furthering the company's "dual functionality" concept, the ML5000 has a flexible group/aux structure, allowing engineers to quickly configure the mixer for FOH, monitor, and combined roles. In FOH mode, the ML5000 provides eight audio groups, eight VCA groups with scene automation, 16 auxes, 8-way matrix, and left/right/center outputs. For more details, call Allen & Heath at 801-568-7660 or visit www.allen-heath.com. Circle EQ free lit. #118.

SPIRITED

Spirit By Soundcraft's 324 Live is a compact digital mixer for the live sound reinforcement engineer. Based on the technology of the Spirit Digital 328, the 324 Live boasts many of the standard features of its sibling. However, to reflect the requirements of live sound mixing for bands, theaters, and houses of worship, the 324 Live incorporates a number of adaptations and new features designed to assist the live sound engineer. Like the 328, the key to the 324 Live's simple operation is its E-Strip — the row of encoders and backlit switches that run through the center of the console. Pressing SEL(ect) on any input channel immediately turns the E-Strip into a conventional horizontal channel strip, giving instant access to all EQ, Aux, and Pan parameters via the row of 16 encoders that run left to right across the console. For more details, call Spirit By Soundcraft at 615-360-0471 or visit www.spirit-by-soundcraft.co.uk. Circle EQ free lit. #119. EQ

World Radio History

EBTECH SWIZZ ARMY CABLE TESTER

Nothing is less glamorous when you don't need it...or more beautiful when something breaks. Yes, I'm talking about test gear, and, unless you're a calibration junkie (like Eddie Ciletti and me), you'll probably never give test gear a second thought. Nonetheless, let's discuss a product that makes your audio life easier.

Few things in audio are more frustrating or cost more downtime than bad cables and connections. It seems bands and sound companies always have a bad cable of some sort at every gig. This is usually because many of us toss suspicious cables back into the wire box and never bother to test anything until the next gig rolls around. That's when the mic lead with the intermittent finds its way into the most important signal path of the show. (Could it be that Murphy's Law was written about live stage work?)

This quick-test device can save your gig — and maybe even your job

So what to do? First, anytime you get a suspicious cable on a gig, tape over both ends with gaff tape and put it in a special place for later testing. Second, get a tester that's easy enough to use and test those cables *before* the next gig. A cable tester is one of the best investments you'll ever make. Moreover, the first time it saves you from delaying, or even losing, a gig due to system failure, it will more than pay for itself.

What to get? Well, for hard-core wire-nuts, you can use a VOM (Volt Ohm Meter). However, they're difficult to use in the heat of battle and don't check for intermittent connections very well. For the price of a microphone, you can buy a dedicated cable tester that will allow you to check a pile of cables in less time than it takes to read one of my long-winded articles.

The best and most comprehensive cable tester I've come across is the Swizz Army 6-in-1 Cable Tester from EBTECH. I just love this thing. It's made from cold-rolled steel, which means you can kick it around stage or throw it in your gig case without fear of failure. In addition, it has a cool belt clip that allows me to keep it close for those frantic minutes when everything seems to go wrong. Finally, I think it tests just about every type of cable and situation you're likely to encounter in a studio or on stage.

Here are the basics of the Swizz Army 6-in-1 Cable Tester. An ultra-cool LED is included to show you which pins are wired to what. You get to see continuity, opens, and

BY MIKE SOKOL

Cleaner Sound. Incredible Bass. Half the Weight. It's PowerWave™ Technology.

POWERWAVE™

Inside the PLX 3402
(3400 watts and only 21 pounds)

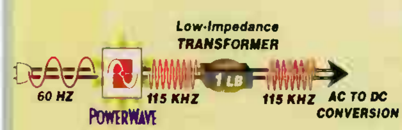
What is PowerWave?

PowerWave™ is QSC's patented power supply technology that not only makes power amplifiers more compact, but also better sounding. You don't have to settle for conventional "lead sled" designs with hum, sagging supplies, and backbreaking weight. PowerWave is a win-win solution, giving you heavyweight audio performance in less than half the size and weight of typical amps.

How does it work?

Conventional power supplies draw 60 Hz AC from the wall directly into the power transformer.

This low frequency requires a massive iron core and hundreds of windings. In fact, a conventional 3000-watt amplifier needs a transformer that weighs *at least 35 pounds*, contributing to as much as two-thirds of the amplifier's total weight.



PowerWave solves this problem by increasing the AC frequency from 60 Hz to 115,000 Hz *before* the transformer. This allows a one-pound transformer to deliver even more clean, efficient power than a 35-pound low frequency design.

Here's an analogy

Today's high-performance 4-cylinder car engines generate more horsepower than yesterday's heavy V8's. How? By doubling the RPM limit, or speed of the motor. Well, imagine increasing the RPM by a *factor of 2000*. That's what PowerWave does for power supplies.

But how does it sound?

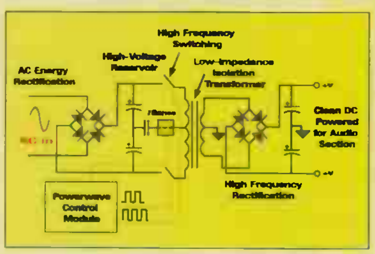
Tighter Bass A PowerWave transformer has lower impedance and greater efficiency because its copper windings are *short* and *thick*. In essence, it provides a "bigger pipe" to get electrical energy to the amp's output circuitry. The result? A stiffer power supply that delivers chest-pounding bass.

Cleaner Sound The PowerWave supply charges the rails 230,000 times per second—a vast improvement over 100–120 times per second in conventional supplies. This high recharge rate minimizes AC ripple that can degrade sonic quality.

Worried about hum?

PowerWave gets rid of it. Completely. The 115-kHz PowerWave supply eliminates 60 Hz fields that can couple into internal or external audio circuitry.

So before you buy yesterday's technology, check out QSC's revolutionary PowerWave-equipped amps at a dealer near you. Whether you're looking for the performance and affordability of the PLX Series or the brute power of the 9,000-watt PowerLight 9.0[PFC], you'll enjoy premium "lead-free" performance.



PowerWave Block Diagram

1. *High-voltage Primary Energy Reservoir*
2. *PowerWave Control Circuit*
3. *High-current IGBT Switches*
4. *Low-impedance PowerWave Transformer*
5. *Secondary Energy Reservoir*

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ROAD Test

MANUFACTURER: EBTECH, 9265 Dowdy Drive, #110, San Diego, CA 92126. Tel: 858-271-9001. Web: www.cymation.com.

APPLICATION: Tests all flavors of audio cables. This one does it all....

SUMMARY: Tests for shorts, opens, and intermittent connections on XLR, 1/4-inch, 1/8-inch, RCA, TT, and MIDI cables. Also includes a tone generator and phantom power detector.

STRENGTHS: Someone used to testing cables while under fire must have designed this, because it does everything logically and quickly.

WEAKNESSES: None.

PRICE: $149.95

EQ FREE LIT. #: 102

shorts for each pin. This is especially useful for checking strange adapter cables such as pin-2 to pin-3 hot swappers for English gear. Oh, and since it uses a tones rather than a D.C. voltage, it will pass right through a transformer splitter and tell you if it's working or not.

The short detector is sensitive enough that it will also show partially shorted cables that have a conductive path between pins 2 or 3 and the shield. This is what often causes Phantom Crackle, where a cable will work perfectly with a passive source, but as soon as you put 48 volts on the line, it starts to crackle. Moisture in the cable end or a conductive plastic shield that's smashed down by the strain relief usually cause Crackle. (Yes, that one took a while to figure out.)

The Test-Tone Generator will put out a 440 Hz or 1 kHz tone at +4 dBu, –10 dBV, or Mic level. This can help ring out a snake with a split in short order or check a mixing console to see if all the input strips are working.

A Phantom Detector looks for D.C. voltage on pins 2 and 3. Now you'll know if you have power available for that condenser mic or active DI box.

The Grounded Shield Detector looks to see if pin 1 (shield) is tied to the XLR shell. It generally should *not* be tied to the shell for a mic lead. I've found that many times when a DI box can't break a ground loop from a stage amplifier, it's not the DI's fault. It's because there's a mic lead with the shell tied to the shield and the ground currents are passing directly through the metal case of the box. Think about it....

The Intermittent Detector lets you test a cable for intermittent shorts and opens while literally swinging the whole mess around your head. (I often do this, so don't get real close to me while I'm testing cables.) A momentary switch resets an LED. You don't have to stare at the display looking for it to "blink" since the circuit will store the failure. Very cool, indeed!

An Installed Cable Tester lets you check a long cable run without having to loop a cable back to the test position. Just plug in a shorting plug on the remote end, and this tester will show you if you have continuity all the way to the far end.

Since you're wondering, the Swizz Army will check all flavors of connectors such as XLR, 1/4-inch TRS, RCA, 1/8-inch stereo, TT patchbay cables, and MIDI. Furthermore, this thing will run for months on a pair of AA batteries. There's even a cool "blinky" mode where all the LEDs chase like a little disco show. It's just one more way to scare the locals and intimidate guitar players.

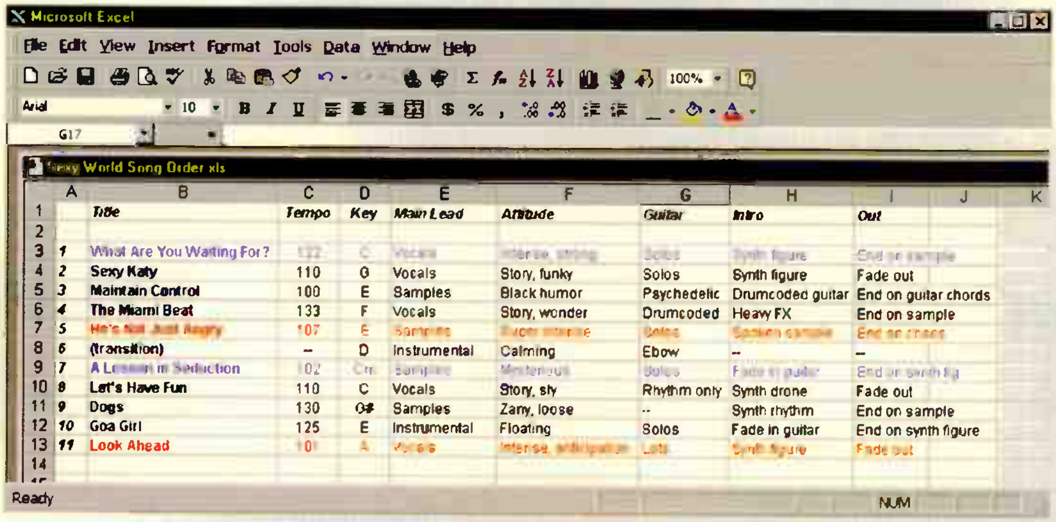
What's not to love? I can't think of anything I don't like about this tester. It gets a perfect 10 from me.

Mike Sokol is a musician/engineer/writer with some 30 years experience on both sides of the microphone. He even did a stint designing calibration gear to test components in missile guidance systems. (Yes, it's the absolute truth.)

Ad Index *For fast and easy information use the **reader response card** in this issue*

EQDrive!

EQDRIVE!



	A	B	C	D	E	F	G	H	I
1		Title	Tempo	Key	Main Lead	Attitude	Guitar	Intro	Out
2									
3	1	What Are You Waiting For?	[illegible]	C	Vocals	Intense, strong	Solos	Synth figure	End on sample
4	2	Sexy Katy	110	G	Vocals	Story, funky	Solos	Synth figure	Fade out
5	3	Maintain Control	100	E	Samples	Black humor	Psychedelic	Drumcoded guitar	End on guitar chords
6	4	The Miami Beat	133	F	Vocals	Story, wonder	Drumcoded	Heavy FX	End on sample
7	5	[illegible]	107	E	Samples	[illegible]	Solos	[illegible]	[illegible]
8	6	(transition)	--	D	Instrumental	Calming	Ebow	--	--
9	7	A [illegible] in Seduction	102	[illegible]	Samples	Mysterious	Solos	Fade in guitar	End on synth fig
10	8	Let's Have Fun	110	C	Vocals	Story, sly	Rhythm only	Synth drone	Fade out
11	9	Dogs	130	G#	Samples	Zany, loose	--	Synth rhythm	End on sample
12	10	Goa Girl	125	E	Instrumental	Floating	Solos	Fade in guitar	End on synth figure
13	11	Look Ahead	101	A	Vocals	Intense, anticipation	Lots	Synth figure	Fade out
14									

FIGURE 1: A typical song order matrix: note the use of color to differentiate the start and end of the CD's two halves.

SONG ORDER VIA SPREADSHEET

BY CRAIG ANDERTON

Determining a CDs song order is never easy — partly because, if you want to know for sure whether the order works or not, you need to listen to the entire project from start to finish. Only then do you realize there are some minor problems — like the first four songs all end in fadeouts, or you have three consecutive songs that feature the same vocalist. So you try another order and listen again...

Recently, it was time to come up with song orders for my own CD, but I tried something new: create a matrix that listed as many song parameters as possible — not just tempo and key — in order to sort out what would give the best flow and coherence. Of course, different types of music require very different parameters, but the point of this column is the general approach, not the specifics of my own situation. Hopefully you can adapt the concept to your own music.

CHOOSING PARAMETERS

The more accurately you can quantify a song's characteristics, the easier it is to come up with a meaningful matrix. Fig. 1 shows how I used a spreadsheet program to create the matrix; let's discuss the parameter descriptions.

• *Title, Tempo,* and *Key* are self-explanatory.

• *Attitude* describes, however inadequately, the song's main emotional qualities. This parameter is needed mostly to avoid bunching up too many songs with the same kind of feel, but also gives an idea of the basic emotional "road map."

• *Main Lead* describes what provides the main lead in the piece. Some of my tunes use actual vocals, some use vocal samples arranged to form a sort of lead line, while others have an instrumental lead (e.g., guitar).

• *Guitar* indicates the degree to which various tunes feature guitar. For example, I didn't want to have all the songs that featured guitar solos to run together.

• *Intro* is how the song starts. This parameter's included because I once came up with a song order where two songs in a row started with sustained guitar fading in; separating the two worked much better.

• *Out* is how the song ends. For example, you don't want all songs that fade out to occur right after another. But also, by looking over the Out and its subsequent Intro, you can get a feel for how the songs hang together.

Also note that the 1st and 7th songs are in blue and the 5th and 11th songs are in red. This is because I tend to think of a CD as having two distinct parts. This isn't just a throw-back to the days of vinyl; by giving each half its own identity, I think it's a lot easier to listen to a CD all the way through, because the experience is more like listening to two shorter CDs back-to-back. On this CD, an "intermission" separates the two halves. This instrumental transition has no real tempo and consists of long, languid lead guitar lines, so it's a good place to "reset" the rhythmic continuity and start over. The second half has a nice climb from 102, to 110, to 130, then a brief dip down to 125 before closing out at a more neutral 101.

Take this unusual approach to help determine the best playlist for your projects

TESTING THE ORDER

There are four main tools for testing order.

• Create one huge sound file with all the cuts, then open this up in a digital audio editor capable of creating a playlist. Use the playlist to try out different orders. You can usually audition the playlist transitions, often with a user-settable pre- and post-roll time.

• Most CD-burning programs make it easy to arrange songs in a particular order, then play through them. Generally, it will also be easy to listen to the transitions between songs. And, of course, once you get the order right, you can burn a CD.

It was time to come up with song orders for my CD, but I tried something new: create a matrix that listed as many song parameters as possible

• You can burn a CD with all the tunes and use a CD player that lets you program a particular song order.

• MiniDiscs make it easy to change song orders around. This method works fabulously when you're on the road and want to determine whether a song order has staying power or not. As you listen, if you want to try a change here and there, you can do so easily, and live with it for a while to see if it works better.

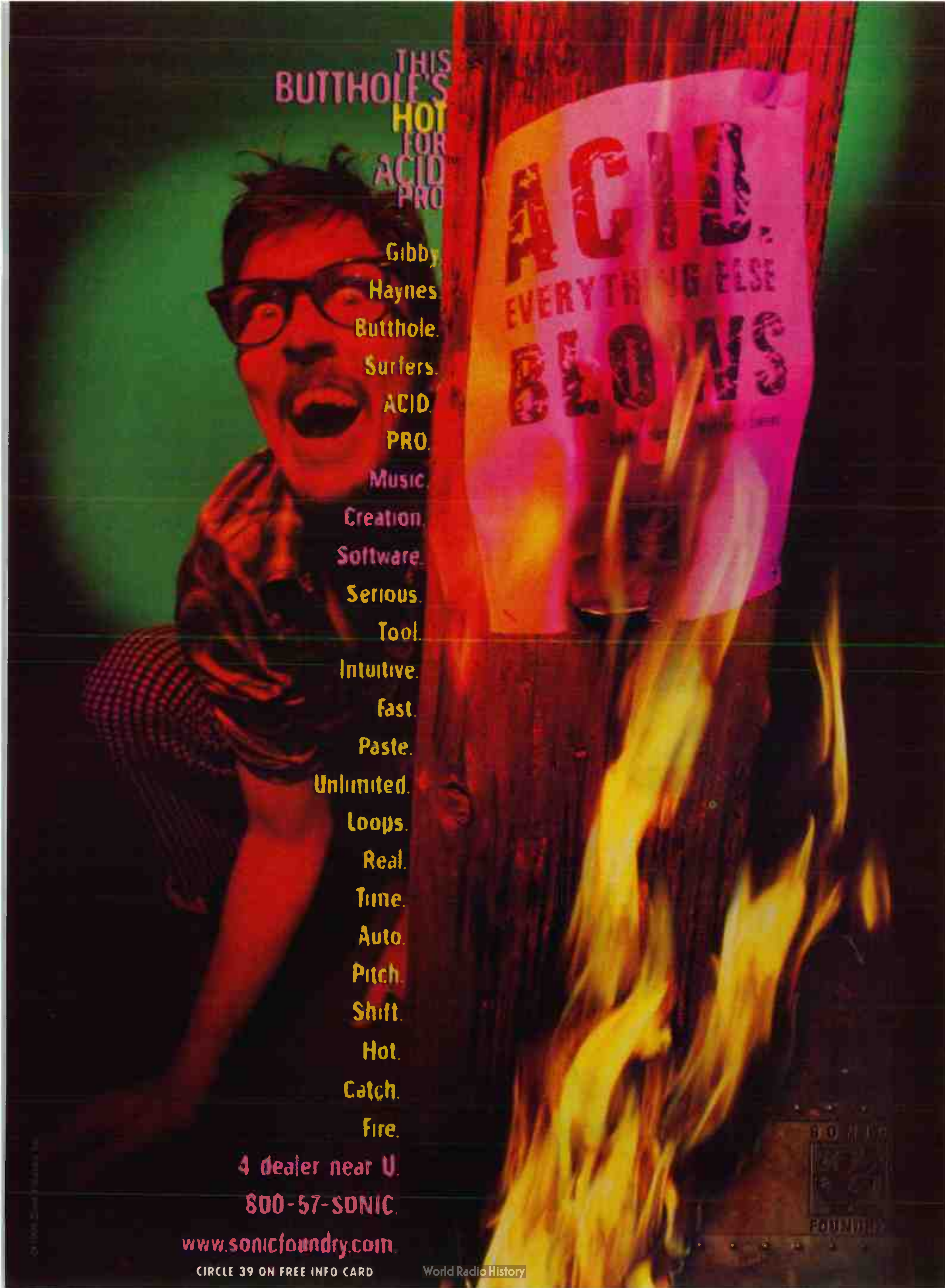
WHAT SPREADSHEET?

It doesn't really matter what spreadsheet you use (the screen shot shows Microsoft's ubiquitous Excel). In fact, you don't really have to use a spreadsheet at all; a word processor will often do the job, or, for that matter, paper and pencil.

SETTING PRIORITIES

This may seem like an overly clinical way to determine song order, but think of it as an idea-starter, not a dictator. At the very least, it will probably help indicate which pairs of songs work well together. The matrix also provides a point of departure, which is always easier than just starting with a "blank page."

The final arbiter of a good order is your ears, but check out this approach and see if it's as helpful to you as it has been to me. EQ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KOBLO SOFTWARE SYNTHESIZERS

BY CRAIG ANDERTON

Get in a fun frame of mind because Koblo's five Mac software synths (three single-voice synths, a "drum machine," and a polyphonic sample-playback synthesizer) are a lot of fun. They may look like they came from a planet where the dominant lifeform is graphic designers, but, underneath those pretty faces, there's some powerful synthesis.

Mac software synths with an analog vibe

Each synth is a document that runs under a host program (called Tokyo) that must be installed first. All synths, along with various samples and presets, are distributed on a single CD-ROM, and all can run in demo mode. Buying an individual program essentially entitles you to a key disk with two installs for hard disk authorization. Floppyless machines use a "challenge/response" system for copy protection, where you e-mail a code to the company and they e-mail a response that unlocks the program. Unfortunately, this method does not allow moving the installation to a different hard drive, as you can when authorizing from a key disk.

Operation is straightforward, and the products include both printed and PDF manuals (software purchased via the Web has only the PDF manuals). An additional manual covers basic synthesis.

You can play the Koblo synths from a MIDI keyboard via OMS, and even my 604e-based, 225 MHz Mac clone didn't exhibit significant latency. In addition, thanks to VST 2.0 compatibility, you can run them as "plug-ins" under a program like Cubase for extremely tight timing.

Perhaps the coolest feature is that *all* the various synth parameters are accessible via MIDI controllers (these are "hard-wired" and not re-assignable to other numbers). Real-time control, such as that provided by a Peavey PC-1600 or KeyFax Phat Boy MIDI fader box, is what makes these things so fun to play — sort of like have a front panel for your software synth.

For example, with the drum machine, I hooked up a keyboard so that pitch bend changed pitch for selected drums and the mod wheel altered bandpass-filter frequency on most of the drums. The variations obtained by diddling just these two controls was enough to sidetrack me for about 45 minutes playing groovacious, real-time loops into a willing DAT machine (also note that all synths except the Vibra 1000 can record what you play to hard disk, in Sound Designer II format).

Let's look at the various synths, which are monophonic except for the Gamma and Stella. You can fatten them by opening up more than one synth, such as a Vibra 6000 and Vibra 1000, or copying several versions of the same synth and opening them all. They will respond to the same MIDI control, and their audio will feed the same output.

Vibra 9000: Koblo's flagship synth (fig. 1) features two hard-syncable oscillators, three syncable multiwaveform LFOs, three ADSR envelopes, multimode filter (including a 48 dB/octave mode), eight modulators where any of 17 sources can modulate any of 32 destinations, arpeggiator, and global section with tune, pan, volume, solo, mute, and similar buttons. If you know synths, working this one is a no-brainer.

Vibra 6000: Similar to the 9000, the 6000 has only one oscillator, two envelopes (one dedicated to level, one to filter cutoff), and no matrix modulation.

Vibra 1000: A scaled-down Vibra 6000, in this synth the two envelopes are attack/decay only, and the filter is exclusively low pass. Nevertheless, this is still a cool synth — particularly because it's free for the download, and you still have that great arpeggiator.

Stella 9000: There are two major differences compared to the otherwise similar Vibra 9000: it plays back SDII-format samples rather than using standard waveforms, and can do eight voices simultaneously. Stella 9000 recognizes a sustain loop (but not multiple loops), and can reverse samples.

Gamma 9000: This is a rhythm box rather than a synth. You program patterns using a step sequencer (like vintage Roland drum machines), or by playing from a MIDI keyboard. In song mode, you can string the patterns together in whatever order you'd like. There are six tracks of individual samples and a sev-

FIGURE 1: The Vibra 9000 synth (foreground) and Gamma 9000 groove box (background).

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enth "keymap" track that loads a complete set of mapped instruments (such as percussion). These are best programmed from a MIDI keyboard.

While there's a decent selection of samples and presets, you can also load any SDII format sample. However, there's no indication how to make your own keymaps, so, presumably, you're limited to what comes on the CD-ROM.

So far so good, but there are also serious mutation capabilities: change sample start point, reverse or loop sample, link tracks 5 and 6 so that new sounds cut off old sounds (good for hihats and cymbal hits), or change the timbre via "modulation" (low-pass filter, resonant filter, and MIDI velocity control over volume, pan, pitch, offset, and low-pass filter). Other modifiers include attack/release envelope for each track, pitch (including individual pitch bend enable), and a mix section with solo, mute, pan, trigger sound, and volume. The best part is arguably the output filter, which is essentially connected in a virtual Aux bus configuration. It can add anything from grit to depth.

There were some occasional weirdnesses. For example, on the Stella 9000, it sure seemed that the envelope responses for envelopes 2 and 3 were inverted, and inverting them changed them to normal. In addition, the explanation of how the arpeggiator works is more or less useless — you have to experiment to understand what the controls do. This is one area where the manual could definitely use some help. Moreover, every now and then a synth would freeze. Although, to be fair, that usually occurred when I was pushing the computer by running other programs at the same time.

TEST Drive

MANUFACTURER: Koblo, Vesterbro Torv 1b, DK-8000 Aarhus C. Denmark. Tel: 011 (45) 8730-1430. Web: www.koblo.com.

APPLICATION: Create synthetic sounds, as well as rhythm patterns, within a virtual Macintosh environment.

SUMMARY: A family of low-cost synths delivers analog soul in a virtual body.

MINIMUM SYSTEM REQUIREMENTS: Mac OS 7.5.6, PowerPC processor, 12 MB RAM.

STRENGTHS: Cost-effective; great graphics and sound quality; every parameter responds to MIDI control; nifty arpeggiator; high fun factor.

WEAKNESSES: Works only with SDII format files (no AIFF or WAV); occasional freezes; tempo varies only in 1 BPM increments; no multisampling with Stella 9000; a few holes in the documentation.

PRICE: Gamma 9000/Stella 9000/Vibra 9000, $189 each; Vibra 6000, $90; Studio 9000 (all four applications), $460; Vibra 1000, free download.

EQ FREE LIT. #: 103

However, those are relatively minor issues. For value, cool attitude, and fun factor, Koblo's synths hit the mark. While writing the review, I came up with enough nifty drum loops and arpeggiated melodies to make me a convert. For quick sample arpeggiation, it's hard to beat. EQ

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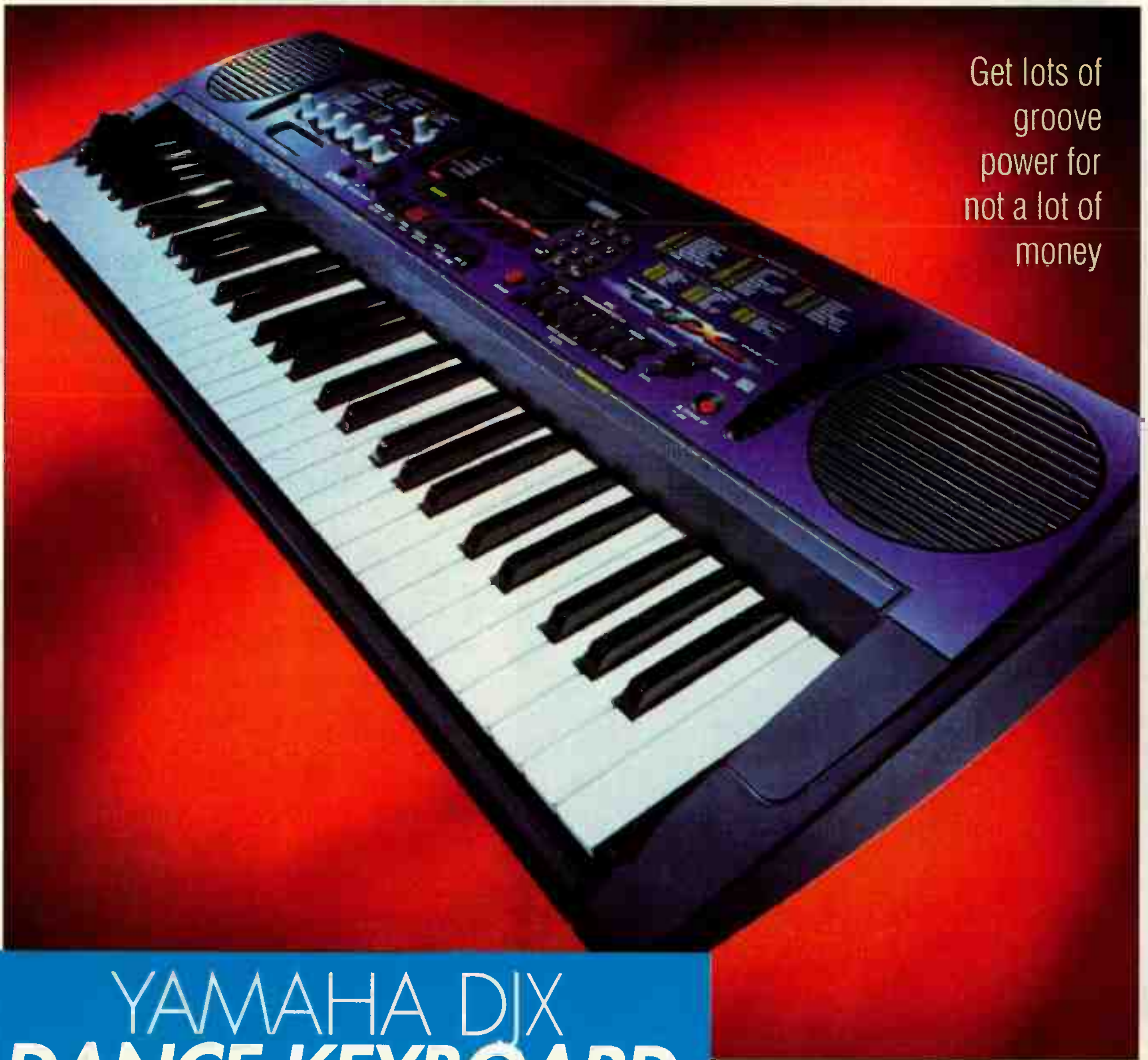

Get lots of groove power for not a lot of money

YAMAHA DJX DANCE KEYBOARD

BY DAVID MILES HUBER

I admit that I'm a dedicated "loopologist." I really like working with looped phrases that can be combined to evoke a mood. I'll also admit that I'm a lot more into the hardware side of looping than the software side. This is because dance synths tend to give me more real-time control over the various lead and percussion parts, serve up loops that are subtler and more interesting, and offer a greater sense of experimental surprise. Personal preference...that's all.

On a recent cross-country trip, I ran into a demo CD from Yamaha that touted the virtues of their new DJ keyboard synth, called the DJX. The disc got the point across that this synth

had tons of features, but when I got my hands on the keyboard, I was totally blown away by this cool tool. Then I found out that it has a street price of under 300 bucks. That did it! I had to have one and I had to share the experience with y'all.

IN THE Groove

MANUFACTURER: Yamaha Corporation of America, 6600 Orangethorpe Avenue, Buena Park, CA 90620. Tel: 714-522-9011. Web: www.yamaha.com.

APPLICATION: Performance groove synth for the stage, street, and studio.

SUMMARY: An amazingly full-featured dance-sequence/loop keyboard with built-in sampler, real-time control knobs, General MIDI, and a host of sound patch, pads, beats, and drum setup banks.

STRENGTHS: Great patches and grooves; easy-to-use; tons of control features; totally addictive for those who are into dance and groove music.

WEAKNESSES: The synth doesn't come with a power adapter (some Yamaha dealers throw in a value-added pack that includes the adapter), so you might have to spring for the one from Yamaha or find a suitable 12V, 2-amp supply; the fact that the only audio out is a 1/4-inch stereo phone jack (instead of two-phone outs and a headphone jack) is a minor inconvenience; yes, the keyboard's a little cheesy, but whatcha expect for the bucks!?

MANUAL: It's so well written and graphically laid-out that I could go to the index, find the page, and get the job done — all without so much as a single *arggghh*! Right on, Yamaha!

PRICE: $395.95 **EQ FREE LIT. #:** 104

THE FEATURES

This lightweight, blue-colored synth has a full-sized, 61-key keyboard, pitch-bend wheel, and large LCD with controls and entry keypad. The DJX also offers a ribbon controller, real-time controller knobs (which can be assigned to a wide range of control parameters), built-in DSP effects, arpeggiator, on-board sampling capabilities, 6-track on-board sequencer, and, oh yeah, an internal set of good-sounding, bass-reflex, stereo speakers.

Sonically, the DJX can be broken down into four basic categories: Voices, Patterns, Songs, and Samples. For the most part, the Voice section contains sound patches playable by a traditional synth. It's made up of 283 great-sounding patches that include synth patches, looped-grooves, percussion kits, and a full General MIDI bankset. If you think they left out the kitchen sink, you're wrong. The first patch (aptly named 000-DJX) is a DJ toolbox that includes a full drumkit (on the lower keys), stabs, hits, male and female vocal hits, sirens (on the upper white keys), and tons of looped grooves (on the upper black keys).

The Pattern section is made up of 100 loops (in various styles, including Rave, Techno, Dance, Rap, and Hip-hop) that can be manipulated in several ways. My favorite is to use the 2nd octave keys on the DJX's keyboard to turn on and off the individual perc and melody parts (or tracks) of a loop. This interactive approach to looping turns the box into a live performance tool that can't be matched by combining simple loop tracks.

The Song section is used to store songs that have been sequenced on the DJX. In effect, it serves two purposes: the first four song banks are actually factory demo sequences, while the three remaining banks let the user record sequences that contain up to six tracks.

The Sample section lets you sample up to 6 seconds of lo-fi sound into patch bank #284. Up to 12 separate samples can be recorded, edited, and mapped (from a mic or line-level input) across the keyboard in such a way that the data is saved in RAM, even after the DJX has been turned off.

CONTROL FOR DAZE

One of the many cool aspects of the DJX is its ability to let you mess around with many of the sound parameters in real-time. I mean, after all, that's what a good beat synth is all about. It's hard to know where to start, so I'll just begin with the ribbon controller. This touch-sensitive bar can be assigned to 15 different parameters (including pitchbend, of course). It's great fun pitching samples and loops down into a beefier range, or changing a hip *huuuaaa!* into one that sounds like a person hurling in the bathroom. I love it! The five, real-time control knobs can be used to alter cutoff frequency, resonance, groove (affects the beat/backbeat style of a rhythm pattern), assign (can be assigned to any of 12 control parameters), and bass-boost (can you guess what this does?).

Like certain other groove synths (such as the now-defunct Quasimidi Raven), the keyboard itself plays an important role in controlling a performance. Starting at the low end, the bottom key acts as a "beat reverse" key (which re-starts and "shakes" up the beat). The remainder of the octave lets you assign the control knobs to any part in a rhythm groove (for example, you could change the resonance on just the snare drum part). Almost an octave up from that are the all-important part on/off keys. The next octave lets you change the pattern's key signature. Finally, the remaining keys can be assigned to any voice patch (for playing melody lines, etc.). Oh yeah, the foot pedal can also be assigned to lots of parameters, including tempo tap entry, effects, hold...the works. Very impressive, at any price.

MY 2 CENTS

I could go on and on about how good the DJX sounds. How it has MIDI and built-in effects. How you can even play it on the streets or in the subway under battery power — not to mention how easy it is to play. But I won't. For pros and casual users alike, just trust me and check it out.

If you wanna get some great sounds, journey into the world of keyboard sampling for the first time, delve into some serious keyboard-based dance sequencing, or just plain have lots of fun on a way-cool keyboard for almost no bucks, I'd suggest that you run right out and pick up a DJX. This little puppy heads my list as the most cost-effective, fun, and powerful deal of the great Y2K turnover. EQ

Comp/Lim Compendium

An overview of compressor/limiters and their guts

BY EDDIE CILETTI AND DAVID HILL

Every compressor/limiter review reminds me that the subject of dynamics processing is very four-dimensional and not so well understood. Conversations with a manufacturer-designer can often be revealing, as was recently the case with Greg Gualtieri of Pendulum Audio. His physicist background literally shed some light on the subject of optical limiters, specifically how he overcame the idiosyncrasies of the devices used in his OCL-2.

I tend to be heavy-handed when testing compressor/limiters, slamming the meters just to see how "bad" things can sound when abused. (To me, the better boxes don't sound that bad under aggressive settings.) My approach did not make the Pendulum 6386 Variable-Mu [see the review in the Dec. '99 issue] "sing" until taking into consideration the VU meter's slow response time. By adjusting for less than a full dB of (displayed) gain reduction, the 6386 came alive, making me realize that older products with VU meters (Fairchild, UREI, Teletronics, or Neve) might also benefit from a kinder and gentler approach. (Translation: More is going on inside than meets the eye outside.)

For any device with a mechanical VU meter — from analog tape machines to signal processors and preamps — the meter's response time must be a consideration, especially when processing "transient rich material." While my goal is to assist in the process of user-education, my own process of evaluation was "adjusted" simply by the good fortune of finding people who are patient enough to answer questions.

I re-learned that it is important to listen — the eyes, a stubborn brain, and an ego can easily get in the way of sonic perception — yet it is equally important for products to accurately display the work that is being done. (My other soapbox is ease of parameter access, especially with digital gear.)

With all of that in mind, I welcomed an e-mail that came from Crane Song's David Hill (former designer of Summit gear) and now creator of the STC-8 and TRAKKER (dynamics manipulators). As a designer-manufacturer, David is not alone in feeling that educated consumers make better customers, and so, with his permission, his "detailed correspondence" was used as the foundation for this column.

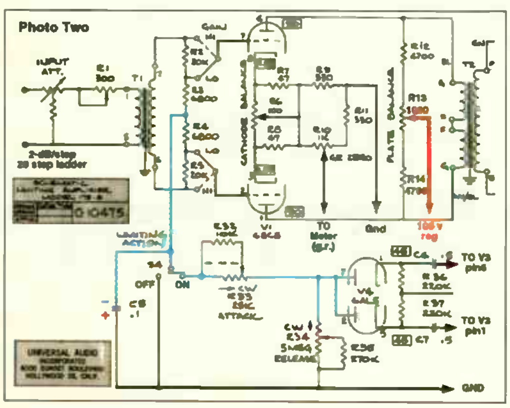
Photo Two

OOPS!

An investigation into our vintage audio heritage is helpful both for the "happy accidents" that occurred (and endured) as well as gaining a greater knowledge of what made these great boxes tick. This article is an overview, with another planned to cover "just" the supporting amplifier technology and its affect on the "sound." If you're new to compressor/limiters and just wanna get a leg up, skip to the paragraph labeled "Questions."

Surely the Teletronix LA-2A is the most famous and beloved optical compressor/limiter. A close second is its transistorized successor, the LA-3A. The heart of both units is the T4 "optical attenuator," consisting of an

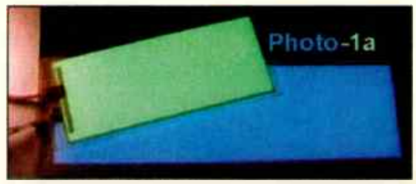
Photo-1a

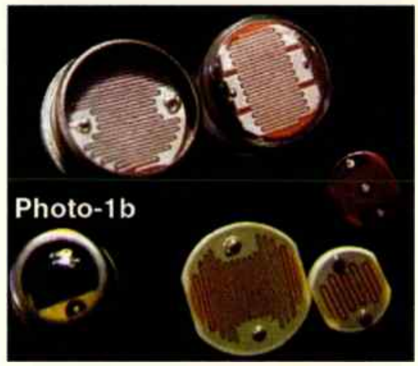
Photo-1b

Electro-Luminescent (EL) panel as light source and a light-dependent resistor (a.k.a., photo-resistor or photocell) as optical receiver (as shown in photo-1a and photo-1b, respectively).

The "other" gain control topologies include a vacuum tube/circuit called "Variable-Mu," the Field Effect Transistor (FET), and the Voltage Controlled Amplifier (VCA). (See table 1 below for a quick comparison.) All will be discussed in this article, "Optical" first, because it's easiest to understand.

TABLE 1: TOPOLOGY GAIN REDUCTION "WINDOW"

Topology	Approximate Range of Gain Reduction*
Variable-Mu	12~25 dB
Optical	25~30 dB
pwm-FET	30 dB
FET	40~50 dB
VCA	100 dB

** Note: The "approximate" range of gain reduction is typical for what is practical for each topology.*

SPEED OF LIGHT

A photocell, responding to light, decreases its resistance as the amount of light increases. The time it takes for this resistance change to occur varies with optical device design and typical production tolerances. Photocells have a built-in "Attack" time-constant, limiting their ability to quickly respond to transient signals. When used as a "limiter," optical devices are not fast enough for overload protection — especially in digital-land, where there's nowhere to go beyond 0 dB Full-Scale (0-dBFS).

In response to a transient, the photocell's recovery is nonlinear (initially fast, then slow). When constantly bombarded with light — as would be the case with heavy compression — the Release time increases, developing a "memory" that, in essence, contours the release curve to be "program dependent." Most users would consider these anomalies "the happy accident," for it is nearly impossible to make a classic optical compressor/limiter sound bad.

The response-time limitations mostly work in the optical device's favor, though, as a gain control device, the photo-resistor is much less flexible than its competitors. It is nearly impossible to speed up the response time of a photo-resistor short of testing and selecting the fastest devices. Temperature and time affect these components more than others, making it difficult to achieve and maintain accurate stereo tracking. One exception is the Pendulum Audio OCL-2, which uses a proprietary approach to speed up and "tame" the optical device (look for a review in a future issue).

OPTICAL TRANSITION

Recent products — both analog and digital — may have more signal-processing ability, but, in many cases, the interface obfuscates the user's ability to take advantage of the available power. By contrast, the LA2-A and LA3-A are simple two-knob devices that are, in many cases, just right for vocals and bass guitar. They serve as a reminder that less truly is more. Inside, the aforementioned "classics" are as basic as they appear on the outside. Only minimal circuitry is needed; one amplifier to drive the EL panel and another amplifier to "make-up" (recover) the gain lost by processing. Oh yeah, let's not forget the power supply!

Later, in the neo-IC age, the designers of the UREI LA-4 substituted an LED for the EL Panel. In order to drive the "transmitting" LED, a "detector" circuit must be added. The perfect lead-in to a more global concept...

THE BLACK BOX

All Gain-Control topologies can be modeled as a three-terminal "black box" with input, output, and control connections. In order to establish "control," it is first necessary to convert the AC signal (via rectification) into a DC control voltage (CV) that corresponds to the variations in signal amplitude. This circuit can be designed to "detect" RMS or "peak" information plus the ability to manipulate parameters such as Attack, Release, Ratio, and Slope. For more information, see the sidebar entitled "Detection and Sidechain."

VARIABLE-MU

The highly treasured Fairchild 670 is one vintage example of the Variable-Mu circuit. More recent versions include the Manley Labs Variable-Mu and the Pendulum Audio 6386. In the latter instance, the model number refers to the vacuum tube used for the task. No longer in production, the 6386 vacuum tube is a Five-Star military-grade dual-triode vacuum tube made only by General Electric (GE). All products currently manufactured are using "new old stock" (NOS) tubes, so the supply is obviously limited.

Photo 2 is a simplified schematic of the front end of the Universal Audio 175 limiting amplifier, the vacuum tube predecessor to the "solid state" 1176 and a Variable-Mu device. Changing the grid-to-cathode voltage results in a corresponding change in "mu," the gain of the circuit. In this case, the blue line indicates the path of the control voltage (CV) — from the detector and sidechain through a voltage divider — to vary the bias of each half of the tube.

For "good" performance over its usable range, adjustments are provided so that each half of the Variable-Mu tube can be balanced to minimize the amount of CV "feed-through" into the audio path (the red arrow). From a purist point of view, CV feed-through is bad, though it can also be part of the "sound" that is desired in some cases. One example would be putting some "attack" back into a kick drum (something that also happens with noise gates). Although the total amount of gain reduction is limited for Variable-Mu when compared to the alternatives, it is easier than optical to make a balanced stereo compressor.

FET CONTROL ELEMENTS

Like optical, the Field Effect Transistor (FET) acts as a voltage dependent resistor. Although it might sound like the perfect answer to a photo-resistor, it's not. As a gain control element, the FET is level sensitive. Very large signals can modulate the device resistance, causing a gain-modulated distortion that is independent of the control voltage used to manipulate the gain. (Translation: Overload the input of a FET-based device and expect some funky, most likely undesirable sound.)

The Universal Audio (later UREI) 1176 is a vintage FET compressor/limiter that is extremely popular with vocalists. The most favored of versions include an input transformer (eliminated in later production runs, both black- and silver-faced), a discrete audio path, and an output transformer. The early Allison Research Gain Brain is also an FET compressor/limiter that is both transformerless and unbalanced.

To make the FET work as a gain control element, the signal level must be kept low, requiring more than the usual amount of gain from the "make-up amplifier." One example is the Input control on a Universal Audio 1176 — a dual-pot configured as a constant-impedance attenuator. To satisfy the input transformer, the attenuator must be *pre*-transformer as well as *pre*-FET. Extra circuits can help minimize, but not eliminate, the inherent distortion of this topology. What remains is part of the sound of any FET-based device. (Discussed in a future article, more than one output amplifier design was used as the 1176 evolved, affecting yet another facet of its sound.)

For stereo operation, FETs must first be matched and the circuit must include adjustments to "find the threshold" of the device so that it operates within its linear region. The Attack and Release times can be much faster than an optical-based product.

VOLTAGE CONTROLLED AMPLIFIERS

Voltage controlled amplifiers, or VCAs, have been around for a while. They are the most common method of gain control, followed by the FET. Early VCA devices did not sound very good; two of the difficulties included matching a half-dozen or so transistors as well as keeping them all at the same temperature. Modern VCAs are much

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DETECTOR AND SIDECHAIN

The function of the detector circuit is to convert (rectify) the audio signal (AC) into a DC control voltage (CV) that corresponds with the signal's changing amplitude. The CV is then used to manipulate the gain of the control device. If this sounds like the workings of an old analog synthesizer, you are correct. Integrated around the rectifier (Peak or RMS detection) is a time-constant circuit to manipulate Attack and Release speed, some type of threshold circuit (soft-knee, hard knee plus ratio), and perhaps an analog computer circuit. The sidechain control-voltage should correspond as closely as possible with the usable range of the gain control element.

VEGAS REVIEW

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editing features, I expect additional functions will be added to the anemic Trimmer window in future versions. Currently, you must pass the audio clip through to an external application (such as Sound Forge, Cool Edit Pro, etc.) for detailed editing or processing. Unfortunately, if you are working in 24-bit depth, Sound Forge Version 4.5 (current at the time of writing) will not accept the file. Stereo tracks are handled thoroughly, but surround formats are not currently supported beyond grouping across multiple tracks to maintain sync. There are aspects of the program — such as AVI support and external DSP support — needing further development than in Version 1.0a, current at the time of writing. However, for a first outing, and considering the cutting-edge performance capability offered, this is a very well-behaved program.

Tracks can have any or all of the four Track Effects chained in a user-defined sequence. These include a 4-band parametric EQ, compressor, noise gate, and dither; each with functional control interfaces (see fig. 4) and a complete feature set. Outputs and effects busses can include both the track effects and DirectX plug-in effects. The output busses can be used as dedicated effects send/return channels, allowing DirectX effects (or native DSP effects if available through DirectX) to be added to individual or groups of channels. With up to 26 stereo busses available, each assignable to specific hardware outputs, users can add a large number of effects.

A single video file can be imported and displayed in a project, and the accompanying audio can be edited and sweetened. The video can be displayed both as a timeline ruler and in a resizable Preview window. To synchronize to video, the program offers support for MIDI timecode, but does not include any MIDI sequencing functions, or support for MIDI controllers (for mixing and transport control from external hardware). A Virtual MIDI Router application that will allow other MIDI timecode applications (such as Acid, Cakewalk, Cubase, etc.) to be synchronized while running concurrently on the same PC is included on the installation CD.

With a keen awareness of the Web-centric production that is becoming common, Sonic Foundry has included support for both Microsoft Windows Media Technologies V4 and RealNetworks G2 streaming file formats, in addition to an optional MP3 plug-in. The result is a solid platform for developing low/variable-bandwidth audio (and audio for video) suited to the most common Web formats. The ability to interleave tracks with audio clips in a variety of sampling rates and bit depths (resampled in real-time during playback) makes Vegas Pro provide an excellent platform for generating multimedia content for online distribution.

Sonic Foundry's long awaited, multichannel audio application is finally here. Vegas Pro offers an extremely direct and efficient method of working with multiple tracks of audio. Don't let the snappy on-screen performance of Vegas Pro fool you, it's not just glitz; this is a hard-working professional.

Wade McGregor is a principal consultant for Mc2 System Design Group, an acoustical consulting firm based in Vancouver, BC. For more info, visit their home page at www.mc-squared.com.

MAINTENANCE

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better because they are "monolithic," a.k.a, an "integrated circuit," or IC. Since all of the transistors are grown on the same piece of silicon, they match and stay at the same temperature.

There are two basic types of VCA, the log/anti-log amplifier and the transconductance amplifier. For the log/anti-log type to work well requires well-matched transistors with a "perfect" relationship between base-emitter voltage and emitter current. Using this over a large control range can run into problems. Increased gain reduction results in less current flow in the circuit, which means high-frequency response decreases as gain reduction increases. Mis-matched transistors in the VCA can create a strange character of distortion than may occur at one gain setting and not another.

The transconductance amplifier looks like a discrete differential amplifier — the gain is manipulated by changing the emitter current. Like the FET, this type of circuit does not tolerate large signals well — distortion sets in — and with large amounts of gain reduction it can also lose high-frequency response. Its well-matched transistors must also have very low noise.

Modern integrated-circuit technology has resulted in great improvement over the years, but the VCA can still suffer from control feed-through and have different types of distortion at different signal levels. A major advantage of using VCA devices is that they can have a very usable dynamic range, which is particularly useful in building automation systems.

PULSE WIDTH MODULATOR

Using Pulse-Width Modulation (PWM) to control an FET is what Crane Song uses in both the STC-8 and TRAKKER compressor/limiters. The technology has been around for about 20 years, having been used by several companies, including EMT. With modern technology, a PWM gain-control circuit can be very fast — from 0 to full gain reduction in 500 nS (0.0005 mS or 0.5 uS) and an audio frequency response to 60 kHz or more.

PWM works as a gate, quickly switching the FET from "off" to "more-on" by varying the pulse-width. With careful selection of components and PC board layout it is possible to have very low control feed-through, a very low noise floor, and an audio-path frequency response that does not vary with gain reduction. This type of circuit requires costly high-speed components and, as a result, there is a fair amount of power consumption (it runs hot). Distortion is very low and does not change with gain reduction.

QUESTIONS?

If you are unsure about how to set the various parameter options — Attack, Release, and Ratio, for example — it's not a bad idea to play it safe. Consider that the optimum range of some gain reduction devices is limited. Rather than over-use a compressor/limiter in one pass, try less-aggressive settings. Process the signal twice, once going to tape (for example) and once coming from tape (or your storage medium of choice).

Combining the fastest Attack with the fastest Release settings can create undesirable distortion. Most compressor/limiters feature a range of Attack times that are faster than the range of Release times making it easy to smash transients. (These same transients are also responsible for "image localization.") To keep the sound "alive" without destroying it, do the reverse of what is more often the default.

On the first pass, go for the fastest Release time and start the Attack time at its slowest setting using a ratio not higher than 2:1. It's OK to increase the Attack speed until it just starts to have a dramatic effect, then back off. This will even out the dynamics so that, on the second pass, conservative but more traditional settings — Fast Attack, Slow Release — will have a more dramatic and consistent effect.

Being "musical" is not necessarily the most desirable end-result. Sometimes what might be considered undesirable artifacts could end up being a cool effect. The best example of this is the vocal effect (of the pitch corrector) on Cher's big summer hit "Believe."

Send e-mail to Eddie at edaudio@tangible-technology.com.

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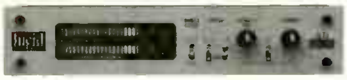

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- Complete user control over SCMS
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- Output trim for XLR balanced analog output
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The new Fostex D-15 features built in 8Mbit of RAM for instant start and scrubbing as well as a host of new features aimed at audio post production and recording studio environments. Optional expansion boards can be added to include SMPTE and RS-422 compatibility, allowing the D-15 to grow as you do.

FEATURES–
- Hold the peak reading on the digital bargraphs with a choice of 5 different settings
- Set cue levels and cue times
- Supports all frame rates including 30df
- Newly designed, 4-motor transport is faster and more efficient (120 minute tape shuttles in about 60 sec.)
- Parallel interface • Front panel trim pots in addition to the level inputs

D-15TC & D-15TCR

The D-15TC comes with the addition of optional chase and sync capability installed. It also includes timecode reading and output. The D-15TCR comes with the further addition of an optional RS-422 port installed, adding timecode and serial control (Sony protocol except vari-speed)

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ACROSS THE BOARD

continued from page 138

processing sounds slightly different, crossfades are done differently, output levels are different because of the extra built-in headroom for more tracks, and probably a hundred other things are slightly different.

Part of the problem with moving up to a bigger and better version of anything is that the manufacturer tells you everything is going to be better. They never tell you about things that may adversely effect you when you change over. Adverse effects should be required like they are with medication advertisements:

"Just released, the new 'Nick-of-Time' digital delay line. The first ever digital delay with negative settings. Yes, that's right, you can set delay times of up to *minus* 1000 milliseconds. Compensate for those pesky hardware delays transferring digital audio in and out of your hard-disk recorder. Erase delay times caused by overweight plug-ins. If you buy enough of the new Nick-of-Time delay units, you could have your project done by yesterday. (Continued use may cause vocal parts to become 3.7 cents flat during breathy vowels, guitar parts unexplainably erased, hihat shuffles to sound like Bernard Purdie, and, in some rare cases, pregnancy by Immaculate Conception)." Of course, in some cases the erased guitar part may be a blessing in disguise.

NEW YEAR

If you are reading this column, then I guess the Y2K bug didn't get *EQ* or you. It is also time for New Year's resolutions. I usually procrastinate when it comes to resolutions. A few years ago, 24 bits was my New Year's resolution. This year I am going to go with Sony Bit Stream as my resolution for the year 2000. You see, Bit Stream basically has no resolution of its own, and you can decide the final outcome later, when you down-convert to 24-bit or 16-bit or 2-bit. Cool.

I made a mistake in my December column dealing with Leap Years. It is actually "any year divisible by four is a Leap Year, unless it is divisible by 100, which is not a leap year, unless it is divisible by 400 when it is a leap year. So 1500 was not a Leap Year, 1600 was, 2000 is, and 1900 was not. That was my only mistake ever! EQ

PATCHING TRICKS

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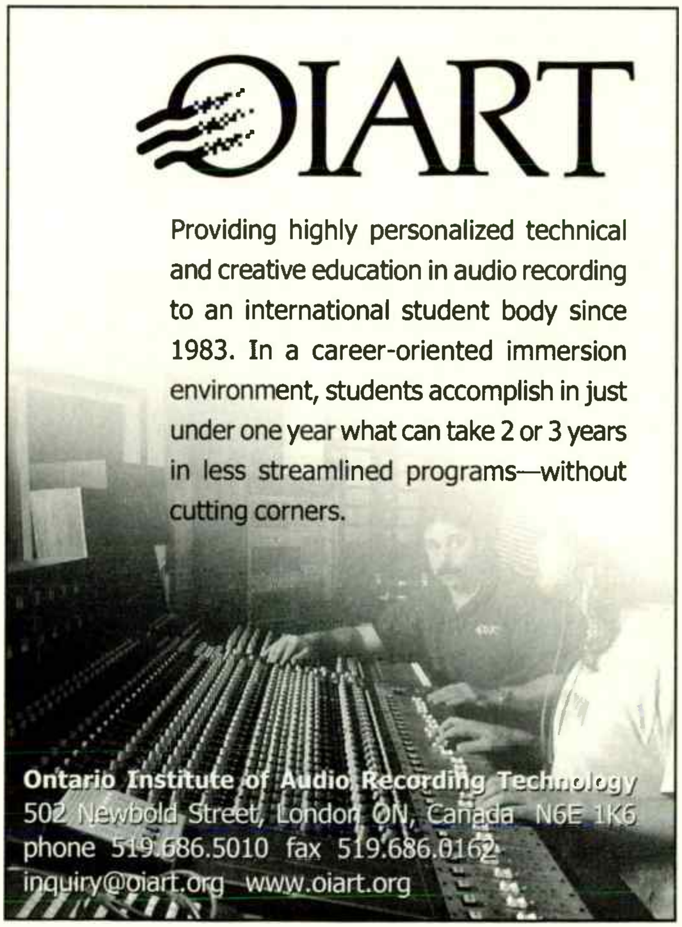
extra, unused channel strips on the console. A few patch cables allowed me to patch out from the aux send of the monitor channel to the spare channel strip and back into the monitor send. Patch in just like you would with an piece of outboard equalizer. Be aware that you'll need to un-assign this "EQ" strip from any L-R or aux sends, since its only function is equalization...just don't assign it to anything and make sure you clearly mark it so you don't get into trouble.

In my case, since the channel strip I was using had a great equalizer with variable low-frequency roll-off and dual sweepable midrange frequencies with variable Q, I was able to finish the gig without any complaints from the act. (Well, there was a catering fiasco, but that's not my gig.)

That's it for this month. So as not to let you feel left out, next month's Tech Tips will deal with old and new miking methods that will help get you ready for surround mixing. You've heard of 5.1 surround, haven't you? See you then. EQ



For about the same price as a traditional sampler, you can own the latest in computer-based sampling and audio workstation technology.

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BitHeadz Unity DS-1

Unity DS-1 is **the absolute cutting edge in sampling**. It's a virtual sampler that lives inside your computer and does MIDI and audio I/O directly with host applications like MOTU's Digital Performer. And Unity **runs circles around costly hardware samplers**. Start with 24-bit audio. Now add more RAM than a traditional sampler could dream about. Next, you get **unsurpassed sound quality** with direct digital I/O to your hard disk recording environment. Unity supports all popular sample library formats — and it can even import SDII and WAV files. That's why Unity DS-1 is showing up on stages, in studios and on the road with more world renowned artists every day. Join the ranks of musicians who have already discovered why the computer has become the sampler of choice.

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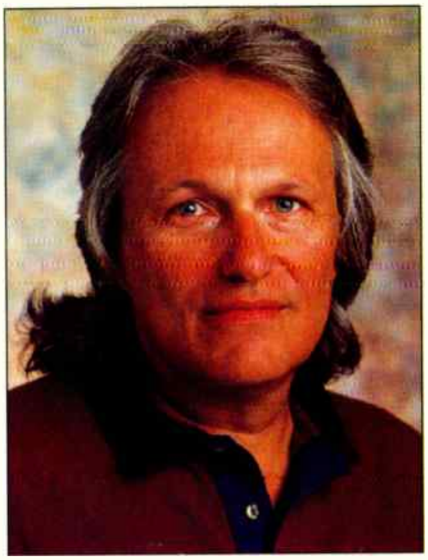
Am I *Done Yet?*



You only think that those new technologies are here to help you...

BY ROGER NICHOLS

Am I done buying high-tech gear yet? Is the latest hard-disk recorder going to make my stuff sound any better than last year's hard-disk recorders? Next year do we have to look forward to 32-bit recorders and 192 kHz sample rates? (Probably *yes*.)

I figured it out. It is the media manufacturers that have brainwashed us into thinking that higher bit depth and faster sample rates are better. Why? Well, because it sells more hard disks and more optical discs and more blank CD-R discs and, soon, DVD-R discs. If you have kept track of how much you spend on blank media, you'll find that even though media prices have been dropping, the amount you spend in any given month remains the same because you need to buy more media to store your "stuff." There are only so many hours in the day to copy your "stuff," so you would think that there would be an upper limit on the amount of media you could use. If you punched 24 or 48 tracks into Record and let them go around the clock, only taking breaks to change tape or hard disks, you could only use so much space, right? Wrong again DVD-R breath!

If the CD-R media people want to sell more blank discs, then they just up the recording speed of the CD-R recorders to 6x or 8x, and now you can burn up to eight times as many disks during a 24-hour period. This was not designed for your convenience. What about recording 48 tracks live to tape or hard disk? You can only do this in real time, so faster recorders are out of the question. But if you are convinced that 24 bits are better than 16 bits, the storage requirements increase by 50 percent. If you have been told that 96 kHz is better than 48 kHz, then the storage requirements double. We are up to three times the storage costs in just two sentences. Just wait — 192 kHz will double the requirements again. Can 32-bit recordings be far behind? If I were you, I would make three or four copies of everything you record just to get used to the future storage requirements.

Think about it. The same thing is happening in the digital camera market. Two years ago, 1-megapixel cameras were top of the line. Last year, 2.1-megapixel cameras were the hot ticket. Now 3.2-megapixel cameras are the "must have" if you want any of your friends to keep talking to you. Think about all of the computer disk storage needed for these pictures. Think of all of the CompactFlash memory cards that you will need to take all of these pictures? I turned on my camera the other day in the highest resolution mode with the RAM card removed to see how many pictures I could take without it. The camera said I had room for *one* picture at the selected resolution. *Wow!*

For you hard-disk recordists, you get the double whammy. Not only do you have to buy more hard disks for all of this high-resolution recording, but what are you going to do when the hard disks get full? You gotta buy backup drives and backup software and backup tape. If hard disks were as cheap as tape, then you would just fill 'em up and buy more. But hard disks are much more expensive than tape, and I just saw a press release from some of the hard-disk manufacturers saying that they are going to start raising the prices of hard disks. They are not making enough profit for R&D. If you want to see terabyte or petabyte drives, then you are going to have to cough up more dough. Maybe we should be talking about the "big-a-byte" these hard-disk purchases are taking out of your paycheck!

TECHNO THIS!

They may be trying to tell you that technology is making things better, but the truth is it takes longer to do everything because of technology. Don't get me wrong, I am the first guy in line for the new Whiz Bang 3000. I then spend half of my time trying to figure out how you are supposed to work it, and the other half of my time figuring out how to change the way I have been working so I can utilize the new features. When the Whiz Bang 4000 hits the streets, they tell you that everything you did with the 3000 will be unusable because the file format has changed in favor of the more optimized 4000 model. Now you really need some of the new 4000 features, but you have built your whole method of operating around the 3000. You have to increase the time it takes to do your project because you have to transfer to the 4000 to use the features you want, and, after you are finished, transfer back to the 3000. You can't use the 4000 only because all of the other hardware and software that you use with the 3000 are no longer supported.

What do you do with all of your old projects that were done on the 3000? You have two choices. Choice one is to keep those projects in 3000 format, but you also need to keep all of the 3000 hardware and the external devices that you bought to use with it. You also have to keep this old hardware in working condition. Choice two is to move the project over to the new 4000 version. If you ever have to work on this project again, it will probably not sound exactly the same on the 4000 system. Internal timings and delays are different,

continued on page 135

This Mic Is Anything But Flat...

The Neumann M 147 Tube

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

Enter the M 147 Tube.

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

...That's Why The Pros Love It.

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Neumann – The Choice of Those Who Can Hear The Difference

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

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

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

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

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

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

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

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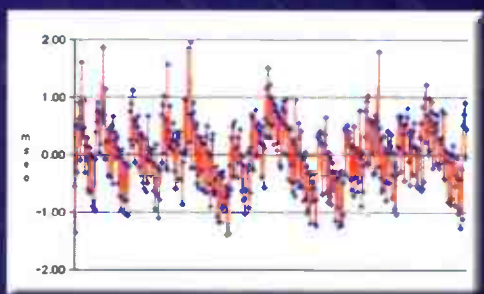
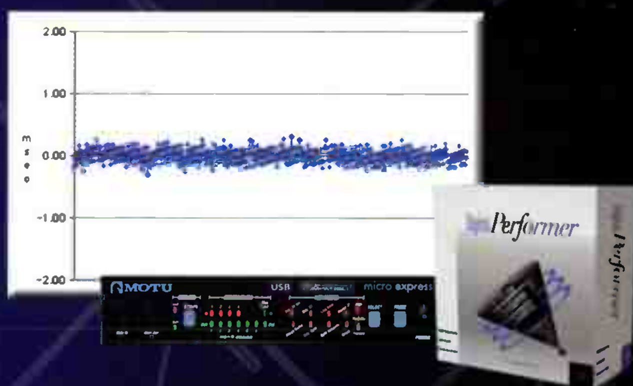
MOTU

Digital Performer now offers the most accurate MIDI timing in the business.

Theirs.



Ours.



Our closest competition in hardware-based MIDI timing has inherent jitter of 1-2 milliseconds, with spikes that are even higher. Digital Performer's MTS timing is as tight as one third of a millisecond — five times better.

MTS MIDI TIME STAMPING

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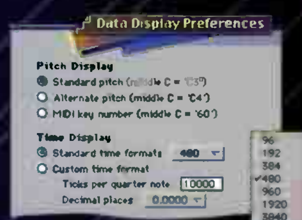

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