Special Report **PROJECT POST**

EQ

THE PROJECT RECORDING & SOUND MAGAZINE

MARCH • 1998

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this ain't no home studio...

project pros

Recording Tips From Top Project Producers:

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The Panasonic DA7 Mixer.
Taking Digital Further.

1-Step Functionality
24 Bit A/D and D/A
Moving Faders
Surround Sound
Automation & Memory

Inspiration can strike you in the strangest places. But, when you've been dreaming of the ideal digital a/v mixer for as long as we have, you jot it down on the nearest piece of paper. Well, the end results of that inspiration have come to pass...Panasonic introduces the RAMSA WR-DA7 digital mixer, and sets an entirely new standard in quality, flexibility, affordability, ease-of-use and value.

TAKE COMMAND... NOW

32 inputs and 6 auxiliary send/returns (for a total of 38 inputs), 8 bus, 24 bit converters, moving faders, instantaneous recall of all settings, surround sound... you'd think nothing this fully featured could be this easy to use or this affordable... but it is!

GREAT SOUND

32-bit internal processing combined with 24 bit A to D and D to A converters, yield an incredible 110 dB dynamic range, putting the DA7's sonic quality in a class by itself.

MAXIMUM FLEXIBILITY

Packed into the DA7 are sixteen analog mic/line inputs and individual access to channels 17-32 through channel flip buttons located above each fader.Twenty faders do triple-duty as level controls for channels 1-16, 17-32, or Aux sends 1-6, Aux returns 1-6, and buses 1-8. We've even added an additional fourth layer, which includes MIDI faders.

EASY-TO-USE

The DA7 features automated, logical layout and intelligent design. Access a channel by pushing its select button, and all parameters for the channel: EQ, bus and aux assignments, and dynamic/delay settings come up on the large backlit LCD screen. To access individual parameters, just touch the appropriate knob in the console's master section. This calls up the sub-menu on the LCD screen and zooms in on the appropriate function. No digging through menus or getting lost in functions; just select... and you're there.

THE POWER TO CONTROL

The EQ section offers four true parametric bands active on every channel, with the top and bottom bands selectable from peaking or shelving, or they can be high and low pass filters, respectively. The frequency bands are overlapping, with the top two bands ranging from 50Hz to 20kHz, and the bottom two bands ranging from 20Hz to 20kHz. Boost or cut for these bands are adjustable in 1/2 dB steps to + or - 15 dB. The bandwidth is adjustable from 0.1 octave to 10 octaves. The DA7 is so full featured, even the Aux returns feature a 2 band parametric equalizer. The dynamics section offers you a choice of a Gate/Compressor/Limiter or an Expander on every channel with variable attack and release times and levels for threshold and ratio. A Delay of up to 300ms is available on every channel. In addition, 50 EQ and 50 Dynamics memories can store your favorite settings for instant recall.


Digital Mixer
RAMSA
SURROUND SOUND AT YOUR FINGERTIPS
The DA7 is equipped to mix 5.1 channel surround through its buses, so you don't have to tie up auxes, controllable by three modes for any channel or combination of channels. All modes provide full dynamic control of panning, and can be copied, stored, and transferred to any other channel. An optional MIDI joystick gives yet a fourth method.
MORE FEATURES THAN WE HAVE ROOM TO TELL YOU ABOUT
The DA7 features four up/down/left/right cursor keys that are switchable to output MIDI Machine Control commands to MDMs, sequencers, or workstations. Data entry is done through the parameter dial or alphanumeric keypad. There's an undo/redo button, solo-mode set, and built-in talkback mic. Honestly, the DA7 is so feature rich, (but still easy to use) that we don't have room to describe it all here. You'll have to test drive it in person!
TAKE ON THE WORLD
The rear panel has 16 analog mic/line inputs (8 XLR with individual software-switched phantom power, and 8 with TRS); 16 channel inserts (preA/D); and 6 auxiliary send/return jacks (1,2 use S/PDIF; the rest, +4dB 1/4inch connectors). Analog outputs include +4dB balanced master outs with XLRs; +4dB balanced record outs on TRS 1/4inch jacks and two +4dB monitor outs on TRS balanced jacks. Digital I/O, via XLR connectors is switchable between AES/EBU and S/PDIF. The rear panel also offers MIDI In and Out, word clock I/Os, plus both a 9-pin RS-422/485 serial port and PC port for Mac/Windows with software support for both, a 1/4 inch footswitch jack for controlling talkback on/off or automatic punch in/out for automation, and a D-15 subconnector for the optional meter bridge.
TAKE IT EVEN FURTHER
3 expansion-card slots allow connection of recorders with ADAT Lightpipe, TASCAM TDIF, and AES/EBU (switchable to S/PDIF) interfaces, with any of the audio cards fitting into any slot. A fourth card provides 8 more analog inputs/outputs via a D-25 subconnector. The third expansion-card slot can be used 3 ways:
• Connect 2 DA7's together with true bi-directionality
• Replace analog inputs 9-16 with digital inputs
• Digital inserts across the 8 buses, six Auxes, and L/R stereo out. An option card provides SMPTE and Video Sync input.
WHEW!
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PROJECT RECORDING
& SOUND TECHNIQUES
VOLUME 9, ISSUE 3
MARCH 1998

FEATURES

EQ LIVE

TECHNIQUES/WORKSHOPS

COLUMNS/DEPARTMENTS

ON THE COVER:
Up close (real close) with David Tickle.
Photo by Edward Colver.

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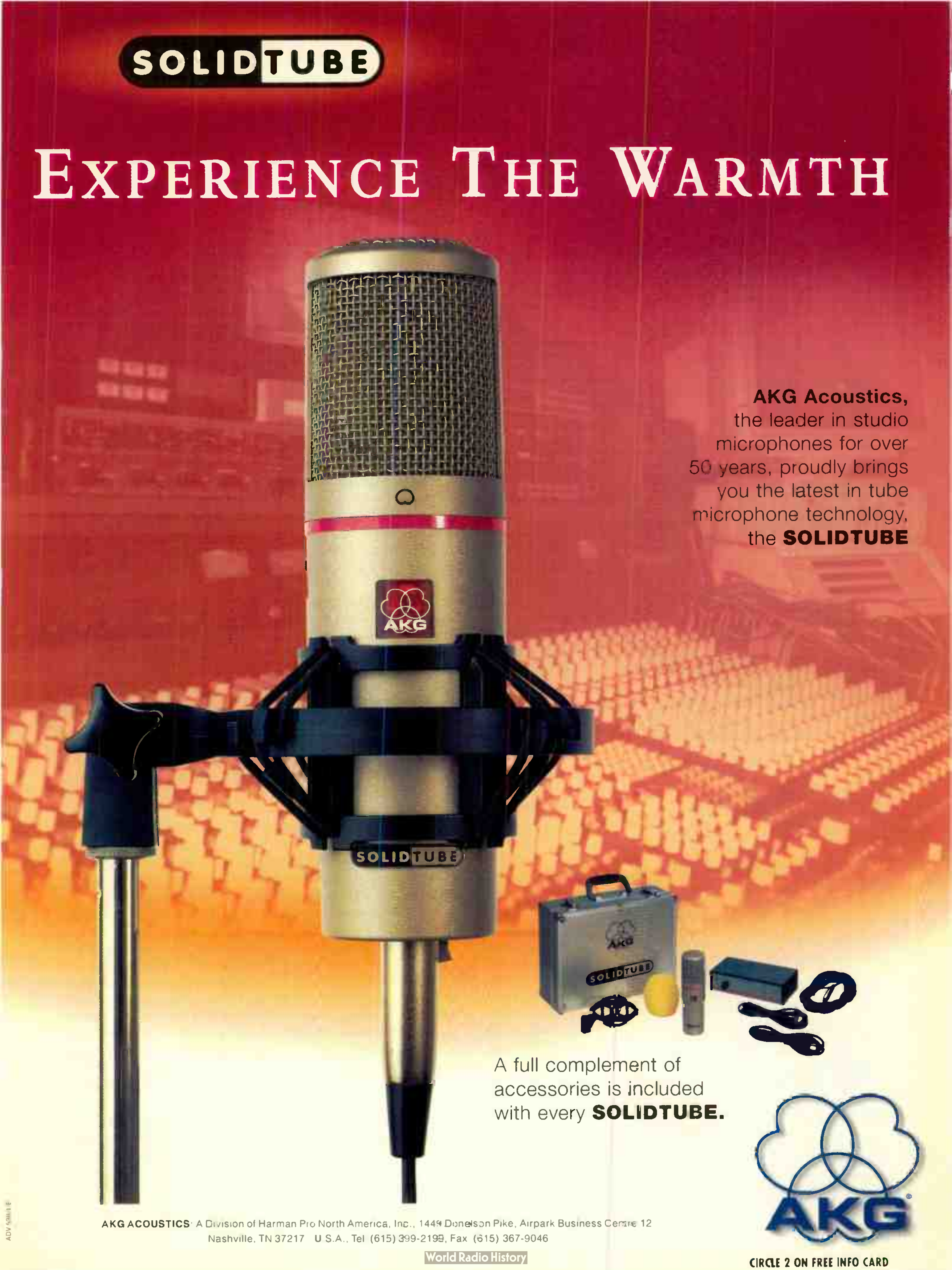
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Miller Freeman

You Can't *Go Home Again*

It's a rare opportunity when I get to vent in the pages of *EQ* (usually I vent on some poor unsuspecting drum kit), so I'm just gonna go for it: I'm sick and tired of wannabe musicians who don't know their input from their output (electronically, biologically, or otherwise), trying to pass themselves off as engineers — and further, trying to pass off their studios as "project" studios. There's a distinct difference between a "project" studio and a "home" studio, and if you have any doubts, let me set it straight for you.

A home studio is a place where nonprofessionals go to make some noise and drink a few beers during recreational hours, usually that noise being some kind of music (I'm not gonna go there...). There are plenty of well-paid, nonmusic professionals with the cash to buy nice gear, which they proceed to toss into a spare room. But these (as good as their intentions may be) are not concerned with (a) whether or not they actually know how to really use all this gear, (b) whether they have any kind of concept about how to connect this stuff so that it works without hums, buzzes, and picking up the local shopping channel, and (c) whether they can make money using it. And you know what else? They insult working professionals like you and me everyday because we do practice a craft at our project studios. Sure, people can record at a home studio — in between the dog barking, the baby crying, police sirens, loud TVs, and sorry we don't do drums because they make too much noise and we only have a little closet to record vocals.

A project studio, on the other hand, is a place where professional engineers like us do serious work such as records (however small the budget), radio commercials, TV and film scoring, editing, voice overs, and even instructional tapes for your son's bar mitzvah (yeah, that's right). We spend far too much of our working income on new gear (we're in therapy for it) because (a) it's a passion, (b) we need to remain competitive, and (c) *it's a tool.*

WE INTERRUPT THIS EDITORIAL...

to congratulate Roger Nichols for winning a Grammy as co-producer of "Best Musical Album for Children" for the John Denver record *All Aboard* (Sony Wonder)

I can always justify purchase of a microphone because to me it's a tool I'll use for the rest of my life. I don't want *just* an AKG C414, I want a *pair,* and I also want a Neumann U87, a Lawson L47MP, and a bunch more. I don't want just a flathead screwdriver. I want Philips, Torx, and tamper-proof. We buy gear because we need it musically, spiritually, and economically (at least that's what we tell our spouses and accountants). Engineers actually learn how to edit the parameters on our reverb units regardless of how poorly written the manual may be. This is not Triple-A. This is what got commercial studio owners pissed off in the first place.

Project studio owners/engineers also spend far too much time trying modifications on old gear to make it sound better (even if it's just a dB or two), we know what it means to align an analog tape machine, and take said alignment way too seriously even though our analog 8-track sits right next to a bunch of new digital 8-track machines. Even though we sneer at guitar players who bring in noisy stomp boxes, we'll still let them record brutally loud tracks through their Marshall's into the wee hours of the morning *because we can* — we know what an STC rating is and we know how to get it high enough so that our neighbors won't hear a thing. For the project studio owner/engineer, this is not a hobby. This is a way of life.

EQ is the only magazine that brings you the project studio professional's way of life, *written* by engineers such as Roger Nichols, Jack Renner, Elliot Scheiner, and Roger Charlesworth. You want the dope? This is where you'll get it. Make no mistake — *EQ* is *the* Project Recording and Sound Magazine.

—Steve "Woody" La Cerra

GRATZI

Thanks to Eddie Ciletti and *EQ* magazine for help with my Sony TCD-D3 portable DAT. You busted one of my nightmares. [See EQ & A column, February '98 issue.] I put my hands on that baby and guess what? By following your detailed instructions and the service manual I definitely fixed that ugly problem! My DAT is now working better than ever. In fact, one of the rollers was really loose, and it had been screwed to its end. I tightened it and then I made the regulation using the 'scope. Then it was simply wonderful. I owe you a dinner here in Italy, so sign up for some good fish!

Ciao, ciao da Lo Zio (means: Ciao, ciao from The Uncle).

Zio Stefano
Italy
via Internet

UP WITH PEOPLE

In reply to Mr. Deutsch's "People Who Need People" letter in the January '98 issue:

Lighten up Ken, maybe Mr. Bonzai's "Barbara Walters" questions to Wendy & Lisa (and others) is his way of getting them to loosen up and open up with candid responses to an otherwise standard operating procedure.

Bonzai's manner has always been a bit tongue-in-cheek, but nevertheless appeals to a lot of us readers. If we take ourselves *too* seriously, we might let the muse slip through our fingers.

If you want strictly facts and figures might I suggest you peruse the local library for journals and stodgy scientific paperwork. I like a little personality once in a while, and always look forward to Mr. Bonzai's "fluff."

Bob Ketchum
via Internet

DIGITAL STAY HOME

I've just finished reading the article in your Jan. '98 issue "Digital Todd With A Twist," regarding the use of the new digital consoles on the road. According to Todd Rundgren's sound tech (Larry Toomey), this is the way to go because you don't have to search through a "whole field of knobs" or you're constantly having to double check things when running a large board, etc.

I don't know...maybe I am old fashioned. I do love the sound of digital, but to me there is something nice about being able to scan over a board while doing a live mix. I can take in the whole console, without having to punch this up or that. To be required to do so would drive me *nuts*. I'm a tweaker — always working on the mix from the start of a show all the way to the last song. Add to this the "glitches" that can occur on many different digital products (haven't seen a perfect one, yet), and I'm not sold.

I work for a local sound company who uses Allen & Heath consoles and I love them. I'd take one of these over a digital console for live work any day. Yeah, there are lots of knobs — and I love 'em!

A final thought: how rugged has the digital gear proven to be on the road? I'd be scared to death to have something like that bouncing around in a road case. A lot of the digital stuff is just not up to life on the road, at least not yet.

Maybe it is the way of the future, but I'll bet it'll be a while.

Mike Pritchard
"Weekend Warrior Sound Tech"
via Internet

WONDER WALL

In response to the "Acoustically Speaking" article of the January issue and specifically to common problems of listening spaces interview with Jay Miller, I detected what I hope was a misprint.

I quote: "Another common problem is rigid wall surfaces. If the walls are not rigid, the 'flex' in the walls will cause bass reflections back into the room."

My experience with physics and acoustical design dictates quite the opposite. A non-rigid wall acts as a diaphragmatic absorber, effectively transforming acoustical energy into motion and then heat. The frequencies absorbed are related to the mass of the panel and the airspace behind the panel. The "Q" of this kind of absorber is dictated by the damping of the airspace. Walls that behave rigidly at low frequencies actually reflect acoustical energy back into the room.

T. Alan Kraus
Sound Advice, Inc.
via Internet

HAVING A BALL

Being the owner of a TASCAM TSR-8 analog recording machine and a project studio of my own, I was curious to look further into the techniques mentioned in the November '97 issue of *EQ* by Terry Hughes of Rubber Ball Productions about how to get big guitar sounds using two amplifiers for one guitar player and the steps to bus it to stereo using a number of microphones and outboard gear.

I was inspired to buy the CD it was used on to hear it. I went to a local record store and purchased a copy of *Riff Raff in the Basement* by This Boys Life and was blown away by the huge guitar sounds. It's hard to believe it's just one guitar in stereo.

Since then I have been in contact with Terry to help re-create that technique in my studio, and although we have different equipment, it is a giant improvement compared to the way I used to record guitars. I even used the Shure SM57 in a wrapping-paper tube!

John DeVoe
Van Nuys, CA

GETTING THE TWANG OF IT

RE: Feb. 98 issue "Searching For Mr. Goodguitar"

Craig Anderton's observation that Strats and Teles tend to have a thinner sound is true. However, I have found that with the new Fender lace sensor pickups, a lot of the "twang" has been nicely curbed without losing the distinctive "Strat" sound. Also, since the country pickers(including myself) really like the heavy "squashed" sound, I need to be a little creative with EQ to make up for the extra compression. I've tried several different effects settings, including the ones mentioned in Craig's article, and have found that the "less is more" mindset works best. Just a little chorus and delay. After all, I want to capture the real guitar. Otherwise, I'll get a sampler.

Rick Sylvester
via Internet

CORRECTION

In the January '98 issue's Room with a VU, the men in the picture were wrongfully identified. Cory Lerios is standing on the right and John D'Andrea is seated on the left.

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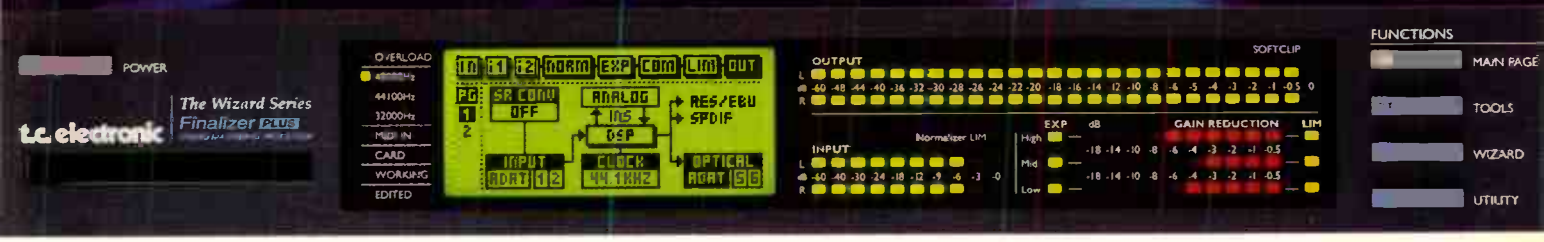

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SCHEMATIC DEPRESSIVE

Q *I built the phantom power circuit that you presented in the July '97 issue of EQ, but it's not working even after double and triple checking it. It's "close," as I'm getting 25 V (about half of 49.5) out. I have 29 VAC from the transformer, but only 38 VDC unregulated, with +19 and –19 across each capacitor.*

I don't think I was fully able to translate your pictorial schematic into a proper circuit, especially where the zener diodes and transistor are concerned. Anyway, I've included a schematic of my understanding of the circuit. Perhaps you could do some comparative analysis and point me in the right direction?

Dave
via Internet

A I modified your drawing and mated it with mine. (See fig. 1.)

The way you describe the problem, I would look at the circuitry after the rectifier. A voltage doubler circuit is not known for its power, so anything wrong will overload the charge built up in the caps. Let's start by disconnecting the transistor and the zener diodes. Leave only the voltage doubler circuit connected to the power transformer. The voltage should now be up to about 75 volts from diode to diode, that is, across both caps.

The next step is the voltage reference created by the two zeners, which should have 24 or 25 volts across each, totaling 48–50 volts. Also, the resistor feeding the zeners will be warm, but not burning hot. All this before the transistor is connected, but do have the cap (C4) across the diodes to ground. If not, one of the zeners may either be shorted or reversed. If the latter, there will only be six-tenths of a volt across it.

Then, once the transistor is connected, it should feed the 470-ohm resistor as well as the resistor feeding the LED. Make sure the transistor you use is properly connected, noting that the pinout may be different from the unconventional type I used. Most modern transistors are, from left to right, B-C-E and not C-B-E as shown. Since I built this supply from parts laying around my shop, even the LED I chose required a bit more current than more "modern" versions. (Readers will find Dave's updated schematic along with the complete version of this project at www.tangible-technology.com.)

Eddie Ciletti
Contributing Editor
EQ magazine

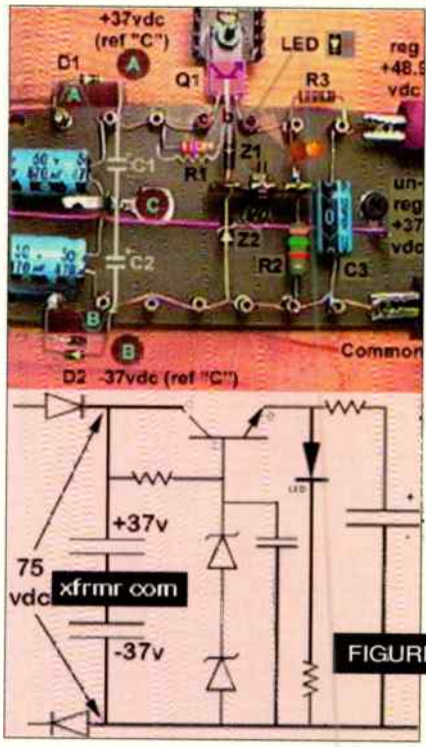

FIGURE 1

WATT'S UP

Q *The owner's manual of my QSC MX1500A power amplifier states that the amp "...delivers 500 FTC watts into 4 ohms per channel, or 600 EIA watts into 4 ohms per channel." It doesn't, however, tell me how the two "volume controls" on the front of the amp correspond to this power. (Obviously, full clockwise is full power.) The manual also states that the "attenuator controls are labeled in dB of attenuation."*

The numbers counterclockwise are: "0-2-4-6-8-10-12-14-18-24-infinity." I think I understand that 3 dB is one half of the power. So does that mean that if I turn the attenuator back from 0 (full) to 3, that the amp is putting out a maximum of only 250 FTC RMS, or 300 EIA RMS? Some help in understanding what (if any) correlation exists between my attenuators and amp power would be helpful. My goal is to know about where my maximum setting would be so that I could use a pair of 150-watt speakers connected in parallel on one channel of this amplifier. I'd rather use the attenuators than use a compressor.

Rod Businga
via Internet

A You've come upon one of the great misconceptions in power amps. The attenuators [gain controls] on the front of an amplifier have nothing to do with output power. That's right! They in no way limit the available power output of the amp. They are strictly gain controls. What they do allow you to do is set the gain of the amplifier so that 0 VU

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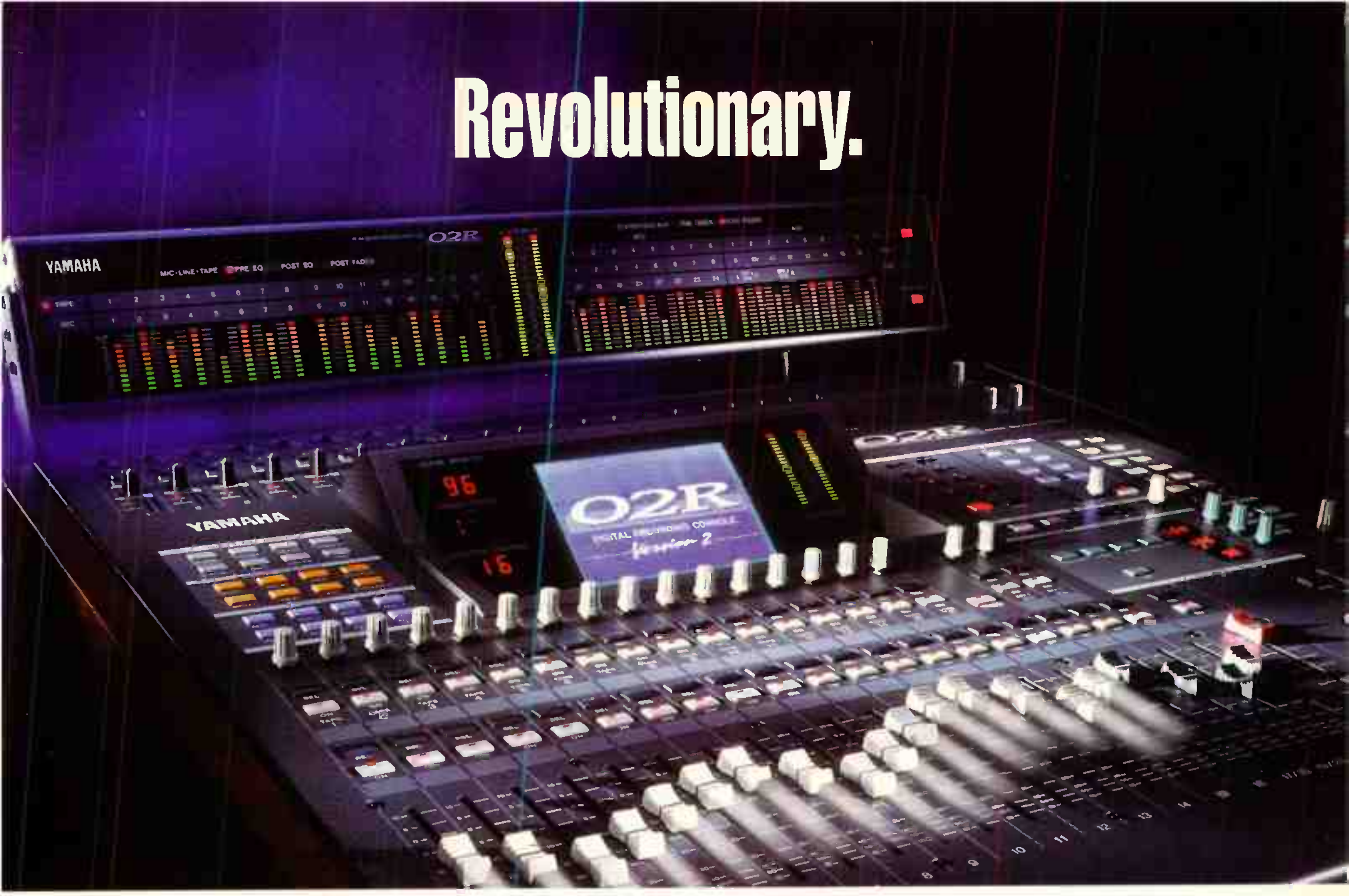

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SMART MIXING

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out of your board corresponds to some lesser output level of the amp. This is certainly useful as an extra safety measure, if you want to use smaller speakers on a big amplifier. But it doesn't act as a final limiter. Therefore, if your system goes into feedback, the amplifier will be putting out full power — with possible dire consequences if the speakers can't stand the heat. Most professional speakers can withstand a lot of abuse for short periods, so I have a feeling you'll be fine with your choice of amplifiers and speakers. Still, the best way to control things is a final compressor/limiter going to the amplifier inputs. But, to be honest, unless there's a guest engineer running my board, I rarely patch in a limiter since I trust my mixing instincts. If you're confident with your own ears, just listen. If it's sounding bad, then turn down the mix 'cause bad things may soon be happening to your speakers.

Mike Sokol
JMS Productions
Hagerstown, MD

CHOMPING AT THE BITS

Q *My 02R console is equipped with version 2 software so I can do 24-bit recordings. And as last month's Yamaha 02R v2 upgrade article in EQ said, "the problem is coming up with [finding] 24-bit data." OK, there's the Rane PaqRat. Do the 02R and the PaqRat integrate seamlessly? It appears that each unit uses a different scheme for recording the digital data onto an ADAT. This doesn't seem to present a problem for getting the 24-bit audio into the 02R — only if it is sent back through the PaqRat and then out of the PaqRat (using AES/EBU digital outs) into, of course, the 02R's AES/EBU card slot. But what if you don't want to use the ADAT optical outs to bring the 24-bit data (recorded via the PaqRat) into the 02R via its ADAT optical card slot? (Naturally, there would be no problem if the 02R were equipped with 24-bit converters, which it is not.)*

Am I missing something, or is there a product available (or in the works) that will make more sense? Perhaps there's a 24-bit A/D converter (2 ch.) that uses ADAT optical in addition to AES/EBU digital outs. A converter upgrade of the 02R's 2-bit inputs would seem appropriate. Do you have any advice?

Anonymous
via Internet

A Your question can be best addressed by breaking it down into three parts:

1. Looking for 24-bit data. As you mention, the 02R version 2 software includes a 24-bit recording mode, which allows recording of 24-bit data onto a 16-bit storage device, such as an Alesis ADAT, or TASCAM DA-88 or 38. While you point out that "the problem is finding 24-bit data," actually, regardless of the dynamic range of the original signal, in the mixing process, new signal is created by using equalization and effects and by mixing multiple signals. The new signal is an entity unto itself, and exists inside the 02R at a higher dynamic range than those signals input into the 02R. Using the 24-bit mode, these signals can be stored at a higher word-length. There are good reasons to do this:

• Even though the current distribution medium, CD, can offer 16 bits of dynamic range, many industry professionals recommend producing material at higher dynamic range, if possible, to maximize the quality of the end product.

• Using the 02R 24-bit mode, you can actually print your finished stereo mixes onto an ADAT as 24-bit data. Storage at a higher word length means the product we create today will be ready for systems capable of higher dynamic range in the future.

2. Schemes for recording 24 bits on a 16-bit recorder. There are several outboard devices that allow recording of 24-bit data on 16-bit storage devices. They accomplish this by splitting the signal to be recorded into two signals, which are recorded onto two tracks of a 16-bit capable recorder. When the two signals are replayed from the 16-bit recorder through this "splitter," they are reassembled to provide the original 24-bit signal. While these products can be used in conjunction with the 02R, they are not necessary, since the 02R now includes the software to split and reassemble signals internally and conveniently for mixdown. Schemes for splitting and outputting the signal onto an ADAT, for instance, differ between these devices. Since the PaqRat, for example, is a 2-channel device with an ADAT interface, it splits the signal onto adjacent tracks of the ADAT. The 02R, how-

continued on page 160

Send your questions to:
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Port Washington, NY 11050
Fax: 516-767-1745
E-mail: EQMagazine@aol.com

EQ PRODUCT VIEWS

V8 POWER

The new self-powered, biamped V8 Shielded Monitor from KRK was designed in response to the growing demand for self-powered monitors at a reasonable price. The V8 features video shielding with an 8-inch Kevlar woofer and a 1-inch silk tweeter. Frequency response is measured at 49 Hz–22 kHz (±2 dB), with a crossover point at 1.6 kHz. The V8's provide 130 watts of maximum power handling and a maximum SPL of 108 dB. V8's carry suggested retail price of $1249 per pair. For more information, contact Group One Ltd., 80 Sea Lane, Farmingdale, NY 11735. Tel: 516-249-1399. Circle EQ free lit. #107.

BUDDY IN A BOX

Offering over 100 distinctive drum kits and 50 bass patterns, Zoom's RhythmTrak•••234 drum machine also offers users real-time recording, a number of advanced performance features, General MIDI options, and more. The RhythmTrak starts with drum sounds that were recorded using vintage pro equipment, famous drum rooms, a number of different drums, and a number of different mic positions to capture room ambience. The machine-based sounds (rap, hip/hop, techno) were sampled, tweaked for more punch, and optimized in the digital domain. Most of the snares are true stereo, and the sample rates of all the main drums are full bandwidth (44 kHz). The bass pads are designed to sustain for as long as a user holds them in Bass mode, and users can tune up any scale — up and down a total of three octaves. The RhythmTrak also boasts a "SoundJammer" slider and a Repeat button for live performance applications. A special Groove Play feature lets users play back drum/bass patterns by hitting the pads and holding them. For more information, contact Samson Technologies Corp., P.O. Box 9031, Syosset, NY 11791-9031. Web: www.samsontech.com. Circle EQ free lit. #108.

THE BIG SQUEEZE

The 565E dual compressor/limiter/expander from Symetrix integrates RMS compression, downward expansion, and peak limiting in a dual channel, 1U rack-mountable unit. All three functions are designed to operate in-line, allowing simultaneous operation for complete dynamic control. The compressor and expander sections of the 565E employs Dynamics Squared circuitry that reduces distortion associated with conventional voltage controlled amplifier topology. The unit's two channels operate in either stereo or dual mono modes. Additional features include individual LED meters for each processing section, balanced and unbalanced inputs and outputs, XLR and 1/4-inch connectors, and an internal power supply. Suggested retail price is $549. For more details, contact Symetrix. 14926 35th Avenue West, Lynnwood, WA 98037. Tel: 425-787-3222. Web: www.symetrixaudio.com. Circle EQ free lit. #109.

V.A.S.T. IMPROVEMENTS

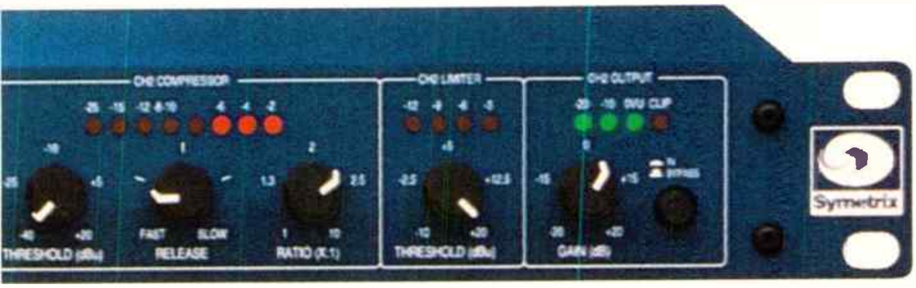
A 61-key 24-voice professional keyboard with V.A.S.T. synthesis modeling technology, a built-in 32-track sequencer, DSP software, and a 3.5-inch high density disk drive, the Kurzweil K2000RVP is a new rack module version of the K2000VP. The K2000VP also offers users more RAM, as well as the new 3.54 operating system. Included with the K2000VP during an introductory period will be the new Analog Collection of synthesizers. The keyboard can be upgraded with all K2000 options and features, including sampling, 64 MB of RAM and PRAM. For more details, contact Kurzweil, P.O. Box 99995, Lakewood, WA 98499-0995. Tel: 253-589-3200. Circle EQ free lit. #110.

DOUBLE YOUR PLEASURE

Yamaha has launched the MD8 multitrack MiniDisc recorder. The 12-input mixer features 3-band EQ with sweepable mids, two aux sends, two mic inputs with balanced XLRs, 48-volt phantom power, and independent cue level and pan controls. Channel flip switches enable up to 20 inputs at mixdown. A single data disc provides up to 18 minutes of 8-track recording, which can be extended using the 4-track (37 minutes per track), 2-track (74 minutes), or mono (74 minutes) modes. Up to eight tracks can be recorded simultaneously. Songs that may have been recorded on the Yamaha MD4 can be converted to eight tracks by using the Copy function with the ability to reorder songs, erase, or duplicate. For further information, contact Yamaha Corporation of America, Pro Audio & Combo Division, Professional Audio Group, P.O. Box 6600, Buena Park, CA 90622. Tel: 714-522-9011. Web: www.yamaha.com. Circle EQ free lit. #111.

SWEET SIXTEEN

Roland's new VS-1680 24-bit digital studio workstation offers 16-track random-access digital audio recording, mixing, editing, and effects processing in a complete table-top workstation. Building on the same technology as Roland's VS-880, the VS-1680 offers 16-track playback, 8-track simultaneous recording, a 26-channel fully automated digital mixer, 256 "virtual" tracks, automated digital mixing, optional CD recording capability, and two optional multieffects boards offering four independent stereo effects processors. For more details, contact Roland Corporation U.S., 7200 Dominion Circle, Los Angeles, CA 90040. Tel: 213-685-5141. Web: www.rolandus.com. Circle EQ free lit. #112.



EQ PRODUCT VIEWS

VIRTUAL CONSTRUCTION KIT

Equipped with up to 900 effect variations, the Virtualizer DSP 1000 from Behringer uses a 24-bit DSP with dual processing engines to produce 32 unique algorithms with both parallel and serial processing available from both engines. The DSP 1000's array of effects include different reverbs, tremolo, chorus, flanging, stereo delays, a vocoder, and more. Additional high and low shelving EQs are also available to fine tune every effect. The Virtualizer allows all values and parameters to be edited from the front panel using a large jog wheel and edits can be stored in a bank of 100 user memories. The unit employs 20-bit A/D and D/A converters and a 48 kHz sampling rate. Suggested retail price for the Virtualizer DSP 1000 is $219. For further details, contact Samson Technologies Corp., P.O. Box 9031, Syosset, NY 11791-9031. Tel: 516-364-2244. Web: www.samsontech.com or www.behringer.de. Circle EQ free lit. #113.

BACK TO SCHOOL

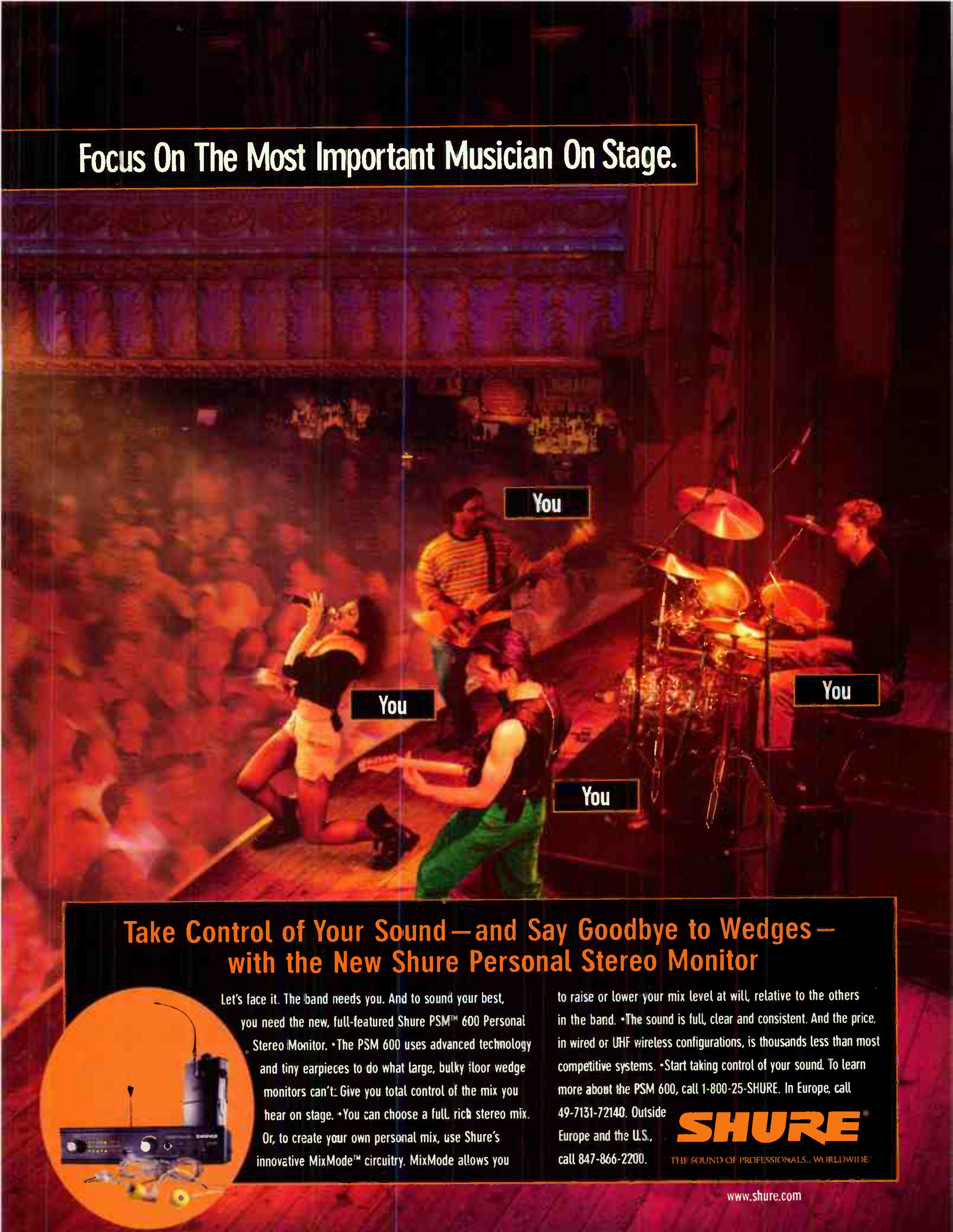
The Conservatory of Recording Arts & Sciences recently announced that it has become an authorized Avid/Digidesign Education Center. The Conservatory is now the only accredited institution in the United States to offer Digidesign Pro Tools Course 135. All Conservatory students who enroll in, and graduate in, the Master Recording Program will now be Pro Tools Certified. In order to meet Avid's requirements to teach the course, the Conservatory has augmented its on-campus hardware with a number of brand new Pro Tools 4.0 workstations. The school has also set aside custom classroom space on campus exclusively for Pro Tools. As part of the Conservatory's certification for this program, three of their instructors were formally trained by Avid/Digidesign representative Joel Krantz — they are three of only 15 authorized instructors in the world.

The Master Recording Program includes study in multitrack music recording, sound reinforcement, MIDI, troubleshooting and electronics, and music business. Conservatory students are required to complete a 280 clock-hour internship in order to graduate.

For more information, contact The Conservatory of Recording Arts and Sciences, 2300 E. Broadway Rd., Tempe, AZ 85282-1707. Tel: 800-562-6383. Circle EQ free lit. #115.

DIGITAL DOMINATION

Spirit by Soundcraft has announced the arrival of the Digital 328 digital console. The Digital 328 comes in a 32/8/2 frame size that includes 16 mic/line input channels with Spirit's Ultramic+ preamps, high-pass filters, and inserts, as well as 16 tape return channels — all routable to groups and mix. The board also offers five pairs of stereo inputs, and mic/line inputs, tape returns, and group and master levels are accessed in banks of 16 via three fader bank buttons. Each input has access to 3-band, fully-parametric EQ, four external effects sends, and access to two internal Lexicon effects units with editable and storable patterns. The Digital 328 also includes two TASCAM TDIF and two ADAT optical interfaces as standard. Two Digital 328s can be cascaded to provide 32-track digital recording and a total of 84 inputs. For more details, contact Spirit by Soundcraft, Inc., 4130 Citrus Ave., Suite 9, Rocklin, CA 95677. Tel: 916-630-3960. Web: www.spiritbysoundcraft.com. Circle EQ free lit. #114.

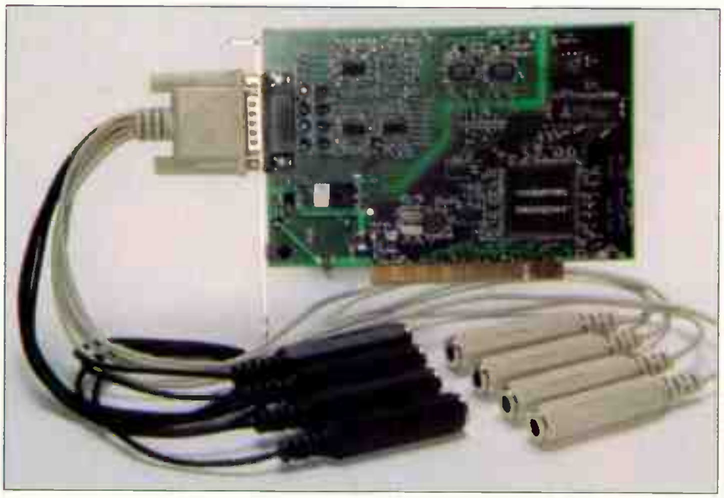


CIRCLE 83 ON FREE INFO CARD

EQ STUDIO WARE

FIRST CLASS UPGRADE

Opcode Systems has announced a major upgrade to Studio Vision Pro. Newly added to the system is full support of Digidesign's Pro Tools 24 recording hardware, as well as the addition of a comprehensive and flexible implementation of the Pro Tools TDM bussing system that allows for the mixing and routing of Pro Tools tracks to any effects and busses in the system. Studio Vision's DSP functions now operate fully on 8- to 24-bit stereo files and functions have also been added for easy creation of professional quality audio crossfades. The program's collection of real-time tools has been expanded to include nondestructive groove quantizing, while the program's "Groove Edit" window provides a flexible way to get rhythm tracks moving quickly via a time-based grid in which events can be quickly sequenced, drum-machine style. A new tool, "The Magic Drumstick," lets users modify, move, and groove elements, all while the beat plays on. Opcode has also added a "Smart Overdub" feature to assignable consoles in Version 4.0, which allows users to record fader moves in a single pass and then during a second pass, record new data only when the fader is moved. For more information on Opcode's Studio Vision Pro, contact Opcode Systems, 3950 Fabian Way, Suite 100, Palo Alto, CA 94303. Tel: 650-856-3333. Web: www.opcode.com. Circle EQ free lit. #116.

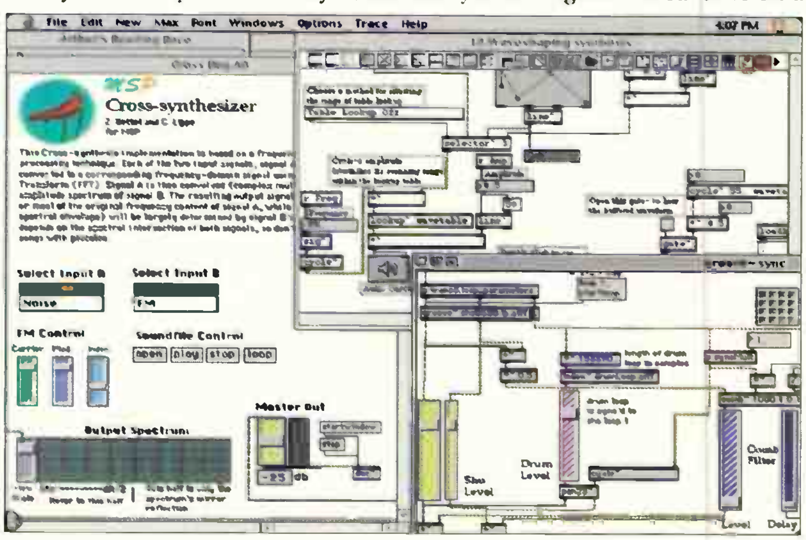


WHO'S THE MAN?

The DMAN 2044 from MIDIMAN is a high-performance PCI digital audio card that supports eight channels of analog audio (4 inputs and 4 outputs). The DMAN 2044's PCI-based design allows all eight channels of audio to operate simultaneously (full-duplex), and its analog audio specifications meet or exceed those of an audio CD. The DMAN offers a number of advanced features, including: 20-bit delta-sigma converters with 128x oversampling; a frequency response of 20 Hz–22 kHz (±0.5 dB); on-board DSP available to perform audio effects such as reverb and chorus on all audio channels; audio connections via break-out cable with unbalanced female 1/4-inch connections; 22, 24, 44.1, and 48 kHz sampling rates; full Windows 95 Plug-And-Play compliance (Windows drivers included). For more details, contact MIDIMAN, 45 E. St. Joseph Street, Arcadia, CA 91006-2861. Tel: 626-445-2842. Web: www.midiman.net. Circle EQ free lit. #117.

RETRO STYLE

Cycling '74 has announced the release of MSP, a set of extensions to Opcode's MAX 3.5 graphical programming environment. MSP consists of over 60 objects that synthesize, process, analyze, and delay audio signals in real time on a Power PC Mac OS computer. The objects can be grabbed and grouped in a graphical manner to create custom applications. A free Runtime version is available that runs any application created with MSP, which includes the complete set of objects, tutorials, and a number of demos. MSP's 60 objects cover synthesis, input/output, signal processing, sampling, math, control, and signal analysis. C programmers can create their own objects and add to this basic collection. MAX 3.5 is required to create and edit MSP applications and is available from Opcode Systems at www.opcode.com. MSP can be downloaded at the Cycling '74 Web site at www.cycling74.com. Users transform the trial version into a fully-functional copy of MSP by purchasing an authorized code online for $295. For more information, contact Cycling '74 at 408-457-0211. Circle EQ free lit. #118.



Gina™ Digital Multitrack.

Get Music Into Your Computer. Get Music Out of Your Computer. Make Records. Have Fun.

You've just written the masterpiece to end all masterpieces. The one that's destined to be the title track of your CD. Do you turn to your trusty four-track recorder? Not a chance. This isn't a demo—this is your life. Okay. So you call up a major studio and book a month or so of premium session time. (Thank goodness you're independently wealthy.)

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Gina Digital Multitrack. Enjoy.

$499, including multitrack recording/editing software.

Gina is 100% compatible with all of the best Windows 95 multitrack recording and editing software, including programs from Steinberg, Sonic Foundry, Cakewalk Music Software, Innovative Quality Software, SEK'D, and Syntrillium Software.

Is your PC ready for Gina? Download the Echo Reporter™ system analysis software from www.event1.com
Gina is designed and manufactured in the U.S. by Echo Digital Audio, an Event Strategic Partner

EVENT ELECTRONICS

CIRCLE 30 ON FREE INFO CARD

IT'S IN THE CARDS

E-mu's Audio Production Studio is a sampling/audio system consisting of a Windows 95 PCI card (the E-Card), a unique Audio Access Bay front panel, and a comprehensive software suite for sample playback, recording, and sound design. The E-Card is based on E-mu's new EMU10K1 multimedia audio processor — a custom chip designed to handle a number of high level audio functions, including: 64-voice wavetable sampling/synthesis, streaming digital audio, sample-rate conversion, and simultaneous multieffect processing. The E-mu Audio Production Studio carries a suggested retail price of under $699. For further information, contact E-mu Systems, Inc., P.O. Box 660015, Scotts Valley, CA 95067-0015. Tel: 408-438-1921. Web: www.emu.com. Circle EQ free lit. #119.

TAKE THE PLUNGE

Wave's recently introduced the EasyWaves entry level native-process bundle. Waves EasyWave includes four processors — 4-band paragraphic EQ, Compression, Gating, and Reverb — and is designed to run on both PC and Mac platforms without any additional hardware. EasyWaves includes: AudioTrack and EZVerb with reverb with presets taken straight from the TrueVerb. EasyWaves is compatible with any Mac software that supports the Adobe Premiere Plug-Ins architecture and DirectX on Windows, including: Premiere, Cubase-VST, Peak, WaveConvert Pro, SoundEdit 16, Deck II, Digital Performer 2.1, Logic Audio 3.0, Studio Vision Pro 3.5, and other platforms supporting the Premiere architecture for the Macintosh; and Sound Forge (v4.0 or above), CakeWalk 6.0, Cubase-VST, WaveLab 1.6, CoolEdit Pro, WaveConvert Pro, and other platforms supporting the DirectX Plug-Ins architecture for the PC. EasyWaves is available for $150 and Waves provides an upgrade path to other Waves' bundles. For more details, contact Waves, 6716 Central Avenue, Suite 8, Knoxville, TN 37912. Tel: 423-689-5395. Web: www.waves.com. Circle EQ free lit. #120.

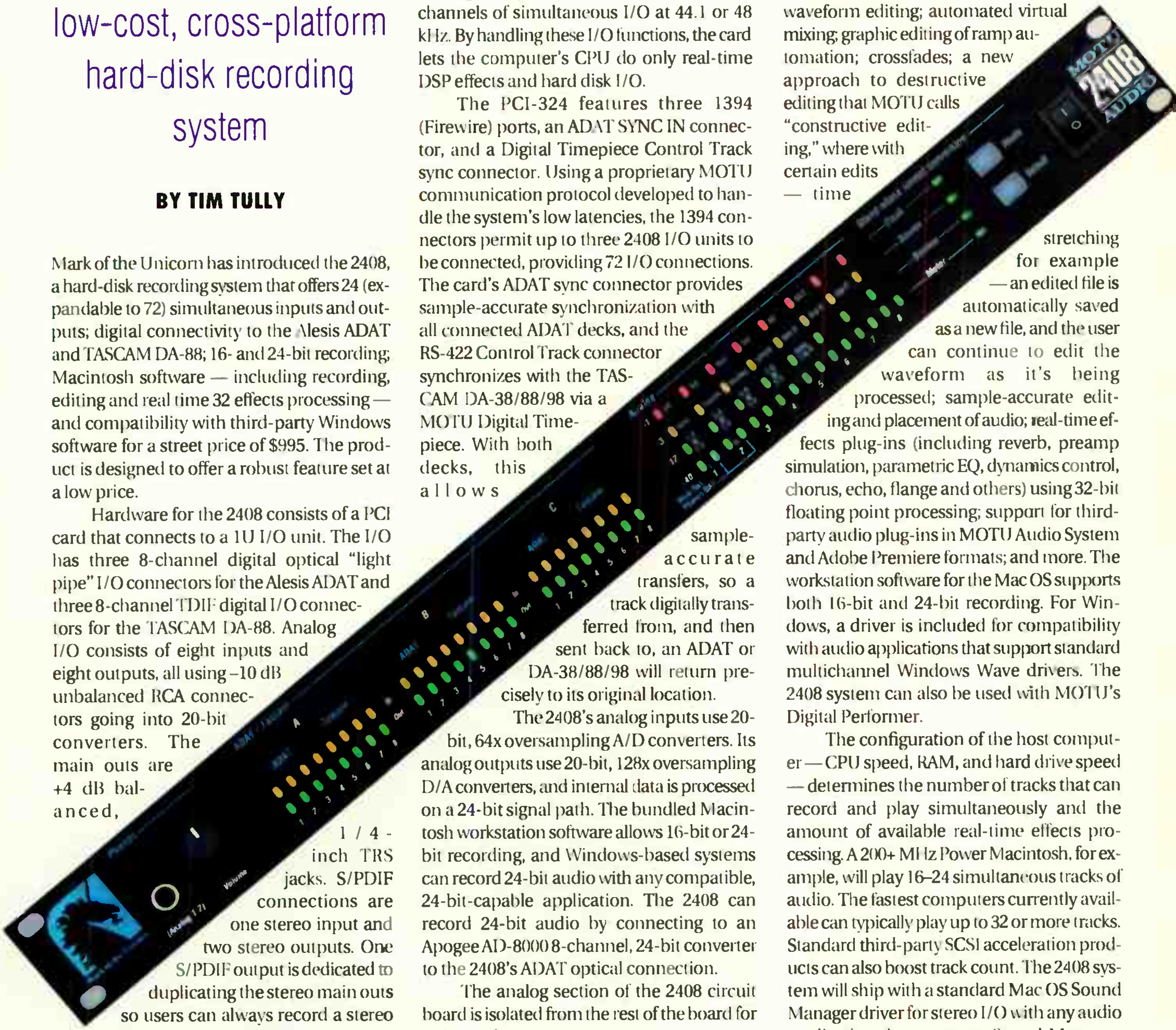
IF YOU BUILD IT...

Audio Architect 3.0 from Audio Software Ltd, re-creates modular analog synthesis on the PC, and, with its latest enhancements, has become a fully functional, playable synth, offering users an array of classic synths in one piece of software. Users can instantly emulate a number of classic synths, from the 303 to the DX7, and users can create their own custom-made networks of synthesis modules to design their own original and unique synthesizers. Version 3.0 can now be played in real time and can take its input from a MIDI device such as a keyboard or a sequencer. The modules are designed to respond to various keyboard controls, including the mod-wheel, pitch bend, and keyboard velocity. Audio Architect offers a wide range of synthesis options, including subtractive, FM, phase distortion, and basic additive. An evaluation version is available to download from the Audio Architect Web site. This can be upgraded with an activation key which is supplied on payment. The full version retails for $200. For more details, contact The Karnataka Group, The Blackfriars Foundry, 156 Blackfriars Road, London, SE1 8EN. Tel: 0171-721-7201. Web: www.audioarchitect.com. Circle EQ free lit. #121.

Mark of the Unicorn 2048

MOTU announces a low-cost, cross-platform hard-disk recording system

BY TIM TULLY



Mark of the Unicorn has introduced the 2408, a hard-disk recording system that offers 24 (expandable to 72) simultaneous inputs and outputs; digital connectivity to the Alesis ADAT and TASCAM DA-88; 16- and 24-bit recording; Macintosh software — including recording, editing and real time 32 effects processing — and compatibility with third-party Windows software for a street price of $995. The product is designed to offer a robust feature set at a low price.

Hardware for the 2408 consists of a PCI card that connects to a 1U I/O unit. The I/O has three 8-channel digital optical "light pipe" I/O connectors for the Alesis ADAT and three 8-channel TDIF digital I/O connectors for the TASCAM DA-88. Analog I/O consists of eight inputs and eight outputs, all using –10 dB unbalanced RCA connectors going into 20-bit converters. The main outs are +4 dB balanced, 1/4-inch TRS jacks. S/PDIF connections are one stereo input and two stereo outputs. One S/PDIF output is dedicated to duplicating the stereo main outs so users can always record a stereo mix to a DAT deck without swapping cables with other S/PDIF devices. The other two S/PDIF connectors serve as independent stereo inputs/outputs. All analog and digital audio connectors are gold-plated. BNC word clock I/O connectors synchronize with standard word clock devices.

The system's PCI card — the PCI-324 — is based on a custom-programmed VLSI chip with two main functions. It acts as a 72 x 72 patchbay, allowing any input/output routing configuration; and it can process 144 channels of simultaneous I/O at 44.1 or 48 kHz. By handling these I/O functions, the card lets the computer's CPU do only real-time DSP effects and hard disk I/O.

The PCI-324 features three 1394 (Firewire) ports, an ADAT SYNC IN connector, and a Digital Timepiece Control Track sync connector. Using a proprietary MOTU communication protocol developed to handle the system's low latencies, the 1394 connectors permit up to three 2408 I/O units to be connected, providing 72 I/O connections. The card's ADAT sync connector provides sample-accurate synchronization with all connected ADAT decks, and the RS-422 Control Track connector synchronizes with the TASCAM DA-38/88/98 via a MOTU Digital Timepiece. With both decks, this allows sample-accurate transfers, so a track digitally transferred from, and then sent back to, an ADAT or DA-38/88/98 will return precisely to its original location.

The 2408's analog inputs use 20-bit, 64x oversampling A/D converters. Its analog outputs use 20-bit, 128x oversampling D/A converters, and internal data is processed on a 24-bit signal path. The bundled Macintosh workstation software allows 16-bit or 24-bit recording, and Windows-based systems can record 24-bit audio with any compatible, 24-bit-capable application. The 2408 can record 24-bit audio by connecting to an Apogee AD-8000 8-channel, 24-bit converter to the 2408's ADAT optical connection.

The analog section of the 2408 circuit board is isolated from the rest of the board for quiet performance.

The included console software (Mac and Windows) uses the 2408's three internal 8-channel I/O busses to set its 24 simultaneous inputs and outputs to any combination of I/O format. For example, eight inputs could be analog, while another eight are set to ADAT optical, and a third set of eight are set to TASCAM TDIF.

The 2408 also includes full-featured audio workstation software for the Mac, including nondestructive, multichannel waveform editing; automated virtual mixing; graphic editing of ramp automation; crossfades; a new approach to destructive editing that MOTU calls "constructive editing," where with certain edits — time stretching for example — an edited file is automatically saved as a new file, and the user can continue to edit the waveform as it's being processed; sample-accurate editing and placement of audio; real-time effects plug-ins (including reverb, preamp simulation, parametric EQ, dynamics control, chorus, echo, flange and others) using 32-bit floating point processing; support for third-party audio plug-ins in MOTU Audio System and Adobe Premiere formats; and more. The workstation software for the Mac OS supports both 16-bit and 24-bit recording. For Windows, a driver is included for compatibility with audio applications that support standard multichannel Windows Wave drivers. The 2408 system can also be used with MOTU's Digital Performer.

The configuration of the host computer — CPU speed, RAM, and hard drive speed — determines the number of tracks that can record and play simultaneously and the amount of available real-time effects processing. A 200+ MHz Power Macintosh, for example, will play 16–24 simultaneous tracks of audio. The fastest computers currently available can typically play up to 32 or more tracks. Standard third-party SCSI acceleration products can also boost track count. The 2408 system will ship with a standard Mac OS Sound Manager driver for stereo I/O with any audio application that supports Sound Manager. MOTU expects to ship the 2408 in the first quarter of 1998.

For more information, contact Mark of the Unicorn, 1280 Massachusetts Ave. Cambridge, MA 02138. Tel: 617-576-2760. Web: www.motu.com info@motu.com. Circle EQ free lit. #130.

Hut Stuff

Spinning records deep in the heart of Brooklyn

STUDIO NAME: DigitalHut Sounds/BML
LOCATION: Brooklyn Heights, Brooklyn, New York
KEY CREW: Joe Natoli and Patrick Messinetti (Atomic Babies)
PROJECTS RECORDED: *DJ Mix Trance Vol. 1*; *DJ Mix Happy Hardcore Vol. 1*; *DJ Mix Trip Hop Vol. 2*; *DJ Mix Drum N Bass Vol. 1*; *DJ Mix House Vol. 1*; *DJ Techno Mix Vol. 1, 2, & 3*; *Digital Empire*; *In the Mix Techno Vol. 1*; *In The Mix Happy Hardcore Vol. 1*, *In The Mix House Vol. 1*; *In The Mix Trance Vol. 1*; *In The Mix Trip Hop Vol. 1*; *In The Mix Drum N Bass Vol. 1*; *Frankie Bones Computer Controlled, Digitized*; *Abracadabra*; *Beat Smart Vol. 1*; *Frankie Bones Factory 101*; *Synchronicity @ Hemp Splash*; *Lemon 714*; *Believe*; and more.
CREDITS: Production for BMG, K-Tel, Simitar, BML; over 30 albums in one year.
CONSOLE: Mackie 32 x 8 x 2 (main), 1604 (submixer)
SOUND MODULES/KEYBOARDS: Roland JP8000, SH101, MC202, and TB303; Yamaha CS1-X; Quasimidi Rave-o-lution 309; Korg M1 and 707; Novation Super Bass Station
SAMPLERS: Roland S330; E-mu SP1200; Ensoniq ASR-X; Casio FZ-1 and FZ-10m
MONITORS: Tannoy PBM 6.5 and PBM8.5
AMPLIFIERS: Alesis RA-100; Crest CA-2; Soundcraftsman PCR 800; Onkyo 300
COMPUTERS AND SOFTWARE: Power Computing (Mac clone) Power Center 132; Digidesign Pro Tools, Masterlist CD, and AudioMedia 3; MOTU Digital Performer
RECORDERS: Nomai 680 RW (CD burner); Power User Pro Hard Drives (4 GB + 1 GB)
DAT MACHINES: TASCAM DA-30 mkII, and DA-20; Sony DTC 690; Pioneer D-900
OUTBOARD GEAR: Ensoniq DP Pro; Alesis Microverb 4, Midiverb 2, and Quadraverb; Ross Systems DL 999; MOTU MTP AV (MIDI interface); Icom Labs DSM 200; Garfield Electronics Mini Doc; Roland SBX-10; RE'AN AP Audio Patchbays [4]; Rolls RA 62; dbx 1046.
MICROPHONES: Røde NT2; Shure SM58
DRUM MACHINES/DRUM MODULES: Roland R8, R8mkII, TR 909, TR 808, TR 606, TR 707, and TR 505; Novation Drum Station; Alesis HR-16; Ensoniq ASR-X; E-mu SP 1200
CD PLAYER: TASCAM CD-201
PRO CASSETTE: TASCAM 302 [3]
OTHER: Techniques SL-1200 mkII (turntable) [4]; Stanton Vestax 500T (DJ mixer); Carvin 3018 (subwoofer); JBL 15-inch 3-way DJ monitors; Klipsch horn (midrange and tweeter driving system DJ Speaker)
STUDIO NOTES: Natoli states: We are adding gear all the time — you know how that goes. Within the next year we feel that our studio will pretty much be complete, at least for a while. Every day new gear is introduced and old gear becomes obsolete; fortunately, we thrive on the old stuff.
FAVORITE GEAR: Natoli continues: We like the Roland R8 and R8mkII drum ma-

chines. A lot of people don't realize the potential of these machines for dance music. If given enough time and devotion, the results, at times, can be unmatched by any other drum units. They offer excellent flexibility.

We like the sound and layout of older samplers. Two of our favorites are the E-mu SP 1200 and the archaic Casio FZ-1 and FZ-10m. There is a "special" sound that comes out of these samplers. We choose to use these rather than newer, more sophisticated units.

STUDIO TIPS: Messinetti states: When producing dance music, we often use our computer sequencer (MOTU Digital Performer) in conjunction with individual units' internal sequencers. For example, we would create a rhythm track inside of the R8 or TR909, a bass line in the 309, and then sequence some pads and synth lines using Performer. The R8/909 and 309 would be set to pattern "play modes" and set to external MIDI sync modes. This yields greater flexibility on mix down and always generates a much more "spontaneous" style in the final track. EQ

BROOKLYN'S BEST: Joe Natoli (left) and Patrick Messinetti.

PHOTO BY CHRIS JOHNSON

Shure SM53

The "hole" story of this multifaceted mic

MICROPHONE NAME: Shure SM53
FROM THE COLLECTION OF: Michael Sosna/ SOS Enterprises
YEAR OF MANUFACTURE: 1969 through 1984
PRICE WHEN NEW: 1969: $150; 1984: $324.50
TYPE OF MIC: Dynamic
POLAR PATTERN: Cardioid
FREQUENCY RESPONSE: 70 Hz to 16,000 Hz
OUTPUT LEVEL: –81 dB (0.09 millivolts) @ 1000 Hz, 0 dB = 1 volt per microbar
IMPEDANCE: 150 ohms
DIMENSIONS: 7.165 inches (length) x 1.5 inches (maximum diameter)
WEIGHT: 8 ounces
MIC NOTES: Shure's SM53 was designed for use in a wide variety of applications including television, radio, recording, sound reinforcement, and film. Unlike most dynamic microphones, the SM53 has a low-frequency rolloff switch built into the mic body; this rolloff starts at around 200 Hz, with a slope of roughly 5 dB per octave. A series of small holes around the circumference of the mic body (near the middle of the body) act as rear-entry ports for obtaining the mic's directional characteristic. According to literature from Shure, only one of these ports need be open for normal operation, enabling the mic to be handheld without worry that the user might cover the ports and change the mic's polar pattern. Front-to-rear rejection of the SM53 is approximately 15 to 20 dB.
USER TIPS: Owner Mike Sosna suggests trying the SM53 along with a Neumann U87 and a Sennheiser MD421 on a combo guitar amp or cabinet. Choose one speaker and place the U87 on the left of center. Place the SM53 perpendicular to the U87, and as close as possible to (without touching) the U87 — placing the SM53 just underneath the U87, pointing at the same speaker. Next place a '421 on a second speaker in the same cabinet, symmetrically to the right of center. Bus the three mics to two tracks, with the '421 panned to 5 o'clock and the U87 to 7 o'clock. The '53 is panned to around 2 o'clock and its phase is reversed to open up the stereo image of the guitar. EQ

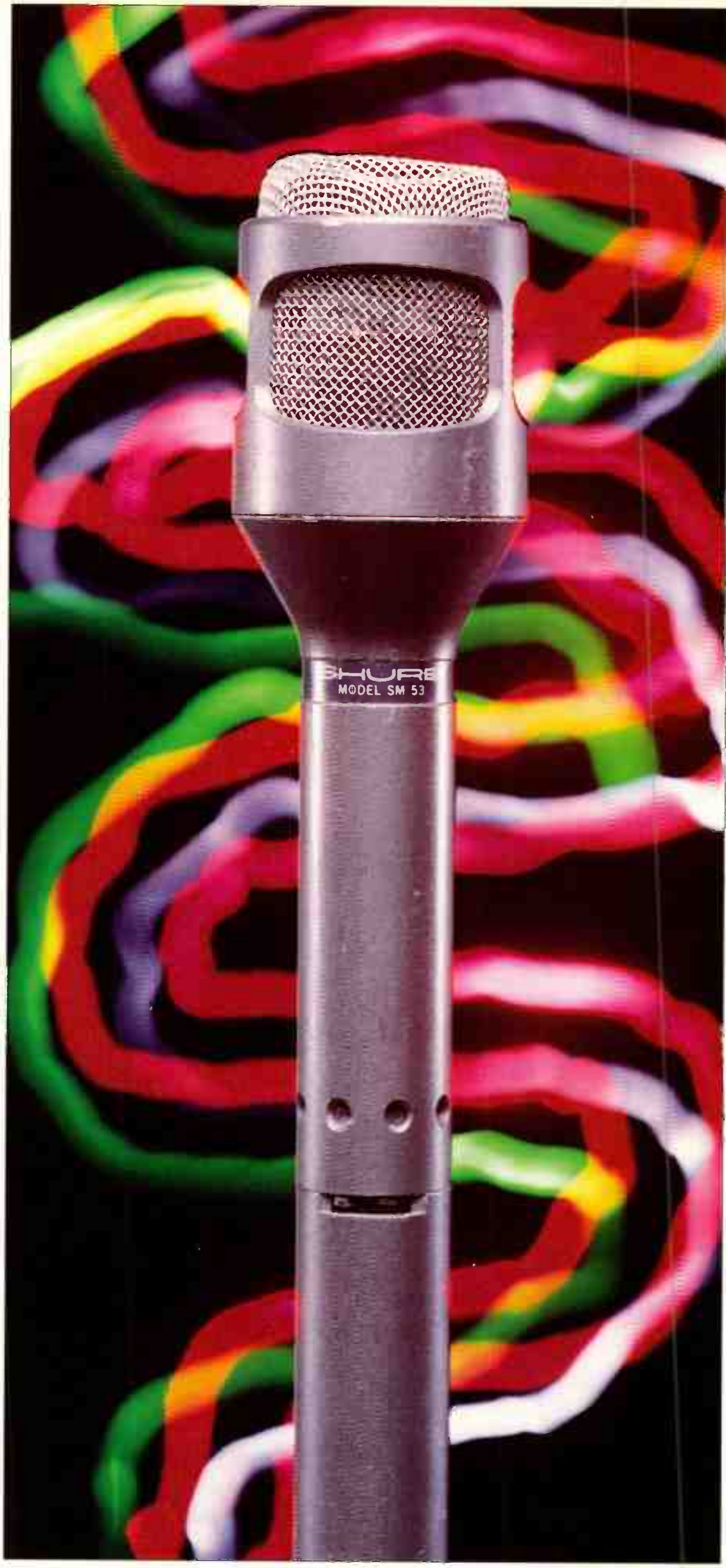


PHOTO BY EDWARD COLVER

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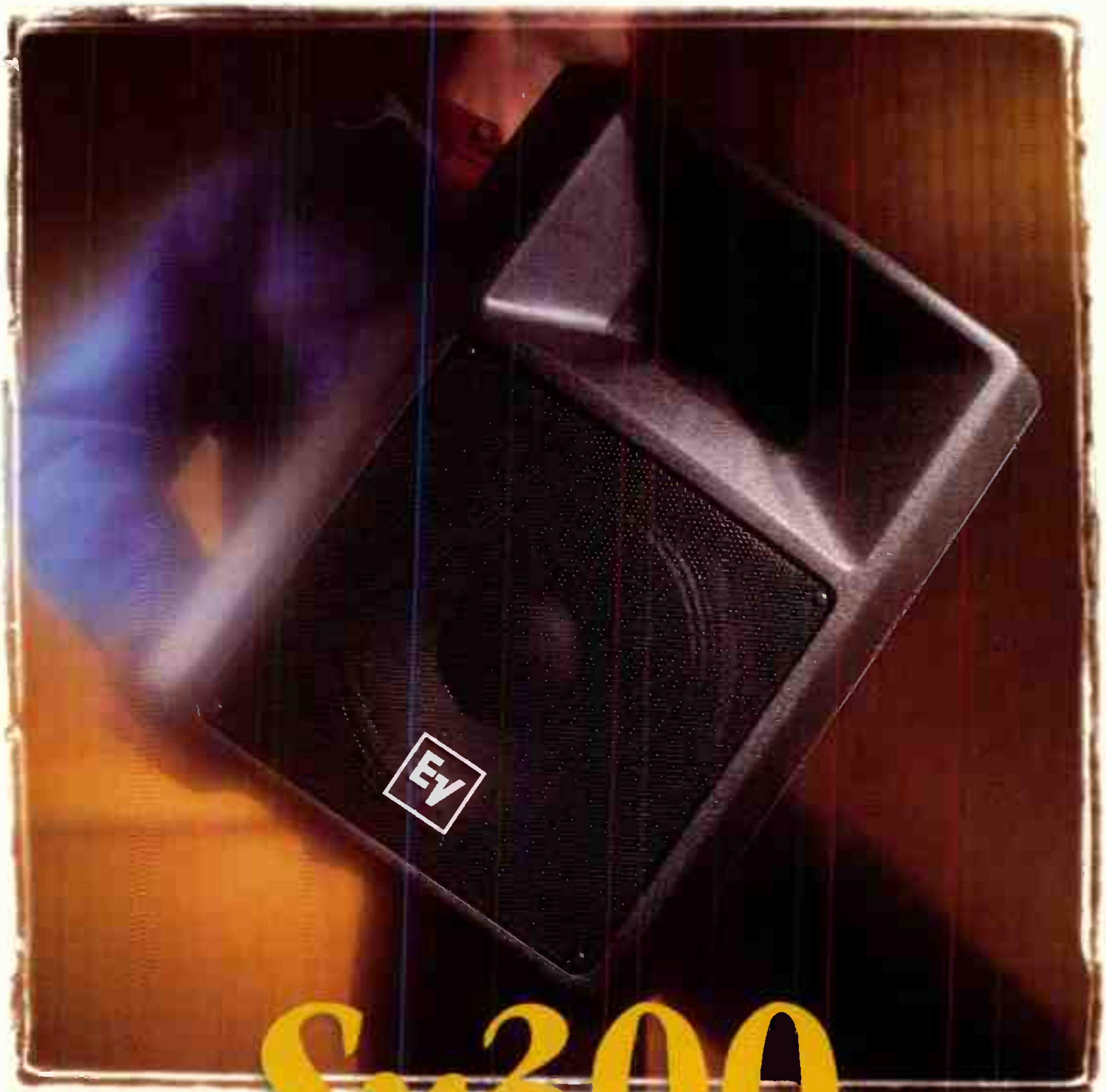

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Yamaha DSP Factory

Now you can turn your computer into a virtual 02R

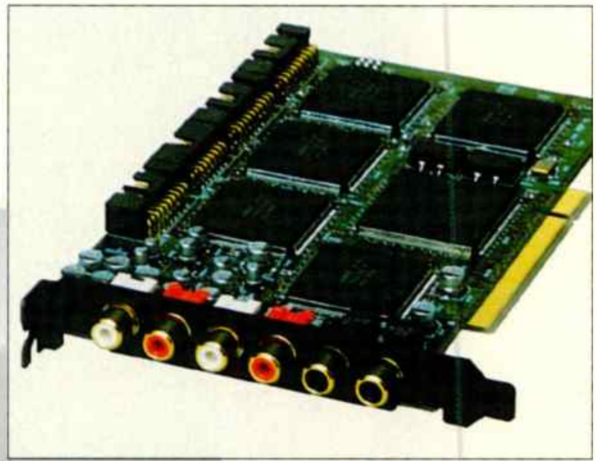
BY STEVE LA CERRA

At the winter '98 NAMM show in Los Angeles, CA, Yamaha made a major announcement: they are entering the computer audio recording market with the introduction of the Yamaha DSP Factory. Designed to enable digital recording, mixing, and editing on a PC, the first available component in the DSP Factory is the DS2416 digital mixing card. An analog audio expansion unit — the AX44 — will follow shortly.

Built on a standard PCI card, the DS2416 (which is PCI revision 2.1-compliant) essentially includes a software version of the Yamaha 02R digital mixing console — 24 channels of 32-bit digital mixing, two Yamaha digital effect processors, 10 bus outputs, 6 aux sends, 104 total bands of parametric EQ, 26 dynamics processors, digital crosspatching of channel inputs and outputs, stereo digital I/O (20- or 24-bit resolution), channel delay, metering, and 2 channels of 20-bit A/D-D/A conversion. Multichannel analog I/O is supported via the AX44. Perhaps most importantly, the DS2416 allows all of this processing to operate simultaneously, and houses five Yamaha DSP chips onboard to handle all functions — increasing performance by relieving the computer's CPU from such tasks. Part of the DS2416's muscle is directed at audio streaming, so that it's possible to simultaneously execute 16-track playback and 8-track recording of 32-bit digital audio.

Gold-plated RCA pin jacks are provided on the DS2416 for analog stereo I/O, as well as stereo S/PDIF I/O. Larger recording/mixing systems may be created in a number of ways. One is by adding an AX44 Audio Expansion Unit to the system. Fitting into the drive bay of any tower-style PC, the AX44 adds four analog inputs, four analog outputs, and a stereo headphone jack. Analog I/Os are –10 dBV unbalanced on 1/4-inch, TS jacks, while the stereo headphone jack is a standard, 1/4-inch TRS connector.

Two of the analog inputs may be switched for either mic- or line-level, allowing you to plug a microphone (albeit with a 1/4-inch connector) directly into the AX44's front panel. Up to two AX44's may be added per DS2416, and if that's not enough audio I/O capability for you, a second DS2416 (supporting another two AX44's) may be added and linked to the first. In the future, Yamaha plans to introduce additional optional interfaces, the first of which will be the AX16-AT designed to provide 16 digital I/Os in ADAT format.

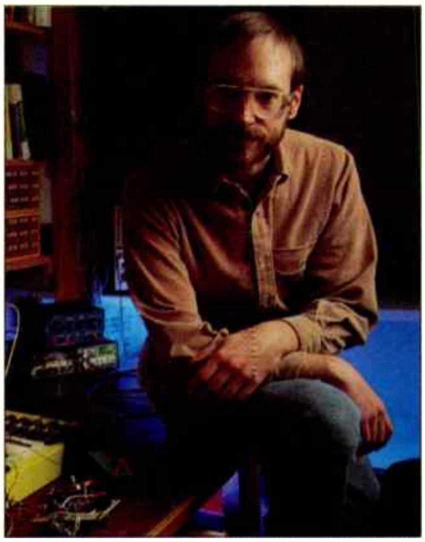


Yamaha has completed and supplied DSP Factory software drivers for the Windows 95 platform; drivers for Apple machines are currently being developed. Operation of the DS2416 card is via existing software. Manufacturers who have officially committed to support the DSP Factory include Cakewalk, Canam Computers, C-mexx, EMagic, Innovative Quality Software, Musicator, SEK'D, Sonic Foundry, and Steinberg. These developers are targeting release of DSP-Factory-compatible software to coincide with Yamaha's introduction of the DS2416 and AX44. EQ

Yamaha expects to ship the DS2416 and AX44 in the spring of this year; MSRP for the DS2416 will be under $1000; AX44, $299. For more information, contact Yamaha at 714-522-9011 or visit its Web site at www.yamaha.com. Circle EQ free lit. #101.

Doing Battle With General MIDI

How to overcome GM's limitations and make a file that will sound good on any sound card



BY CRAIG ANDERTON

If you thought that converting your songs to mono/8-bit/22 kHz was bad, are you ready for General MIDI?

Seriously, though, I'm not here to diss General MIDI. The concept is great: a standardized set of sounds, fed by an extremely compact data stream that requires neither a fast computer nor gobs of memory for playback. Besides, I'm always interested in any new application that can help keep a project studio afloat. Creating GM-format MIDI sequences or converting tunes into GM might help bring in some extra bucks.

With the proliferation of GM-compatible sound cards, we will probably see more GM sequences embedded in everything from Web sites to e-mail. For example, last Christmas I received an online greeting card that fired up the MIDI player within my browser to play a short classical piece over the PC's speakers. Very cool. And recently I was tapped to convert some of my songs into GM format for incorporation into a piece of interactive music software.

However, GM has some limitations, primarily because it's a consumer-oriented (read: low cost) solution. Compromises must be made in a GM chip set compared to pro-level synths. The challenge in working with GM is not only to overcome the spec's limitations, but to create a piece that can play back on any sound card and still sound good. Here are some tips for doing just that.

HEADERS & FOOTERS

With GM sequences, your main palette of tools consists of notes, along with commands for program change, modulation (controller 1), volume (controller 7), pan (controller 10), and pitch bend. You can safely assume that any sound card supports these.

Once a device receives a MIDI controller value, it will maintain that value until changed, so it's important to initialize all parameters at the end of a sequence (the "footer"). You also need a setup measure ("header") that sets the initial values for the above parameters. Fig. 1 shows a typical header and footer.

Regarding the header, here are a few tips:

• Place the program change first, as changing programs may reset other parameters. Try to avoid changing programs in the middle of a track, but if you must, do so during silence.

• Decide on a standard initial volume level. The recommended default is 100, but I set everything to maximum (127) and bring down levels for instruments that need to be quieter.

• As the setup measure includes many events, to avoid stressing out the synth, don't place them all on the same beat.

SEQUENCE DEVELOPMENT TIPS

• Many pro synths offer a GM mode, but always play your sequences back through a sound card for a more accurate picture of the end result.

• Check your sequencer's program-change numbering scheme. It may number programs as 1–128 or 0–127. With the latter, you need to enter a program change of, for example, 34 to call up program 35 on the sound card.

• Set your sequencer's maximum bend range to ±2 semitones, as that will be the target sound card's default.

• Be sparing with quantization. Given that with GM you're often playing back substandard sounds over substandard speakers, anything you can to do make the sound more "humanized" is worth it. In one recent tune I General-MIDIfied, no solo line was quantized, and the parts that were quantized used 80-percent strength (i.e., they were moved 80 percent closer to the correct beat, not right on top of it).

POLYPHONY

A card's standard 24-note polyphony is

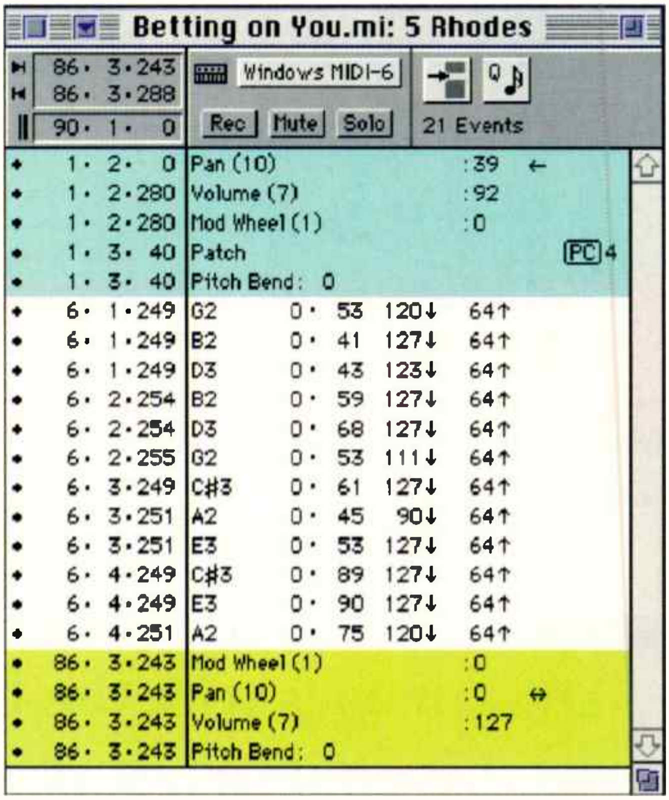


FIGURE 1: The header is highlighted in blue and the footer in yellow.

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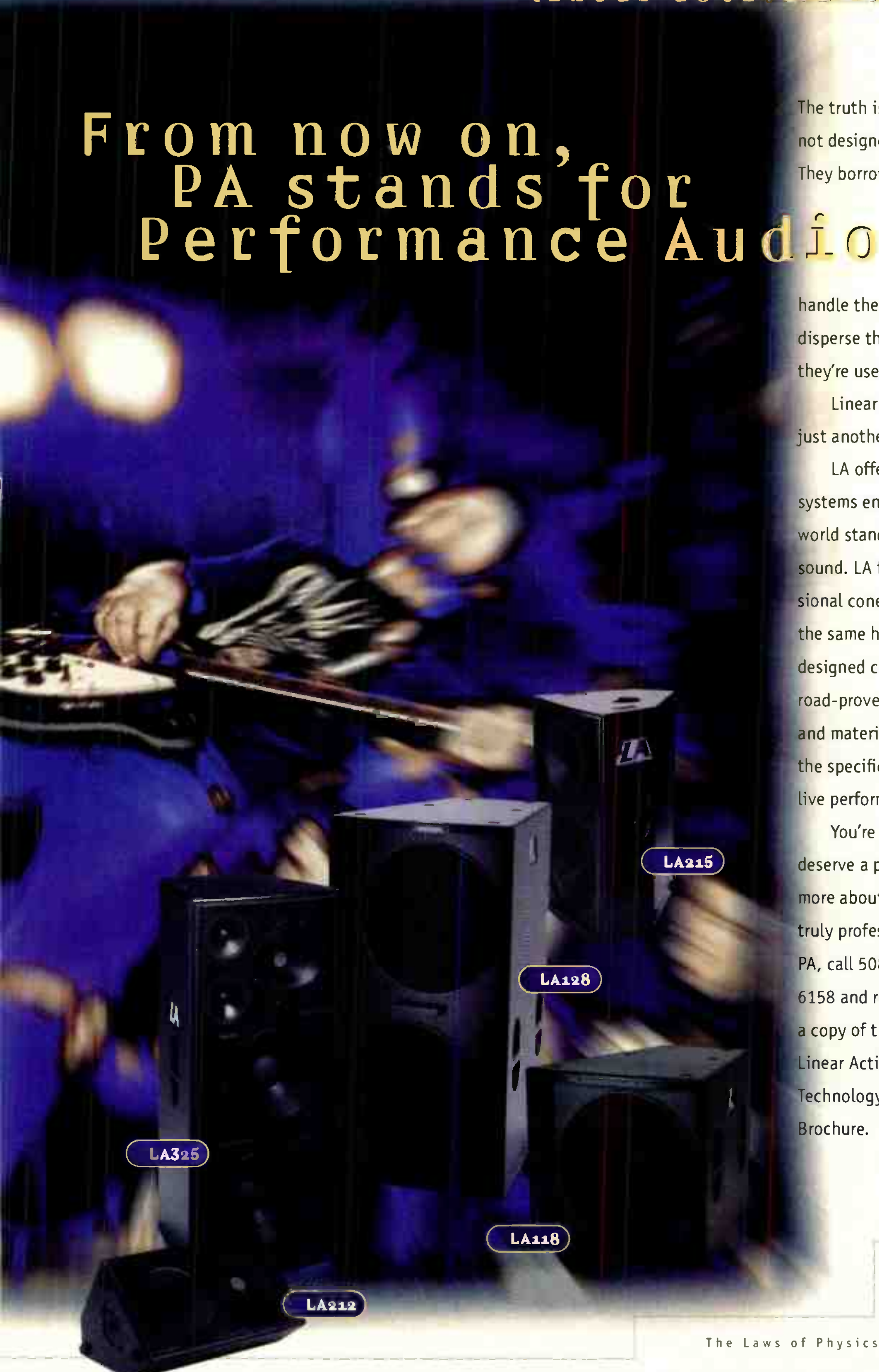
LA

From now on, PA stands for Performance Audio



The truth is that most PA systems are not designed for live performance. They borrow cone drivers from instrument amps and crossovers from home hi-fi sets. They can't handle the power and they don't disperse the sound evenly. Basically, they're useless.

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good enough for most situations. However, to minimize note stealing and improve timing, change the duration of all drum hits to a non-rhythmic value. For example, if your sequencer runs at 96 pulses per quarter note, set all drum note durations to 21 "ticks." This ensures that the note-off command doesn't fall on the beat, where other, more important data might be happening.

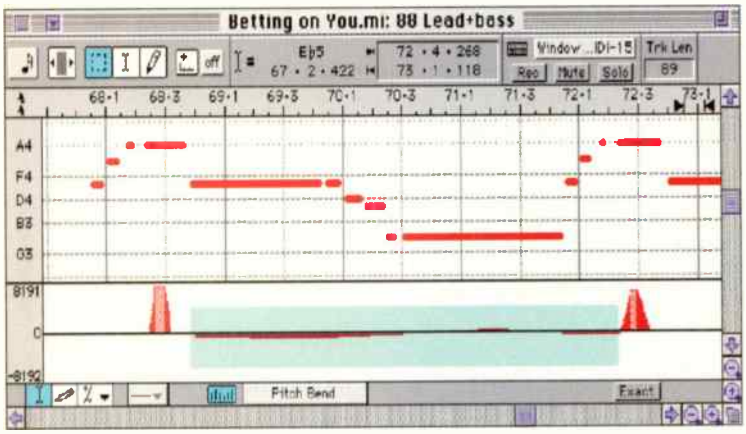


FIGURE 2: Adding slight amounts of pitch bend (highlighted in blue) to an instrument that's doubling another part adds a sort of chorusing effect.

LAYERING

One hundred and twenty-eight instrumental sounds may seem limited, but layering can help. For example, I was looking for a round, full, articulated bass sound, but found that Fingered Electric Bass (program 34) was too bright, and Fretless Bass (program 36) lacked definition. Layering the two, however, sounded great.

Another trick that's worked well for me is transposing a sound upward and layering it to provide an attack transient. Vibes, glock, harp, and acoustic guitar all work well for this.

Layering is particularly important with leads, which often sound rather wimpoid. Try layering a neutral, sine-wavy sound (e.g., piccolo or recorder) with more substantial sounds, such as distorted guitar, to add interest. Furthermore, adding a tiny amount of slow pitch bend (fig. 2) to one of the parts creates enough detuning to give somewhat of a chorus effect.

VOLUME ISSUES

Volume changes can be expressive, but be careful, because not all volume curves are the same. I often keep volume more or less constant and change velocity for dynamics. Even then, I'll compress velocity values for a more uniform response. Steinberg's Cubase has a compress velocity feature, but if your sequencer doesn't, either add a constant to all velocities to do the MIDI equivalent of limiting, or "compress" by dividing all values by a constant (e.g., 2 for 2:1 compression), then adding a constant to bring up all velocity values to the maximum possible level (127).

Layering can also help create dynamics. For example, on the tune with the layered basses, I used mostly one bass timbre for the verse, the other for the bridge, and layered the two together at equal volume for the chorus.

REALITY CHECK

Before signing off on a GM sequence, play it

BEYOND GM

The DLS (downloadable samples) extension approved by the MIDI Manufacturers Association allows adding custom sounds to the GM set by loading samples into sound card RAM. If you need a medieval sackbut sound (definitely not part of the standard 128 GM synth patches), create a downloadable sample version that loads into the sound card when you load a GM file. This overcomes existing instrument limitations, but also allows adding any digital audio to a composition, even vocals.

When GM simply isn't good enough, there's Yamaha's XG specification, which ups the number of sounds and adds fairly sophisticated signal processing. Although not as common as GM, XG has settled into a comfortable groove as a higher-level alternative. Many sequences are now available that take advantage of the XG format, and since XG is a superset of GM, GM files will play back on any XG-compatible sound module. If your sound card doesn't hack it, but you aren't quite ready to leave your high-end synth permanently connected to your computer's MIDI Out port, then consider an XG-compatible sound card such as the AudioTrix or the Yamaha MU-80 outboard box. Either one does a fine job of bridging the gap between consumer and pro performance.

through various sound cards. The Creative Labs AWE64 is useful in this respect, as it offers several playback synth engines, including: GM, 2-op FM, MT-32 synth, GS (a Roland superset of GM), and — assuming you have RAM on the sound card for loading in custom sounds — a couple of large (2 MB and 4 MB) synths. And yes, I do check for compatibility with the old 2-op FM cards. If a sequence sounds good on that, it will sound good on anything!

Note: For more information, check out the book *General MIDI* by Stanley Jungleib (ISBN 0-89579-310-5; published by A-R Editions, 801 Deming Way, Madison, WI 53717. Tel: 800-736-0070 or 608/836-9000).

Craig Anderton is the author of Home Recording for Musicians and Multieffects for Musicians *(both published by AMSCO), and* Do It Yourself Project for Guitarists *(Miller-Freeman). In addition to recording and writing, he also lectures around the world on the subject of music technology.*

Deep Soul Music: *Casualty of the '90s*

PHOTO BY JIM HERRINGTON

Today's soul music is missing a key ingredient — soul

BY AL KOOPER

Soul music, an off-shoot of blues and gospel music, reached the mass market in the '60s, primarily courtesy of Stax-Volt and Motown. Its heroes numbered Marvin Gaye, Otis Redding, Al Green, William Bell, Wilson Pickett, Smokey Robinson, Ann Peebles, Aretha Franklin, and James Brown to name but a few. This genre of music was killed off dead by the "new" R & B, a synthesized, virtual soul music, piloted by machines, greed, and hardly any songwriters. All the records from the '60s and '70s are being sampled in this new medium. Why? It would seem it's because they can't even come up with their own modern grooves to match their tech.

Now don't get me wrong — there are some good writers and artists out there now, but they are surrounded by ice cold, calculated machine musicians who don't file union contracts. What would happen if Babyface, Teddy Riley, Mariah Carey, and Boyz II Men went in the studios with real human musicians and pumped some warmth into some of those good new songs they're writing? The heartbeat would come back into the music, and, believe me, gang, it's sorely missing.

You can't call today's R & B "soul music." In Europe they've renamed the music I grew up on that I mentioned before: they call it Deep Soul; probably to differentiate it from today's "shallow" soul music, I'm guessing. Now I use machines myself, but I am always trying to emulate the human groove. I consider my machine powered tracks, demos. I can't wait to play them for real rhythm sections, so that I can gain the input of a great drummer, bass player, and guitarist.

My age group, and I know many of you readers are far away from it, left a pretty decent legacy of music for your generation to grow up with. Without Dylan, there wouldn't be half of the alternative bands on the radio today including The Wallflowers. And the same with The Velvet Underground, Jimi, and The Who.

So why won't the R&B community pay reverential respect and tribute to their legacy? Smokey, Al Green, and James Brown did it the hard way. They sampled themselves from rehearsing. Yeah, yeah, I know it's not cost-effective, but it's soul effective. Go compare the best of the '60s and '70s R&B with today's no-song, MIDI mindlessness and you will *have* to agree with me.

What would happen if Boyz II Men went in the studios with real human musicians and pumped some warmth into some of those good new songs they're writing?

There's a lot of talent out there. There are great artists on the scene today. Record companies are forcing them to work with producers who pigeonhole them into the morass that is called soul today. Somebody be brave enough to go in and take a time-honored tradition and try it on today's great material. It might just work. Let's go back into the studio with great songs and great musicians and put the soul back in soul music...I'm beggin' ya!!! EQ

Bruce Swedien

PHOTO BY ED FREEMAN

Learn the latest exploits of the legendary producer/engineer

BY MR. BONZAI

Bonzai: What have you been working on lately?
Swedien: I just finished mixing the debut album by Nicole Renee on Atlantic. She's the producer and composer, with one track produced by David Foster. This young lady is 22 years old, from Philadelphia — mature beyond belief, and the music is a mixture of ultra-modern hip-hop and some of the best singing I've ever heard. Reminds me of the early stuff I did with Michael.

As producer, what have you been up to?
I just finished a BMG/Germany album for Sweetbox, a hugely successful band in Europe and Japan. Classical music with a lady rapper — great fun and right up my alley. I'm also working on an orchestral/Indian dance album with my daughter, Roberta, who lives in New Delhi.

Why all of this international work?
Could be a result of my lecture tour, which has taken me to Tokyo, Hamburg, Cologne, Munich, Gothenburg, Mexico City, and Orlando — so far. I talk about my approach to music, which is a bit different. I learned recording with Basie, Ellington, and Quincy Jones. They were very supportive of my experimentation during a period when the engineer's imagination was not thought to be part of the recording process. I gained a few insights into what makes the emotion of recorded music work, manifested in my recording of real stereo images, rather than what I call "2-channel mono."

What are the main pieces of gear in your project studio?
Harrison 32 Series console, the same kind that I did *Thriller* on, with new Virtual Faders automation. Summing amps are from an API desk, and I've got some Neve components as well. The EQ and filters are nearly identical to those in the new Harrison series 12. I have a 24-track MCI JH-24 analog machine, 32 tracks of ADAT, 105 microphones in 15 Anvil cases, EMT 250 and 252, TC M-5000, Spatializer, and two 7-foot racks that go with me everywhere.

How tall are you?
[Sings] 6-foot-2, eyes of blue, koochi-koochi-koochi-koo.

Any tips for people recording at home?
The musical values are always the same. The only difference in recording at home is that you have to keep the dogs quiet.

What do you think of the newer digital formats?
I just used the new Pro Tools 24 system on the Atlantic project and, sonically, it's wonderful. A giant step forward. I've also heard the new Sony single-bit architecture system. Great sound. High resolu-

Suspect: Bruce Swedien

Occupation: Producer, Engineer, Composer

Hobbies: Part-time tugboat captain, scuba diver, and agriculturist

Spouse: Bea Swedien; "My business manager for 34 years — and my best friend."

Residence: Westviking Farm, CT

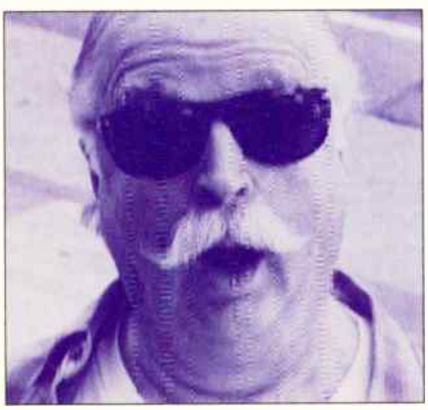
Vehicle: John Deere 950 diesel 4 wheel drive farm tractor with manure spreader

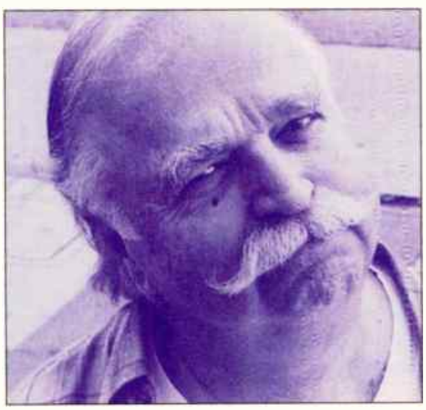
Pets: Great Dane, "Teddy"; Border Collie, "Gordy"; 25 chickens, three horses, 23 cats, one peacock, and ten turkeys

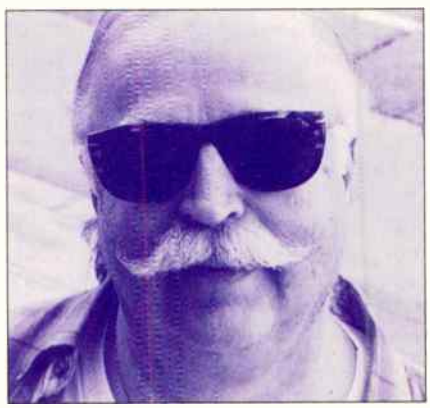
Pet Peeves: "Recording people that reach for the EQ knobs and start twisting and twirling before they've heard the sound source."

Credits: 5 Grammys for Best Engineered Album: *Thriller, Bad, Back on the Block, Dangerous, Q's Jook Joint*; TEC Award for Lifetime Achievement.

Notes: Swedien's career began with the music of the post-swing era and continues today with megatrack multimedia digital technology. He captured some of the finer moments of the '50s with greats like Duke Ellington, Count Basie, Nat King Cole, Stan Kenton, and Woody Herman. The '60s yielded such pop classics as "Big Girls Don't Cry," "The Duke of Earl," and "Higher and Higher." He has recorded, mixed, produced, and composed for records, TV, and film, with hundreds of artists. Swedien's creative skills have produced the most listened to records in history. He's a record breakin' engineer!







PHOTOS BY MR. BONZAI

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tion is the way to go, but I still prefer to originate on analog.

What's your favorite console?

The SSL 9000 J, which to me is the most powerful and incredible sounding analog desk in the world. There is none better.

What's your favorite microphone?

All of them — you can't have a favorite. Mic choice and technique are dictated by the music.

What's the secret of the sound on the Michael Jackson records?

A lot of care and a lot of imagination. Michael and I have a saying: "The quality goes in before the name goes on." Quincy and Michael have always left me a tremendous amount of room for my own creative input — the soundfields start in the imagination.

If you were a musical instrument, which would you be?

An upright double bass.

What's wrong with the music industry?

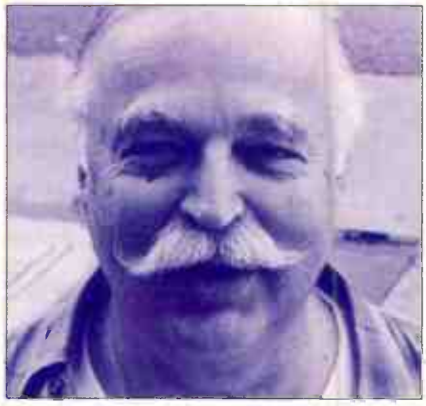
I'll have to take the Fifth Amendment on that one.

What kind of music do you listen to while you're driving?

I listen to "good" music, and that could mean classical, country, R&B, jazz, or pop. I don't believe in pigeonholing music by style — good music rises above categorization.

What is the first music you remember hearing?

Debussy's "La Mer." My mother sang in the women's chorus with the Minneapolis symphony, and I also remember hearing the works of Gustav Mahler. I was only nine or ten years old, but I learned a lot of things about orchestral set-up that I still use today. I wasn't thinking in terms of stereo, of course,



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–Teddy Riley

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Now, you're probably thinking we had to pay Teddy to say all this. But the truth is, he's been using Pro Tools on every project since Michael Jackson's *Dangerous* — simply because it's the best tool for the job. Based on the number of hits he's cranked out, his career and life have been nothing short of remarkable. Or, as Teddy would say…"*complete.*"

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but it made a lasting impression on me.

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

I believe the first musical instrument was the voice. I would love to hear prehistoric voices — before any false values were set.

What did you learn from Quincy Jones?

He told me once, "It's much easier to be done than to be satisfied." I drive everybody nuts with that philosophy, of course. I also learned about Quincy's kaleidoscopic approach to music — examining it from one angle, then you turn it and look from another angle.

Is there anyone else in the world you would like to record?

Absolutely. Nobody that I know of, though. Working with this young lady, Nicole Renee, has restored my faith in young people. I'd also love to record some really big orchestras — I love doing it and I'm very good at it, but nobody calls me [laughs] except Michael or Quincy.

Do you know any interesting business tips?

Keep it in the family — someone you can trust.

What was your most outrageous experience in a recording studio?

I was recording a well-known female artist in the '60s, and she got too warm in the vocal booth and began to remove articles of her clothing. The booth was off to the right of the control room, so I got the perfect view. The band didn't see it at first, but when they got a look, the session broke down. She was wearing a beautiful black satin brassiere.

What animal do you identify with?

The dog, for two reasons. First, the dog has incredible hearing powers. Second, dogs can't lie about love.

What makes a great producer?

A great producer is like a great director in motion pictures — he knows how to cast the work at hand, and chooses the right musicians, engineers, orchestrators, copyists. Then he gives them the freedom to do what they do well.

Have you ever witnessed a miracle?

There are no miracles in recording studios — only talent and hard work.

Who's the most amazing artist you've worked with?

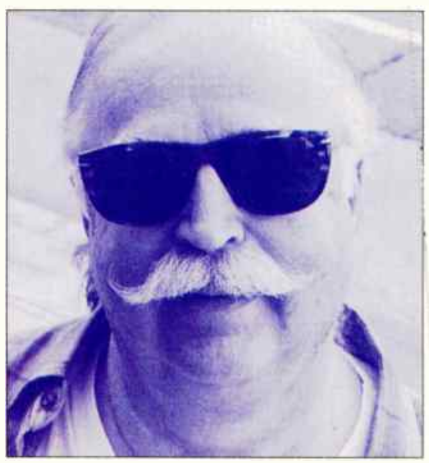
Who do you think?

What is the biggest mistake of your life?

Not marrying Bea sooner.

Any advice for getting a good start as an engineer?

Listen to a variety of live music, a lot. Develop in your mind's ear a benchmark based on live music, real music. Learn how to identify a good musical balance in an acoustical situation — all music is conceived to be heard with acoustical support. By acoustical support, I don't mean reverb or echo. Then pick a good school. And, finally, the most difficult thing for young people to learn: trust and follow your instincts. EQ



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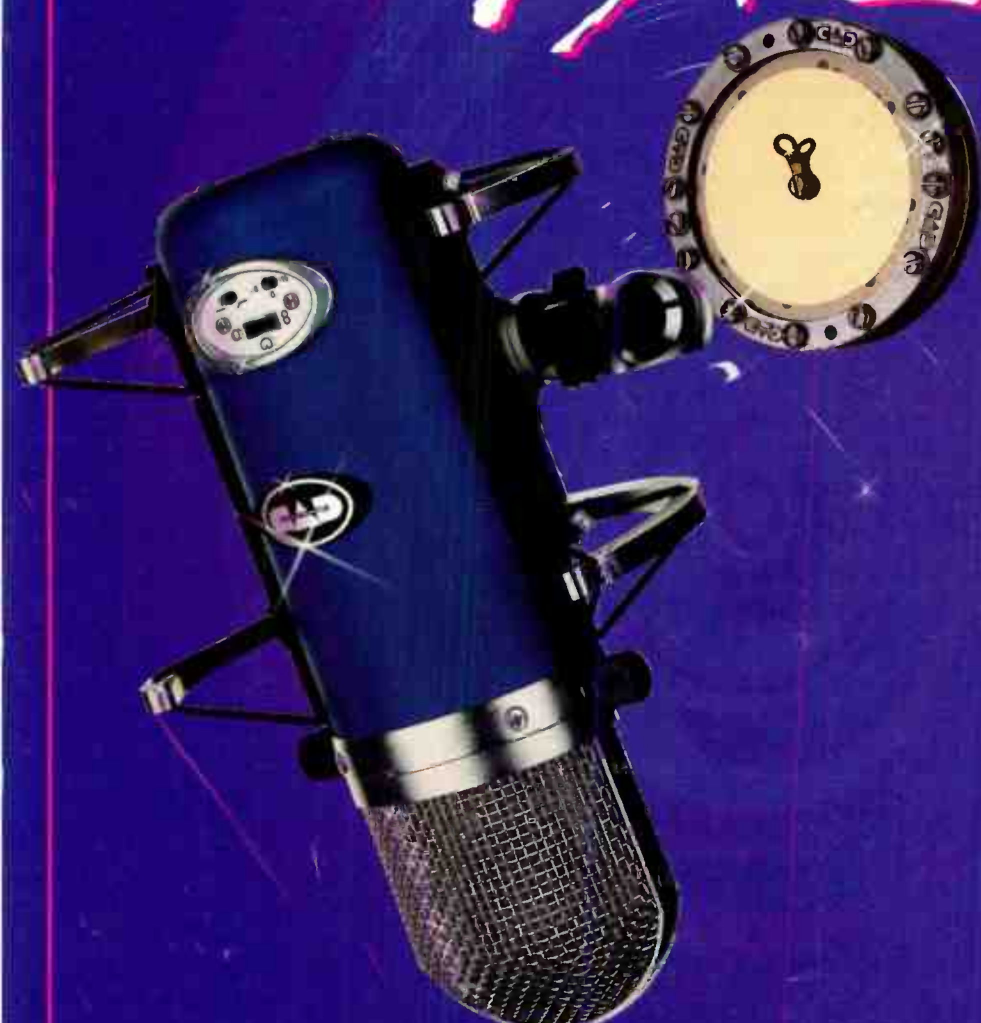

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Long Distance Project Recording

Recording a project with two studios, two DA-88's, and plenty of miles between them all

BY STEVE LA CERRA

Owning a digital multitrack tape machine is a beautiful thing, not only because it sounds good, but because of the myriad of possibilities for the way records can be made using them. Two of the more interesting projects I've worked on recently involved producer/guitarist Thomas Atkins, who until recently, lived nowhere near my studio (the Wood Shop).

HEADS OR TAILS

The first of these projects was *Heads Or Tails* (High Strung Productions) with Tom's band Reason, a four-piece alternative/rock band based in Dobbs Ferry, NY. Our goal was to create a high-quality, full-length CD on a modest budget. We were thinking in terms of 16 tracks because Tom owns a Tascam DA-88 and there's a single DA-88 at the Wood Shop. We figured that we could use the two machines together, but there were several problems to overcome. I was scheduled to be on the road with another band for several weeks smack in the middle of this project, and the modest budget supplied by the band's label required us to get it done as inexpensively as possible. Since the band lived and rehearsed in Dobbs Ferry, we thought that we'd do some of the recording at the Wood Shop and some of it at the band's studio. Then, when the tracks were finished, we'd mix at the Shop.

Tom and I agreed that it made sense to cut drums at the Wood Shop, since we have a small but great-sounding booth and a good selection of mics. This studio is very small: the control room is roughly 12 x 15 feet and the booth/live room is about 12 x 8 feet (we also have a small lounge area, which came into play). That's basically it. Setting up the entire band and recording live (with amps miked) would be difficult or impossible, and recording the guitar and bass direct was not an option for us, sonically speaking.

Our solution was to bring Tom's DA-88 to the Wood Shop to cut "real" drum tracks and roughs for the bass, guitar, and vocal. Then the band could take their machine back to Dobbs Ferry and proceed with overdubs at their rehearsal studio. Drummer Scott Sosa

GO THE DISTANCE: Two studios separated by many miles and an engineer off on tour presented producer Tom Atkins (above) with some complications during the recording of Reason's *Heads Or Tails*.

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brought his kit to the Wood Shop and set up. We had a tough time getting sufficient low end while working on the kick sound. Patching in a graphic EQ didn't really help because we were wrestling with the sound of the drum itself. After some retuning of the drum to drop the pitch, a '57 (oddly enough) turned out to be the solution. The snare was miked with a Shure SM98A, toms with Beyer M201's, and floor toms with Sennheiser 421's. Scott also had two smaller concert toms that were both miked with a single Sony C48 set to bidirectional. We used an AKG C414 on hat and a Beyer MC 833 stereo mic for the cymbals. The MC 833 was set to XY pickup (90 degrees) and placed approximately two feet above the cymbals, pointing down towards the snare drum.

All tracks were recorded onto the DA-88's through a Mackie 24•8 and, wherever possible, signal paths were kept short. The Beyer stereo mic went to a Demeter VTMP-2a mic pre, through a pair of Valley Commanders for a bit of compression and then directly to the DA-88. The rest of the drums were routed through the Mackie. Direct outs from the kick and snare channels were patched into dbx 160x compressors and then to tracks on the DA-88. The two rack toms and the two concert toms were grouped to a stereo pair of busses on the Mackie and then routed to a pair of tracks (no processing), while the two floor toms were each recorded on their own track via direct outs. This gave us a set of eight drums tracks, all on what would become the "master" tape.

At the same time, bassist Dave Fontaine was recorded direct through a Countryman Type 85 direct box to a Symetrix 501 compressor and to tape. Tom's guitar went into an ADA MP-1 guitar pre and then straight to tape (no cabinet for the roughs). Since the bass and guitar were direct, we wouldn't have to deal with the isolation problems inevitable in such a small studio. Tom and Dave would play rough tracks in the control room for a live feel, monitoring on a pair of KRK 7000B's. Vocalist Kai Lee was set up in the lounge with a pair of Fostex T20 headphones for monitoring, and a Shure SM58 (we used a Radio Shack headphone extension cable plugged into "phones 2" of the Mackie and ran it out to the lounge. The "phones 1" mix was for Scott). The '58 went into the Mackie, then a Behringer Composer, and to the DA-88. Tom, Dave, and Kai were strictly performing reference tracks for Scott's benefit. These three tracks were recorded onto the slave machine. This entire setup procedure was done on the night before the actual recording was to take place. After getting some rough sounds and levels, we called it a night. Scott came in fresh on the second day to start cutting his tracks and finished up on the third day.

Our next step was to prep a tape that would return to the band's studio for overdubs. With the second DA-88 locked to the master, we bounced a rough mix of the drums from the master to an open track on the slave tape. A total of four tracks were now on the slave tape: keeper drums and reference bass, guitar, and vocal. The band took the slave tape and their DA-88 back up to their rehearsal studio, and I hit the road.

MAKING TRACKS

Tom recorded his guitar tracks in stereo using Shure SM58's. His cabinet was a Boogie 4x12 stereo box, so he used two 58's close-up to the cones, plus one in the back of the open cabinet (phase-inverted) and another mic about two feet away from the front of the cabinet for a bit of room tone. For clean guitar sounds, Tom added a second microphone in the rear of the cab. A Samson 2242 mixer was used for signal routing and monitors were Tannoy PBM 6.5's driven by a Samson Servo 5000 amplifier. Compression came from a dbx Project One 266 comp/gate.

When it was time to record Dave's bass, Tom used a Shure SM57 close up to the speaker, and another mic slightly farther away to get a bit of air into the bass. Working in this manner really allowed Tom and Dave to take their time, getting both their tones and performances to sound the way they wanted. This was especially important for Tom, who really takes a lot of care in his guitar sound. By the time Tom and Dave were finished tracking bass and guitar, I was back from the road gig and ready to continue the project.

CAREFUL WITH THAT ERASE BUTTON

Tom returned to the Wood Shop with his DA-88 and the slave tapes,

BOOGIE NIGHTS: The guitar was recorded using a Boogie 4x12 stereo box miked in front by a Shure SM58 and also with another mic in the back of the open cabinet.

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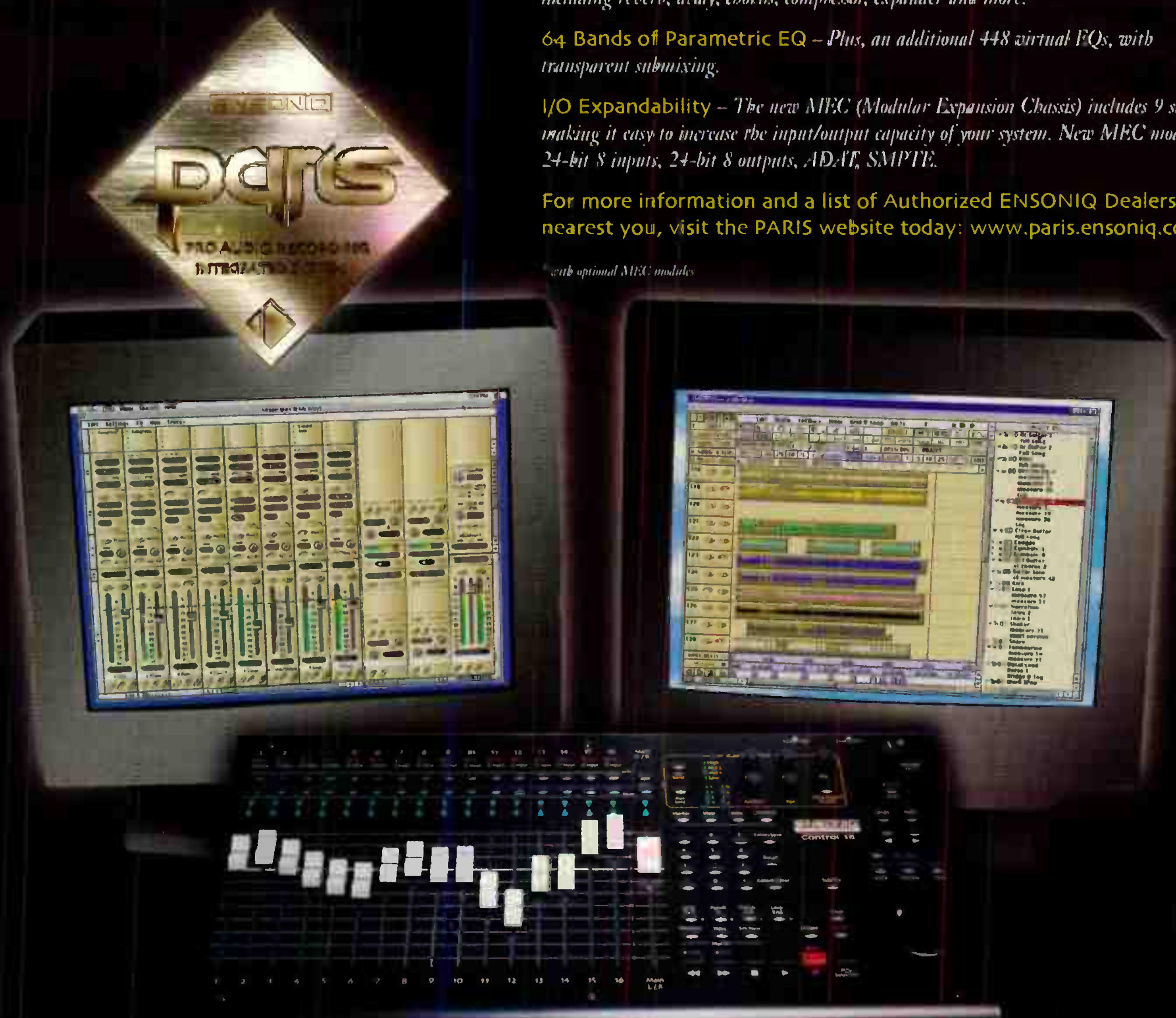

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which typically had five finished guitar and bass tracks per song. We locked the two DA-88's together and erased the rough drum mix, leaving three tracks for the vocals. Without the rest of the band around to distract us (which was always a distinct possibility), Kai and I began recording the vocals, almost exclusively through a Peavey PVM-T9000 tube mic (which really sounded great on Kai). This was patched to the Demeter pre, Valley Commander, and then to tape. Kai really has a strong voice, so we used the LF rolloff (but not the pad) on the mic, and also put a stocking-type windscreen in front of the mic. Kai and I comped many of the vocals, recording tracks while he was loose and then editing afterwards by picking sections off the work tracks and dumping them via bus to the comp track. When the vocals were finished we were ready to mix.

All mixes were done manually on the Mackie and recorded to a Sony DAT recorder. The kick drum continued to present a problem in the mix, so for a few of the songs, we triggered a sample and mixed that in with the real kick. In the patchbay, the kick track was patched to a mult. One side of the split went to a tape return on the Mackie. The other side of the split was patched to a Valley Kepex gate, through a DOD R830B 15-band graphic EQ, and then to a dbx 463 gate. The Valley gate and the EQ were used to remove as much extraneous sound from the kick track as possible, and to create a strong spike for the trigger. The second gate helped clean up any sounds that the first gate missed (which could have caused a mistrigger). Output from the 463x went to the trigger input on an Alesis D4 drum module; audio output from the D4 was returned to the console, where it was mixed in with the real kick sound.

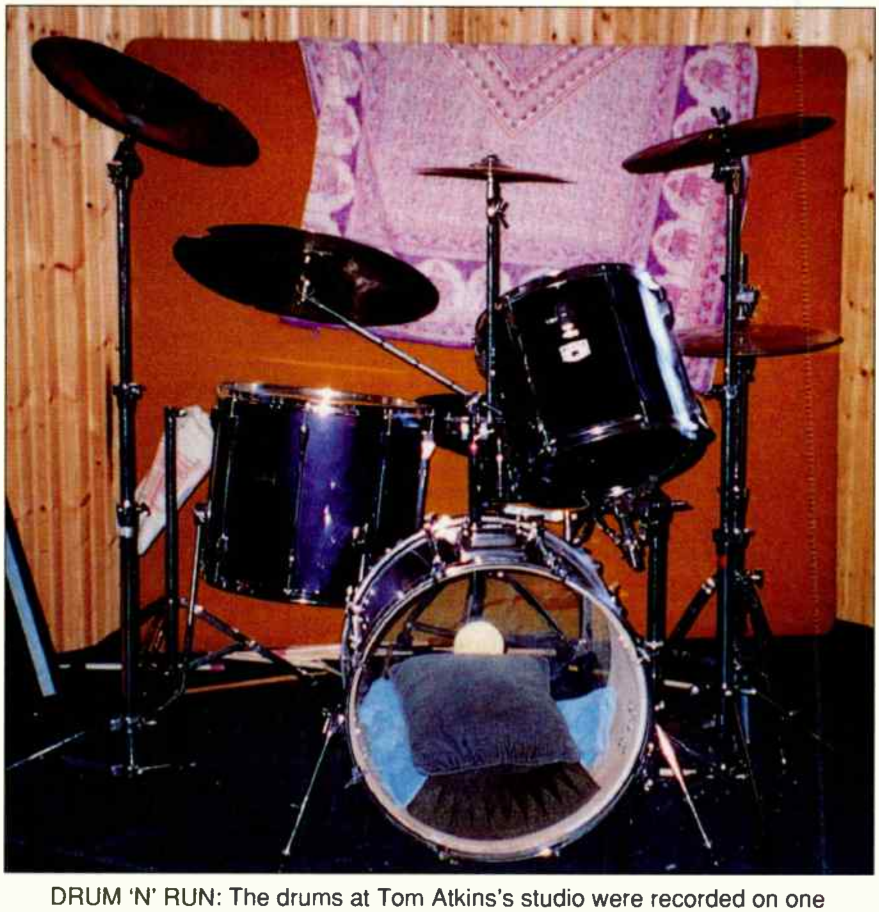
One of the cool things about working with Reason was that Kai asked me to come up with a unique vocal sound for every song. Naturally, I did my best to oblige and play with some toys. Some of the songs have a combination of reverb and delay from a Lexicon PCM80 or reverb-only from a Lexicon PCM90. On one song we used a Lexicon LXP1 set to a small room. On several of the songs, I used an ART SGE Mach II to create some very strange sounds. In particular, the song "Indian" employs a distortion program ("Turbo Dist 3") along with EQ, flange, and a reverse-gated reverb ("Rvrs 1A"). This sounds very creepy and was added in with the straight vocal in the verse. For the opening song, "Head," we used a flange from the ART unit that sweeps very subtly as Kai's vocal lines develop. A one-octave pitch shift fattens up Kai's verse in the song "Fat." Mixes were mastered by Roger Lian at Masterdisk (NY) and then released by High Strung Productions on CD.

More recently, Tom has been working on a project with vocalist John Grignon called John-Thomas (imagine that!). During the recording of this project, Tom was living up near Albany, NY and the budget was even tighter than that for *Heads or Tails.* Having purchased a Mackie 8•Bus and built a studio, Tom now had the ability to do all of the recording on his own time. The problem (again) was that he only had one DA-88, but needed to record more than eight tracks.

In his studio, Tom has Opcode Studio Vision Pro 3.5 running on a PowerMac 8500/180 MHz with an AudioMedia III card for audio I/O. Recording tracks in Studio Vision and locking the PowerMac to tape would be an option, but Tom does not have SMPTE lockup capability in his studio. And, besides, we wouldn't be able to process the Studio Vision tracks individually, due to the nature of the I/O card (two analog and two digital outs).

Our solution was rather unorthodox. Tom recorded the drums onto "master" DA-88 tapes (a total of two) across eight tracks. Two different drummers played on the John-Thomas project — Pat Knittel and Bob Naperski. The drums were miked mostly with Shure SM57's through a Mackie 8•Bus — the exception being an Audio-Technica AT421 stereo condenser mic for the overheads and an AKG D112 for the kick. Pat's kit had more toms, so one '57 was used for each pair of toms. When drum tracks were completed, Tom made a mix of them and bounced them directly into Studio Vision Pro.

This rough drum mix was then bounced back out to a fresh piece of DA-88 tape, thus creating a slave tape. Tom and John proceeded to add their overdubs to this slave tape, working on



DRUM 'N' RUN: The drums at Tom Atkins's studio were recorded on one DA-88 tape, across all eight tracks.

their finished tracks while monitoring a rough mix of the drums. Bass was recorded direct and guitars were recorded with SM57's on a multi-amp setup (the ADA/Boogie rack system previously described and a Boogie Mark III combo amp). When the recording process was finished, Tom and John brought the tapes and the DA-88 to the Wood Shop for mixing.

LOCK AND LOAD

For this project, we approached the remixes differently — both in technical and creative terms. On the song "Peeping Tom," the bass had actually been doubled. In the mix we would end up running one bass track through an ART Tube MP preamp, overloading the input for a bit of attitude and mixing that in with the "straight" bass track. And on the song "Told You First," John sang some very cool background tracks that we chorused through a TC Electronic M2000 to make the vox stack up fat.

Our technical adventure at mixdown began by connecting the second DA-88 into the studio but *not* locking it to the first. Tom had logged start times for each song, both on the master drum tapes and the slave tapes. We picked a song to mix first, manually cued both tapes to the start of that song, and noted the locations of the start point for each machine. Then we did a bit of math and calculated the difference between the machine locations. The resulting number would be our offset.

There was absolutely no consistent relationship between the location of a song on the master drum tape and that song's location on the slave/overdub tape, so some of the offsets were positive while others were negative. We'd lock the machines together, hit play, and listen to the drums, hearing both the master drum tracks and the slave's rough drums. If there was a delay going on, we knew the offset was wrong. By switching the slave DA-88's display to "offset" we could adjust the offset while the tapes were playing and continue to listen. When we heard a flange between the two machines (or heard the kick drum get thin from phase inversion), we knew the offset was right. Just as a quick note: It's probably not a good idea to do this with one instrument's tracks spread across two machines. If the machines are a hair out of sync (say, a few frames) it will be OK musically, but it would cause definite phase problems between (for example) the kick mic and the overhead mic tracks.

Every song had a count at the beginning, so in some cases, we'd listen to the countoff and hit the MEMO 1 button on the DA-88 at the first note of the song. We'd do this for both machines. By hitting LOC 1, the machines would go to the first note of the song and park. We could then use those locate positions to do our math and figure out a rough offset. Then we'd lock the machines and fine-tune the offset while listening. Once we had the correct offset, the machines were run in the normal fashion and the song was mixed. After doing this a few times, it became more and more easy. We mixed seven songs for the John-Thomas CD in this manner. Although getting the offsets was tricky, the whole thing worked!

In addition to being an audio mercenary, Steve La Cerra is also the senior editor of EQ magazine. You can hear some of the music he discussed in this article at the High Strung Web site: www.highstrungpro.com.

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It's All In the Mix

Mix engineer Andy Johns reveals his secrets for building a time-honored mix

BY BOBBY OWSINSKI

Andy Johns should need no introduction to EQ *readers because we've been listening to the music that he's helped create for most of our lives. With credits such as Led Zeppelin, Free, Traffic, Blind Faith, The Rolling Stones, and, most recently, Van Halen (to name just a few), Andy has set a standard that most mixers are still trying to reach. Following are some of Andy's thoughts on mixing, which, by the way, is excerpted from my upcoming book,* Mixdown.

EQ: Where do you start when you're building a mix?
Johns: I don't build mixes, I just go, "Here it is." [*Laughs heartily.*] Actually, I start with everything. Most of the people that listen to and tweak one instrument at a time get crap. You've just got to go through it with the whole thing [mix] up because every sound affects every other sound. Suppose, for example, you're modifying a 12-string acoustic guitar that's in the rhythm section. If you put it up by itself, you might be tempted to put more bottom on it, but the more bottom you put on it, the more bottom it covers up on something else.

The same thing happens with echo. If you have the drums playing by themselves, you'll hear the echo on them. You put the other instruments in and the echo's gone because the holes are covered up.

Do you have a method for setting levels?
That's all crap. That's rubbish. There was a famous engineer some years ago that said, "I can mix by just looking at the meters." He was obviously an upstart wanker. If you stare at meters long enough, which is what I did for the first 15 years, you find they don't mean anything. It's what's in your soul. You hope that your ears are working with your soul along with your objectivity, but truly you can never be sure.

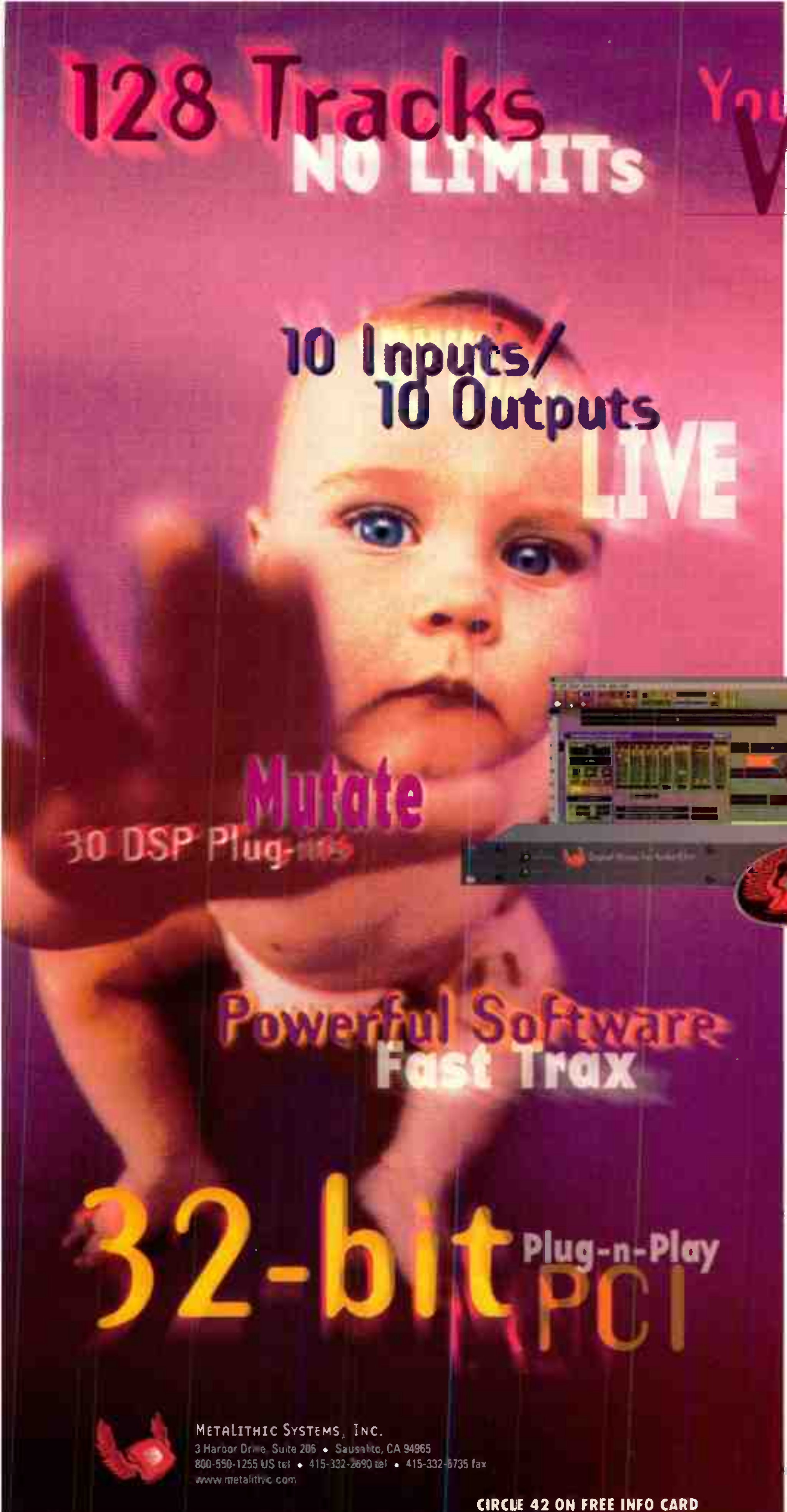
The only way that you can get a proper mix is if you have a hand in the arrangement, because, if you don't, people might play the wrong thing or play in the wrong place. How can you mix that? It's impossible.

The way that I really learned about music is through mixing, because if the bass part is wrong, how can you hold up the bottom end? So you learn how to make the bass player play the right parts so you can actually mix. It's kinda backwards. I've been into other people's control rooms where you see them working on a horn part on its own. And they're playing with the DDLs and echoes and I'm thinking; "What are these people doing?" Because when you put the rest of the tracks up, it's totally different and they think that they can fix it by moving some faders up and down. When that happens, they're screwed. About the only thing that should move is the melody and the occasional other part here and there in support of the melody.

Does the fact that you started by working on 4-track machines affect the way you now work?
Yes, because I learned how to balance things properly to begin with. Nowadays, because you have this luxury of the computer and virtually as many tracks as you want, you don't think that way any more,

PHOTO COURTESY OF THE MICHAEL OCHS ARCHIVES

THE MAN BEHIND THE MIX: Andy Johns is responsible for some of the songs you love from Led Zeppelin (above), The Rolling Stones, Blind Faith, and many others.



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but having to do it that way was a great learning experience.

You know why *Sargent Pepper* sounds so good? Or why *Are You Experienced?* sounds so good? Almost better than what we can do now? Because, when you were doing the 4 to 4 [bouncing down from one 4-track machine to another], you mixed as you went. There was a mix on two tracks of the second 4-track machine and you filled up the open tracks and did the same thing again. Listen to "We Love You." Listen to "Hole In My Shoe" by Traffic. You mixed as you went along; therefore, after you got the sounds that would fit with each other, all you had to do was adjust the melodies.

What's your approach to using EQ?

You don't get your sound out of a console; you get your sound from the room. You choose the right instruments and the right amplifiers for the track. If you have a guitar sound that's not working with the track properly, you don't use EQ to make it work. You choose another guitar and/or amplifier so it fits better in the track. It might take a day and it might take four or five different setups, but in the end you don't have to worry about EQ because you made the right acoustic choices while recording.

With drum sounds, even though where you put the mics is reasonably important, it's the way you make the drums sound in the [tracking] room. The way you tweak them [at the source], that's where the sound comes from. The sounds come from the instrument and not from the mixer. On rare occasion, if you run into real trouble, maybe you can get away with using a bunch of EQ, but you can fiddle for days making something that was wrong in the first place just different.

How do you use compression?

I use compression because it's the only way that you can truly modify a sound. That's because whatever the most predominant frequency is, the more you compress it the more predominate that frequency will be. Suppose the predominant frequencies are 1 to 3k. Put a compressor on it and the bottom end goes away, the top end disappears and you're left with *"Ehhhhh"* [makes a nasal sound]. So for me, compressors can modify the sound more than anything else can. If it's a bass guitar, you put the compressor before your EQ because if you do it the other way around, you'll lose the top and mids when the compressor emphasizes the spot that you EQ'd. If you compress it first, then add bottom, then you're gonna hear it better.

What are your listening levels?

If I'm listening on small speakers, I've got to turn them up to where they're at the threshold of breaking up but without any distortion, or I listen very quietly. If you turn it way down low, you can hear everything much better. If you turn it as far as it will go before the speakers freak out, then it pumps. I can't do it in the middle. It's just not rock 'n' roll to me.

Got any listening tricks?

Obviously the idea is to make a mix work on all systems. You listen on the big speakers, the NS10's, out in the car, plus your own speakers, then you go home and listen again. This is a lot of work, but it's the only way to go.

I tend to bring JBL model 4310's, 12's, 13's, and 12A's, and I put those out in the actual studio. But you know, I don't care how close you think you've got it that night,

you take it home and play it back in the morning, and every time there are two or three things that you must fix. It's never happened to me where I've come home and said, "That's it." You hear it at home and you jump back down to the studio and sure enough, you hear what you hadn't noticed before on all the systems there as well. So every system you listen on, the more information you get. You can even turn up the little speaker in the Studer [tape machine] to hear if your mix will work in mono.

Do you listen in mono much?

No, but I'll tell you this. If you've got a fantastic stereo mix, it will work in mono as well. For example, "Jumpin' Jack Flash" is a stereo mix released in mono. People don't listen in mono any more, but that used to be the big test. It was harder to do and you had to be a bloody expert to make it work. In the old days we did mono mixes first, then did a quick one for stereo. We'd spend eight hours on the mono mix and half an hour on the stereo.

When do you add effects in the mix?

I have some standard things that I do that more or less always work. I always need a great plate like an EMT 140 and a short 25 to 32 ms delay just in back of the vocal. If it's kind of a mid tempo tune, then I'll use a longer delay that you don't hear because it's subliminal. It doesn't always have to be timed to the track; sometimes it can go in the hole so you can hear it. I've been talked out of putting reverb on electric guitars, but "Start Me Up" has a gorgeous EMT 140 plate on it. Most studios you go into don't even have one anymore.

So you usually predelay the plate?

Usually, but not always. In the old days like on the Zeppelin stuff, you'll hear very long predelays on vocals. You know what that was? That was a 3M tape machine, which, because it originally was designed to do video, had about a 9-inch gap between the heads as opposed to the 2 1/4-inch-gap on a Studer or Ampex. Sometimes I'd even put it at 7 1/2 ips. Another thing we used was the old Binson Echorec. Listen to "When the Levee Breaks." That was me putting two M160's on the second floor with no other microphones at all because I wanted to capture [drummer] John Bonham the way he actually sounded, and it worked. Jimmy Page would say that he made me do it, but he was down at the pub. He did, though, bring me his Binson Echorec for the track.

Do you prefer analog or digital?

What I like is the sound that's coming into the mixer. I don't want it modified by some tape machine. I've always fought with analog. I've always fought with vinyl. With digital, the sound that's coming in, you get it back. It's much truer than any analog machine ever was. If you've got to smooth out your sound with some analog machine, then you're in trouble to start with. With analog, the noise factor is like a security blanket in that the hiss can cover up some weasely things.

Which automation system do you use, or do prefer to mix manually?

They're all shitty because you're fighting a machine. I suppose the GML is the easiest, but I still have to have somebody there with me to help. That's the part of the job that pisses me off. You've now got to be a bloody scientist. Sometimes it makes you too clever for your own good. If you just learn the tune, then you're in tune with the tune. You let it flow through you. Still, though, you might listen to it years later and say, "I think I missed that one." Or, you might go, "Fucking hell, I wish I was that guy again. That could not be any better. Who was that man...?" EQ

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CIRCLE 82 ON FREE INFO CARD

Laid-Back Multitrack

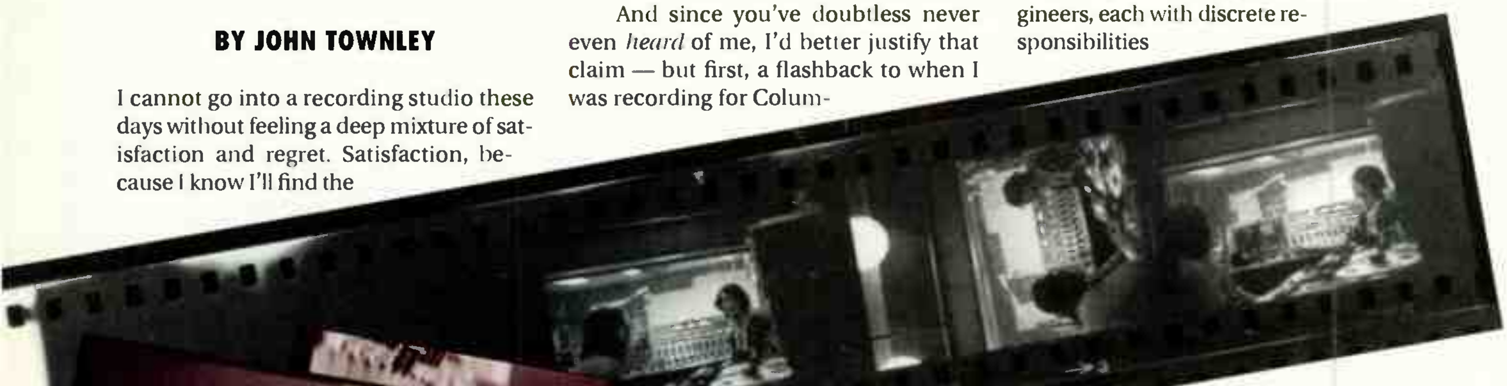
The downtown birth of modern recording studio style

BY JOHN TOWNLEY

I cannot go into a recording studio these days without feeling a deep mixture of satisfaction and regret. Satisfaction, because I know I'll find the kind of equipment I need and the atmosphere of creativity I want to get the job done. Regret, that you can't copyright an idea — otherwise, I'd be rich, since the marriage of equipment and atmosphere is something I invented way back when it all started...

And since you've doubtless never even *heard* of me, I'd better justify that claim — but first, a flashback to when I was recording for Columbia back in '65 with a New York group called the Magicians, an act that almost made it, but...well, our producer Charlie Koppelman had fatter fishes to fry, as did Columbia's Goddard Lieberson. Roy Halee was our engineer (producer/engineer for Simon and Garfunkel), and with the exception of his incredible talent, recording was about as unpleasant a chore as you could imagine. Not only was it user-unfriendly, it was ludicrous. The uptown Columbia studio itself was hideously brightly lit, so every smudge on the dirty, pegboard-lined walls stood out and erased whatever chemistry you might have prepared on the way in. Definitely several tokes under the line...the very foundation for justifiable musician paranoia, because...

There were only eight tracks to work with (you wonder how the Beatles survived so well on that — lots of money/time helped), and there was no separate cue system for overdubbing (honest!), so you listened to whatever mix the engineer needed to do his trip. And it was a union shop, with three separate engineers, each with discrete responsibilities with which no one, but no one, was allowed to interfere. So, doing a take was kind of like a WWII submarine movie: "Take one!" cries Roy. "Tape is rolling," informs the man in charge of starting the 8-track, in an entirely separate room. "Echo on!" finishes the third in a yet more distant room, punching on the 2-track tape delay. After that, you got to start playing — all that was missing was the wait to hear if the torpedo had found its mark.



Mixing was equally as crazy, as you still had all three to deal with, and you were never — I mean *never* — allowed to touch a fader, lest you get your fingers slapped. You had to tell each engineer what you wanted and hope he understood. All the major record company studios were like that, and independent studios weren't much different.

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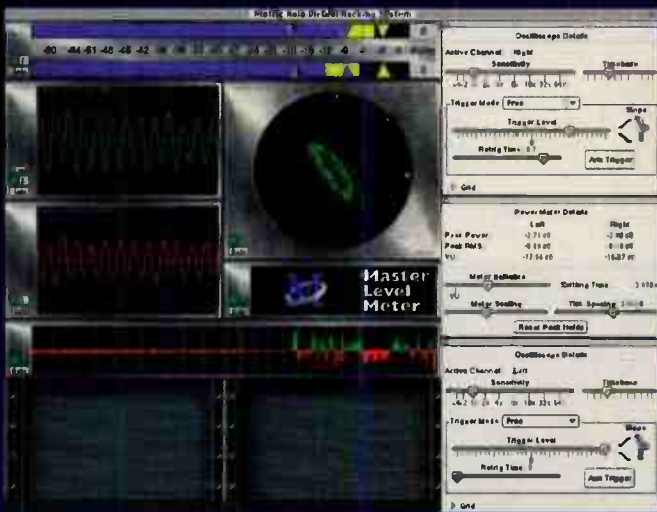

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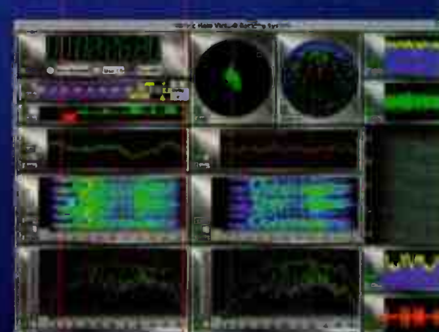

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CIRCLE 46 ON FREE INFO CARD

By the fall of '66, I'd had it with group, recording, and first marriage, so I took some money from a small inheritance and idealistically launched out to change it all. What was amazing — I did!

First, I picked a downtown location in the Village, as that's where all the musicians were. Be where you're comfortable. After all, only the execs were uptown. It was a loft building on 10th St. near Broadway with a hand-operated freight elevator its only access, one where the elevator had no walls and worked by pulling a cable to make it start and stop. That was a golden opening opportunity — our artist/elevator man Nicky Osborne soon had the entire darkened six-story shaft detailed in black-light psychedelic murals, and his personal welcome in full Viking costume definitely made a serious start on the way up. Way up. We often wondered what the clients of the very-tolerant baseball cap sales company on the 6th floor thought of it.

The rest of the studio followed suit, with totally controllable theater lighting throughout, so whatever the state of your insides, you could make the environment match. "No smoking" signs in every respect were a thing of the future, indeed.

The equipment matched the decor, user-friendly to the possible max. I insisted on the first independent cue system ever on any mixing console, (built by Lou Lindauer, his Automated Processes' opening gambit, who brilliantly nursed us through our demanding technology changes). And, of course, more tracks (how quickly you run out at only eight) — we

HISTORIC HARDWARE IN SOFT FOCUS

By Richard Kunc, former Apostolic Studios engineer

I recall the API console as a sea of blue Formica, a wondrous machine praised by Frank Zappa, with sparkling new arc-shaped British Painton faders. Sort of "proto-faders," actually, the Painton controlled a linear series of many individual make-and-break contacts. On very quiet passages you could actually hear the tiny "bip-bip-bip" as it went from contact to contact.

Each input position had rudimentary equalization available, but it amounted to little more than glorified bass and treble controls. We had a pair of Lang equalizers mounted externally to help the cause. The input positions were normalled to their corresponding tape tracks, but you could reassign any input to any tape track via the patchbay. For mixing, you could also assign each input position to either left, center, or right. These were hard switch selections. There were no pan pots on the input positions. You could patch into the console's two independent pan pots but that was it.

Three Melchor compressors did give us a smidgen of ceiling control in extreme cases. They were better known for their pronounced and dramatic "breathing" effects in which the momentarily suppressed background sound comes rushing back up in volume after the peak has passed. Also external to the console was a very fast (for its day) limiter that used a light source coupled to an optical detector to do its work.

The prototype Scully 12-track machine used 1-inch tape. It had twelve sets of their normal full-size rack-mounted electronics, the ones they put in their mono and 2-track machines — imagine twelve of those babies, each one with a complete set of knobs and full-size meter! It was just huge — but it worked. One problem with the machine was bleed-over while recording on adjacent tracks, so we'd record on odd-numbered tracks during the first pass, and then overdub in between the initial tracks from then on.

The machine had one neat trick, though. You could take a 1-inch tape with eight tracks recorded on it by an 8-track machine, put it on our 12-track machine, and add four more tracks! You ended up with twelve very mixable tracks, all of which still had very acceptable signal to noise ratios, and it made us completely compatible with other 8-track studios.

The primary "echo chamber" was one of those "state-of-the-art" EMT vibrating steel plate deals. Inside a huge wooden case a sort of loudspeaker was attached to one corner of a metal plate that must have measured maybe four feet by eight feet. When you fed a signal to this "speaker" it sent waves through the plate, sort of like ripples on a pond. A transducer at the opposite corner picked up these waves, equalized them, and sent them back as echo. There were also some experiments with live speakers and microphones in the stairwell of Apostolic's ancient building, much to the dismay of the residents of the other floors, and whose complaints effectively squelched our efforts.

What gave the console its saving-grace versatility was its extremely comprehensive patchbay. It was the old tip-ring-sleeve variety, left over from telephone company technology, though more than one unrepeatable take was destroyed by some bit of studio dust that got in there...

And for microphones — a few Neumann U-67 condenser mics with their accompanying power supplies, a few borrowed Sennheiser ribbon mics, a smattering of assorted decent dynamics, including those warm and indestructible cone-headed D-202s, and a motley gang of one-of-a-kind items including an indestructible Altec "salt shaker" routinely used for drums and a giant, mellow vintage RCA mike that in its youth must have recorded Bing Crosby or the Andrews Sisters.

In the control room were large hard-edged Altec monitor speakers, located behind the console. Plus the favored (by us) KLH-6's up front that blew out all the time when the clients said, "Louder," not to mention delightful Bose clones built by Gus Andrews, our first reggae recording artist.

Apostolic was a real Viking ship, a gutsy voyage into the uncharted depths of new toys and new ideas from which came some of the cleanest recordings of that era. I'd give a lot to sit again at that console and look through the glass at those original Mothers of Invention. Rest well, Apostolic.

Richard Kunc graduated from Apostolic to become director of recording for Frank Zappa's various record companies. He now heads a marketing agency specializing in aviation and is the author of many articles on recording, flying, and electronics.

went to twelve because that was as many as Scully could crowd onto a relatively editable 1-inch tape transport in this first-of-a-kind machine they built to our specs. And faders, instead of pots (goodbye '50's sci-fi flicks), also a first, and the newfangled solenoids...best of all, everyone could touch them all, no union, except don't spill your Coke on the board! All that, and teenage whiz engineer Tony Bongiovi (early producer of Jon Bon Jovi and past owner of the now-defunct Power Station) were enough to launch us into the ozone.

When we opened in the spring of '67,

everyone in "the biz" said we were crazy: no one would come downtown, nobody needed twelve tracks, and our whole style was *very* unbusinesslike...our name was Apostolic Studios, after our twelve tracks and unabashedly spiritual (though not particularly Christian) tilt, and clearly we were nuts.

Well, inside three months we were booked solid. The Critters, Spanky And Our Gang, the Serendipity Singers, and, most of all, Frank Zappa and the Mothers Of Invention found the qualities of user-friendly tech and musician-friendly ambience to be just what the doctor ordered. Six months after that, Gary Kellgren, who had made an early reconnaissance visit to Apostolic, opened Record Plant with an identical 12-track Scully and similar board and ambience, and shortly thereafter Jimi Hendrix built Electric Ladyland downtown on 8th St., just blocks away.

Flushed with success, Apostolic went on to open another 12-track operation (Pacific High in San Francisco, the first on the West Coast), to become a record/publishing/management company (Larry Coryell, Jim Pepper, and the like), and to see our "Witchi-Tai-To" Cherokee trance

continued on page 140

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CIRCLE 55 ON FREE INFO CARD

Dylan's State **of Mind**

Daniel Lanois on producing Bob Dylan's Grammy-winning *Time Out Of Mind*

BY ALAN DI PERNA

"It's mostly about darkness — just those deep tones that I love."

Daniel Lanois is talking about his knack for crafting evocative sonic ambiences. It's a thread that runs throughout his work as a producer, engineer, guitarist, film composer, and solo artist. Lanois started out working with ambient music pioneers like Brian and Roger Eno, Harold Budd, and Jon Hassell. "Those lessons have always stayed with me," he says. "I don't know if I consciously use any specific techniques from those days, but I do have a love for those records. Those inspirations are still there."

Those ambient inspirations have echoed throughout a career that encompasses two solo albums, film projects like *Sling Blade* and *The Last of the Mohicans*, and powerhouse album productions for U2, Peter Gabriel, Robbie Robertson, the Neville Brothers, EmmyLou Harris, Luscious Jackson, and others. Lanois first worked with Bob Dylan in 1989, when he produced the latter's *Oh Mercy* album. They reunited to record Dylan's 1997 masterpiece *Time Out Of Mind*. Lanois's moody sonic aesthetic was perfectly attuned to Dylan's elliptical, world-weary songs. *Time Out of Mind* entered the charts at number 10 when it was released in October of last year. It topped numerous critics' polls and has received the Album of the Year Grammy Award. Fresh from his collaboration with American's premier songsmith, Lanois can remember the *Time Out Of Mind* project in all its glorious technical detail.

Is Bob Dylan's voice a difficult one to capture on tape?

Lanois: Not at all. He has this amazing punch and midrange presence in his voice. Barking range. You can hear it in Bob's voice when he's speaking in the room. His voice has a lot of timbre. His expression ranges considerably when you study his body of work, from the '60s to now. The vocal sound on *Oh Mercy* is really different than what we did on *Time Out of Mind*. Most of the vocals are live vocals on *Time Out of Mind*. Bob was in the corner [of the studio]. And all the musicians were in a horseshoe around him. So you have the advantage of the band spilling into the vocal mic, which can be a very exciting sound. Especially if you use quite a bit of compression on the mic, as we did.

This is one of the most processed vocal sounds he's ever done on record.

The way we did it was we would record a processed vocal on one track and a clean vocal on a separate track. And then we would blend them to taste during the mix. So the processing often went down to tape while Bob was performing. The processing would help shape his vocal performance. It's kind of the equivalent of a Chicago blues harp, where you overdrive it. It just gives you a kind of sound that inspires you to play or sing in a certain manner. We tried to encourage that as much as possible.

PHOTOS BY MATTHEW WELCH/MONTAGE

CAT IN THE HAT: Daniel Lanois reveals his recording techniques for capturing Dylan's unique vocal stylings.

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CIRCLE 37 ON FREE INFO CARD

What kind of gear were you using on his voice?

The vocal processing varies from track to track. "Love Sick," for example, is a totally modern bit of processing. I think it was an Eventide 3500 with some unlikely kind of stereo phasing sound. If someone were to describe that to me I would say, "Well, I don't think you should use that on Bob's voice." But it happened to come up and there was something cool and "transmissional" about it. It almost sounds like it was being beamed in from a satellite or something. Some of the other vocal sounds were more crude — piping the vocals back through a PA and remiking.

It sounds like slap echo, maybe, on "Standing in the Doorway," "Million Miles," and some others.

Yeah, we used what we call the "Elvis echo" on a few tracks, which Bob likes. That '50s, sound, you know?

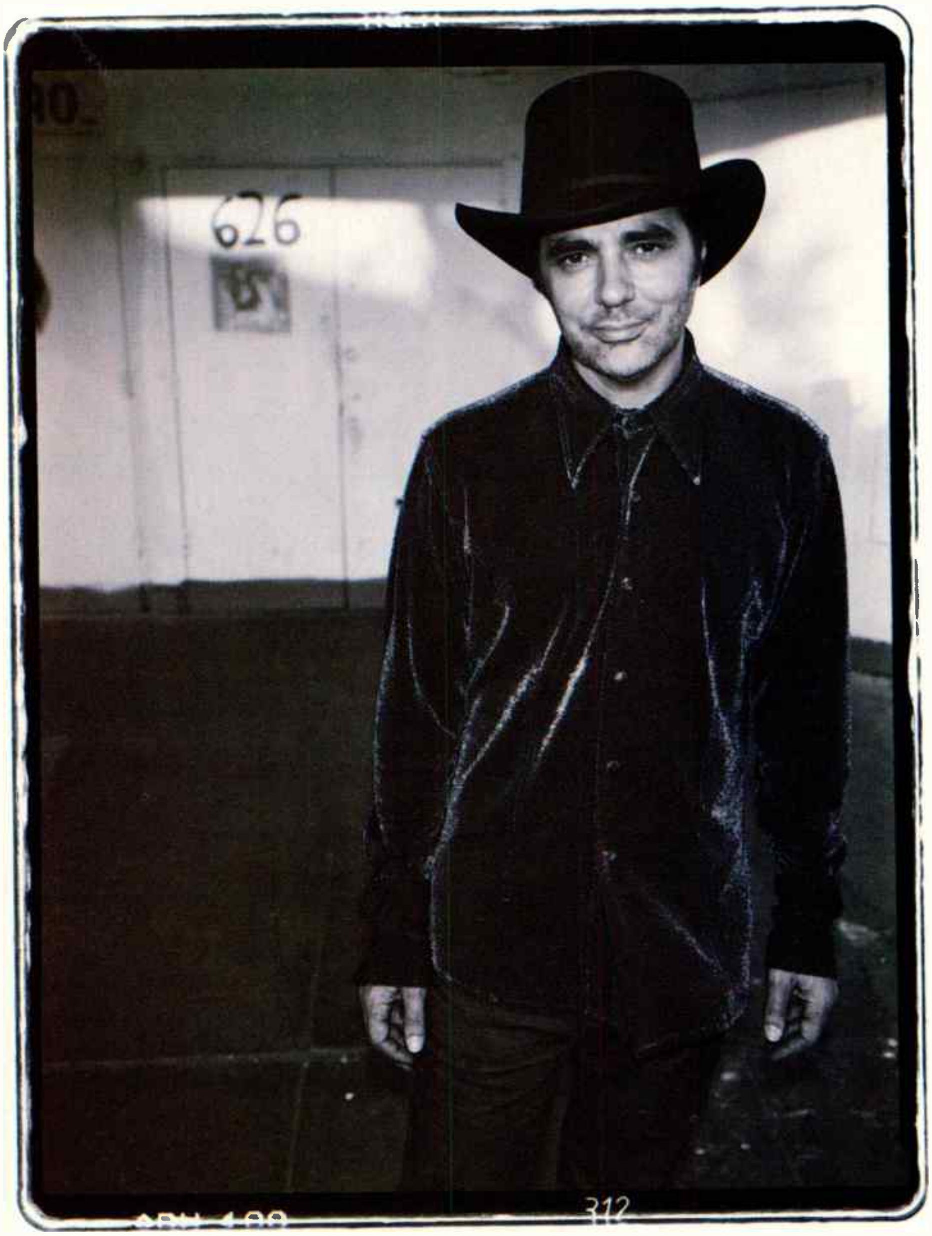
Actual tape echo?

No, we never actually rolled a tape. I think it was just an AMS harmonizer™ with about an 180 millisecond delay on it. A 180 millisecond delay is a mimic of a 7-1/2 ips on an old Ampex machine from the '50s. The exact delay time depends on the width of the heads and that stuff, but it's right around 180.

And you said you used a lot of compression on the vocals.

Quite a bit. The mic was the very same one that I also used on *Oh Mercy*: a Sony C37A. It still had Bob's name on it. A great mic. It doesn't have a lot of high frequency, but it has a lot of body. We technical people often forget, in our quest for high frequency, that there's a lot to be had out of low mids. That microphone has that built in. And the compressor is also the same one we used on Bob on *Oh Mercy*: an LA-2A. I should mention that Mark Howard, the engineer, did a lot of great work on Bob's vocals.

And I should explain that when Bob would do his vocal, we would take him out of the monitors. He was singing acoustically in the room. He was not using any headphones or any PA, other than the overdrive through the PA stack that was sitting in the corner. This was in case he wanted to change a lyric line, which he often does. He overwrites, in the sense that he'll generally have more lyrics than what ultimately ends up on the record. Often he'll say, "Let me try this other vocal line." So we had a pair of stereo speakers out in the room and we'd pipe all the music back in the room while Bob was singing the lines he wanted to punch in. This would simulate the band being present in the room, so that the spill into the vocal overdub line would match the live performance. If you've recorded a live performance vocal and you try to do an overdub with headphones, it's never gonna match.



The overall sonic ambience of *Time Out of Mind* is so distinctive. It reverses the usual notions of background and foreground. Was that kind of stuff discussed with Bob right at the start of the project?

Bob has a fascination with records from the '40s and '50s and even further back. We listened to some of these old recordings just to see what it was about them that made them compelling. I concluded that what was interesting about those records is what I'm calling depth of field, with something very much in the foreground, something else further back, and something else further back than that. Obviously that depth of field wasn't artificially created. They didn't have multitracking back in those days, so I can only assume that those results were arrived at by placement of people and microphones in the room. We decided this was something to pursue. I knew Bob had a feeling for that sort of thing.

What was your approach like to recording guitars?

For Bob, we used this little baby Martin acoustic of mine: a small bodied acoustic from the '30s, with a Lawrence pickup which just snaps in the soundhole. And we put that into a late '50s tweed Fender Deluxe and miked that with a dynamic mic: a Sennheiser 409. The reasons for that, rather than just miking the guitar acoustically in the room, is that it solves a lot of isolation problems. If Bob needs to change a lyric line later, we don't have the old lyric still in the acoustic guitar mic. We put the amp near him, behind a little gobo wall, and mic it. We did the same thing on *Oh Mercy*. It still sounds like an acoustic guitar, only it's got a little more life. You can't miss with those old tweed Deluxes. Neil Young uses those for some of his guitar sounds. I use them all the time.

What about your own guitar contributions to the album?

I play guitar on all the tracks. My rig is really simple: a '50s gold top Les Paul with a single coil P-90 pickup through a Vox AC30. I would sit as close as I could to Bob and we'd bash it out. My tone is very dark, which I like a lot. And the same type of microphone was used on my amp as Bob's: a Sennheiser 409.

Did the other guitar players, like Duke Robillard, overdub their parts later on, or did they play live, too?

Duke was only there for a few days. When he came, we just set him up in the room. We put him in a chair, stuck his amp there, and he played along with the rest of the band.

Do you have any general favorite microphones for guitar?

Dynamic mics sound best for close miking. I think the currently made dynamic mics are terrific: Shure, Beyer, and, of course, the Sennheiser 409. And for further back I use all the classic old tube mics: Neumann U47's and a couple of great AKG C24's that I have. If you're having "thin sound" problems with guitar, there's an excellent ribbon mic called the Coles that will get you out of trouble. And some of the old RCA ribbon mics will solve the thin problem. They have a lot of body, those ribbon mics. I used the Coles for the bass drum on Bob's record.

Back from the head a little?

Yeah, a couple of feet from the drum. A drummer like Brian Blade, who played on the record, has an amazing bass drum sound. That kind of microphone does his tone a lot of justice.

You get really organic drum sounds that fit nicely into mixes.

Well, it's not that mysterious. We just didn't mic the drums too close. There are times when I love close-miked drum sounds, like on some of my favorite reggae records. But those sounds did not have a lot to do with ambience; they were more about hihats right in your face. But if you're trying to get an organic sound, like a jazz drum sound, distant miking is your friend. On Bob's record the snare and hihat were kind of treated as one and miked with a U47 a few feet away.

Were you doing a lot of ambient miking of the room itself to get some of those depth-of-field type of effects?

We didn't do a whole bunch of ambient miking, because you get a lot of that naturally if you've got a live vocal and the drums are near the vocal. A lot of times we were using two drum kits, with overheads on those, so there were a lot of microphones in the room. We didn't have to deliberately hang ambient mics up high.

Is it the panning that helps contribute to that unique sense of sonic space on the record? It seems like there's lots of extreme side information.

Well, the two drum kits were pretty widely panned, so anything bleeding into those mics would also be panned.

A lot of the organ and guitar parts, in particular, seem to be living at the extreme left and right edges of the mix.

In the case of the organ, it was pretty isolated. Even though it was screaming in the room, we had the Leslie stuck in a little booth. So we were able to pan that pretty radically. For that matter, the electric guitars were tight miked, too. So even though they were spilling into the vocals and drums, we had access to pretty wide panning.

The record was mixed at your own studio in Oxnard, CA?

You know, a lot of the rough mixes that were done in Miami turned out to be the finals. The balance of a rough mix has a practical reason for being the way it is. Let's say you're fine tuning the vocal and you have a mix up. That balance exists because you're trying to make the singer as comfortable as possible. So it's not like some kind of mix fabrication. It's actually up for a practical purpose. Often, those blends are highly musical. I've noticed that on many records. And Bob made the decision that he really liked some of the Miami rough mixes and they're the ones that got on the record. There's often an excitement in those early rough mixes that goes beyond explanation. It's spirit of the moment stuff.

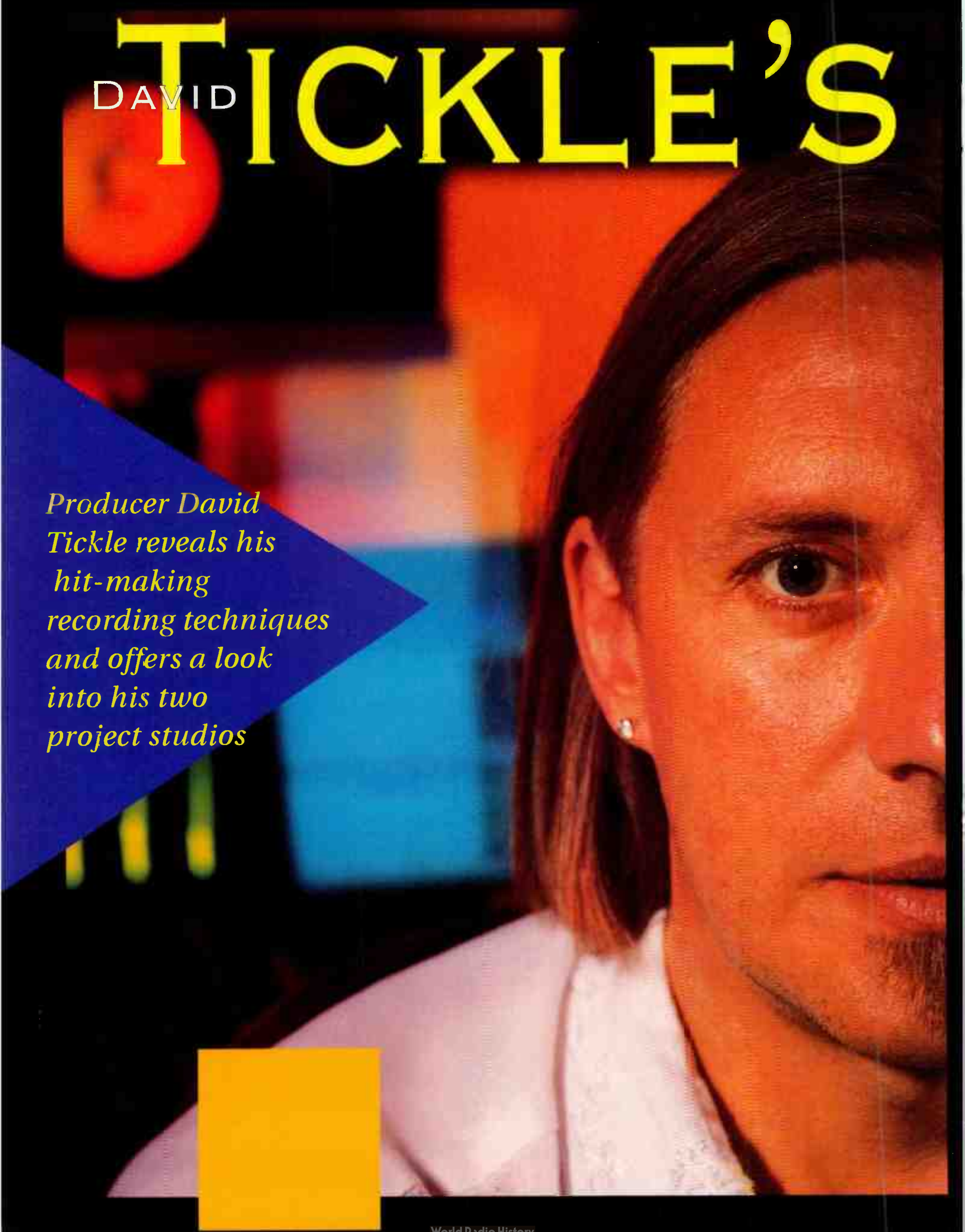
And then some of the other tracks were really worked on in Oxnard, after Miami. For example, "Can't Wait" was radically modified. Bob decided that when the song went to the IV chord, he wanted it to be minor rather than major. So I had to replay all the instruments with that chord change, including Augie's [Meyers' vintage Vox] organ. I punched in all the parts. It was too late to call the players back. Everybody was scattered all over the map.

Wow, punching in matching the tone of the original!

Yeah, it was wild. But I think, as a result, it has a certain kind of sound. EQ

DAVID TICKLE'S

Producer David Tickle reveals his hit-making recording techniques and offers a look into his two project studios



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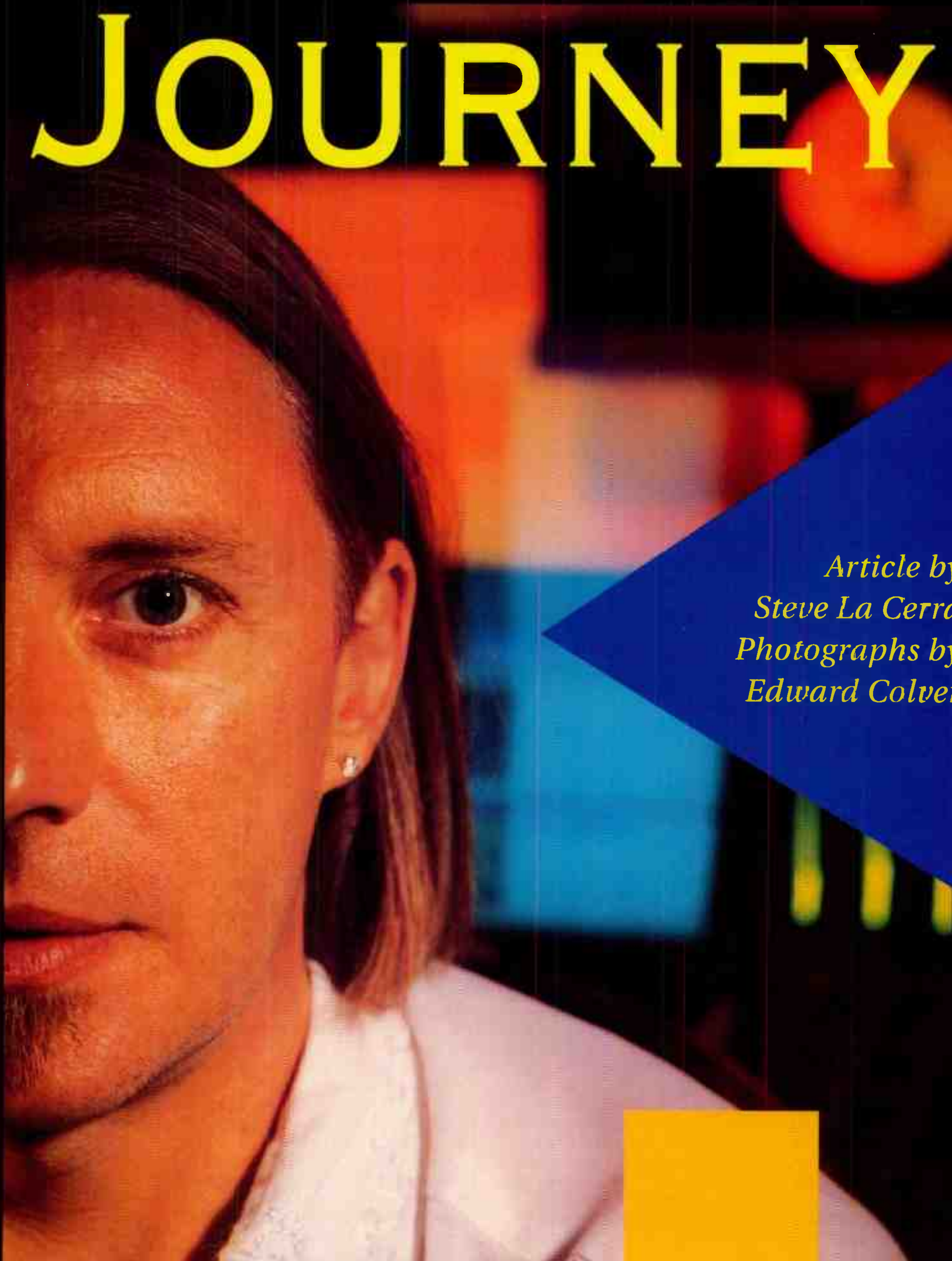

Article by
Steve La Cerra
Photographs by
Edward Colver

Tickle's Journey

As a producer and engineer, David Tickle is in the midst of an incredibly varied and exciting career. From his work with artists like Blondie and Split Enz to U2 and Joe Cocker, from "The Artist" to Four Non-Blondes, David has quietly been making a deep mark on the modern music scene — not to mention what he's done in 5.1 surround. David recently spoke to EQ *regarding his production philosophy and techniques, as well as his latest endeavor to form his own record label.*

Having produced a variety of solo artists, as well as bands, what do you feel are the necessary differences in the production approach?

There is definitely going to be a different approach with each; solo artists often have a clear idea of what they want to achieve. It can be very exciting in its own way, going out to cast the project with the talent you feel is most appropriate for achieving those goals. But the other dynamic is when you have a "cast of characters" that come with the package known as a band. That's what makes them distinctive — every band is different and has its own dynamic and personality.

What was your first production?

Split Enz was the first band I produced. When I began working with this band, I had already made quite a few successful records, working with Mike Chapman and engineering for Blondie, The Knack, and other artists. Split Enz' *True Colors* was my first production. The dynamic of that band was incredible. There was a tension between Neil (Finn, vocalist and guitarist) and Tim (Finn, lead singer and songwriter). But they're also loving brothers who wanted to create music within the same band. Neil was the newest member when I started working with the band; *True Colors* was the second record in which Neil had been involved.

Both Neil and Tim wrote songs and sang, so there were these two personalities within the band, going on record. Each would support the other with harmony vocals, and the band had a richness because of the two lead vocals. I remember one song called "What's The Matter With You?" (written by Neil), which didn't sound super-exciting. Tim suggested the idea of himself singing the verses while Neil sang the choruses, for a sort of duet. We thought it was a good idea, so we tried it out. What happened was that Neil and Tim were singing very much in the comfortable range of their vocal abilities. But with the lyric being, "what's the matter with you?" and the whole vibe of the song, I decided to switch them around so that each sang the opposite part — Neil sang the verses and Tim sang the choruses. With the key that the song was in, this became a strain for each of them.

And you knew that would be the case...

Absolutely. It was a deliberate move. It added this tension to the vocals because they were both pushing harder. It had all this excitement and it was thrilling — having that kind of vocal depth in the band.

For this particular production of *True Colors* I would listen to the songs and lyrics to get an overall sense of what was being evoked. Then I'd basically storyboard the songs to come up with a visual concept of what I was seeing in the music. The keyboards were especially easy to ma-

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Looking back over the projects I've done, I find that working with a solo artist can be a big relief if you are working with a lot of bands

nipulate, and Eddie (Rayner, keyboardist) could create any sound. We were using synthesizers such as the Prophet, which offered a lot of control to really paint colors and textures with the keyboards. An example would be in the song "I Got You." In the verses there is this sort of ominous keyboard tone, sort of an "eoowwwww" from the Prophet — a very distinctive sound. Behind that in the verses I wanted some kind of pad, but I wanted it to have an "orangey," misty feel. The other keyboards and guitars in the verses had this sort of "purpley" tone. I was looking for a pad almost like a fog with an orange hue. It was the last element that we added to the song, and when we did, everyone just went "wow." It glued the song together. The Enz' were a really interesting band with an outstanding dynamic and I really enjoyed it. I never worked with a band that had quite that kind of chemistry.

"Coral Sea" (an instrumental from that record) is another example of that production technique. It started as a jam and we were trying to figure out our approach. We took a break one evening and on the TV came Jacques Cousteau with all of this wonderful underwater photography. The sound was off and I was watching the screen. I played the track and all of a sudden I realized that "Coral Sea" could be a perfect soundtrack for the video. We did all the overdubs as schools of fish that move past from left to right — we thought of the various instruments as different water creatures moving through the whole track. I think it goes back to when I was seven and my father took me to see *Fantasia*. That was probably one of the biggest creative openings for me, listening to the orchestra and seeing pictures come to life with the sound. As soon as that happened, I had this understanding of music. From there on I have always visualized what music is saying to me.

Visualization became a cornerstone for your productions...

It did and still is.

Do you prefer the dynamic of the band situation over the solo artist?

Looking back over the projects I've done, I find that working with a solo artist can be a big relief if you are working with a lot of bands consistently, but I really enjoy bands. There's something about the synergy of people who have lived together, worked together, and maybe been on the road together; something about all of them owning the music. As you approach the recording, you can help pull out and direct the parts and ideas that the band comes up with. I can *identify* what needs to be put into a song, but I try as much as possible to inspire the *band* to come up with the solution.

Quite often it might be a matter of separating who is playing what at the moment, so that everyone has a focused part. Collectively the parts must work together. If the band revises their parts — depending on their talent, likes, and dislikes — they will come up with their version of what that means. One of those versions will be "the one," but they will own it because it came from within the band, as opposed to me specifying the solution. I may say, "OK, we need to do something about this part," but I won't say, "You need to go to a B-flat minor here." I prefer to explore it with the musician. Otherwise, I'd end up as a musician with a certain set of formulas, whereas it's great to be fresh and open to an accident. Working with a band, there's a lot of that type of interaction.

Take for example, a band like Four Non-Blondes. When I went to see them on stage in San Francisco, Linda (Perry, lead vocalist) had a ton of charisma. I really enjoyed their songs and when "What's Up?" was played, it was a phenomena that I hadn't seen very often with unsigned bands — the entire place was singing the lyrics. I knew it was a very special song. But the level of musicianship had some weaknesses, so I tried to spend several weeks rehearsing the band rather than going straight into the studio. I wanted to get the best feel and performances that we could when we went into the studio.

Unfortunately one of the band members, Shana (the guitar player) started to freeze up in the studio. I did my best to encourage her and support her during the tracking process, but it got to the point where I had to get the tracking done, everyone else was performing great, but the guitar was quite lacking. I continued tracking the album and spent three weeks just doing guitars over tracks that took ten days for the entire album. After the three weeks of really solid work, I sat back to review the guitar work and it was just not coming up to the standard. It was really painful because you have a responsibility to make a great record for the band while also delivering to the record company.

After the three weeks, I had to cut her loose and bring in someone else who turned out to be Louis Matoya. This was tricky because I had to cast someone that wasn't coming in as a session player and some of the players who came for a try were very session-oriented. The project really needed to have the band vibe or it wouldn't work. Luckily in Louis we found that vibe. When he came in and played some parts on our tracks, you could hear that the marriage and energy were there. All of a sudden there was excitement in the room and spirits were lifting. In about eight days, we pretty much had the album done. Having the right person was a dramatic difference.

Reflecting back on it, one of the reasons the band chose me to produce was that most producers who had seen the band had made the point of lack of musicianship, but their overall feeling was to let go of the drummer and guitar players and bring in session players. That scared the band and was one of the things I didn't do. I pointed out what was

You walk into some studios and see that huge SSL. Doesn't it look impressive? But the fact is that if you are in there all day, it can be wearing on you.

lacking but also came up with a solution to overcome that problem. I encouraged the musicians and wasn't about to toss them all away. That's the kind of depth I like to go to, because great careers can come out of great records. An example is Pete Best and The Beatles-syndrome. If these people came together as a unit, you have to try to maintain that as much as possible.

The last time we spoke you were in the process of building a new studio...

I have a studio in Los Angeles that's been up and running for about eight years now, called the Journey Room. Four years ago, we upgraded with a Euphonix console and added digital recording. Currently I'm in the process of building a studio in Hawaii because I'm moving there. It's a year away from being completed, but I am really excited. New gear is getting to the point where it's actually doing what it's supposed to do. For example, the new Ramsa console is digital, 24-bit, accommodates 5.1, and has moving faders and dynamics on every channel. Now I haven't heard it, but they are saying it's going to be available for under $5000.

In designing the Journey Room and my new place, I paid close attention to the atmosphere of the room. Of course, I tried to make it work well sonically — it has to have those characteristics — but the room becomes more important than just the equipment in it. You walk into some studios and see that huge SSL. Doesn't it look impressive? But the fact is that if you are in there all day, it can be wearing on you. There's no windows, a hard floor, and air conditioning. It's a very sanitized environment. Not an organic place. I have gotten quite allergic to these sanitized kinds of places!

I really like openness, and Peter Gabriel's studio was an inspiration for that. My designs aren't quite so huge as his, but have similar ideas, allowing in light. You have a sense of the time of day and I find it really stimulates your work. If we're cutting a new song, it's the morning when I get serious work done and get everything sorted out. Everyone rehearses and gets the song down. In the early afternoon, we have the arrangement and cut the tracks. After a break around 6 or 7 for dinner, it's evening and the atmosphere is totally different. You might overdub or fix parts, but it becomes more cerebral as far as the exploration of sound. And yet in the day, I feel very industrious — I want to get in there and get stuff done.

Do you have natural light coming in both the Journey Room and the new room?

Yes.

Did it present any sonic challenges?

No. I even have sliding glass doors on each side of the control room, so you can exit to the outside. The control room is about 21 x 22 feet and the walls are slightly flared towards the back of the room, so that there's no parallel surfaces, as such. I do have theater curtains in there that can be pulled if I want to keep it dead. You'll notice that 90 percent of studios have a hard floor surface where the console is, because you have these consoles that are 12 or 14 feet long. You need the hard floor to roll your chair around just to work the console. Then there's a lot of trapping around the room and in the ceiling. I have a cedar, pitched roof, 14-foot high in the middle, and the floor is carpeted. With the Euphonix console, I don't have to move the chair all over the place so the carpet works. I have some trapping on the back wall and an oversized couch.

Yes, the couch is a very comfortable place to sit, but in fact it's the bass trap, using the same volume of material in the couch that you might use in wall trappings. It's not just what the sound hits, but what's in the room. If something is in the room, it will alter and absorb sound. The fact

The couch is a very comfortable place to sit, but in fact it's the bass trap, using the same material that is used in wall trappings.

that there are theater curtains in there means they are absorbing a certain amount of sound.

Regardless of whether they are drawn or not.

Exactly. If you clap your hands in the room, you hear a certain liveness. You wouldn't hear a ping, but a liveness. If you laugh or raise your voice, you hear it becoming amplified like you would in your living room. I am listening in a more natural environment. When I take my mixes out, they are so much closer than other places I have worked, because I am mixing in a room that sounds closer to a normal living room.

Do you take the same approach with the studio room?

Again, I prefer to be unorthodox. I think it is a good idea to have a controlled area for doing vocals, certain types of percussion or solo instruments where you want it relatively dead, whether it's through trapping, curtains, or whatever. Ten by eight feet is plenty big. As for recording guitars and drums, the more perfect the tone of the room, the more I find that the instruments become very clear-sounding, but have less personality.

The studio I have is a little different. I have a 9-foot ceiling and I hung a parachute from it. The parachute covers the whole ceiling and looks as if it is coming down. That disperses the high end and cymbals quite nicely, and takes the edge off. It's a tile floor on concrete, a wood wall, and quite a bit of glass, which makes it bright. I find it much easier to deaden a live room than to try make a dead room live. I might use packing blankets to cut down cymbal splash — that high end ping. The room is about 30 by 30 feet, and it has tremendous concussion because it's not a big vaulted room. It might not have a long decay as far as RT, but it's extremely percussive and present, so the kit jumps and snaps out.

I think twice about how I do things for 5.1 because you'll hear the flaws more. It is forgiving in that if you do have a flaw, it can sound very honest and real

Contrary to the way people record drums, over the last few years I have been recording drums in smaller environments rather than larger. It gives me so much more control over what I am doing by using certain compression on the ambience mics. I say "certain compression" because I had this box made with about a 160:1 compression ratio and I put that on the room mics. I call it the "juicer." Even in a simple little room, when you hear the drums with that compressor on the room mics, it sounds like this huge powerful warehouse. If I don't have the ambient mics up (these are separate from the overheads), the drums sound pretty much like they're recorded in a dry environment because there's no echo coming back to the mics. With the overhead mics pulled back a bit so they become more cymbal mics than overheads, I have tremendous control over the sound. I can mute the ambient mics, and all of a sudden it sounds like a dead drum kit. I open them and it sounds like we're back in Bearsville.

What kind of mics do you use for the ambient mics?

Neumann TLM 170's.

Where do you place them?

Usually in the upper corners of the room. I pick two corners furthest away from the kit and put the mics about a foot away from the corners of the room, where the ceiling and two walls all join together. You'll get a low-end buildup in the corner, which is nice. And I face the microphones towards the *wall* and not at the drums. If you face the mics at the kit — depending on the microphones — the mics will feature the cymbals. What I'm really trying to do is feature the sound of the room. It's like using a camera — I'm more interested in the environment that the subject is standing in than the subject itself.

And then you'll run the mics through the Juicer?

Yes. Very rarely do I let the mics go very straight to tape. If I do that, then I have more of a country-sounding record, which is great, but not necessarily what I want.

What would be a good example of a popular song where you have used this technique?

Check out Adam Ant's *Wonderful* album. It's all over it in many different forms. Or Joan Armatrading's album *What's Inside*.

How do you find that your recording technique has changed in light of 5.1?

It's probably becoming more cautious. In some past recordings, I have been more reckless with the sound — in a good way. For example, if I needed to get a certain bass sound, I might just put it through a SansAmp, crank it up, and get some fuzz on it to make it sound like it's going through a cabinet. In stereo, that might sound great. But in 5.1, the resolution is much higher. I can hear the deficiency of using something like the SansAmp to get that sound. It might be better for me to use an Ampeg combo; a 15-inch with an amp to actually capture the real sound.

In the later '80s with Walkmans, ghetto blasters, and Japanese tweeters everywhere, it seemed I could get away with this stuff without a problem. It was a part of the sound. I think twice now about how I do things for 5.1 because you'll hear the flaws more. It is forgiving though, in that if you do have a flaw, it can actually sound very honest and real, and become very endearing. But it might have the opposite effect as well.

Will your control room in Hawaii be 5.1 equipped?

Yes.

At the Journey Room, what do you record onto?

We use three formats, the main being an Ampex MM1200 — good 'ol Betsy (2-inch, 24-track). I have made so many hit records on MM1200's. I love the sound of the machine, and it keeps on rockin'. There's not much to them, so once you get one tweaked to work properly, it just keeps on going. I have Dolby SR and run it at 30 ips. I usually track on the 2-inch and then overdub on TASCAM DA-88's (I have four). The DA-88's are locked to tape via the SY-88 sync card; a MicroLynx serves as the interface between the Ampex and the DA-88's. I still believe that analog sounds better, but it's impractical due to space and expense to use two 24-track analog machines at all times. DA-88's are easy to make slaves with and there's a lot of practicality to them. I always keep my drums on analog because I haven't been successful in getting them to sound right on a digital machine.

If the guitar is really important and grungy, I would use the analog machine to get more grease on the sound — and that's important. Vocals still sound sweeter on the 24-track, but by the time you fiddle around with comping on analog with slaves and "b" reels, it actually slows a project down. I am able to bring life to vocals off the DA-88 anyway with the Neve and Fairchild, so it's a lot more favorable for me to use the DA-88 as a production tool rather than use the analog machine. But I still think that 2-inch sounds better. I also have an 8-track Pro Tools system that we essentially use as a tool.

What's your latest project?

As you know, I've been doing a lot of 5.1 and I'm continuing work in that area. Also, I am acting as executive producer for a record called *Out Of The Blue*, which is for a charity called UCI: The Universal Cetacean Institute for the protection of mammals of the ocean — dolphins and whales — to keep them there, as opposed to capturing them for the display industry. The main mandate is to educate people so they know that when they go to a place like Sea World and think it's great because Shamu is there and looks really healthy...well that's Shamu number 12 in 15 years. People don't know that. They think it's the same Shamu. If you saw how these poor creatures are actually caught, it's quite ugly (we have serious video of this). The record will come out in late spring/early summer/late fall with artists like Peter Gabriel, Daniel Lanois, Julian Lennon, Belinda Carlisle, Jackson Browne, and Van Brand 3000 participating. I'm doing this record as a labor of love. I want to get it out there, but I want to do it with maximum integrity, so I am forming a record label called Planet Blue. Hopefully, *Out Of The Blue* will be our first release.

In addition to *Out Of The Blue*, I'm also starting this label to offer a different approach to artists. We plan to develop artists without setting the pressure to have the instant hit, so they have to perform (in an economic sense) on the first album or be dropped. When a label does that, the artist doesn't get another chance because the label won't put that kind of money into them again (if it didn't work the first time). It's better to talk about making several records and build a career, doing the first one in an economical fashion because the music is great. Then put the act on the road and let people hear them doing what they really love to do: perform. My favorite bands weren't flash in the pans — bands like REM or U2. They work seriously hard, but had great support from their record companies, allowing them to grow. That's where the future of the industry is. I want to create an environment where we'll be in partnership with the artist to achieve that goal. **EQ**

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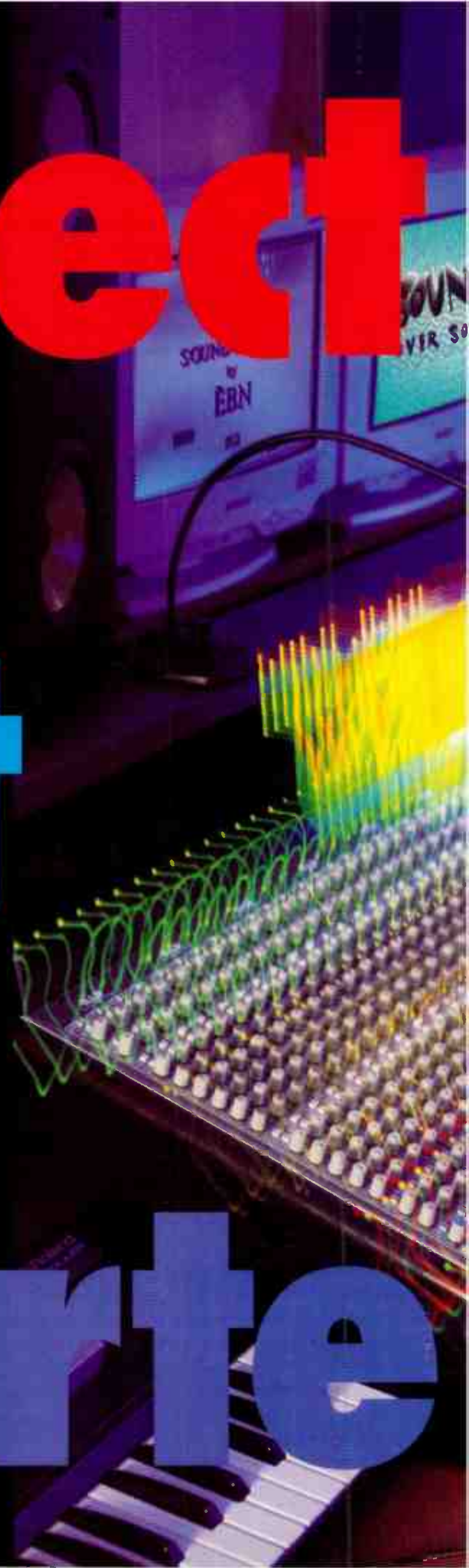


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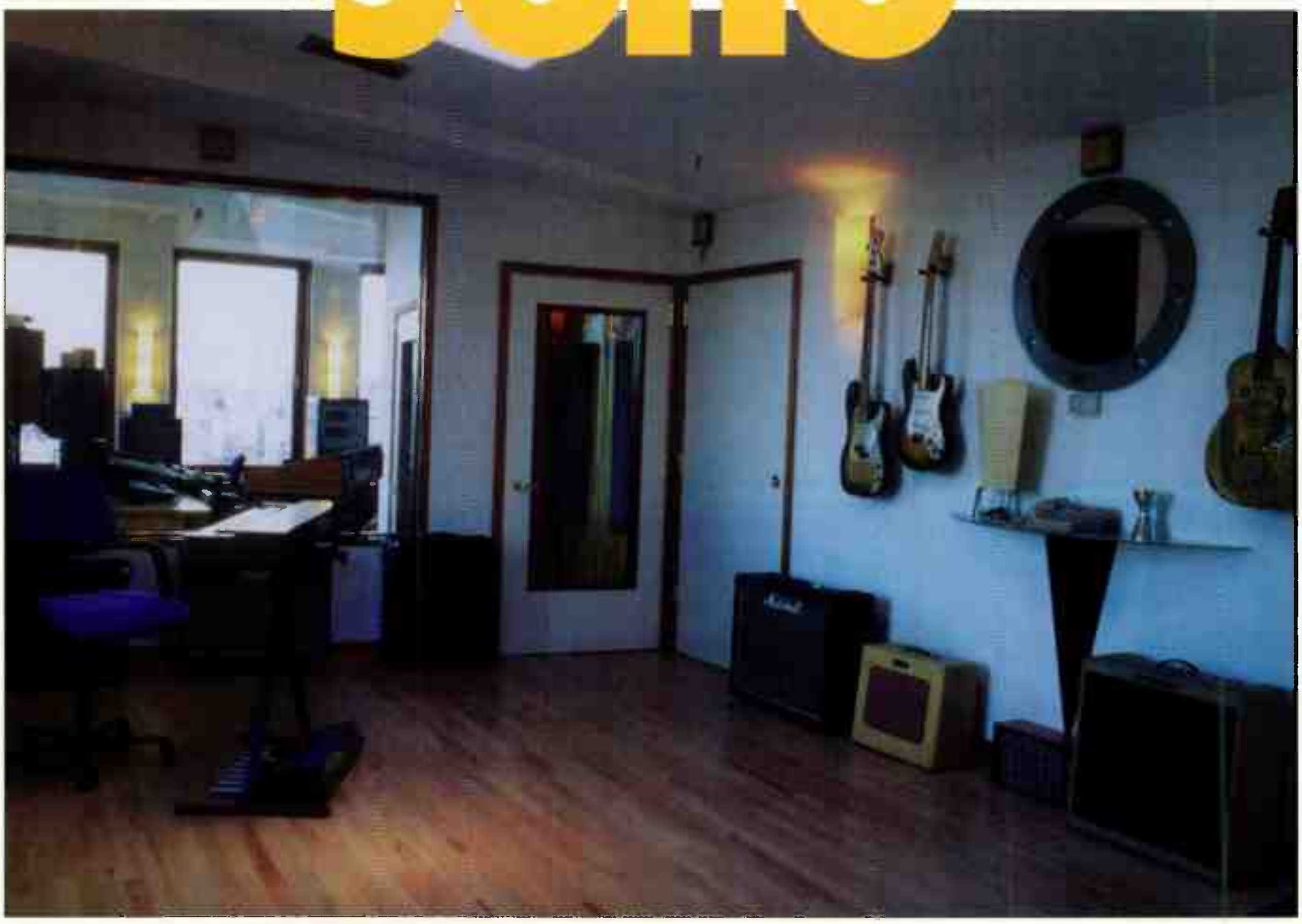
SIXTY SECONDS OVER SOHO

Using hard-disk recording techniques to make producing radio spots easier

By EBN

Recording artist/producer/composer/sound designer EBN has been active in the music industry for more than 30 years. A multi-instrumentalist, he plays guitar, bass Dobro, and drums. As an artist, he's had three major-label records — two with his band Riff-Raff and one with his band EBN/OZN. In addition to having worked with producers Phil Ramone, Arif Mardin, Keith Diamond, and Robert Margouleff, he has worked with such varied artists as Ravi Shankar, Scritti Politti, James Ingram, Michael Bolton, and Peter Tosh. At his second studio, Sundragon, the Talking Heads recorded their first album. EBN was one of the first artists to use the Fairlight/CMI to make an all-sample-based pop album and participates in beta-testing soft-



ware for many digital audio/sequencing manufacturers.

EBN recently completed building a studio called Sound Over Soho in New York City, which serves as his new home base for music and record production, film scoring, commercial production, and sound design. Some of his commercial clients include Miller, Canon, Burger King, Goodyear, Panasonic, Covergirl, and Kool Aid. Here, he discusses his studio and the recent production of a radio spot for Ricoh Business Systems.

Sound Over Soho is actually my fourth studio and serves as a facility in which I can do a wide variety of music production. I was involved in all aspects of design and oversaw the construction of my new facility, realizing that this would be the best way for me to get what I really wanted — an acoustically sound facility with great natural lighting and a view of New York City. John Storyk consulted with me and was very instrumental in helping me realize these goals (he has a wonderful way of taking what seems like an impossible set of conflicting parameters and making them work in perfect sync). Because of our busy schedules, John and I did a lot of communicating via e-mail. Sound Over Soho may be one of the first studios designed in cyberspace...yeah technology.

Since my goal was to be able to produce live music using the latest technologies, as well as to produce MIDI-based music, I have assembled an interesting arsenal of equipment. Some of that gear includes vintage Pultecs and LA-2A's (which I purchased for my first studio back in 1969), my new G3 PowerMac, an automated, 56-input CAD Maxcon console, a Roger Mayer 18-input class-A recording console, Studer and Otari analog tape machines, and Pro Tools, Logic Audio and TASCAM DA-88's for digital recording. Having such a wide range of equipment allows me to work in the manner best suited to a particular project. We do many kinds of projects here at Sound Over Soho: yesterday I was composing and producing tracks for an upcoming solo album from Eric St. Michaels (BMG Records) and today I composed and recorded a 60-second radio commercial for Ricoh.

WHAT DO YOU WANT FROM ME?

Before I could begin the composing and recording process for the spot, I needed a clear idea of what kind of music my client wanted. We held a preproduction meeting where they played a few pieces of music for me that had the vibe they had in mind. Hopefully, when you do a commercial, the client has an idea of what they want, which makes our lives as creative people much easier because you have something to relate to. It becomes difficult when you walk away with an unclear feeling of what's required. You don't want to be writing Lithuanian folk music when they want the Clash! This was a score, not a jingle, and it had pretty much wall-to-wall copy, so the music was really there to accent and drive the spot along, creating energy underneath to keep you listening. Musically, to my ears, it's kind of like Little Feat meets Seinfeld.

Having clearly understood what kind of sound they wanted, I began writing a 60-second piece of music that employed a combination of live Dobro guitar, synthesizers, and samplers. Using Logic Audio I recorded a MIDI-based background rhythm track of bass and drums. A Kurzweil K2000 supplied the bass sound, a Roland JV1080 was used for the drums (specifically the program "Brush Set"), and a Roland VG-8 was used for the electric guitar parts (more on that later). The Dobro was the featured instrument. Part of the commercial (the voice-over) would later be recorded in ProTools, but I used Logic Audio for the rhythm section because Pro Tools doesn't offer the kind of MIDI recording and editing I require. Logic Audio is generally where I live (it rules!) for MIDI and combination audio/MIDI projects.

THINKING LOGICALLY

At the time, I was running Logic Audio on a PowerMac 8100/110 MHz. I have stayed with the 8100 a little longer than I might otherwise have because I own a Digidesign expansion chassis fully-loaded with two SampleCell's, two disk I/O's, four TDM farm cards, and four Lexicon NuVerb's (which I love). I'm continuing to use this system, though I am now getting an Apple G3 300 MHz PowerPC (give me speed!), 24-bit Pro Tools, and two SampleCell cards. I always prefer to use two SampleCell cards because I can dedicate one to the drums and the second to other instruments and effects.

I use SampleCell frequently in my general setup, but I also use Kurzweil K2000's and K2500's equally as much, depending on the project (I have a massive library of sounds — thank God for Samplesearch!). I also have a Fairlight Series III, which to me is still the biggest, best, fattest-sounding sampler ever made (sort of like the miniMoog of samplers), but it wasn't used on this particular recording. Among these instruments, you could say I'm "well-sampled."

After working out the backing rhythm track via MIDI, I was ready to record the main instrument for the track — a Dobro guitar. I have a 1939 Dobro that sounds wonderful; basically all you have to do to record it is put a mic up. I used an Audix SCX-1 microphone with a cardioid capsule plugged into an ART Pro MPA tube preamp. The mic was about eight inches from the body of the Dobro and four inches above, placed at a 45-degree angle. I find that by using some of my tube gear such as the MPA, VLA, LA-2A's, or Pultec's, I can add some personality to recordings and remove some of the "digital-ness."

On that particular day I was working alone, so I set up in the control room, put on some headphones and — knowing that I wouldn't need to EQ the sound of the Dobro on the record side — connected the Pro MPA directly to a Digidesign 888 interface. The Dobro was recorded into Logic Audio at 44.1 kHz via the 888, without compression on the way in (I had used this setup recently for an album project, so I knew I wouldn't need to compress or EQ it.)

For all you TDM fanatics out there, I like to use Arboretum Systems "Hyperprism" stereo plug-in a single-miked sound like this. It can be subtle, but it really spreads the sound and gives it more sparkle (in fact, I find the whole Hyperprism Suite to be extremely useful). I also use the MDT plug-in by Antares Systems quite often. It's a very musical compressor and can be extremely potent in its sound changing/shaping abilities. The Q-Sys Plug-ins are very effective in widening and stereo image, but (like any great enhancer) should be used with taste!

RUSH, RUSH, RUSH

When you produce music for commercials, things move at a fast pace due to deadlines. There are often revisions, so repeatability is extremely important if you don't want to go totally crazy. (Aren't we already totally crazy?) A client might say to you, "That's a great sounding part. Can you try it with this kind of a feel, but let's keep the sound the same?" Now this could happen a day and a half or a week and a half later, towards the end of the project. You need to be in a position where you can get that sound up, change the groove of the part, and not be moving mics and tweaking knobs for and hour and a half. Clients don't like to sit around for that, and everybody can become a little anxious. (Remember Mel Brooks's *High Anxiety*?)

Changing the MIDI parts while retaining the sound is easy, but it could be more difficult to re-create electric guitar parts, so I've been using the Roland VG-8, of which I'm a big fan. I can quickly tweak it around and play with the parameters to get very authentic

tones, and they are repeatable. I have a lot of vintage guitars and amps, so I know what they're supposed to sound like anyway — which has helped me in getting the final result on the VG-8 to sound very similar to the real thing. I'm not saying that it's the way to go when you're cutting your album and that you shouldn't bring out your vintage Marshall, Hiwatt, Vox, or Fender amps (or whatever kind of amps you love). On the other hand you can push the VG-8 technology to obtain tones that are impossible to get with vintage gear, creating very cool and never-before-heard tones!

For media music technology, it can bring the end result very close to where you need it to be, while saving a lot of time (for anyone breaking into the commercial world: time is very important). The bottom line is that it's a joy for quickly getting guitar sounds with minimal sacrifice in tone. I used a "Hendrixy" Strat-type of sound from the VG-8, to round out the rhythm section.

FIVE TIMES OVER

When it came time to record the voice-over, I used Pro Tools because it's very well suited to this task. The VO artist spoke through a U87, and we made the recording in my vocal booth, which is very dry-sounding acoustically. There are a lot of open, ambient-sounding areas at Sound Over Soho that are great for recording things like drums, guitars and rocket launches, but you don't want to do a VO in a room like that because you can't control the reflections and you lose intelligibility. For a standard VO, you want the room to be tight, dry, present, and big. In the mix, you can always add a bit of space to it. I added a bit of [Eventide] H3000 in the mix for some depth, a modified version of the "Magic Air" program, which is a little too big, so I tightened it up. The effect is used in a very subtle manner, but it makes the vocal step out and surround you a bit more in the stereo mix.

In producing this commercial, there were actually five different spots: the middle section for two of them was the same and the tags on three were the same (these spots were running in different cities), but it was imperative that all of the VOs were the right length. I always find it amazing how a good VO artist will have this built-in time clock, so if they're 2 1/2 seconds over (i.e., 62 seconds) and you let them know, on the next read they'll come in at 59.5 seconds. Using Pro Tools negates the need for using a stopwatch to determine the length of the VO, since you can just mark across the VO, and the program's counter will tell you the length of the file.

There's another reason I used Pro Tools in this case: Why make the VO artist record these parts five times when all I need is one good take. In the end, I can cut and paste these parts into the different versions — which is exactly what I did. Along the way, I edited out some (but not all) of the breaths by selecting what I wanted to remove and deleting it. Note that in Pro Tools you should be in "slip" mode, *not* "shuffle" mode to do this type of deletion because, if you're in shuffle mode, the program sucks the part out and moves the surrounding regions together.

Once I had all five VOs cut, copied, and pasted the way everyone wanted, I then did a bounce to disk to create one complete Sound Designer II file of each VO without all of the cuts. Then I closed Pro Tools, reopened the VO file(s) in Logic Audio, and slipped the VO to the correct start point in relation to the music. In this particular case, the spot was completed in Logic Audio, though I can also start in Logic Audio and finish in Pro Tools, depending upon the project. It made sense to bring the VO back to Logic Audio because, at that point, I was not finished working on the music. Once the VO artist left, everyone went home except the creative director and myself.

Being that this was a radio spot, the mix was not particularly complicated and we didn't need to use the automation — just careful listening (if it had been for a TV spot, there might have been a lot of level rides and sound effect starts and stops to coincide with the picture. That would have required automation). It was just good old-fashioned mixing by hand, which can be a lot of fun.

I feel it's very important to listen on the right speakers, so I have the full gamut here, including the old Fostex LS3's, which I am particularly fond of (these have a 15-inch LF driver and use a wooden horn). I trust them, and they've become like an old friend. They're very useful for doing all the things you do on big speakers, such as placement of instruments in the stereo field and getting the bottom end correct on music mixes. But for this type of mix, I also like to use something smaller, so I used a pair of Audix Nile 10's. They are very true and don't make music sound sensational. When I mix on them and bring the mixes elsewhere, I find that my mixes sound the same. I also listen on ROR's (good Auratones) to check radio- and TV-playback sound.

Because I was under a time crunch, I mixed straight to DAT. Since I have 32 channels in my Digidesign hardware, I usually mix to tracks 31 and 32, then when I archive the whole project, it includes the final mixes (I use a Retrospect 4mm TurboDAT drive for backup). But this time I did it to DAT because I had to get the tape FedEx'd for a deadline. I made a clone of the DAT and then cloned that back to my DAW before backing up the project. These spots for Ricoh Business Systems are now running on national radio.

And always remember: No matter what — have fun when you're making music!

Credits for the Ricoh commercial:
Agency: Gigante Vaz Partners
Client: Ricoh Business Systems
Written by Alan Platt Creative Director at Gigante Vaz
Music and Sound Design by EBN

As this issue was going to press, we regretfully learned that EBN had passed away. EQ sends our deepest condolences to his family, friends, and associates. His contributions will not be forgotten.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The DOD Dimension 12 puts incredible sampling power at your fingertips. The D12 boasts 4 banks of 6 second sampling, or 2 banks of 12 second sampling. A stutter button gives you hit real time control. Editing your sample is a snap. A large jog/shuttle wheel puts edit points right where you need them. A reverse function and an LFO twist your samples into submission.
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The DOD Dimension 3 has the least complicated operating system made. There's an effect for all applications and they are all at your fingertips with the turn of a knob. You can have the perfect sound in a matter of seconds. There is nothing complicated with the D3. Select an effect and dial it in.
DOD Dimension 6
DOD's NEW Dimension 6 is basically all the processing power of the Dimension 3 . . . times two. The D6 has broad, spacious reverbs, thick chorus and flange, and tight clean delays. The operating system is as simple as the turn of a knob. You want more effect? Grab a knob. Need to adjust delay time? Grab a knob. Exceptional real time control from DOD
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CIRCLE 23 ON FREE INFO CARD

EFFECTIVE SOUNDING SOUND EFFECTS

Some ideas on using your project studio to get started in this creative field

By Simon Amarasingham

What kind of sounds do flying devils and laser cannons make? Electronic and computer game sound effects can really put a sound designer's imagination to work. If you're interested in expanding your creative and business horizons by writing music for multimedia applications, adding sound effects to your list of services can help get the gigs, and project studio gear can do the job. Here are some ideas to get you started.

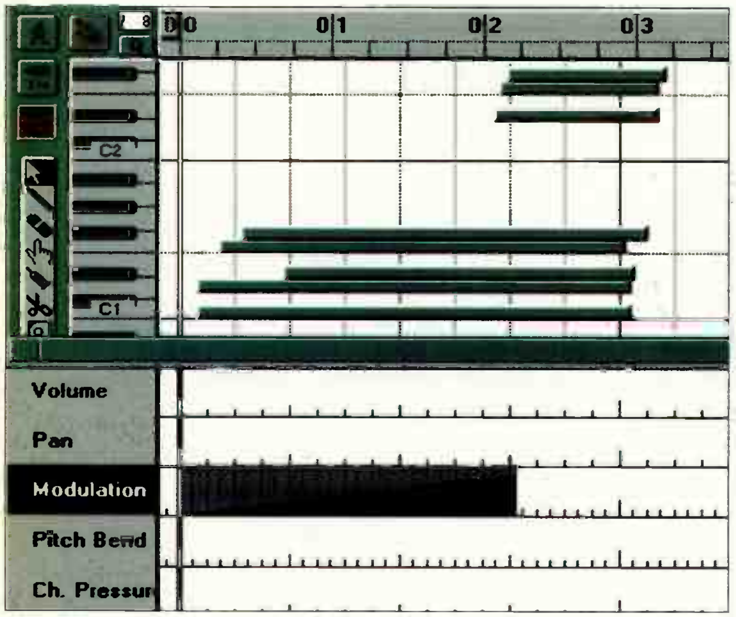

FIGURE 1: The Stone Door Opening sound effect as seen in a sequencer's piano roll editor. Each sound has been played much lower than the original sample and multiple pitches are layered on top of each other.

GETTING THE BASE SOUNDS

Unusual sounds often start with something more mundane, such as recording household objects in your studio. As you familiarize yourself with how equipment can alter a sound (see "Processing Sounds"), you can bang, scrape, and crumple all kinds of things in front of a microphone and extrapolate in your mind's ear how that sound might be transformed. For example, to make the sound of a heavy stone door opening — a deep scraping of stone against stone — substantially lowering the pitch of pottery cups or bricks scraped against each other might work. Experimentation is key.

If you have a portable recorder (e.g., a DAT), the great outdoors can yield lots of sounds. However, since you can't control the recording environment (e.g., silencing all the tourists at the zoo while you try to record the gorilla breathing), treat these expeditions as a way to build up a collection of sounds for possible later use. Relying on this method to get usable sounds that a client wants tomorrow seldom works.

SOUND EFFECTS LIBRARIES

For an action/combat game, it takes a lot of time, energy and money to record explosions, heavy artillery, and military aircraft — but luckily you don't have to. Professional sound effects CDs run about $50 per CD when purchased in collections of 10–20 CDs; single CDs are also available, and these are a good way to get started (see sidebar). Just make sure to read the fine print regarding usage and licensing.

Whether the sounds are original or come from a library, the real fun starts when you transform them into something new.

PROCESSING SOUNDS

Two main sound processing tools are samplers and a computer with audio editing software. With some extra work, audio editing software can do the sampler's job, but for the moment let's say we've sampled two bricks scraping against each other to make a stone door opening sound.

Most samplers can assign a sound to a keyboard range below the original recorded pitch. Fig. 1 indicates that the brick-scrape has been assigned to the range C1–B1 with its original pitch set to C4, so that the sample will play 2–3 octaves lower. As well as dropping the pitch, multiple brick-scrapes can be played simultaneously.

So far we have a stone-against-stone scraping sound, but we also need a huge door hitting the wall as it comes to rest. A sound effects library might have plenty of door closing sounds, but let's try something different — like dropping the pitch of a slam-dunked basketball smacking through a hoop. Not only does this give a huge slamming sound, it's original and ear-catching.

Record from a CD (digitally, if your sampler and CD play-

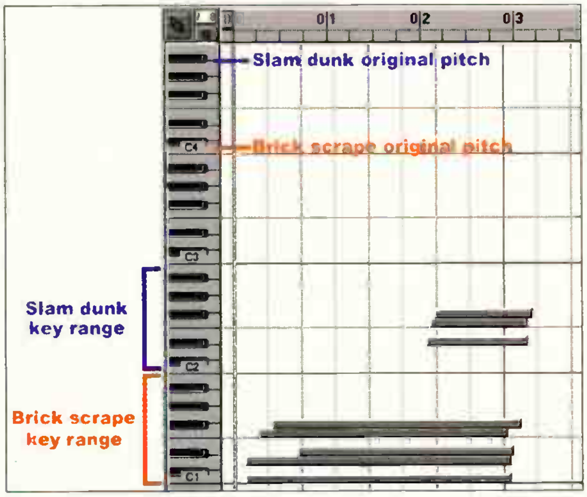

FIGURE 2: Editing the mod wheel movement can open the filter fully by the time the slam dunk sound appears.

er allow it), then assign the sample to a one-octave range, 1.5 octaves below the original pitch (fig. 1).

After assigning the samples across the keyboard, "perform" the effect. After a little practice this could be good enough for the final version, and you may want to record it directly to the computer. However, by first recording the performance into a sequencer, you can lengthen, shorten, and move notes to get the best result (fig. 1).

Layering different pitches is an easy, effective way of altering a sample, but also consider processing the brick-scraping sample with a low-pass filter, and assigning the filter frequency to the mod wheel. Editing the mod wheel movements in the sequencer can open up the filter (so the scraping sound gets brighter) as the door opens (fig. 2).

Once the sampler-sequencer team has been tweaked to perform the best possible door opening, record the final result into the digital editing software.

USING DIGITAL EDITING SOFTWARE

Programs such as Sound Designer II, Peak, Sound Forge, and Cool Edit Pro have functions that can do the jobs just described for samplers (although it's more work). Pitch shifting, filtering, mixing, and many other effects are in there along with tools for changing sample rates, converting between audio formats and other functions that are indispensable for delivering audio for games. If your budget is extremely tight, a shareware Windows version of Cool Edit (www.syntrillium.com) is a great start.

The other piece of the puzzle is actually getting the audio into your computer. There are now many different sound cards and hard-disk recording systems available, so if you're uncertain which one is for you, keep reading this magazine!

As the sequence plays the sampler via MIDI, record the sampler's audio output into the computer (press Record in the audio editing application, then press Play at the sequencer, and finally press Stop in both applications after capturing the sound).

For synchronizing the sound effect with an animation, ideally we'd have an AVI or QuickTime movie of the stone door opening before we started creating the sound, but the artists can be working for months before coming up with any finalized animations. Rather than wait for them, it's better to work up some sounds from descriptions, then fit them to the animations as they become available.

Say the stone door animation is 1.667 seconds long, but our door sound is over 3 seconds long. Time-compressing a sound by 50 percent will create audible artifacts that may or may not produce an interesting result. Cutting, pasting, and

continued on page 140

SOUND EFFECTS RESOURCES

The Hollywood Edge [www.hollywoodedge.com] offers several sound effects collections, including two double-CD collections called *Super Singles Vols. I & II* ($60 each). These have a large number of shorter sounds (good for games) that cover a wide spectrum. Also check out Sonic Science [www.sonicscience.com]. East West's Web site [www.eastwestsounds.com] is particularly interesting because sounds are in a searchable database, and can be previewed and purchased individually over the Internet ($2.95 each with discounts for bulk purchases). East West says that The Hollywood Edge's collection will soon be available from their site, too. A limited number of free sound effects samples are available from Craig Anderton's "Sound, Studio, and Stage" site on AOL (keyword SSS).

Powerful 24-Bit Sound

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Power On/Off Speaker "Pop" Protection
Multi-Channel 2 In 4 Out Capability
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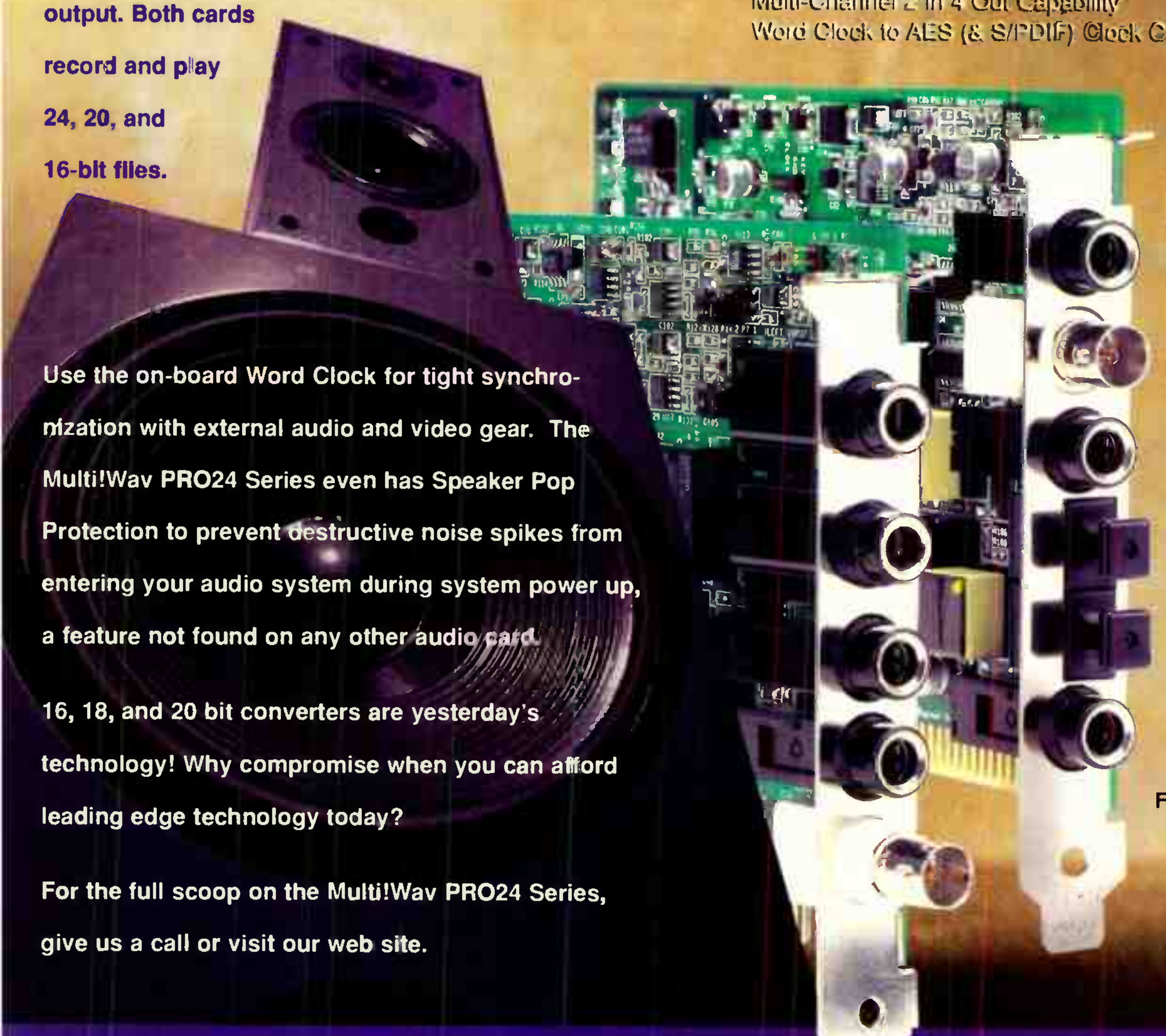

Use the on-board Word Clock for tight synchronization with external audio and video gear. The Multi!Wav PRO24 Series even has Speaker Pop Protection to prevent destructive noise spikes from entering your audio system during system power up, a feature not found on any other audio card.

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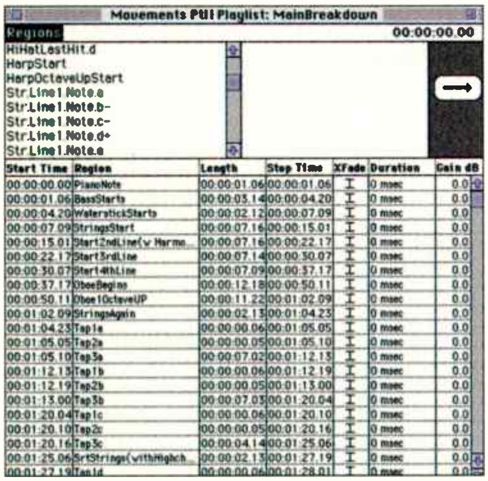
SOUND FOR PICTURE ON A BUDGET, PART 2

More on getting your studio geared up for multimedia

BY TONA OHAMA

In part I of this article (November '97), I described the software and hardware I use to do audio-for-video in my studio. This issue we'll put all that fun stuff to use.

The first project I attempted was my own — a multimedia live show at the Alberta Science Centre Planetarium (see *EQ*, 1995, issues 1 & 2). My next project was for STARS (Shock Trauma Air Rescue), creating a multimedia show for a fund-raiser. They also wanted the video portion of the show to work as a stand-alone promotional video for use throughout the year. This was the first time I would actually edit a video in Adobe Premiere and sync sound directly to it. During the video shoot, Chris Daniels of The Audio Lab handled location recording and Dan Parrish of Parrish and Company headed up the video crew. We decided to split the video and audio, without timecode, and sync them up later. No problem!



Movements PUI Playlist: MainBreakdown

Regions 00:00:00.00

HiHatLastHit.d
HarpStart
HarpOctaveUpStart
Str.Line1.Note.a
Str.Line1.Note.b-
Str.Line1.Note.c-
Str.Line1.Note.d+
Str.Line1.Note.e

Start Time	Region	Length	Stop Time	XFade	Duration	Gain dB
00:00:00.00	PianoNote	00:00:01.06	00:00:01.06	I	0 msec	0.0
00:00:01.06	BassStarts	00:00:03.14	00:00:04.20	I	0 msec	0.0
00:00:04.20	WaterstickStarts	00:00:02.12	00:00:07.09	I	0 msec	0.0
00:00:07.09	StringsStart	00:00:07.16	00:00:15.01	I	0 msec	0.0
00:00:15.01	Start2ndLine(w Harmo	00:00:07.16	00:00:22.17	I	0 msec	0.0
00:00:22.17	Start3rdLine	00:00:07.14	00:00:30.07	I	0 msec	0.0
00:00:30.07	Start4thLine	00:00:07.09	00:00:37.17	I	0 msec	0.0
00:00:37.17	OboeBegins	00:00:12.18	00:00:50.11	I	0 msec	0.0
00:00:50.11	Oboe1OctaveUP	00:00:11.22	00:01:02.09	I	0 msec	0.0
00:01:02.09	StringsAgain	00:00:02.13	00:01:04.23	I	0 msec	0.0
00:01:04.23	Tap1a	00:00:00.06	00:01:05.05	I	0 msec	0.0
00:01:05.05	Tap2a	00:00:00.05	00:01:05.10	I	0 msec	0.0
00:01:05.10	Tap3a	00:00:07.02	00:01:12.13	I	0 msec	0.0
00:01:12.13	Tap1b	00:00:00.06	00:01:12.19	I	0 msec	0.0
00:01:12.19	Tap2b	00:00:00.05	00:01:13.00	I	0 msec	0.0
00:01:13.00	Tap3b	00:00:07.03	00:01:20.04	I	0 msec	0.0
00:01:20.04	Tap1c	00:00:00.06	00:01:20.10	I	0 msec	0.0
00:01:20.10	Tap2c	00:00:00.05	00:01:20.16	I	0 msec	0.0
00:01:20.16	Tap3c	00:00:04.14	00:01:25.06	I	0 msec	0.0
00:01:25.06	SrtStrings(withhighch	00:00:02.13	00:01:27.19	I	0 msec	0.0
00:01:27.19	Tap1d	00:00:00.06	00:01:28.01	I	0 msec	0.0

CHECK IT TWICE: A sample list of a breakdown done with Sound Designer II.

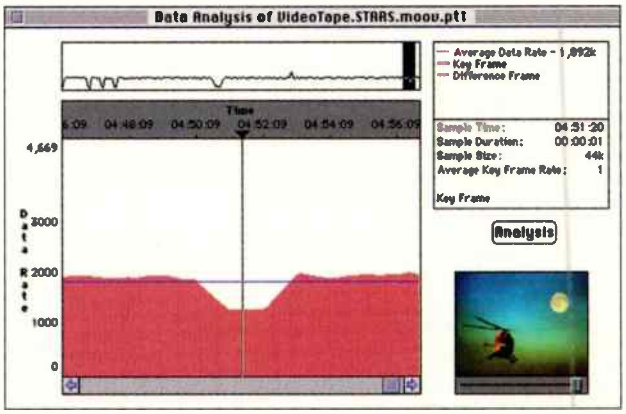
We captured video into Premiere via the Radius VideoVision Studio card (VVS). Dan was very organized, and knew which clips he wanted. This was a key factor that allowed me to do the project because I didn't have the hard-drive space to digitize everything and let the director choose from numerous takes. I managed to get a rate of 1.7 MB/sec on my Mac Quadra. This is about half the 3.5 MB/sec that is used on a professional broadcast system, but the quality was deemed acceptable because the source was S-VHS, not Betacam. Note that it is impossible to play back audio and video simultaneously at this rate on a Quadra without dropping video frames.

I began with a clean drive, captured with "abort on dropped frames" selected, and then used the "movie analysis" tool in Premiere to verify. I've found that once a QuickTime movie has a jerky area in it, it's there for good. Defragging after the fact will not fix it. The cause of these hiccups appears to be discontinuities in the compression data rate. If you have several clips and "splice" them together without recompressing, this can occur.

Once the video edit was complete, I gave Chris a copy on VHS with timecode burn-in plus an audio SMPTE track. He mixed the audio with SAW on a PC and transferred to DA-88's for the mix. We rented a Betacam and transferred the video directly from my Mac without timecode. To get QuickTime movies in Premiere locked to timecode, the solution is Pipeline's ProVTR, but I didn't know that at the time. We were able to sync the DA-88's to the Betacam by adjusting the SMPTE offset. This got us to within a frame of sync, not "frame edge" accurate, but sufficient for the project.

I've done a number of commercials, and I admit I was intimidated the first time someone asked, "Can you do broadcast-quality audio?" Well, it turns out that "broadcast quality" is a vague term. If it's playing off a Betacam, people assume it's great, but a Betacam only produces what is put in, and I've heard sound that would make your flesh crawl. I'm not saying you can get away with lousy sound, but I am saying don't be intimidated when someone asks if you can do broadcast quality. If you're doing CD masters or high-quality demo tapes, then you already are.

The types of commercials I've done are typically low-budget, short turn-around affairs where speed really counts. I request a VHS tape with timecode burn-in and capture it at 400k/sec using a consumer level VHS player. The video is only a guide, so quality isn't as important as smooth playback. In fact, most computers can only play back 400–700k/sec quality levels while simultaneously running something like Pro Tools.



DIGITAL ANALYST: With the Data Analysis tool in Premiere you can check for changes in the movie compression rate.

SR24•4/32•4

From the van to your studio.

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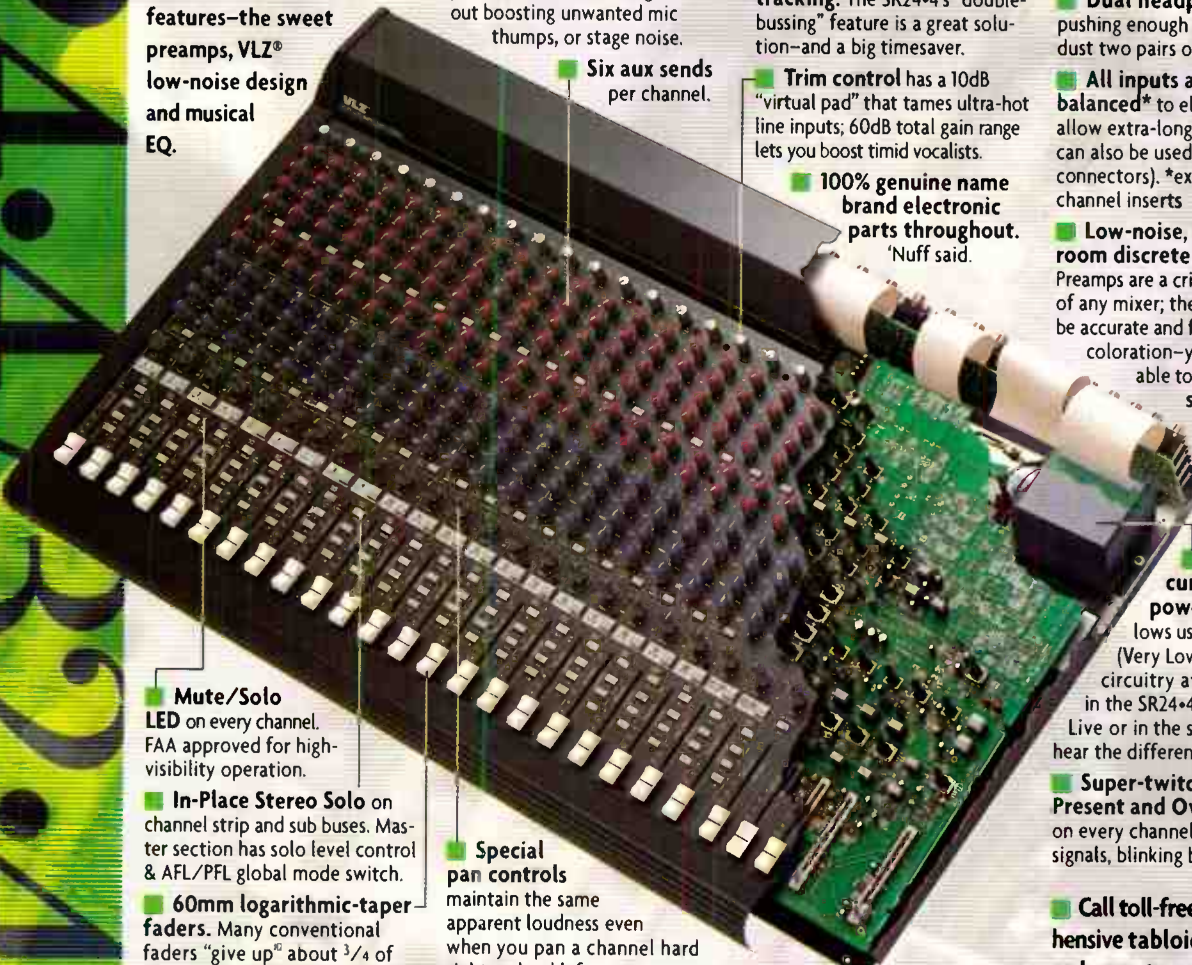
Few bands have the bucks for a separate studio mixer. So instead of cutting corners, we made the SR24•4 a "down-sized" Mackie 8•Bus with much of its circuitry and many of the same cool features–the sweet preamps, VLZ® low-noise design and musical EQ.



Flexible, creativity-enhancing equalization. Mono mic/line channel's swept midrange has a super-wide 100Hz-8kHz sweep range (and a broad, natural-sounding 1.5-octave wide curve). You also get a sharp, 18dB per octave **low-cut filter** that lets you use the Low shelving EQ without boosting unwanted mic thumps, or stage noise.

Six aux sends per channel.

Inserts on all mono channels– plus submix and main stereo mix inserts.

Double tape outputs eliminate repatching during tracking. The SR24•4's "double-bussing" feature is a great solution–and a big timesaver.

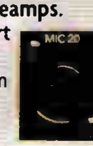
Trim control has a 10dB "virtual pad" that tames ultra-hot line inputs; 60dB total gain range lets you boost timid vocalists.

100% genuine name brand electronic parts throughout. 'Nuff said.

Dual headphone outputs pushing enough amperage to dust two pairs of eardrums.

All inputs and outputs are balanced* to eliminate hum and allow extra-long cable runs (they can also be used with unbalanced connectors). *except RCAs and channel inserts

Low-noise, high-headroom discrete mic preamps. Preamps are a critical part of any mixer; they must be accurate and free from coloration–yet be able to handle screaming vocalists and close-miked kick drums without overloading.



Large, high-current internal power supply allows us to use VLZ® (Very Low Impedance) circuitry at critical points in the SR24•4 and SR32•4. Live or in the studio, you'll hear the difference.

Super-twitchy Signal Present and Overload LED's on every channel. Subtly senses signals, blinking boldly bright.

Mute/Solo LED on every channel. FAA approved for high-visibility operation.

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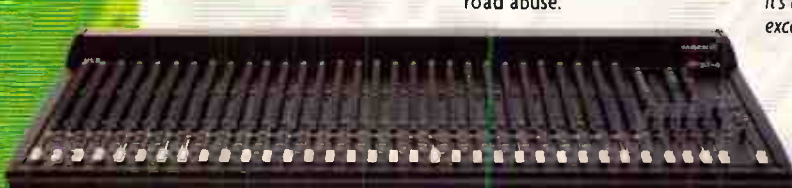
60mm logarithmic-taper faders. Many conventional faders "give up" about 3/4 of the way down. Ours are audibly tapered to zero. Each fader's wiper surface is derived from automotive sensor technology and won't develop "the scratchies" even after years of compulsive tweaking.

Special pan controls maintain the same apparent loudness even when you pan a channel hard right or hard left.

Solid steel chassis, gold-plated internal interconnects and sealed rotary controls keep the SR24•4 working through dust, smoke, moisture and road abuse.

Ultra-high "AIR" EQ on submix buses centered at 16kHz. As one magazine review put it, *"The AIR controls turned out to be effective in adding top end clarity... it's almost an 'exciter' kind of effect, except without the harshness."*

Call toll-free for a comprehensive tabloid brochure, or log onto our Web site for the full story of the SR24•4 and its big brother, the SR32•4. They look good outside. But more important, they SOUND good inside.



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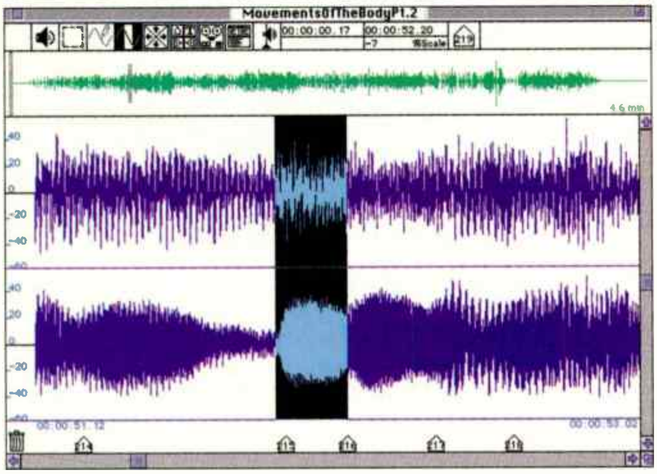
World Radio History

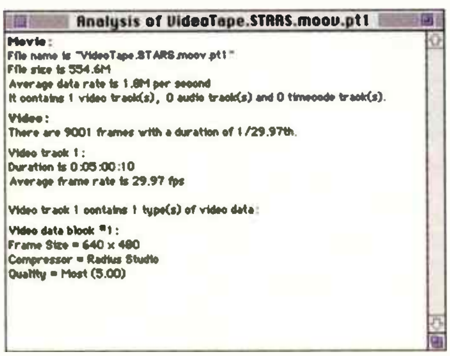
CIRCLE 77 ON FREE INFO CARD



MAKE YOUR MARK: Using the ENTER key while playing a sound file in Sound Designer II will drop in markers.

After capture, I conform movies (Adobe Premiere's definition) to 30 fps. I don't use 29.97 fps because, when using Digital Performer, I like to use the arrow keys to increment by frame, and it won't do that at 29.97 but will at 30. (This could be a QuickTime issue, but I really don't know.) I always verify that Adobe Premiere's frame count matches the burn-in numbers to catch dropped frames or incorrect frame-rate settings. Usually the finished mix is delivered without timecode on DAT at 48 kHz with a set of –15 dB/1 kHz/1 frame sync pulses at the top. Some clients want the audio on timecoded DAT or on Betacam, in which case I'll either rent or take my DAT machine to their studio and insert the audio. I make sure I know clearly before I start what they expect for a delivery medium. Commercials are ideal for my setup because they are so short, and sync becomes less of an issue.

I've worked on many animated films, and the audio has been handled differently for every project. On one film, the animator gave me exposure sheets, which are a frame-by-frame listing of all the action. I used these numbers to create a list of markers in MOTU's Digital Performer, with a frame rate of 24 frames per second. Then it was simply a matter of gathering all the audio, creating sound bites, and dropping them in at the appropriate marker. The final mix was recorded to the TASCAM DA-30 DAT with sync pulses, then transferred to the film so this soundtrack was created entirely without seeing any visuals at all!

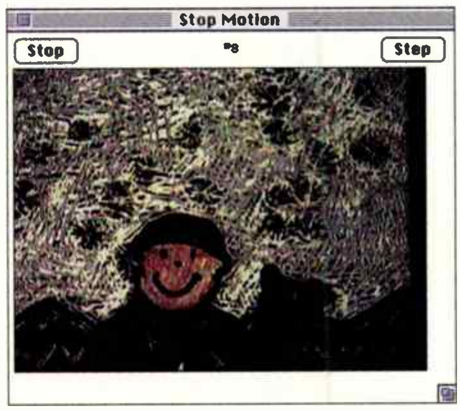


CHECK IT OUT: The "Analyze Movie" command in Premiere can be used to check for dropped frames or frame rate.

Some animators want a soundtrack before their film is shot, without exposure sheets. For Wayne Traudt's *Movements of the Body*, all the audio was created first, and so I had to do a breakdown for the animator. A breakdown is a list of start/end frames for every bit of sound. For the MIDI tracks, all I had to do was print a note list with timecode, but for the audio tracks, the points had to identified manually.

First I mixed the soundtrack to DAT and reloaded into Sound Designer II, creating a sound file equal in duration to the entire soundtrack. Then I played the track, hitting Enter repeatedly to get markers close to the spots I wanted in the breakdown. Each marker was manually dragged to the nearest frame. His film required over 1000 points, so I wrote a macro in Quickeys that used the markers to create regions. A playlist of appropriately named regions gave Wayne the numbers he needed.

TREES! was created on film, but delivered on VHS to 600 rural elementary schools and Mennonite communities in Alberta, Canada. The animators, Carol Beecher and Kevin Kurytinek, used the cameraless technique: 2500 images were etched by hand directly into the emulsion of opaque 35 mm motion picture film stock and colored with markers. The film was rear-projected and silver-mirrored onto a sheet of velum stretched over the compound table of an Oxberry animation stand. Each projected image was step-captured using a Hi8 camera and VVS/Premiere using no compression, as a series of PICT files. These were then converted to a QT movie, played at half-speed and compressed using the Radius JPEG codec (compressor/decompressor) at about 1.2 MB/sec. It worked well, except the pixelization due to compression was very apparent during the credits. Large, flat areas of color do not compress well, whereas lots of moving colors do.

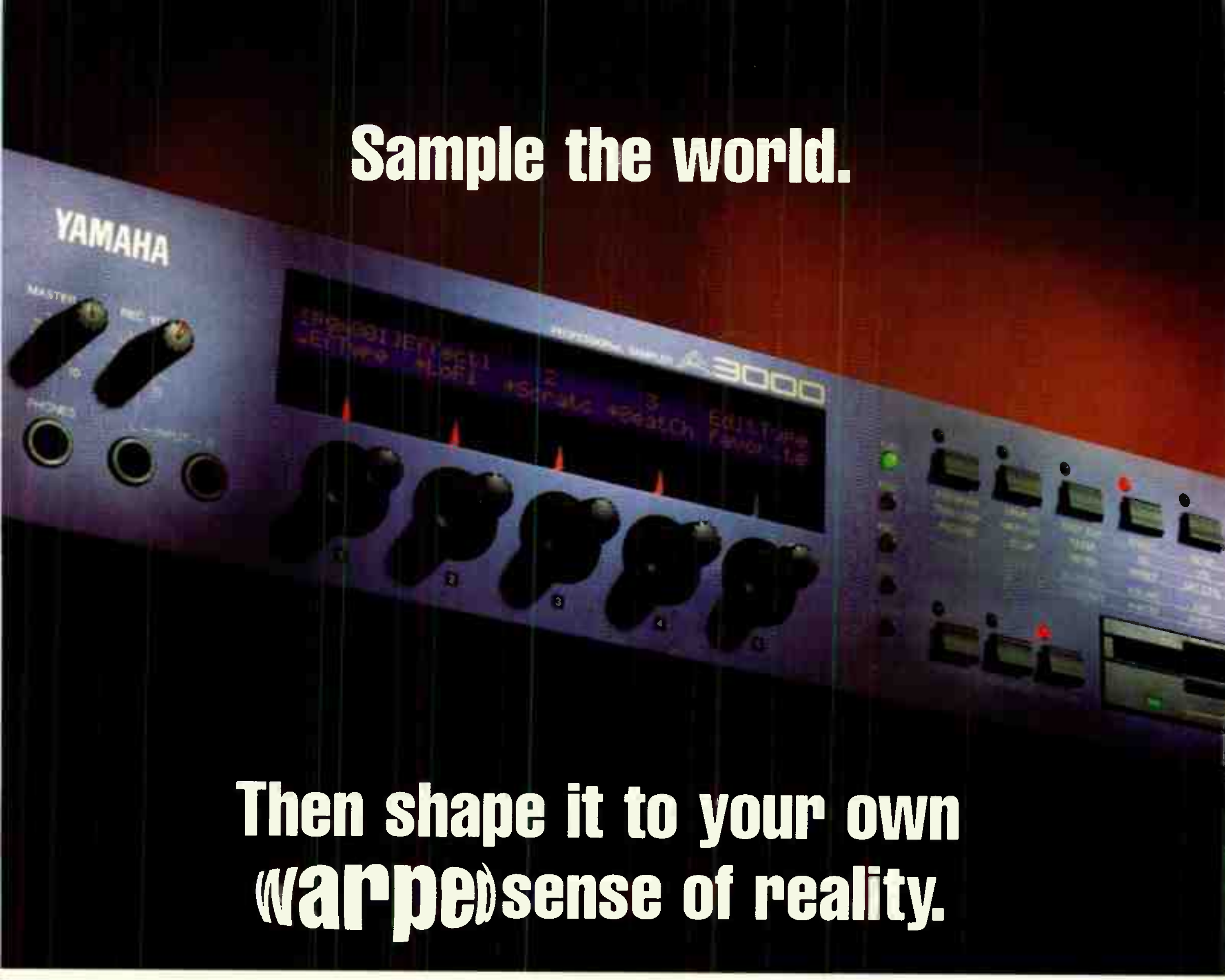


STOP 'N' STRING: Adobe Premiere's "Stop Motion" feature lets you capture a series of still images and string them together into a movie. (Image from *TREES!*)

I mixed all the audio in Premiere, then I tried Digidesign's Sound Drive Extension to get the audio out through the Pro Tools Interface, but it did not work. I converted to 22 kHz, 8-bit mono and sent the audio out through the Mac mini port. Not great quality, but it was acceptable for mono VHS. I had to jump through some hoops to get everything to sync. First I saved the video edit to a single QT movie, without sound. Then I transferred the audio mix to DAT and re-imported the mix to Premiere via the Radius interface.

I now had two QT files — one audio, one video. I tried creating a single QT movie out of these, but on playback the audio would drift no matter what I did. The solution was to create a new Premiere project and import the audio and video clips. By using "preview" rather than create a new movie, I was able to get the movie to play in sync, and transferred this to Betacam.

The "video purists" out there are probably cringing over my methods, because I'm the video equivalent of a studio doing CD masters off a cassette recorder. All I know is that my clients have been happy, the studio has widened its range of services, and I'm learning something new everyday. Many studios will already own much of the hardware and software required to get started, so just fill in the missing pieces and try it out! With streaming video clips on the Internet and DVD just around the corner, the world of sound-for-picture is only going to get bigger! EQ



The Yamaha A3000 gives you the power to capture any sound and stretch, warp, duplicate, or blast it into any form that you can imagine.

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Speaking of capacity, store your warped world on the A3000's internal floppy disk drive, optional internal hard drive or even an external hard drive for unlimited storage capacity.

The Yamaha A3000 isn't just powerful, it's easy to use. Turn the knobs, push a few buttons and you can accomplish anything, including internally resampling to include effects with the sample data. Applying 4-band total EQ. Adding parametric EQ to each sample. Or freely mapping EQ to key and velocity ranges with layers and/or splits.

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All of this power comes at a price. Fortunately, it's exceptionally low—just $1999 MSRP. This much power has never been available to you for anything close. So get a Yamaha A3000 sampler today and shape the world's sound to your own warped sense of reality.

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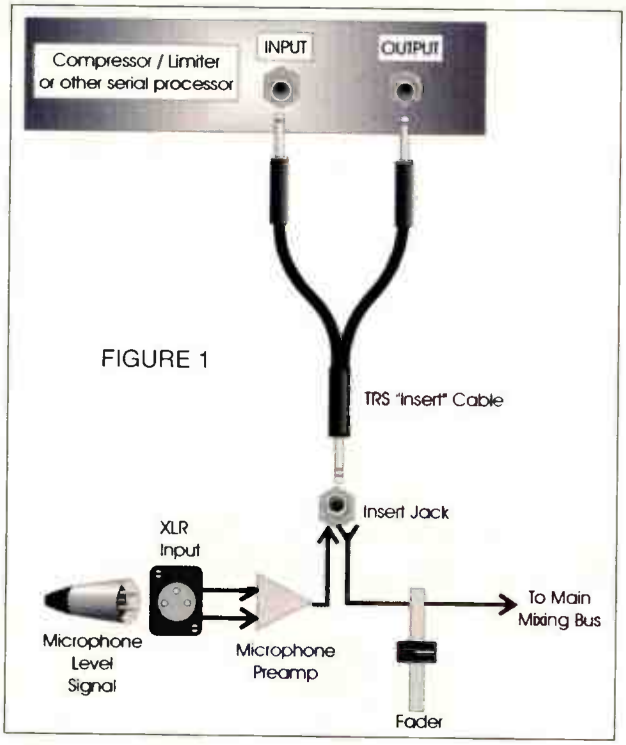
Our Friend The Compressor-Limiter

You can spend a lifetime experimenting with compressors, and, indeed, it certainly seems like I have. If you're not very familiar with their usage, then what follows will set you on the right path to learning more about them. If, on the other hand, compressors are your sonic constant companions, then you'll certainly have your own war stories and favorite configurations.

My first run-in with audio compressors was 20-something years ago when I made a deal with a local radio station to play a spot promoting my band. Since I had a small project studio used to make demo recordings of our act, we naturally wanted artistic and creative control of the spots. After agonizing over the text and mix for several days, we finally had a spot ready to go. Since we were a cool techno-rock act, we used lots of flanging, phased bass, and a big "whip crack" throughout the spot. It sounded awesome over the studio monitors, but hearing the much-anticipated spot over the radio was a real letdown. It sounded awful! The stereo image squirmed all over the place, and the level of the narration ducked on every whip crack.



I called the radio station and complained about how they messed up our spot, but they claimed it was fine. After talking to the station engineer I found out what had happened. The compressor-limiter on the radio station transmitter didn't like the out-of-phase bass we thought sounded so cool; plus, it went into a spasm on every whip crack, causing the whole program to suck down and come up for air every few seconds. A few days of remixing experiments did fix the problem, but not without me first eating crow served up by the engineering staff.

Thus began my love-hate relationship with various audio level controllers (hereafter called "compressors"). These are not to be confused with digital data reduction programs or CODEX units, which are technically "lossy data compression," where you're trying to reduce the digital bandwidth and storage size of audio files without affecting the program dynamics. No, we're talking about gain-controlling devices. Some sound great, while others sound terrible. And nothing sounds worse than a compressor in the hands of someone who doesn't know how to operate it.

Over the last two and a half decades, I've used compressor-limiters in lots of different ways. Herein are a number of examples that have worked for me in many situations. Of course, these are starting points, and you need to experiment and listen to the results. Contrary to what some manufacturers may lead you to believe, there are no hard and fast rules to using compression.

WHY WE COMPRESS

In a perfect world, we wouldn't need compressors. In fact, on a lot of the symphonic recordings that I do, it's a big no-no because we're trying to capture as much of the dynamics as possible. Therefore, any compression is seen as altering the reality of the performance. But you probably can't get and don't want real-life dynamics on tape. I'll usually sneak in a little compression on the strings, or whatever is not staying up in the mix. When you're in a "live"-listening situation, your brain can dynamically ride the gain. (That's why your ears

BY MIKE SOKOL

nd The
or-Limite

A guide to knowing and using your friendly comp-lim

can be sensitive enough to be bothered by a drip in the middle of the night, but still tolerate fairly loud music without pain.) You also have some pretty cool binaural processing that allows you to concentrate on a single conversation in a crowded room. Microphones and tape recorders can't do that, and you have to artificially control the dynamics or they'll overload the tape or dominate the mix. Radio stations started compressing the audio as a way to raise the average volume levels (and hence the perceived volume) of the music without overmodulating their transmitters and bringing the wrath of the FCC upon them. In the pop music world, it's a way to get a soft-voiced singer up into the mix while protecting your ears and sound system when he screams, "Hello Cincinnati!" to the crowd.

A FEW DEFINITIONS

Compressors and limiters are basically the same thing; it's really just a matter of degree. Both units have common hardware. First, there's a detector stage that "listens" to the audio signal, some controls that set the attack and decay rates, and a variable amplifier that's used to dynamically change the gain of the unit. The compression ratio is the rate at which the output will change with an input level change. For instance: a 10:1 compression ratio will only allow the output level to change 1 decibel for every 10 decibels of input increase once the threshold level is reached. Likewise, a 3:1 ratio will compress a change of 3 decibels on the input down to 1 decibel on the output. (That's why it's called compression.)

Traditionally, ratios from 2:1 to 10:1 are defined as compression, while ratios from 10:1 to infinity are defined as limiting. Compression is normally used to "tighten up" the dynamic level of an instrument, while limiters are used as safety stops to keep the level from getting out of control. So while compressors act over a wide range of audio dynamics, limiters are generally set to avoid too high a level; for instance, to prevent digital splatter in A/D converters or keep amplifiers from clipping and frying speakers. Again, it's really a matter of degree and what you expect the gain reduction to accomplish.

SIDECHAINS

These are an insert into the detector circuitry of the compressor. By placing an equalizer into this loop, you can tailor a specific group of frequencies to be compressed while ignoring the rest of the spectrum. Note that you're not really affecting the EQ of the sound routing through the compressor — you're only changing how it interprets the signal. So by booting the equalizer by 6 dB at 5000 Hz, you'll essentially be compressing any 5000 Hz sounds by an extra 6 dB. This is useful for removing sibilance from a vocal. You can also experiment to control a bass guitar's string slap while leaving the bottom frequencies alone. If you want to be really creative, you can route the signal from another source into the sidechain to make the dynamics of the processed channel do strange things. For instance, vocal ducking is a standard radio station process where the voice of the DJ causes the music to "duck" whenever the voice is present. I've also heard the kick drum routed this way, which caused the cymbals to "suck down" on every kick beat and slowly crawl back up before being hit again. It was a little disorienting, but interesting.

LINKING

If you have a stereo compressor (or two mono units), you may need to link the channels together. This cross-connects the two detector circuits so that compression on one channel is matched by the second channel. This is only necessary if you're using it to process the final stereo mix and want the stereo image to remain stable and not dance from side to side. If you're using one side for vocals and other side for bass guitar, then engaging the Link button would have unpredictable and unmusical results.

ATTACK AND DECAY

The Attack setting is how fast the compressor recognizes a level increase and does something about it. Too slow a setting will let the peaks through, which can defeat the whole idea to begin with; too fast a setting will flatten the dynamics and sound artificial. I start with around 1 to 5 milliseconds for voice and maybe a little longer for percussion that I want to sound big but not too big. By watching a peak level meter, I adjust the attack time to start trimming back the LED meter while still keeping the character of the sound.

The Decay setting is how quickly the compressor returns the signal to normal. This can usually be set from 50 milliseconds up to several seconds or more. A very short decay can cause pumping where you hear the signal level changing up and down

Featuring reviews from dbx, Requisite, and Focusrite. Plus a Peavey First Look.

with each beat, while a very long decay acts more like an auto gain control. For radio voice work I use about 100 to 300 milliseconds decay to start, which is pretty quick, but helps to keep the talent from stepping on their own next word. Bass guitar is tricky, since low-frequency notes themselves can modulate the compressor if the decay isn't set long enough. Usually around 300 to 500 milliseconds works for most applications, but again you must let your ears be your guide. For general overall compression you can use up to 1 second or more of decay time, since this will simulate someone riding the faders on a final mix.

HOOKUPS

Like any signal processor, a compressor must be hooked up into the audio chain correctly. Unlike parallel processors such as reverbs, however, compressors need to work serially, in-line with the audio. This requires you to "break" the signal path and insert the compressor. Usually this is accomplished with a 1/4-inch phone TRS (Tip-Ring-Sleeve) plug. A "Y" cable is plugged into the input and output jacks of the compressor and the TRS end is hooked into the board. (See fig. 1.) You usually have several areas in the signal path you can use. Which one you pick depends on what you're trying to accomplish. Note that some compressors have two selectable operating levels, with the –10 setting being for the channel inserts in prosumer boards, and the +4 setting for the line-level outputs and channel inserts on professional boards.

CHANNEL STRIP INSERT

The first possibility is the preamp channel strip. This insert is generally after any pad or trim controls on the channel strip, but before the fader. This keeps the levels feeding the compressor constant, even though you may be moving the fader. In this mode, it's one compressor per instrument, so it's normally used for things like bass guitar, lead vocals, and kick drum. Remember that usually the prefader monitor sends from each channel will probably be affected by any changes you do to the compression, which will be heard by the musicians in their monitors. This can be a bad thing if you try to patch in a compressor on a problem vocalist after a soundcheck or in the middle of a show.

SUBGROUP INSERT

Another possibility is inserting your compressor into the subgroup bus. The quintessential application uses a Mackie 8-bus board. Compressors can be inserted into each of the subgroup busses and then various instruments are assigned to each bus as desired. For instance, you could have bus 1 assigned as your vocal subgroup. Any vocals that need compression are sent to bus 1 first, where they are processed by the compressor and then sent on to the output bus. Bus 2 could be for bass guitar, bus 3 for acoustic guitars, and so on down the line. This works very well in a live situation since you can quickly "patch in" in a compressor on the fly without crawling around the back of the board, and also avoid affecting the sound level being sent to any prefader monitor sends, which, as mentioned before, can cause rebellion on stage.

In a perfect world, we wouldn't need compressors. In fact, on a lot of the symphonic recordings that I do, it's a big no-no because we're trying to capture as much of the dynamics as possible. Therefore, any compression is seen as altering the reality of the performance. But you probably can't get and don't want real-life dynamics on tape.

STEREO BUS INSERT

Another place to put your compressors is in the stereo output bus. You can either do this with a traditional insert cable, or externally with a compressor that has XLR connectors and a +4 dBm operating level. You probably want to place the compressor before any equalizers or other enhancers such as a BBE or Aphex unit so it won't exaggerate certain frequencies. If you're setting it up as a control on the sound system level for visiting engineers, though, then placing a brick wall limiter just ahead of the crossovers may be what's called for. Be aware that this can be made to sound pretty awful, but it's a lot better than reconing speakers.

LEARNING TO HEAR COMPRESSION

To learn how to hear compression, you just need a peak responding meter of some sort (an LED array on your mixing board or DAT recorder works fine), a regular mechanical VU meter (which shows the average level), and your ears. Insert the compressor so that it affects what's going to both your monitors and the meters. With the compression set at minimum, play back a CD that has little or no compression, such as a live Celtic act or percussion group. You'll see that the meters really dance. Notice that while the peaks can be hitting very close to 0 dB on the LED, the slower acting VU meter is very low, maybe at –20 VU. As you bring up the compression, the peak levels shown by the LEDs will be reduced, while the average levels on the VU meter will rise. If done properly and in the correct amount, it will be unnoticed by most listeners. In fact, it may be perceived to be a higher quality mix, since most "regular" people are used to listening to the very compressed sound of radio stations. It has now become the "nominal" sound you need to emulate in your live performances.

THE OTHER COMPRESSORS

Every electric guitar player has a compressor — and may not even know it. You probably guessed it — guitar amplifiers (especially tube-based designs) are really big compressors run in true limiting mode. Sure they add in a lot of distortion and equalization, but the sustain craved by guitarists is only there because the amplifier has been topped out and won't go any louder. In the "old days" — before master volume controls — this could be a pretty painful since a 100-watt Marshall amplifier run into clipping was pretty darn loud indeed. Now, though, modern guitar amps use a master level control, which allows you to run the preamplifier into clipping first to get the limiting action desired, but at a low sound pressure level since the output stage of the amplifier doesn't need to be driven into clipping.

THAT'S ALL FOLKS

We all need to listen, listen, listen. Examples of good sounding compression are all around us. If you have a favorite mixing engineer (I happen to really like the mixes from Bob Clearmountain), it's because they've got their levels (among other things) under control. It's all in the ears. ➤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Compressor-Limiter

dbx Blue Series 160SL

BY WADE MCGREGOR

If you take compression seriously, then you must check out the new dbx 160SL 2-channel compressor-limiter. The newest addition to the Blue Series of dbx products brings no-compromise performance and creative control over compressor-limiter parameters to the working professional.

The distinctive blue dbx 160SL follows on a long history of excellent compressors, but makes use of the latest VCA technology from dbx, the V8. The VCA is the heart of this analog dynamic processor, and the dual proprietary V8 VCA modules feature an extended dynamic range (127 dB) and extremely low distortion. The result is a high-performance unit that will provide long-term service in any demanding audio production environment. In fact, the audio performance and control features of this compressor will enable the 160SL to easily co-exist with the rapidly expanding dynamic range of digital audio, without any apologies.

MANUFACTURER: dbx Professional Products, 8760 South Sandy Parkway, Sandy, UT, 84070. Tel: 801-568-7660. Web: www.dbxpro.com

APPLICATIONS: Full-featured compression for any audio production — live, studio, or post-production.

SUMMARY: Intended for sophisticated compression users, the 160SL provides all the necessary controls for both the creative uses of compression and quick fixes to overly dynamic mixes.

STRENGTHS: An excellent AutoVelocity® mode that allows control over the sound quality of compression; capable of +30 dBu output levels; no-compromise audio design; ultra-fast attack and release times; excellent interface; not cheap.

WEAKNESSES: No allowance for –10 dBV inserts; not cheap

PRICE: $2799.95

EQ FREE LIT. #: 126

The 160SL has the fastest attack and release parameters of any compressor I've ever encountered. This opens a whole new area for creative application of dynamic control, especially for percussion instruments. The attack and release can be set so fast (attacking at rates up to 400 dB/millisecond and releasing at up to 4000 dB/second) that the transient attacks of a hand drum can be reduced without the usual pumping of the ringing drumhead. You can actually adjust the relationship between the hand slap and the ring using the Ratio control (variable between 1:1 and infinity:1). You can also adjust the amount of attack in piano or guitar parts, independently of the body of the note. This is especially important when recording digitally, as the transient will set the absolute level for recording the instrument but the tone will determine the subjective level of the instrument in the mix. This recording situation generally creates a compromise when you want a percussive instrument to feature prominently in the mix.

The two-rack space, stereo unit features the switchable Overeasy® compression threshold that dbx introduced to the audio industry many years ago. It also includes a threshold control for a more recent dbx feature called PeakStop®. This offers a convenient method of limiting the absolute output level of the unit, independent of any compression settings. PeakStopPlus® is a switchable mode that adds Instantaneous Transient Clamp (no peaks rise over 2 dB above the PeakStop threshold) to what dbx terms "Intelligent Predictive Limiting" — a method of comparing the short-term signal history to reduce the audibility of this aggressive limiting function. PeakStop is well suited to processing analog signals prior to A/D conversion, as it reduces the potential for clipping while allowing the overall signal level to be as high as possible. PeakStop can be set anywhere between +4 dBu and +30 dBu relative to the unit's output (post make-up gain) and offers both protection and potential for generating interesting sounds from acoustic guitars, raspy wind instruments, bright snare drums, and so on.

Despite offering controls for each parameter, the unit also has the inevitable Auto mode for the set-and-forget signal-dependent attack and release time constants. Unlike other compressors, the 160SL offers the user control over this useful operating mode. With the new AutoVelocity® mode engaged, the Attack and Release controls set the fastest allowed attack and release settings, but the signal waveform will determine when this upper limit is necessary. The result is a compressor that can selectively compress fast transients from, say, a percussive bass playing style without any audible pumping of the sustained low notes.

Adding compression to a complete mix can now be done using AutoVelocity, allowing the user to determine the speed of transients that should be untouched by compression while still preventing low-frequency fundamentals or sustained notes from determining the upper limit of the attack and release times. For those who prefer the more conventional auto-mode, an internal switch (for each channel) will bypass this feature. AutoVelocity is available as an upgrade to owners of the 160SL and may alone be enough to convince some people they need this unit!

The blue front panel is made from aircraft aluminum. All of the solid aluminum control knobs are clearly identified and easy to set. These controls also behave the way you would expect on a high-end unit. For example, threshold does not affect the input level; stereo-coupled mode allows all parameters to be set on channel 1 (channel 2 controls, except output level, are disabled, with channel 2 LED indicators going off, just to make it obvious); and all the pushbuttons have LED indicators to show the unit's status from a distance. Two nice, big analog VU meters with peak indicators are independently selectable among input, output or gain reduction.

The rear panel is also fully featured, with gold-plated Neutrik XLR connectors for inputs and outputs including the sidechains for each channel. Pushbuttons allow the selection of connecting XLR Pin 1 of the actively balanced input to the ground bus, unbalancing the output transformer (connecting XLR Pin 3 to Pin 1, thereby reducing output level 6 dB) and lifting Pin 1 on the output XLR. There is a pair of binding posts with a jumper that allows the installer to choose between grounding the audio to the unit's chassis or separating these two ground planes. The power cord is a detachable IEC-type with the fuse accessible immediately below, and is connected to a well-shielded, substantial +/–24 V DC power supply inside. A

BASF

STUDIO OBSERVER

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VOLUME ONE NUMBER FOUR

When the Stones Come to Your Studio

STUDIO LIFE DURING THE MAKING OF BRIDGES TO BABYLON

Let's face it: The Rolling Stones are not your normal recording outfit. After selling a zillion albums and establishing a reputation that stretches from London to the moon and back, the Stones don't need to abide by conventional approaches to anything. Of course, they never have.

Reading the liner notes to their latest studio collection, *Bridges to Babylon*, it appears that about half the recording industry was involved in the event. Can you recall a single CD that draws together names like Don Was, Ed Cherney, the Dust Brothers, Babyface, Rob Fraboni, Tom Lord-Alge, Bob Clearmountain, Jim Scott, Dan Bosworth and a comparable cadre of session musicians? That's the Stones – a one-of-a-kind operation, and everyone wants to be involved. The same is true for studios, with credits going to Ocean Way, Record Plant, Conway, Encore, Maison Rouge, Pie, Sarm West and Westside studios.

Ocean Way's Allen Sides and Claris Sayadian adjusted to 'vampire hours' for the Stones project.

So imagine that the Stones want to record their next CD at your studio. What would you do?

Studio Manager Claris Sayadian of renowned Ocean Way Recording in Los Angeles first began to sense an X-Files-worthy adventure brewing last January when she was approached by super-producer Don Was.

"You know there's a major production coming up when Don Was personally comes to your office and checks the schedule," Sayadian says. "He was very vague about it. He just said, 'Hold on to three of your rooms for a few months.' When he finally told us it was the Stones, it was very much, 'Let's just keep this hush-hush.' It kind of felt like being pregnant with triplets and having to hide it."

Why were the Stones coming to Los Angeles to make a record? According to the band's publicist, Mick felt there would be fewer distractions than in New York. L.A. is a quieter town at night, and they thought this would allow them to get more work done.

There was also a desire to make this a more communal record, as Keith had wanted to expand from their usual isolated world in the studio to include more musical ideas. And the ideas came roaring in, from some of finest and most diverse session players in Los Angeles, including Me'Shell Ndegeocello, Jim Keltner, Blondie Chaplin, Waddy Wachtel, Wayne Shorter, Kenny Aronoff, Benmont Tench and Billy Preston.

Running the studio around such an intense project was no small logistical concern, according to Sayadian. "Once they came in I talked to Pierre (de Beauport), Keith's right-hand man, and Don's assistant Jane Oppenheimer to plan some measures for the security and privacy of everybody, including their many guests," she says. "That meant everything from setting up valet parking to literally giving them the key to the building. All three rooms were pretty much in full production for about four months. We locked out the entire studio for them the whole time.

"We had to adjust to what they called vampire working hours," Sayadian adds. "Basically, they started late in the afternoon and would go until five or six in the morning. They were tireless. In a way it was easier that they had the whole building, rather than if they were in

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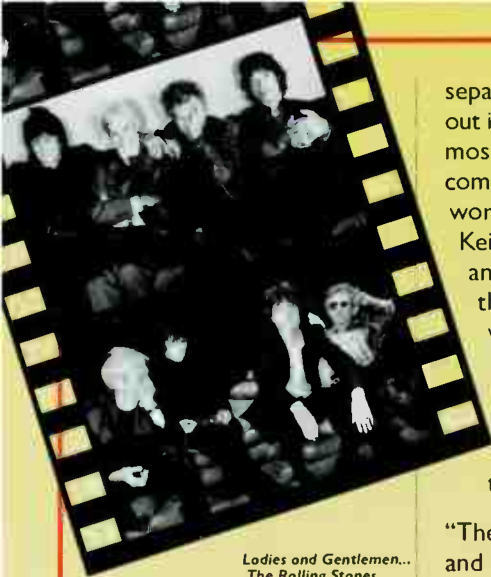

Ladies and Gentlemen... The Rolling Stones.

Continued from previous page

one or two rooms and we had another client in, because we were literally doing a 24-hour shift around them."

Don Was chose his frequent engineering partner, Ed Cherney, to get the sessions started. "Don called to ask me if I would start the band out tracking while they started working out songs and arrangements," Cherney says. "At first he just wanted me to come in and work for a week. But that turned into five weeks.

"Basically Don just wanted me to do my thing and capture the band. I've had a lot of experience of late with the Stones because I went on tour with them and recorded the *Stripped* album. We had recorded in everything from a little studio in Tokyo to hundred-thousand-seat stadiums. I had a lot of experience recording them in live situations, and that was the idea on this record – setting up the band and recording them live."

Cherney continues, "Initially it was going to be Keith and Charlie and Darryl Jones, the bass player, and Mick coming in and just putting together what this album was going to be. For at least the last year, they were in separate studios working out ideas for their songs, but most of the songs weren't completed when we started working on them. Maybe Keith would have a verse and a chorus, and then in the studio they would work out things like Keith's signature licks, and find the hook, and work on the structure and arrangements of the songs.

"They basically just set up and let me do my thing. If it didn't sound right, Keith would hit me over the head with his Stratocaster, and if it did, it was 'Ed, we love you.' Typically, I was just to capture them and not miss anything they were doing. That was really the mandate – to record every lick that everyone played.

"You really had to be on your toes because you never knew what was coming next," Cherney adds. "You never really had the opportunity to sit with someone and get a sound. You didn't know who was going to show up or what song was going to be played. There wouldn't be a countoff. The song could start from anywhere in the room on any instrument, and you'd better catch it. There was certainly a lot of chasing the music."

"We never knew what to be ready for," says Sayadian. "At 3am Mick would be working in Studio Two with his producer and they'd decide that they wanted to change the tracking session to Studio One. So you can imagine the runners and the assistants frantically running the mic stands and the mics into the next room. It was very spontaneous for the most part. We had a couple assistants who requested early retirement after those sessions."

Fundamentally, though, it was a straightfoward rock and roll session, according to Cherney. "Get the guitars roaring in front of an amplifier without any effects, and get Charlie pumped up and sounding good. Give Mick a good mic with enough isolation characteristics that he can move around in the room with the musicians, and just try to capture what the Rolling Stones do."

About 400 reels of two-inch tape were used during the sessions at Ocean Way because nearly everything went down live in all three rooms. "We recorded to analog on the Ampex ATR124 (audio tape recorder)," says Cherney, "and started out with Ampex 499 tape. As it progressed we discovered the BASF 900 and moved over to that. The Ampex is a fine product but, for this project, everyone kind of felt that the 900 sounded better. In the end it turned out that about half the album was recorded on 499 and the rest on 900.

Engineer Ed Cherney in the thick of it

"When we got takes we were going to be using, we transferred over to the Sony 3348 digital to do overdubs and manipulate it digitally. I mixed down to half-inch 30 ips analog onto BASF 900 tape, with a DAT backup.

Considering that a studio like Ocean Way and an engineer like Ed Cherney spend their time working with top artists from around the world, is there still something extra-special about being in the studio with the Stones? "It just doesn't get any bigger than that," says Cherney. "The sounds they make are so identifiable, it's kind of like your life flashes before your eyes when they're playing in the studio.

"Sitting in front of them, Don and I would sometimes look at each other and start to giggle – afraid that we were going to be found out as imposters and summarily thrown out." ■

– David Schwartz

TROY HIGHTOWER Selected Discography

SWV
On & On
BMG /RCA Records

Jodeci
"Get on Up" (remi
Uptown/MCA Recor

Warren G.
"I Shot the Sheriff" (
G Funk/Def Jam Re

Redman
Muddy Waters (gold
Def Jam Recor

D'Angelo
"Those Dreamy Eyes of
EMI Recor

George Cli
"Everybody Get Funk
East West/Sony

LL Cool J (w/ B
"Hey Lover"
Def Jam R

IN SESSION WITH TROY HIGHTOWER

Troy Hightower is a busy man. In the last three years he has engineered 980 recording sessions for an encyclopedia of rap, hip-hop R&B and dance artists. **Onyx, SWV, Keith Murray, D'Angelo, Redman, LL Cool J, Erik Sermon and Methodman** all have looked to New York-based Hightower to work his recording and mixing magic on their music.

Hightower, whose album credits bear an uncanny resemblance to the **Billboard** rap and R&B charts, records and mixes on analog tape. "Analog tape sounds warmer than digital, and I especially like the newer high-output tapes. I choose the best tools I need to do my job. That's why I like to record on, and mix to, analog tape. It just sounds better."

Studio Observer asked Hightower about his experiences with **BASF** products. "In 1996 I mixed an album in Europe for Portuguese rap group **Mind Da Gap**," he recalls. "It was all done on **BASF** tape – **SM 911**, I believe. I liked what I heard, and the tape handled really well. Later I had a chance to work with **SM 900**, as well as the other brands of high-output tape. This tape, **SM 900**, does something to the mix that makes it sound better, meatier, on the low end. Yet it still sounds cleaner than the others."

Hightower continues, "The last album I did for **Common Sense**, *One Day It Will All Make Sense*, was mixed to **SM 900**, and it just sounds great. I'm about to begin work on a new album for European artist **Boss AC**, and I plan to record it entirely with **SM 900**. It's my first choice in recording tape."

Hightower got his break in the business while working as an assistant engineer at **The Apollo's Harlem recording studios**. One night, he was asked to fill in for an absentee engineer. The song he recorded and mixed that night became the gold single **"Throw Ya Gunz"** by **Onyx**. The work hasn't stopped since.

Although Hightower is well known in **New York** as one of the hottest engineers for rap, **R&B** and hip hop artists, he is driven to expand his base of clients. "There's still Europe and the rest of the world," he says. "And I haven't won a **Grammy** yet."

"Yet" is the operative word. Armed with abundant talent and skill, limitless determination, a winning personality and **BASF** tape, Hightower is sure to achieve his goals.

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FROM THE TOP

Do you know which company invented commercially produced magnetic tape? Or which company manufactures tape and data media in full compliance with the world quality standard ISO 9001?

The answer to these questions is EMTEC Magnetics, maker of BASF-brand studio, broadcast, duplication and data media. With worldwide sales of more than $1 billion, EMTEC is a leader in production and development of the highest-quality media products available anywhere.

Now, the sales and distribution team for the United States, Canada and Latin America, formerly known as JR Pro Sales, has been brought under the umbrella of EMTEC Magnetics GmbH and has commenced doing business as EMTEC Pro Media Inc., as of January 1, 1998. What does this mean to our customers? It means a stronger organization with more resources to meet your needs today and tomorrow.

If you use BASF-brand tape products, thanks for your confidence in us. If you haven't tried the BASF brand, visit your nearest BASF dealer or our Web site (www.emtec-usa.com) to request information.

I promise you, we're listening.

Joe Ryan

President
EMTEC Pro Media, Inc.

TAPE TALK

JEAN TARDIBUONO

National Sales and Marketing Manager
Studio and Broadcast Products

SURF CITY

Why do people surf the Internet? Entertainment and information are the two main reasons. We at EMTEC have a Web site that serves as a solid resource for anyone involved in recording today. In addition to basic information about the range of BASF professional studio, broadcast and duplication products we offer, there's information about the best way to store your tapes, back issues of The Studio Observer newsletter, press releases about our users and products, user spotlights and links to a number of other sites. There's also a company history, including a soundbyte from the world's first music recording made to tape in 1936; recorded to BASF tape, it still plays back today.

We've also added a page for our popular "Win a Case of BASF Tape" contest, so you can enter the contest via e-mail. I hope you'll take a few minutes and check us out on the Net (www.emtec-usa.com). Supplying the best media and the best information – that's our goal. ■

WIN A CASE OF BASF TAPE!

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It's easy to win a case of BASF ADAT, DTRS, DAT or Reference Master tape. Simply send us a letter, fax or e-mail letting us know two things:

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AND

2) Why you are using it (price, quality, availability, etc.)

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FAX: (805)295-5554 · EMAIL: info@emtec-usa.com

Five entrants will be drawn at random on April 30, 1998 and will receive one case of tape in any of the following formats: ADAT Master, DTRS Master, DAT Master or Reference Master.

Employees of EMTEC or any authorized BASF dealers are not eligible to participate. Limit one winner per studio or household.

large ground lug completes the brutish broadcast quality of the rear panel. A small blank panel on the rear of the unit is reserved for 24-bit AES/EBU interface that should be available this spring.

MANUFACTURER'S SPECIFICATIONS

INPUT CIRCUIT
THD: Less than 0.002%
Input CMRR: Greater than 100 dB
Maximum Input Signal Level: Greater than 30 dBu
Bandwidth: Greater than 200 kHz

V8 VCA
Dynamic Range: 127 dB max (unity gain)
THD + Noise: Less than 0.007%
Bandwidth: Greater than 200 kHz

OUTPUT CIRCUIT
Output Drive Capability: Greater than 30 dBm
Output Stage THD: Unmeasurable (What was the limit of the measurement system?)
Earth Potential Isolation: Greater than 200 Volts DC
Output CMRR: Greater than 110 dB
Bandwidth: Greater than 200 kHz

OVERALL UNIT SPECIFICATIONS
Dynamic Range: Greater than 125 dB
Head Room: 26 dB (relative to +4 dBu)
Maximum Signal Level: 30 dBu input or output
Signal to Noise: 99 dB reference = +4 dBu
THD + Noise: Less than 0.008%
Frequency Response: +0/–3 dB, 2 Hz–200 kHz; +0/–0.25 dB, 20 Hz–20 kHz
Deviation from Linear Phase: Less than 0.5degrees, 20 Hz–20 kHz

Over the years, we have seen dbx produce a lot of compressor-limiter products, many of which were focused on making this useful process more accessible to those with minimal experience with dynamic processing. However, the dbx 160SL offers an excellent compressor-limiter to those who know exactly what they want. The excellent audio, colorless signal chain, and extensive parameters offered will give even those with a full complement of processing a unit worth listening to. You know you are serious about compression, if you can't let the demo unit go back after your first few sessions....

Wade McGregor is a principal consultant for MC Squared System Design Group based in North Vancouver, BC.

Requisite Audio Engineering L1 Tube Optical Limiter

BY MIKE SOKOL

I bought a UREI LA3A about 10 years ago. Some guy was getting rid of it for a hundred bucks 'cause it didn't "have 1/4-inch phono jacks on the back, just those darn terminal strips...." I had some "semi-pro" compressors in my studio rack at the time that had been used on a bunch of projects ranging from vocals to percussion. The first time I patched in the LA3A I was amazed at how far the compression meter would swing without the music actually "sounding" compressed. You could have the meter pegged at –20 dB and the sound just seemed to get fatter, not flatter. Now, for those with a nose for classic gear, the transistor-based LA3A was preceded by the LA2A, a tube-based version with an optical-based gain path, so named because the compression gain control is via a photo-resistor and light source. [*See Eddie Ciletti's Maintenance column in this issue.*]

Requisite Audio Engineering has entered the classic compressor field with a beautifully executed version of the original LA2A theme. First off, let me say that this thing's heavy — 18 pounds to be exact — due largely to a huge toroidal power transformer and its needed support. All wiring is hand soldered point-to-point and gold-plated tube sockets are used. In the back is a rectangular pocket that

MANUFACTURER: Requisite Audio Engineering, 2645 East Glenoaks Boulevard, Glendale, CA 91206. Tel: 818-247-2047. Web: www.requisiteaudio.com

APPLICATION: Mono optical compressor-limiter for studio use.

SUMMARY: If you like the sound of a UREI LA2A or LA3A comp-limiter, you're gonna love this.

STRENGTHS: Opto-isolator gives classic LA2A sound, but with improved frequency response during heavy compression; balanced input and output transformers with +4 levels for professional interface; Tube Access Channel allows tubes to be replaced without removing cover and improves cooling so that units can be vertically packed in a rack without heat buildup.

WEAKNESSES: Tubes don't have retainers, which could allow them to loosen in a road rack.

PRICE: $1895
EQ FREE LIT. #: 127

Compressor-Limiter

they call a TAC (Tube Access Channel) that not only allows you to change the tubes without removing any covers, but provides for ventilation from the rear. I only wish they put those little wire hold-downs on the tubes like on some of my early tube amps — the sockets do seem to substantially grab the tube pins, so maybe I'm just paranoid about loose tubes rolling around.

The front panel has Peak Reduction and Output Level controls (just like on a LA-2A), plus a selector switch for 3:1 compression or 10:1 limiting, Bypass/On switch, and power On/Off. The limiter setting has a step in the knee of the compressor curve, where it functions as a 3:1 compressor from about –30 dB to –20 dB and then goes into 10:1 limiter mode, the so-called "soft knee" curve. This is one of the reasons you can get away with huge amounts of compression and still sound musical. A link connector on the back panel (and associated switch on the front) allows a pair of L1's to be ganged together for stereo use. As one would expect, there's a custom, illuminated Sifam VU meter that can be switched between output level and gain reduction modes. Six-hundred-ohm input and output transformers and XLR In/Out connectors complete the package. The front panel appears to be a gold anodized, brushed aluminum plate that's two rack units high [3/16-inch milled aluminum and steel chassis].

Since the designers claim the L1 sounds like an improved LA2A, I patched it into a stereo feed with my LA3A on one side and the L1 on the other. My thoughts were that the L1 should match up to the improved frequency response of my LA3A while still offering some of the benefits of tube-based processing, and I was right. After running a variety of solo program material though it, including Dobro, guitar, narrative voice, and singing, I was satisfied that I could get the really a deep limiting action in excess of 30 decibels without any noticeable ar-

FIRST LOOK

PEAVEY VC/L-2 VALVE COMPRESSOR

BY STEVE LA CERRA

Peavey's VC/L-2 vacuum tube compressor/limiter joins the company's VMP-2 as the second valve-based processor in their AMR (Audio Media Research) line. A 2-channel unit (in a two-rackspace chassis), there are no solid-state devices in the audio path of the VC/L-2, and the circuit design features Peavey's exclusive OptoDynamics topology. According to Peavey, [using] OptoDynamics results in a more transparent sound than could be otherwise obtained by using opto-isolators, opto-couplers, or opto-diodes to accomplish gain reduction. Instead of using one of the more typical, aforementioned devices, the VC/L-2 employs an EL84 tube and an electro-luminescent panel. Peavey maintains that this approach overcomes several traditional compression problems, including poor response to transients in bulb-based gain reduction circuits (bulbs take time to heat up and produce light), and erratic response curves in LED-based gain reduction circuits (LEDs turn on and off abruptly).

An electro-luminescent panel is basically a high-voltage light source that becomes brighter with increasing voltage input. Unlike LEDs or opto-devices, the time constants (attack and decay) of its operation are favorable to audio applications — an electro-luminescent panel provides faster response to transients than a bulb, yet a more smooth attack and decay characteristic than an LED. In the VC/L-2, voltage level of the audio input is stepped up to the neighborhood of 100 volts via one of the tubes in the input stage, so that audio can drive the panel. As audio input modulates, so does the brightness of the panel. Gain of the audio circuit is controlled by a CDS (cadmium sulfide) cell that acts as a light-sensitive resistor circuit. When the panel's brightness varies, so does the resistance of the CDS and we have a gain change in the circuit. Why go to all that trouble? Because classic compressors such as the LA2A used electro-luminescent panels to accomplish compression — and they sounded great!

In addition to the OptoDynamic system, Peavey uses two 12AX7 tubes (configured in four stages) on the input side, and a two-stage 12AT7 for the output section. The output stage is capable of producing a maximum level of +20 dBm. Front-panel controls for each channel include a VU meter (switchable for output or gain reduction levels), rotary controls for threshold and gain, and a compress/limit switch. On the rear panel of the VC/L-2 are a set of transformer-balanced input and output connectors, as well as 1/4-inch instrument- or line-level I/Os.

The Peavey VC/L-2 Valve Compressor Limiter is now shipping at a suggested retail price of $1249. For more information, contact Peavey Electronics at 601-483-5365, Web: www.peavey.com. Circle EQ free lit. #128.

tifacts such as distortion or frequency response problems. I then patched it into the stereo output bus in 3:1 compression mode and tried a variety of premixed stereo programs. As a fin-

MANUFACTURER'S SPECIFICATIONS

Frequency Response: 10 Hz–110 kHz
CMRR: Greater than 60 dB (@ 10 kHz)
THD: Less than 0.05% @ +4 output
S/N: –79 dB @ maximum gain
Total Gain: 30 dB
XLR input: 600-ohm balanced
Attack Time: 10 µSec.
Release Time: approx. 0.07 sec. for 50%, 0.5 sec. to 5 sec. for complete release program dependent
Size: 19" x 3.5" x 8"

ishing compressor, it was outstanding. A pair of them would make a nice "mastering" compressor to tame those nasty peaks before you go to CD. The sound was really effortless, which is a tribute to both the original design and contemporary execution of the L1.

Does the world really need another LA2A clone. Maybe, maybe not. Nonetheless, Requisite Audio certainly makes a strong case for updating and improving a classic design. And, after talking to the designer-builders, their efforts seem to be a true labor of love. If you can find an original LA2A, you'll likely pay more for it than for a new L1, and you still may have to deal with dried out caps and dirty connections. With the Requisite L1, you can have your cake (tubes?) and eat it too.

Focusrite Green 4

BY MIKE SOKOL

Compressor-limiters are one of the most important tools in your audio arsenal. Remember: All audio levels will, eventually, be compressed, either by you, the engineer, riding levels during a live production or recording session, or by a dedicated compressor/limiter in your recording chain — or at a radio station's transmitter a few nanoseconds before broadcast. Compression happens.

Properly applied, a good compressor can really reduce an engineer's workload during a performance. Once you have the desired levels dialed in (the so-called "mix"), the only way to keep from chasing all the level changes of a live performance is to patch in enough compressors on the proper channels and subgroups. A good compressor is a thing of beauty. Alas (did I just say, "Alas"?), compressors are also one of the least forgiving links in an audio chain. Since they are artificially distorting the dynamic range of the original audio program, there's lots of ways make things sound really bad. The most obvious compression artifacts are an audible "pumping" of the audio program, or a harmonic distortion during heavy compression, which tends to tax the internal headroom of the compressor's circuitry.

I've got a dbx 160, a UREI LA3A, and a UREI 1176LN in my normal processing chain. All three can provide outrageous levels of effortless compression without introducing audible problems. I didn't realize how good I had it until I borrowed some inexpensive compressors for a few projects. All of a sudden my previously carefree usage with the classic compressors turned into a real struggle using the inexpensive consumer products. It was no

Compressor-Limite

fun at all. With the bargain units there was a very small window-of-opportunity to get a decent sound that didn't "sound" compressed. The UREI's take anything you throw at them and ask for more. That seems to be the main quality in a great compressor; effortless level control that doesn't sound like it's being compressed.

In compressors, you usually get what you pay for, and if I want to buy any more UREI compressors, the going price is about 30 times what I paid for them six years ago. So when *EQ*'s magnanimous executive director, Hector La Torre (yes, we are instructed to genuflect in His presence), offered to send me one of the new Focusrite Green Series compressor/limiters, I was more than a little interested. This was going to be fun. [*Sorry Mike, nice try, but you can not keep the unit....—H.G.L.*]

Focusrite's Green series of audio processors is promoted as a more affordable version of their highly acclaimed Red series. This still

MANUFACTURER: Focusrite Audio Engineering Ltd., England. Tel: +44 (0) 1628 819456/44 1494 462246. Distributed by: Group One Audio, 80 Sea Lane, Farmingdale, NY 11735. Tel: 516-249-1399. Web: www.glltd.com.

APPLICATION: Two-channel compressor-limiter for field and studio use.

SUMMARY: Focusrite has taken the design expertise from their Red Series product line and produced the Green Line Series at a lower price. Compact, 2-channel, 1 RU chassis design. Brings classic sounding compression to the masses at an affordable price. Tastes great, less bucks.

STRENGTHS: Class-A circuitry for transparent sound with tons of headroom; offers deep compression without audible artifacts; on-board sidechain equalizer for de-essing without external patching; separate limiter and compressor functions; selectable +4 and –10 dBU proper level.

WEAKNESSES: The rack-mount ears use only two screws on diagonal corners, which looks artsy, but four screws make for better road usage; lots o' heat as the class-A circuitry draws some serious power for a signal processor (about 35 watts); the small heat sink on the back of the chassis needs ventilation, so don't close-pack this unit in a tight rack.

PRICE: $1599
EQ FREE LIT. #: 129

doesn't make the price cheap, but certainly more affordable. In short, you still get a very professionally built product with a world-class sound, but on a budget.

THE BASICS

The Focusrite Green-4 compressor-limiter is a 2-channel, 1 RU processor with all the


www.dynaudioacoustics.com
IF IT'S THERE
YOU'LL HEAR IT
dynaudioacoustics
Ultra Precision Studio Monitors
CIRCLE 11 ON FREE INFO CARD

standard controls — threshold, ratio, and make-up, as well as attack and release times — expected in a professional unit. A Soft Knee switch changes the compression action from a hard-knee absolute curve to a kinder, gentler curve, allowing higher levels of compression while still sounding natural. (The action of the soft-knee compression reminds me a lot of my dbx 160X.) An Auto-Release switch provides a release time that changes with the program dynamics, effectively slowing down the release of very loud material to reduce audible pumping, and doing a quicker release on less dynamic program. There's a separate control for hard limiter action, which effectively allows you to do compression below a certain audio level but still acts like a brick-wall limiter as a safety net.

The really cool knobs are the sidechain filters labeled "S/C Filter." This is a built-in sidechain equalizer for working on problem material. These two knobs form a variable-frequency low-pass and high-pass notch filter, allowing you to select a frequency band that needs help, while leaving the frequencies outside of the pass-band unaffected. Traditionally, a sidechain is a breakout point in the sensor part of the compressor's VCA (Voltage Controlled Amplifier) circuitry. By patching this port into an external equalizer, you can vary the compression at various frequencies without affecting the equalization of the sound passing through the processing device. For instance, one application might be to de-ess vocals.

On the back panel are balanced XLR connectors with a switch for –10 or +4 dBU operation. (According to the manual, a version with 1/4-inch phone jacks is also available.) On the front panel are switches for stereo-link and output-level or gain-reduction meter selection. All in all, it includes all the appropriate switches and connectors one would expect on a professional compressor.

THE SONIC VERDICT

How does this baby sound? Wonderful! In fact, I recently worked with the guest engineer for the act "Eddie from Ohio," who had had some experience with the new BSS compressors. This engineer said he really liked the Focusrite Green 4 better. Patched in on vocals, we could apply very aggressive compression without hearing any artifacts at all. You really had to keep your eyes on the meter to know how much compression you were dialing in 'cause you just couldn't hear what it was doing until you were completely to the wall.

I also tried it on acoustic guitar on a studio session with equally impressive results. Most of my de-essing processing is done in my digital workstation, but I did try the S/C filters on voice in the studio, and they worked as expected, but without having to patch in an external equalizer. You could easily set this up for acoustic or electric bass to limit string popping or to smooth out the lower register dynamics as desired.

This Green 4 would also be a great compressor for classical music where you really need level control to limit the dynamics for live recording or sound reinforcement, but don't want anyone to know what you're doing.

For the money, the Green 4 is a great deal. If you want to get a client's attention and raise the quality of your productions, either in the studio or on the road, put one or more of these in your rack. The Focusrite name will get their attention, and the sound will keep them happy. It's that simple. **EQ**

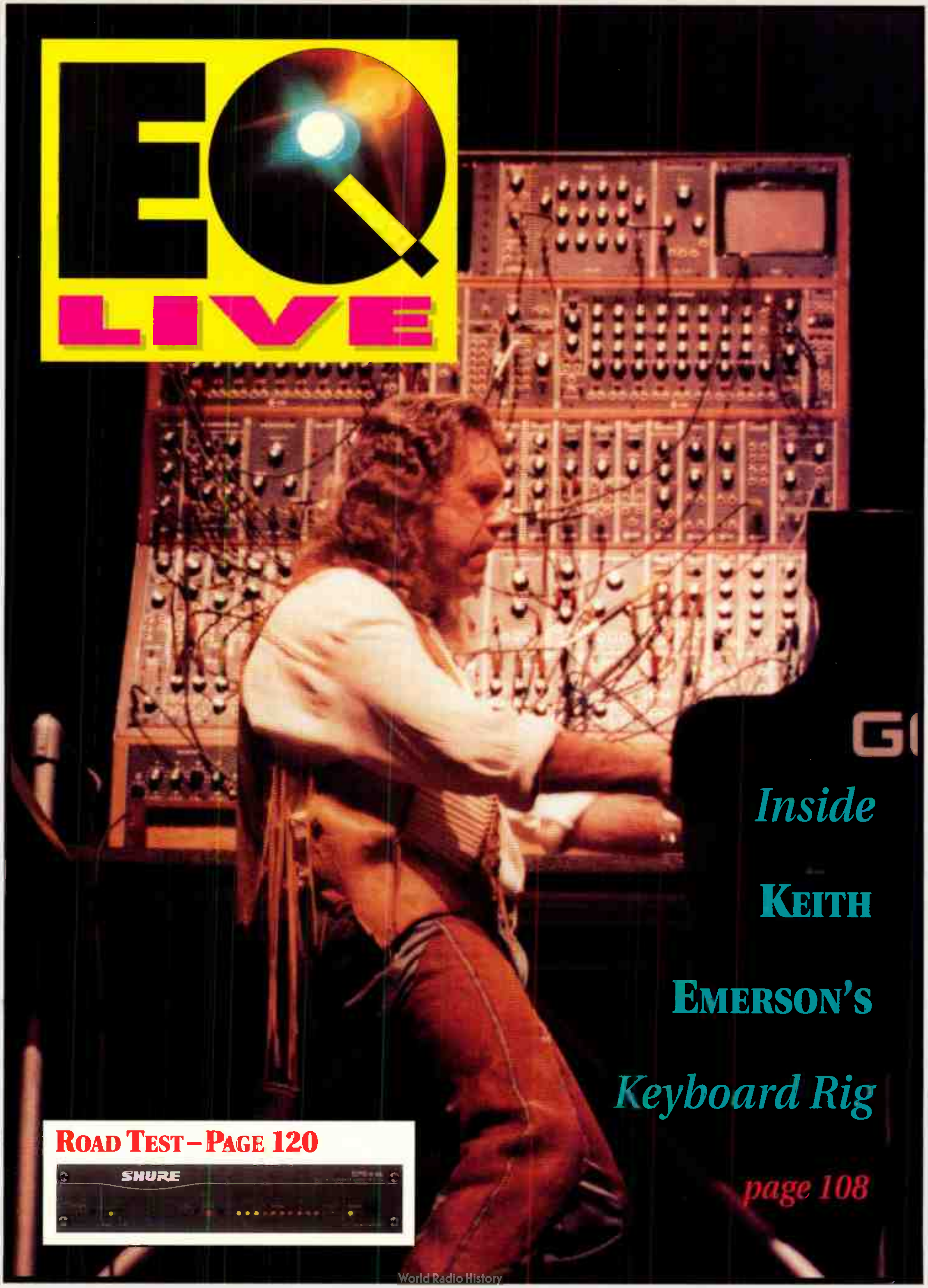

PHOTO BY MARYANN BURNS

EQ LIVE

On 'Board With Keith Emerson

KEYBOARD TECH WILL ALEXANDER REVEALS THE SECRETS BEHIND THE LEGENDARY KEYBOARDIST'S LIVE RACK

PHOTOS BY MARYANN BURNS

By Tony Di Lorenzo

When people talk about keyboards and synthesizers, a few names just have to pop up. Moog, Keith Emerson, Hammond, Will Alexander. Wait, who is Will Alexander? He is the man who has been entrusted with the monumental task of caring for a keyboard rig that's brought us such rock and roll classics as "Lucky Man," "Hoedown," "From The Beginning," "Tarkus," "Pictures At An Exhibition," and "Karn Evil 9" — just to name a few. It is Will Alexander who keeps Keith Emerson's keyboard rig alive and well. EQ recently talked to Will about Emerson's legendary rig and believe me, as a fellow keyboard player, I had more than a few questions! While you're surfing cyberspace you'll need to stop in at www.picaso.net/emerson, but since you're here, read on...

Does someone go to school for a gig like yours?
If you call work experience "schooling," I worked for Oberheim in 1977 or 1978 building OB-X's, Four Voice systems, and OB-1's. I had a math background from college, so electronics came naturally to me. Then when I first heard the Fairlight, and learned that it was a computer making music, I was really intrigued. It was the first commercially available computer sampling instrument. I had an early Apple IIc (before Macintosh) with a couple of floppies in the days before hard disk, and I thought I was flying. Then there were the Alpha Syntauri voice cards that made little buzzy noises, but this (Fairlight) was a serious instrument.

What keyboards and samplers are ELP using today?
For Keith's live rig we're using the General Music Pro2 digital piano, which we both feel has the most realistic digital piano sound, period. The dynamics, the hammer sound, and the dampening modeling on the Pro2 is incredible. We use two Korg Trinity's:

a Trinity Plus and a Trinity Pro, an Alesis QS8, and, of course, the modular Moog, along with a set of Elka DMP18 dynamic MIDI pedals for bass notes, effects, and other things — sort of the third-hand things that need to be done.

There's also an off-stage station that I operate. During the show I do all the program changes because Keith's hands are way too busy. There are no pauses in the music, so things need to be done on the fly. In the rack we have an Alesis QSr and an S4, a Korg Wavestation SR, and a Studio Electronics SE1. On stage it's necessary to do fast program changes, so rather than doing all the programming on the Moog, I use the SE1 for some of the classic Moog sounds. I get a very authentic representation of the Moog sound from that instrument. We're also using an Ensoniq MR rack that has really great French horns. It's all controlled by an Opcode Studio 5LX that I program using Opcode Studio Patches, and run from a Power Book 1400.

What controllers is Keith using on stage?

All of the keyboards except for the Hammond and the Moog are MIDI controllers.

So at any given time he could reach for any one?

When he's playing the piano, he'll play either the QS8 or the General Music piano. The only thing that you can count on is that when he's playing the Hammond or the Moog it's really the Hammond or the Moog. If he's playing any of the other controllers on stage it may be functioning as the instrument or it may just be functioning as a MIDI controller, or it may have a whole bunch of things patched or mapped across the keyboard in order to fit the necessary program for the piece of music that's being played.

Anyone who has ever seen the band live has noticed that Keith certainly has his hands full on stage. The music is complex and so is the sound design. How active are you during the show in helping out with the sounds and the patches?

Everything is programmed. I sequence all the program changes, which is fairly easy. It's more like being in a recording studio and having to punch in on time, all the time. It's actually very easy except for the nervous tension that you have in doing the show.

So you could say that you've become another member of the band. You must need to be as tightly rehearsed as the rest of the band...

[*Laughing.*] I'm not gonna go there. All the people in the crew have a lot to do with what happens on stage and the way that it's presented to an audience.

Do you use any special technique for miking the Leslie?

This time out we're using these new Sennheiser MD504 clip-on mics. We've also been using them on Carl's drum kit. We just clip them right onto the cabinet and we get an excellent sound. I've done all kinds of things with Leslie miking in the studio — all kinds of very interesting combinations. I think one time we even had eight mics on it at different distances and different angles.

You mentioned sequencing patch changes. ELP albums have always been beautifully arranged and, in some cases, heavily orchestrated. How much, if any, sequencing is necessary to get that same sound on stage?

On the past two tours we've used no sequencing at all. It's live. It's the real thing. That's the way ELP likes to do it. In '92 and '93, when we did the Black Moon tour, we used three or four sequences to create the lushness of the *Black Moon* album.

I seem to remember this huge, light-up metronome at the front of the stage and it seemed like the band would just watch that.

That was the count off for the sequences.

Once that was counted off, did they have some kind of feed in the monitors?

Carl (Palmer, drummer) would wear headphones at that time. Once Carl picked it up, the band would follow Carl and Carl would be listening to the metronome.

Will Alexander tries to make sense of all the wires and MIDI commands coming from Emerson's rig.

Keith's Hammond has been modified by the Geoff company. Could you tell us a little about the modifications?

The one that we're using now is the original C3 that Keith bought around 1967 or so. It was the organ that he played on all the ELP tours from the '70s. Technology has come a long way from where it was in the '60s. Parts like capacitors have a much greater shelf life or active life.

Eventually it becomes necessary to take all the old parts out. Capacitors leak or break down, leads break off, and you need to upgrade tone generators. We've had modified percussion and modified vibratos put in. There's an EQ circuit that you can change in a Hammond to beef up the percussion. Al Geoff put in new electronics, a custom interface out to the Leslies, and improved the gain structure on the preamps.

Has the C3 been MIDIfied at all?

We have a MIDIfied C3 that we used on the Black Moon Tour. It's a whole other can of worms to deal with, especially in a live performance, especially the way Keith plays because there's a whole new bus bar with dual-touch contacts. Then you have a J-wire type system. It's like having a whole MIDI keyboard inside the keyboard of the Hammond.

Is the band using any samples?

Yes, we do use samples. In the Alesis QS8 you can burn samples onto

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PCMCIA cards and plug them in the back. So all the samples we're using are stored on PCMCIA cards and plugged into the back of the QS8 or the QSr.

What type of samples?

We use choirs, and we've done some custom sampling. For instance, in the tune "Iconoclast," there is a downward moving line at the beginning of the piece. We worked with Eric Norlander at Alesis and recorded this line with all kinds of special effects, so when it comes to that point all Keith has to do it stomp on the DMP18 bass pedals and this really complex line comes out. It's really great-sounding, with this barber pole-kind of thing happening plus the downward moving scale.

The Modular Moog was in pretty bad shape a few years ago, and you restored it.

I received the Moog in 1990, and that's when I started working on it. It's kind of like working on an old car — there is always something to do on it. It hadn't been operational in almost ten years when I received it.

So the Moog was retired for ten years. What brought it out of retirement?

Keith feels it's an icon in the keyboard world. Even today it has a sense of mystery. People come up and look at it and they don't care about the technology. They could care less about samplers and synths, but they look at that Moog and say, "Wow!" When I got the Moog it was in pretty bad shape. It hadn't been used. There was lots of corrosion on all the contacts and modules didn't work. I have a friend named Gene Stopp who is also a Moog enthusiast, so we brought it into my living room for eight months.

Big living room?

Not really. It was pretty dominant — a great conversation piece, I must say. Our first goal was to get it working as well as possible, as it stood. That meant repairing all the modules and going through it. Obviously there are a lot of 1/4-inch jacks on the thing and a large number of them were in pretty bad shape. The plating on the contacts is a lot better today than it was in '69 or the 70s.

We cleaned all the contacts, fixed the power supplies — they were in awful shape. All the pots on the sequencers were dirty and those are really difficult to clean because they're sealed pots. Just getting everything working electronically was the first thing. There's a programming system that can be used on it to control the filters, oscillator frequencies, mixers, and time constants on the envelope generators. That was in incredible disarray. It's a huge bank of pots that need to be adjusted and they also have voltage followers on them to drive the bus. Those go to the appropriate parameter that needs to be controlled.

So it seems like there was calibration after calibration.

Oh yeah, and the thing is that you'll get everything working, turn the programmer, and the load to the programmer changes the tuning of everything so you'd have to go back and retune it. As you plugged more things into the keyboard control voltage, the tuning changed so we ended up building power supplies in the keyboard so that the keyboard wasn't driven off the power supply of the synth.

So there was no drain on the synth itself?

Exactly. The keyboard is no load on the synth

Why didn't Bob Moog think of that?

You can design something and then a third party can look at it and make improvements. Bob Moog told Keith that it would never work live and that he was out of his mind for trying to take it out on the road.

And here we are over 25 years later...

...And it's still out on the road. That's the same synth that was on all the early recordings — it's just been expanded upon. It started out as a ONE C console with the one slanted console. Now it's the equivalent of a THREE C with an extra tier at the top and a few modules and programmers in there.

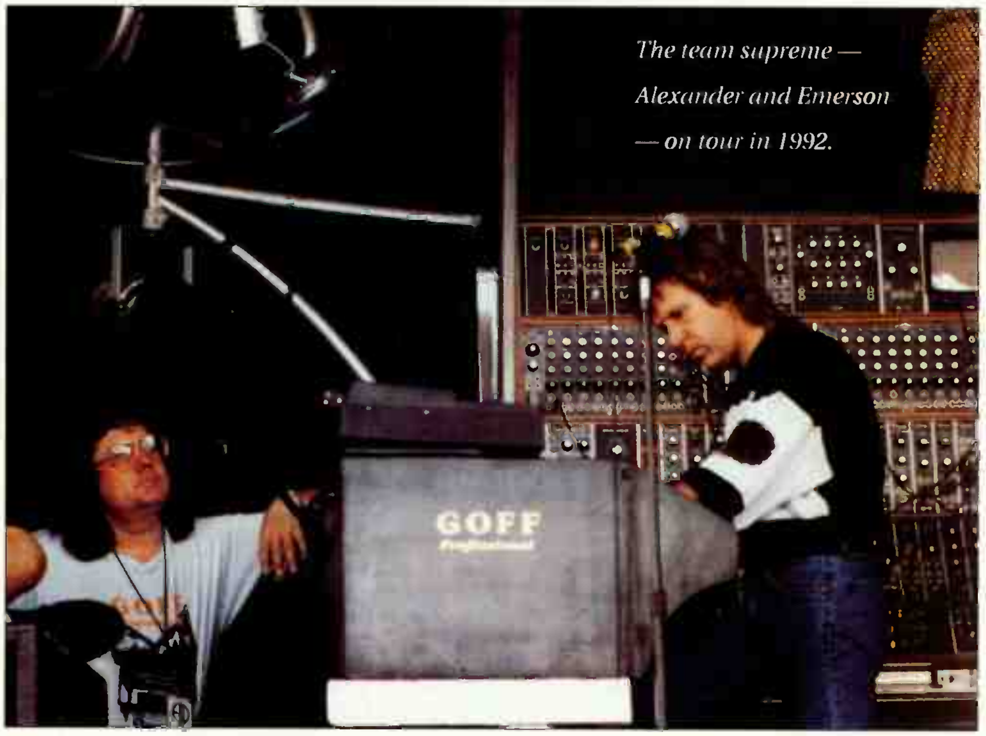
And that 'scope?

Well, the 'scope is just sort of a visual effect at this point in time.

Why cart around a modular Moog? Why not just sample it or possibly trigger one or two MIDI'd miniMoogs? After all, the Modular wasn't on the ELP/Tull tour, so why cart around something that requires so much extra labor?

The reason the Moog wasn't on the ELP/Tull tour was because ELP was the opening act for the Tull tour. We had sixty minutes to do the set and get off the stage. To go through all the trouble to set the Moog up for that short period of time wasn't worth the effort. Now that we're doing a full two-hour set, it becomes worth the effort.

You can program or sample, and Cameo International has a Keith Emerson CD of Moog and his Hammond. And, yes, you can use the SE1



The team supreme — Alexander and Emerson — on tour in 1992.

for convenience because it's programmable by MIDI, you can send program changes and you can capture the essence of the Moog. But the Moog in itself is a musical instrument with it's own sound quality. As I said, its an icon. People want to see it. People love seeing it. It's not, "Why is he dragging that thing around?" The amount of people who would criticize us for dragging the Moog around are outnumbered by the amount of people who love seeing it. It's history. Aside from the excitement of "will it work?" that instrument makes it more of a live show. One of the things that makes ELP exciting is the fact that things do screw up. For instance, in Boston on this last tour, an L100 that we've been using since 1992 caught fire on stage right in the middle of "Tarkus." This organ now resides at the Rock And Roll Hall Of Fame. We dropped it off there and said, "Here's part of your ELP exhibit."

Now what is Keith beating up during "Fanfare?"

It is now a T100. The only difference is the cabinet style. And we have other L100's in the locker waiting in line. Everybody thinks that there is only one L100, but there have been quite a few. Eventually they go. Mostly the cabinetry can't take it.

I've seen ELP several times, and I know about the standing on the organ, and the pushing, the swinging, and the knives into the keyboard.

When it stops working, he usually throws it down and I'm thinking, "Don't do that, I'm the one that has to make it work the next day."

Look at it as job security.

I'm absolutely confident of that. [*Laughs.*]

It seems that a synth like the Modular Moog needs so much attention on the road. Just how much maintenance does it need?

It depends. When we have our own truck, we treat the gear very well and it's been extremely reliable. When we were in South America, they were flying the gear everywhere. In that case, we're not in control. It goes on and off trucks and planes and more trucks — then it's different. There are over 300 screws in the front panel. Those screws are always working themselves loose and modules are falling out, so I'm always having to tighten everything up.

Do you have a daily check list for keeping the Moog in line?

Yes, it's kind of like a preflight check on a plane. After a while you know where all the vulnerable parts of the instrument are, and you can go check those out. It's kind of a routine now. All the things that could have gone wrong have gone wrong, and I know where those things are.

On this last tour we set up the Moog and there were five things that didn't work. And when you have an hour-and-a-half till show time and the synth has five things wrong with it, you have to go in and fix it quickly.

What if you couldn't solve those problems within the hour-and-a-half.

Then we couldn't use the Moog. We have a "Plan B" for everything. All the sounds that the Moog makes are online through other things.

Have any of the other synths been modified by you?

continued on page 140

Road Gear

Samson Series One

Wireless Wonder

Operating in the 800–806 MHz band, Samson's new UHF Series One employs new patent-pending PLL (Phase-Lock Loop) transmitter technology that "locks" in the signal for reliable wireless transmission. The system also features a comprehensive front-panel control scheme with a multi-segment RF level meter, Squelch control, and audio peak and antenna A/B indicators. The UHF Series One's compact "beeper-sized" beltpack transmitter and lightweight handheld transmitter feature a 3-segment LED "battery-life" level meter. The UHF Series One starts at $449 for a guitar/bass system. For more information, contact Samson Technologies Corp., P.O. Box 9031, Syosset, NY 11791-9031. Tel: 516-364-2244. Web: www.samsontech.com. Circle EQ free lit. #122.

It's All About Networking...

The UA888 Networking Interface System from Shure can be used to remotely control and monitor up to 32 Shure UHF wireless receivers. The system is comprised of a rack-mountable interface module, Windows-based PC software, and all the necessary connecting cables. Built-in monitoring features incorporated within the UA888 keep tabs on RF level, diversity signal strength, audio levels, and available battery power at the transmitters. Both monitoring and control capabilities are offered for group/channel selection, frequency, user name, squelch, and lock/unlocked status. The UA888's software allows scenes to be setup, stored, and recalled for future use. The UA888 Networking Interface System carries a suggested retail price of $1990. For more details, contact Shure Brothers, Inc., 222 Hartrey Ave., Evanston, IL 60202-3696. Tel: 847-866-2200. Web: www.shure.com. Circle EQ free lit. #123.

Shure UA888

USR-Friendly

In addition to microprocessor control, Telex's USR-100 UHF wireless microphone system offers users factory tuning, enhanced sound, and the availability of 100 different channels. Microprocessor control allows for the automatic self-tuning of exact factory set frequency information, which enables the full retention and recall of those settings. The USR-100 works in the 668.1 to 679.9 and 734.1 to 745.9 MHz frequency range, offering 100 transmission channels. The SH-100 handheld version of the USR-100 system employs a Telex dynamic mic element, a Telex condenser element, or the Audix OM3 element, which is aimed at vocal performers. A lapel mic version is also available. The LT-100 belt pack transmitter is designed to work on two AA batteries. For further information, contact Telex Communications, inc., 9600 Aldrich Avenue South, Minneapolis, MN 55420. Tel: 612-884-0043. Circle EQ free lit. #124.

Telex USR-100

Climb Every Mountain

The latest addition to the Soundcraft "K" family, the K2, is a professional 8-bus, 48-channel sound reinforcement console designed for all types of live sound applications. All of the circuitry is situated in individual vertical channel PCBs, with high-quality jack and XLR connectors on the rear of the console and other features, including a built-in VU output meter bridge and individual LED prefade input meters, and a comprehensive Mute control section. The K2 is available in a range of sizes, from 24 to 48 mono input channels, with each size including an additional four stereo inputs. For more details, contact Soundcraft, 1449 Donelson Pike, Nashville, TN 37217. Tel: 615-399-2199. Web: www.soundcraft.com. Circle EQ free lit. #125. EQ

Soundcraft K2

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ar stopped at the heat sinks." (By the way, this incident led to a complete housecleaning of all audio gear.)

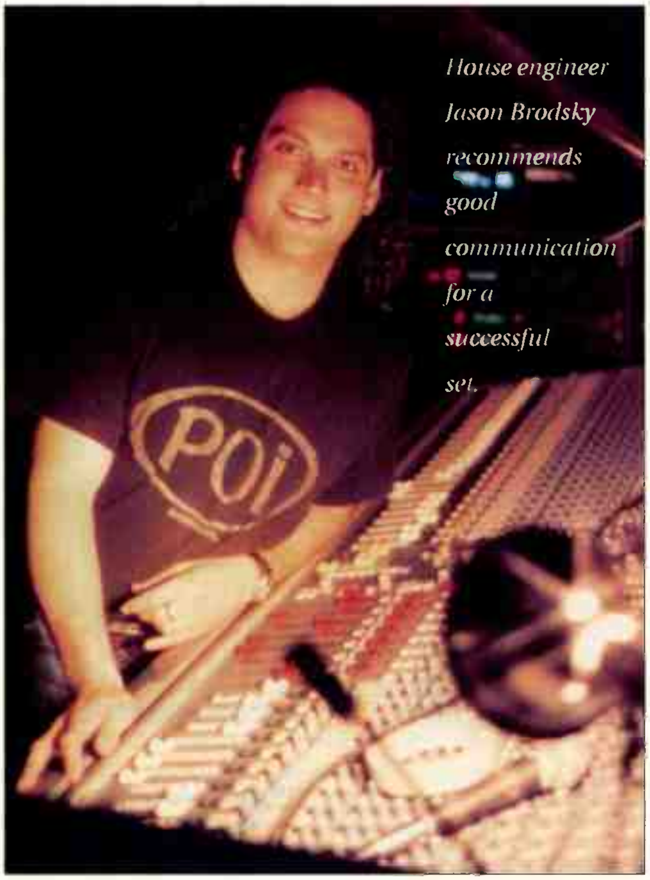
The JBL power amps are used to drive JBL SR-Series II 4894 cabinets, flown two per side. Each bi-amplified box is loaded with two 14-inch drivers and a high-frequency horn. Low end is produced by JBL AS1028 cabinets with two 18-inch drivers in each box; there are two subwoofer boxes per side, for a total of eight 18's pumping out the low end. "Originally," Alex explains, "we thought we'd need three 4894 boxes per side, and we did actually put them up. But the direction that the outside cabinets were pointed in created a lot of unnecessary reflections off of the side walls. So we pulled the extra 4894's down, left up two per side and used the remaining two cabinets for side fills."

MONITOR LAND

A total of eight mixes can be created from the Yamaha MC3210 monitor console that lives on stage right. Wedges are JBL 4890's, with a total of six available. Drum fill comes from a biamped JBL Cabaret SR4733A cabinet with two 15-inch drivers and one 2-inch horn, crossed over at 1 kHz. The remaining monitors are crossed over at 1.25 kHz using BSS FDS 318 crossovers, and all monitors are bi-amped with JBL MPX's. Completing the monitor system are the aforementioned 4894's, one each in a stereo side-fill arrangement on the 29 x 16 foot stage (an 8-foot square moveable drum riser is available).

Slim's has a complement of microphones that begins with the typical suspects such as three Shure Beta 58's, four SM58's, two Beta 57's, ten SM57's, three SM81's, three Sennheiser 409's (Jason says "these kick ass" on electric guitar cabinets), one Sennhesier 421, one AKG D112, and one Audio-Technica ATM25 — which Jason prefers for kick drum. Some of the more unusual suspects in the mic locker include two Audio-Technica Pro 37R's, an AKG C1000, and one mystery condenser cryptically labeled MR0120 (anyone with information leading to the identification of this mic, please contact Slim's or *EQ* magazine...).

Direct boxes include four Whirlwind Directors, one Missing Link active unit, and two of the new BSS AR133 actives. A Barcus Berry Pro 4000 contact pickup is on hand for the very rare occasion when an acoustic piano is brought in. The club had its own acoustic piano in the days when they hosted more R&B-oriented acts, but these days acoustic ivories in the room are scarce.


House engineer Jason Brodsky recommends good communication for a successful set.

As the house monitor engineer at Slim's for almost three years, Alex reveals that "you can't allow too much lower-mid into the monitor system because then you really start messing with the front-of-house mix. The monitor system is so powerful that it's easy to encroach upon the house mix. As a result, I tweak down the lower-mid frequencies on the monitors in the 315 and 500 Hz area. Of course, you also have to watch 1.25 and 2 kHz. Due to the quality of the equipment, the monitor system is easy to work with and we have complete control over what is happening on the stage. For me, the biggest problem is when someone is singing very softly and the band is playing loudly, which can be helped by use of personal monitor systems. Having the advantage of the side fills can help in a situation like that, as can asking the band to turn down their volume levels."

HOUSE MIX TIPS

Jason explains, "A combination of good attitude and good communication skills can make any show run more smoothly. I find that 90 percent of the problems I come across as an engineer are due to excessive stage volume. If I have established a rapport with the artist and I ask them to turn down, they know it's not a personal attack but concern with ensuring a quality performance.

"As far as technical tips, the most important thing in any mixing situation is understanding how to utilize proper gain structure and to not rely so much on what you're *seeing* as opposed to what you're *hearing*. For example, many engineers like to see the master fader parked at zero, but if we ran our master at zero, the system would be way too loud for this room. Utilize the expertise of the house crew because they are in the room every night and can significantly speed up the soundcheck process — assuming that the 'house sound guy' is not the bartender.

"Specifically in Slim's, I find standout problem frequencies typically occur around 100 Hz, 160 Hz, 315 Hz, 500 Hz, 800 Hz, 1 kHz, and 2.5 kHz. If a visiting engineer were to cut three dB at all of those frequencies, they would have a good starting point. An engineer soundchecking at Slim's when the room is empty would probably notice a washy quality to the room, usually around 500 Hz and 1.25 kHz, but these problems disappear as the room fills up with people." EQ

Shure DFR11EQ Digital Feedback Reducer and EQ

By **WADE MCGREGOR**

The little Shure DFR11EQ is so simple looking that, at first glance, you may simply dismiss the unit as another version of an automatic feedback filter. The front panel gives very little away regarding the highly sophisticated parametric or graphic equalization available from the unit. Without plugging a computer into the rear-panel serial port, the unit will act as an automatic seek-and-destroy feedback filter. If you have a PC connected, the supplied software provides access to the parametric or graphic EQ, signal delay (1 to 100 ms), and all of the unit's setup parameters.

The features of the single-channel DFR11EQ include the detection of a feedback howl (squeal, ring or screech) and the automatic adjustment of up to 10 notch filters to reduce the feedback, and software-addressable equalization and signal delay. The software (Version 4.2 in this review) is Windows-based and provides access to all the parameters except those set by the rear-panel DIP switch. The excellent software interface for the unit allows full control over all filter settings, using both graphical and numerical entry. The parametric filters go one step beyond the typical graphical controls to include the Q (bandwidth of the filter) for each of the 10 parametric filters. The interface even allows the numeric adjustment of the center frequency and depth of the automatically set feedback filters. You can even use these filters to manually add additional notches in the response. The result is a powerful box of filters that is easy to control and where you can easily visualize (see fig. 1) the individual or combined response of any of the filters.

The DFR11EQ was able to detect feedback quickly and insert a –3 dB notch at that frequency. If the feedback persists, the notch is increased in 3-dB steps up to a maximum of +18 dB. Like all other automatic feedback filters I have tested, the DFR11EQ will mistake some notes from certain musical instruments for feedback and add filters unnecessarily. This is why there are two modes of operation in the popular units (including this one): Dynamic and Fixed. The Dynamic mode allows the filters to be automatically set, deepened, or reset (when all filters are in use) as feedback is detected. Dynamic filters are very useful in situations where you are not concerned with any false triggering by music and must run the system close to the threshold of feedback. It is also helpful in applications where the system may run unattended or the microphones are likely to be moved too close to a reinforcement loudspeaker. The fixed filters can be set by gently raising the system gain, while audio, preferably broadband and impulsive, energizes the system. The user can vary the ratio of fixed and dynamic filters in the control software (factory default is five of each).

Once adequate gain-before-feedback is achieved, or you have run out of notches (a clue that this is not

ROAD TEST

MANUFACTURER: Shure Brothers Incorporated, 222 Hartrey Avenue, Evanston, IL 60202-3696. Tel: 847-866-2200. Web: www.shure.com.

APPLICATIONS: Equalization, signal delay, and adaptive filters for sound system setup and active feedback reduction.

SUMMARY: An excellent parametric equalizer and signal delay that also handles automatic null-filtering of feedback-prone systems.

STRENGTHS: Excellent software interface for control of one or more units; reliable feedback reduction; very good audio quality.

WEAKNESSES: Limited to +18 dB maximum input or output; difficult to assess status of equalization from front panel.

PRICE: $735 **EQ FREE LIT. #:** 102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the appropriate solution for your situation), then you can lock all the active filters and the unit cannot false trigger on music. This is done by simply pressing the front-panel push-button marked Lock, which disables the feedback detection circuitry and all the current filter settings (fixed or dynamic) are maintained.

Systems that include multichannel monitor mixes are difficult to assist with the Dynamic mode, as a vocalist's mic feeding back will often be in all the mixes, but only fixable in one of them. A DFR11EQ on each mix will add the feedback notch filter to all the mixes, simply because there is no way of the unit knowing which mix was the source of the feedback. A savvy monitor mix engineer can handle this situation more appropriately using conventional equalization, even though the notches produced by the DFR11EQ are much less audible than an equivalent cut by a 1/3-octave graphic EQ.

The unit is constructed of black-brushed aluminum in a half-rack format. The neatly labeled front panel uses very small letters to name functions and indicators, and these can be a little difficult to see in low-light conditions. Thankfully, all audio connections are balanced and provide both XLR and 1/4-inch phone jack connectors. A standard IEC AC power connector is provided, so there's no wall wart to deal with. Connection to a computer is via a DB-9 RS-232 serial connection, while communication between multiple units is via the Shure Link 5-pin DIN connectors (MIDI cables will work, but this is not a MIDI port). The rear panel also includes: a row of recessed DIP switches for setting modes; input and output levels; front panel lockout; and (if you have more than one unit linked via the Shure Link port) the device ID number.

An excellent feature of the DFR11EQ is the choice of modes for storing the fixed feedback filter setup. In Hold mode, only filters set before this mode are engaged and restored each time the unit is powered up. In Update mode, the unit stores the current feedback filter settings so that when the unit is powered up, it continues to update the filters already applied in previous use. The Update mode is typical of automatic feedback filtering devices where new feedback rings will stack more filters or adjust existing dynamic filters if all available filters are in use. The Hold mode, however, allows the basic "ring modes" of a sound system to be stored, while clearing any temporary filters that are enabled by transient feedback situations — for example, a lapel mic worn by someone who has occasionally walked in front of the loudspeakers.

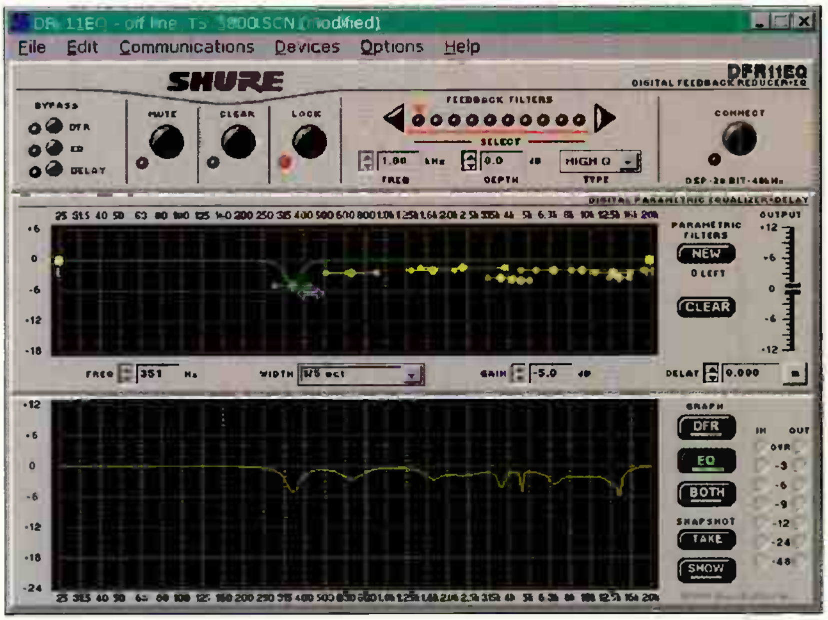

FIGURE 1: The DFR11EQ offers all the software controls on a single screen. Both the keyboard and mouse can adjust any value, including the Parametric mode equalization shown here. The bandwidth of the active filter (green line) is being adjusted with the mouse (double arrow) on the upper trace, while the lower trace shows the resulting frequency response of all the active filters.

The dynamic filter settings, in many applications, will not be required for the next use of the system, as the status of the mics has changed (mic was moved, different gain, etc.). By allowing these dynamic filters to be reset without changing the fixed filters, each new use of the system can start with a minimum of feedback filtering.

The graphic equalization is adjusted in the software, using your mouse to move any of the 32 sliders and supplemented by a graph of the resulting response. An unusual feature is the choice of combining or "true 1/3-octave" modes. The combining filters (default) will produce more even frequency and phase re-

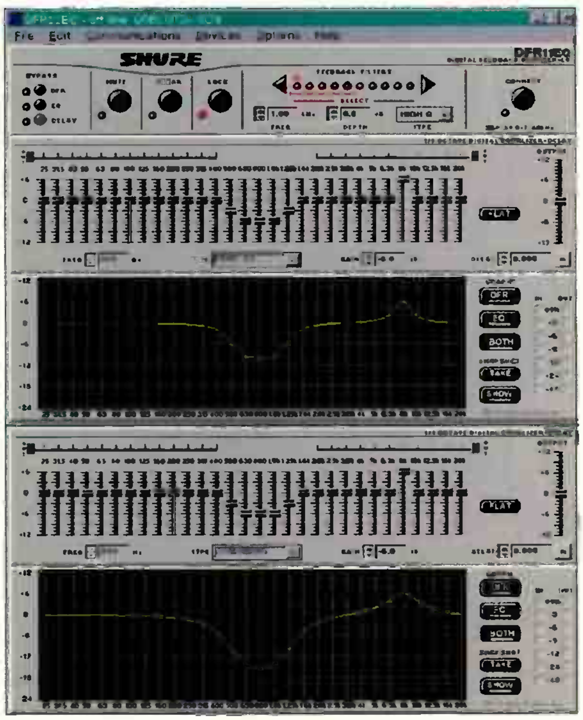

FIGURE 2: A comparison of the same equalization settings in the two 1/3-octave equalization modes. Upper trace is "True 1/3" and the lower trace is the default "Combining" 1/3-octave filters.

sponse changes, while the "true" mode will yield a more predictable (if you are just going by +6 and –12 dB slider position scale), albeit much bumpier, response (see fig. 2). As the unit includes the choice to display the resulting response, the user can see the resulting effects (at least the frequency domain) and select the type of filter that best suits their application. I prefer to use the parametric EQ, anyway.

The parametric filters and signal delay features of the DFR11EQ are only available in the Version 4 software. This software is shipping with all new units and will also enable these new features on older (silver front panel) units. Shure automatically sends the new version to existing registered users, but those that have not registered their product should contact Shure for this substantial software upgrade.

MANUFACTURER'S SPECIFICATIONS

Frequency Response: 20 Hz to 20 kHz ±1.0 dB re 1 kHz
Dynamic Range: 104 dB minimum, A-weighted, 20 Hz to 20 kHz
Sampling Rate: 48 kHz
Digital-to-Analog, Analog-to-Digital Conversion: 20-bit resolution
Input Clipping Level: +18 dBu minimum (at +4 dBu setting); +6 dBu minimum (at –10 dBV setting)
Output Clipping Level: +18 dBu minimum (at +4 dBu setting); +6 dBu minimum (at –10 dBV setting)
Total Harmonic Distortion: less than 0.05% at 1 kHz, +4 dBu, 20 Hz to 20 kHz
LED Signal Indicators: Clip: 6 dB down from input clipping
Dimensions: 219 mm x 137 mm x 40 mm (8 5/8" x 5 3/8" x 1 3/4")
Weight: 930 g (2.5 lb.)

Shure has developed the DFR11EQ to be an excellent DAP-based filter box for use in live sound applications. The ability to hide sophisticated system equalization from busy fingers is a major advantage to both rental and fixed install sound systems. The basic feedback reduction functions are easy to use and work reliably (given the above caveat), and the addition of signal delay is a bonus that, in some situations, can also help to reduce feedback. The 20-bit A/D/A and excellent filter algorithms provide a quiet unit that sounds good, while offering the capability to make very significant improvements in the overall sound quality of the system (using the parametric EQ) and substantial improvements in the gain-before-feedback of any well-implemented sound system.

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

TELEX 450-SERIES WIRELESS MICROPHONE SYSTEM

By **STEVE LA CERRA**

The 450-Series UHF wireless system is a recent introduction from Telex. Consisting of the FMR-450 receiver and the HT-450 handheld microphone transmitter, this system offers a choice of three different microphone elements: the HT-450/65ELE (Telex 65ELE), the HT-450/OM3 (Audix OM-3xb), and the HT-450/EV 757 (Electro-Voice N/D757). Although up to 30 FMR-450 systems may be used simultaneously, each system is factory-set to a single frequency and is not user-adjustable.

The FMR-450 receiver is a one-space, half-rack unit with an outboard power supply. Front-panel features include a power switch and three meters designed to help you monitor system status. One indicates diversity status (0 or 180 degrees); the second shows RF strength; and the third — an AF display — indicates relative modulation of the system. We found the AF display essential for setting proper gain structure with the various mic elements.

The rear panel of the FMR-450 has two coax antenna jacks; a balanced mic-level XLR audio output; an audio output adjust for the latter; and a power receptacle for the 13-volt wall wart (*ugh!*). Although we hate wall warts, we'll give Telex credit for a few things: [1] the cable used for the power supply was more robust than usual; [2] it had an AC ground pin; and [3] there's a rear-panel strain-relief clip to ensure that the plug does not accidentally pull out of the jack.

Next to the antenna jacks are small colored dots that indicate which antenna should be used with the system; the proper antenna (supplied) has a complementary color band that should match the dots. The final control on the rear panel is a compander in/out switch, which we left "in" as recommended in the HT-450 manual (Telex supplies thorough manuals that explain wireless setup clearly and concisely).

The HT-450 is, operationally, a rather simple transmitter: unscrew the handle to insert a 9-volt battery (contacts are clearly marked), then screw the mic closed. On the mic base are: an antenna receptacle with a clever bayonet-style mount, a power switch, an audio switch, and a red LED that blinks when power is switched on. The LED will light steadily when you have about one hour of battery life remaining. While Telex specs battery life at 8 to 10 hours for an alkaline cell, we found it to be slightly shorter — about 6-7 hours, or three shows including soundchecks.

Telex sent us the FMR-450 receiver and all three HT-450's [mic elements]. We took the system out on the road for a major two-week club tour that included stops in California, Washington, and Oregon. All components were packed inside a two-space rack bag, and the receiver was not rack mounted (we packed foam around it). After flying out to the West Coast in an overhead compartment, the rack bag spent the remainder of the tour bouncing around in a truck. In other words, we didn't baby the system. Although we didn't rack mount our unit, Telex does offer a rack mount kit (FMX-450) that can house either one or two receivers.

ROAD TEST

MANUFACTURER: Telex Communications, Inc., 9600 Aldrich Avenue South, Minneapolis, MN 55420. Tel: 612-884-4051. Web: www.telex.com.

APPLICATION: Handheld, wireless vocal mic for stage use.

SUMMARY: Telex has nailed the technology on the head.

STRENGTHS: Impeccable RF performance even under critical situations; easy to use.

WEAKNESSES: Not frequency-agile; wall-wart power supply.

PRICE: FMR-450, $875; HT-450/65ELE, $746; HT-450/OM3, $988; HT-450/EV 757, $835.

EQ FREE LIT. #: 131

Proper setup of the 450 system is easy, but crucial to attaining high performance. Basically, we'd tweak the system by turning on the receiver and transmitter and yelling into the mic while watching the AF display. Then we'd adjust the gain pot on the HT-450 until the AF display showed the yellow LED, indicating optimum dynamic range and S/N ratio without clipping.

Since the FMR-450 may be used with any of the three capsules, gain adjustment is critical (we found that AF levels from the various HT-450's were significantly different). Then we'd use the audio output adjust on the rear panel of the receiver to set an output level that made the consoles (monitor and FOH) happy. Telex supplies a small plastic tool for making these adjustments. The FMR-450 has plenty of gas to drive any input; our unit came set to maximum output, which easily overloaded the mic preamp of a Yamaha monitor desk. A setting of roughly 11 o'clock proved to be more than enough for most preamps. The fact that the FMR-450 uses an XLR output is a double-bonus: quiet operation and a secure connection.

Our following comments regarding RF performance of the system apply to all three versions of the HT-450: transmission and reception were virtually impeccable. In more than 12 different locations, including downtown San Francisco and Seattle, this system never coughed, hiccuped, dropped out, or lost audio — not even once. This included a performance at Seattle's King Kat Theater, where the house engineer described his room as "RF hell" (in fact, the opening act's guitar rig made quite the monitor for the local taxi service!).

Although we did a walk-around during soundcheck for dead spots, they were rare. Audio quality was very good to excellent, depending upon the element in the HT-450: the male vocalist liked the EV 757 capsule best, describing it as the one that cut through the monitors best. One monitor engineer described it as having very good feedback rejection.

Out at FOH, I liked the sound of the Audio OM3xb capsule best because it was smooth sounding over a wide frequency range, didn't build up in the LF range, and had a bit more depth in the lower mids than the EV capsule. Proximity effect was a bit of a problem in the version with the Telex 65ELE capsule, so this might be best suited to anemic male vocalists, female vocals, or for announce purposes. Even though the system operates on a fixed frequency, we didn't miss the ability to change frequencies because the RF performance was so good.

The Telex FMR-450/HT-450 wireless system is a clear winner. It's easy to use, well made, and could fool you into thinking that there's an invisible wire on the mic. Though we normally prefer frequency-agile systems for touring purposes, we never found the system's fixed frequency to be a limitation. This wireless system isn't cheap, but it's worth every penny. EQ

MANUFACTURER'S SPECIFICATIONS

Audio Bandwidth: 50 Hz to 15,000 Hz, ±1 dB (transmitter)
Receiver S/N Ratio: 104 dB typical
Frequency Range: 524 to 608 MHz and 614 to 746 MHz
Transmitter Power Requirements: one 9-volt battery
Range: 1000 feet open field, 250 feet adverse conditions

Event Electronics

Gina Digital Multitrack Audio Card

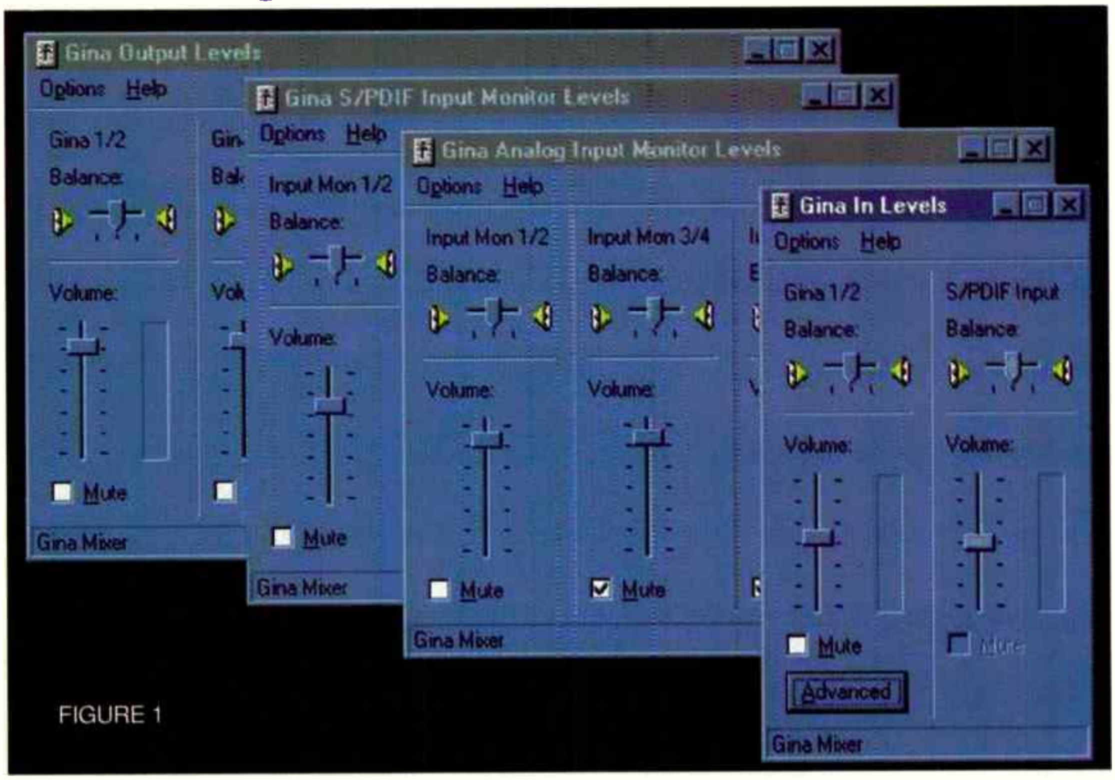

FIGURE 1

Upgrade your PC with a 24-bit A/D-D/A converter

BY TIM TULLY

In the past year or so, Event Electronics has announced three new Windows 95-based multitrack digital audio recorders that claim a superior bang for the buck. This review covers Gina, which occupies the middle of the Event line between the budget Darla and yet-to-be-released, high-end Layla.

THE HARDWARE

Gina consists of a PCI adapter card that connects to a breakout box via a 3-foot, custom cable that closely resembles a SCSI cable. The 3- x 2- x 6-inch breakout box has ten 1/4-inch mono jacks — 2 inputs and 8 outputs — and the card has RCA stereo S/PDIF in and out jacks. Gina provides 20-bit A-D and D-A converters with 128X oversampling; all internal processing, including S/PDIF, is 24-bit (though the device can write a 32-bit word when required by the DSP operations of certain software). Supported sample rates include 44.1 and 48 kHz, although the card's hardware can resample to 11.025 and 22.05 kHz. Gina's S/PDIF ports, by the way, let you select either consumer or pro modes — apparently some DAT recorders are picky about what kind of header data they see. Nonetheless, Gina never records nor transmits SCMS data, so you only have that problem when you record to a consumer DAT that writes its own SCMS code to tape.

Gina is based on a Motorola 56301 DSP chip that, according to Event, has roughly the equivalent processing power of a Digidesign DSP farm containing four 56002 chips.

SOFTWARE

Like the other Event products, Gina is the fruit of Event's "lets-give-them-good-I/O-cheap" philosophy and its association with ASIC heavyweight, Echo, which designed the hardware and drivers. To maximize compatibility with both system software and popular audio applications, Event designed its drivers to look just like the Windows Wave Device Driver (WDD). This way, both Windows and applications like Cakewalk Pro Audio, Musicator, Samplitude, SAW, Sound Forge, and others are sure to work with Gina. (Macintosh drivers are in development.) Since the

EQ LAB REPORT

MANUFACTURER: Event Electronics, Box 4189, Santa Barbara, CA 93140-4189. Tel: 805-566-7777. E-mail: info@event1.com. Web: www.event1.com.

APPLICATION: High-quality, 2-in, 8-out A/D-D/A conversion for Windows 95 systems.

SUMMARY: Easy-to-install PCI card and breakout box that works with most Windows 2-track and multitrack audio programs.

STRENGTHS: Excellent aural quality; good connectivity (lots of I/O); wide compatibility with software; low price.

WEAKNESS: Temporary dependence on Windows applet for gain control.

SYSTEM REQUIREMENTS: IBM compatible computer with PCI architecture expansion slots (version 2.1 PCI BIOS) running Windows 95, 16 MB of RAM, and a fast, high-capacity IDE or SCSI hard drive.

PRICE: $499 **EQ FREE LIT. #:** 103

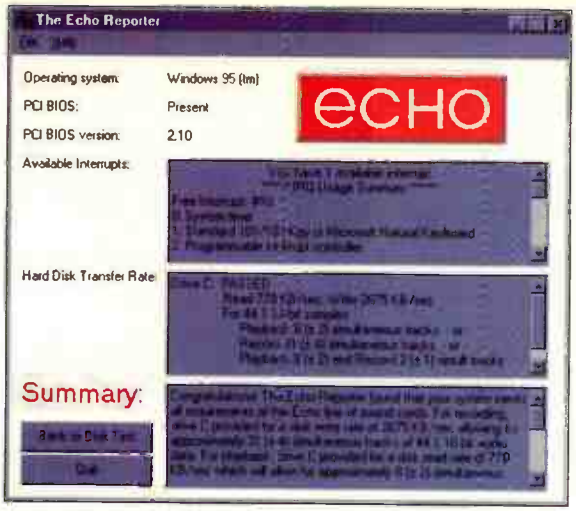

FIGURE 2

WDD only supports stereo devices, Gina is designed so each pair of its ins and outs looks like an independent stereo device to Windows. For pro performance, each device is sync'd to the others with sample accuracy.

Though the compatibility this offers is a considerable benefit (particularly in light of the difficulties of installing third-party cards in Windows), there is a small downside to it. To notify Windows that Gina is the sound card you want to use (as opposed to the one that came with your system) and to control the level and pan of each pair of Gina ins and outs, you have to use the Windows applet, SNDVOL32. Unfortunately, SNDVOL32 was designed by Microsoft for consumer control of consumer sound cards and, therefore, fits poorly into a professional environment. SNDVOL32 has separate windows for adjusting input, output, and monitor levels, and going from one to the other, as well as initially getting to the app, is a somewhat tedious, multistep process.

In fairness, you shouldn't have to use SNDVOL32 all that often, particularly if your recording application has level and pan controls. Fortunately, Event is writing the Echo Console, an application that contains all of SNDVOL32's functions in one coherent interface, and which will be available soon. Meanwhile, an Event representative pointed out that Windows can open multiple iterations of SNDVOL32, allowing you to open all four of the control panels at once (see fig. 1). If you need to make level and pan changes as you work, this is not a bad solution.

It's important to note that both SNDVOL32 and the upcoming Echo Console do not change volume by altering the digital data, but rather control Gina's analog circuitry. This means that attenuating an input or output signal does not diminish its bit-resolution — which would compromise sound quality — but acts like a fader on an analog console, and allows I/O to operate as close to the 20-bit ideal as possible. In fact, to take maximum advantage of Gina's 20-bit I/O, Event bundles the Easy Trim software. This application analyzes the material you're recording and takes advantage of a circuit in Gina to set the maximum input level possible for that material. This means you aren't leaving any more headroom than necessary for each track you record, and you always get the best possible bit resolution.

The Gina bundle includes Echo Reporter software. As the manual suggested, I ran this software before installing the card, and I now believe that Reporter should come with every adapter card for the PC.

The application automatically checked my system and reported that, in my computer, IRQs 10 and 11 were unused, and listed what devices were occupying my other IRQs. Reporter also tested the internal IDE hard drive that came with my system (a 200 MHz Micron Millenia Mxe) and described in detail its probable performance at multitracking. For 16-bit, 44.1 kHz operation, it reported I could: play back 17 (±4) simultaneous tracks; or record 33 (±4) simultaneous tracks; or play back 16 (±3)

continued on page 140

Quasimidi's Rave-O-Lution 309 *Drum/Percussion/Pattern Machine*

An all-in-one device for all your groove needs

BY DAVID MILES HUBER

Look! In the sky! It's a real-time groove machine. It's an analog-like synthesizer. It's a sound processor. It's the Rave-O-Lution 309 beat-box from Quasimidi! OK, I'll cool down, but really, folks, this little German puppy takes the concept of "drum machine" to a whole new level. After taking the 309 out of the box, plugging it in, and pressing Play, it took about 2.9 seconds to realize that the Rave-O-Lution is chock-full-o-killer beats and just reeks of real-time parameter control. OK, enough drooling, let's get down to business.

In its standard configuration, the 309 contains five-independent tone generator sections (Kick, Snare, Hihat, Percussion Kit, and Bass-Lead Synth), which create and control sound using A.E.S. (Analog Emulation Synthesis). The 309's first four synth sections derive their sounds from prerecorded samples (no news here). However, the Bass-Lead Synthesizer section is unique in that it doesn't use samples at all, but rather digitally generates waveforms that are then filtered and processed. All five sections are routed to their own set of "virtually" controlled oscillators, filters, amplifiers, and a 4-stage envelope generator section that very convincingly emulates analog control parameters. Although all of the usual synth edit parameters are available from the 309's LCD screen, some of the more pertinent controls have been dedicated to front-panel knobs for easy access. In addition, each section also has a Mute button (for muting individual instruments) and a Select button (that actively selects the instrument for editing).

Before being routed to the system's final output jacks, the overall mix can be sent to any of the 309's three effects sections: FX-1 (reverb algorithms); FX-2 (modulation and delay effects); and FX-3 (a digital, 2-band parametric equalizer). The folks at Quasimidi also saw fit to design a cool "Overblast" control that boosts the output's overall bass and adds to the instrument's perceived loudness and distortion.

EQ LAB REPORT

MANUFACTURER: Radikal Technologies, 1119 North Wilson Avenue, Teaneck, New Jersey 07666. Tel: 201-836-5116. Quasimidi Products, Elsenbahnstr. 13, D-35274, Kirchhain 1, Germany. Web: www.quasimidi.com.

APPLICATION: A factory- and user-programmable groove-type beat box for the studio, stage, or DJ booth.

SUMMARY: A multipurpose rhythm unit that combines a polyphonic drum module with a monophonic bass/lead synthesizer, and also features an integrated sequencer, complete with Real-Time and Step-Record modes.

STRENGTHS: Killer grooves and sounds; uses an incredibly innovative approach to creating real-time percussion and groove sequencing, not to mention that a great monophonic analog-like lead synth gets thrown into the deal.

WEAKNESSES: Currently, the timing will automatically revert to 140 BPM when switching between pattern banks, which could cause some serious "speed bumps" during a live performance.

MANUAL: Very straightforward. It gives you what you need to know without getting bogged down in excessive jargon. I particularly like the headphone tip: "Prolonged exposure to excessive sound levels can cause permanent hearing damage."

PRICE: $1295; Audio-Expansion module, $349.95; Synth-Expansion, $299; Drum-Expansion, $299; rack-mount ears, $30.

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CIRCLE 93 ON FREE INFO CARD

LET'S PLAY!

Hit the Play button, and you're immediately slapped with a groove pattern that's pretty awesome right out of the box. Turning the Value-Tempo wheel lets you vary the tempo to the desired BPM (beats per minute), and pressing any of the eight Pattern-Pads (while cycling through the Song-Pattern banks) gives you access to 99 ROM grooves and 100 user-programmable pattern locations. Once the current pattern has cycled through its entire loop, a newly selected groove will seamlessly begin playing. Although the BPM remains constant between the patterns within a bank (0–9), be forewarned that the timing will automatically revert to 140 BPM when switching between the various patterns. This tempo jump could be a serious setback when switching between banks during a live performance.

The next step is to begin pressing buttons and twiddling knobs. At the 309's lower right side are eight "Special Loop Track" buttons. These are pretty radical in that whenever one or more is pressed during a groove, a series of percussion variations and accents is introduced into the groove that serves to vary the pattern during a performance. Pressing the Mute button on any instrument also lets you combine percussion sets into a seemingly endless series of combinations. In addition to being able to transpose a groove's key over an octave, you can endlessly change the pattern on individual instruments while a groove is still in progress.

Once you've decided to make the jump and begin creating your own sequenced grooves, building your own patterns into the 309's internal sequencer is as simple as playing a riff on the Pattern buttons while listening to the existing groove and guide click track. Although the process isn't difficult, I'm not even going to pretend that I've gotten the process down yet, as there seems to be an endless number of ways to create and edit a percussion pattern.

LET'S TALK MIDI

Each of the 309's instrument sections is assigned to its own MIDI channel (Ch. 1 thru 5 by default). This means that if you want to sequence all of the instruments at once, you'll need to simultaneously record all five channels. Quasimidi cleverly designed the 309 so that whenever the control knobs on any instrument are varied, corresponding continuous controller messages will be transmitted on the appropriate channel, allowing you to encode parameter changes directly into the sequence.

One of the biggest bonuses is the ability to route MIDI to the bass-lead synthesizer (Ch. 5) will actually allow you to play the instrument as though it were a monophonic analog synth. No lie, folks, this synth section sounds killer and allows you total, real-time control over the instruments' synthesis and tonal parameters. Some might find this feature alone worth the asking price.

EXPAND-O-MATIC

An optional Audio-Expansion card adds two major functions to the 309. The first of these is an extra set of output channels (out 3 and out 4). Any of the groove voices can be assigned to these jacks; however, the effects processors can only be assigned to the main stereo outs (this generally isn't a problem, since the whole idea of a separate out is often to assign an instrument to an external effects device or dry recorded track). Secondly, the expansion card lets you inject a stereo pair of audio inputs directly into the bass-lead synthesizer's path. This high-quality feature means that you can create some interesting and wacky real-time effects by modifying an audio signal using the synth's filters. In addition, you can even assign these inputs to the box's internal effects processors, meaning that the 309 could be used as an extra, high-quality effects box.

Oh yeah, as if all this weren't enough, you can add an optional Synth-Expansion (that gives you access to two additional synth engines, which can be routed to outputs 3 and 4) and Drum-Expansion chip that gives you tons of extra sounds and grooves. The expanded sounds are fat, punchy and knocked me on my butt.

MY 2-CENTS

As if you couldn't tell, I'm a big-time Quasimidi fan. They're one of the most innovative companies around. The Rave-O-Lution 309 is just that: "revolutionary." It's full-O-phat sounds, has tons of great grooves that can be varied in real-time, is extremely versatile, and even throws in a killer-sounding monophonic analog-type synth (or two, with the Synth-Expansion module) for good measure. If you're into techno, dance, or progressive modern beats, you must look into adding the 309 to your groove toolbox. This is simply one of the most versatile, amazing tools I've come across. I'd definitely place the 309 smack-dab in the "best all-round buy on the block" category. You've truly got to check this one out! EQ

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Akai DPS12 *Hard-Disk Recording System*

An EQ Exclusive: Akai's latest entry in the computer-less hard-disk recording field comes on strong

BY CRAIG ANDERTON

The Battle of the Self-Contained (No Computer Required) Hard-Disk Recording Studios continues, and the DPS12 is Akai's latest offering. It combines a digital mixer, 1 GB Jaz (or fixed 2 MB) hard-disk transport, and actual controls — fader and pan for 12 channels, input gain for six channels, jog/shuttle wheel, cursor buttons, seven transport controls, and 39 buttons for control and selection. Like the competition, it's trim and compact, and has a "serious" rather than toy-like ambience.

GETTING STARTED

Before opening the manual, I was able to start a project, record basic tracks, overdub vocals, set EQ, assign effects sends and parameters, and, of course, mix and monitor. I finally had to crack the book when dealing with effects returns. Although deeper features require a bit of study, this box is very easy to use for basic recording, if you understand the basics of how to route signals (of course, all routing is done "virtually" on the LCD instead of with patch cords).

The DPS12 has 12 channels with fader and pan that are dedicated to mixing down the 12 tracks stored on hard disk. You can

EQ LAB REPORT

MANUFACTURER: Akai, 4710 Mercantile Dr., Ft. Worth, TX 76137. Tel: 817-831-9203. Web: www.akai.com/akaipro.

APPLICATION: Record, process, and mix multitrack digital audio in a self-contained device.

SUMMARY: Makes a unique contribution to a crowded field via cost-effectiveness, ease of use, and sound quality at a reasonable price.

STRENGTHS: No audio data compression to alter sound; provides an easy interface, solo function, variable speed, and 250 virtual tracks; can slave to MIDI (MTC and MMC) or serve as MIDI master with clock and song pointer; tempo and time signature map; optional effects board; digital bouncing and editing.

WEAKNESSES: Automation handles neither EQ nor effects, and scenes cannot be recalled during playback; manual has no index and limited tutorials; lacks any XLR mic inputs; despite the 250 virtual tracks, it's awkward to create composite tracks.

PRICE: $1499 (base unit); $1849 with internal drive (either 1 GB Jaz removable or 2 GB fixed hard drive). Add $299 for EB2M effects board.

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data from the DPS12, as you can capture dynamic changes for the faders, pans, and sends, as well as send a MIDI data "snapshot" at any moment by hitting a button to transmit controller data representing fader, pan, and aux send settings (if the Aux mode is stereo, pan is recorded as well). Unfortunately, EQ settings are neither transmitted nor recallable via MIDI; you have to recall scenes manually to change these parameters. I'd like to see a future generation of DPS12 software that could assign program change commands to scenes so that they could be recalled via MIDI.

OVER & OUT

Useful details include the flexibility of being able to send aux tracks to outboard gear and bring them back in via the Thru Mix channels, and multiple levels of undo (up to 250 — but remember that more Undos eat up more hard-disk space). The optional effects board is pretty cool, too, with chorus, flanger, delay, tape echo, pitch shifter, dynamics, phase shifter, rotary speaker, triggered and auto pan, reverb, 3-band parametric, auto and envelope-controlled wah, and even some effects pairs (chorus/delay etc.). They sound quite good, and reinforce the studio-in-a-box concept.

So, overall, we're definitely in thumbs-up land. One caution: If you're using the version with the removable drive, factor in the cost of an uninterruptible power supply. If the power goes on vacation while accessing a Jaz disk, damage could result. Another concern is the use of Jaz media. Some have complained of spotty reliability, and hard-disk recording really exercises a drive. *Note:* According to Akai, the DPS12 Jaz drive is a version developed by Iomega that is currently not available to the public and is optimized for multimedia. If you have an older Jaz drive, consider getting a DPS12 with the internal hard drive, and using the Jaz for backup.

I was very impressed by (and had fun with) the DPS12 — not necessarily because of hot-shot, groundbreaking technology, but because it makes hard-disk recording painless and obvious while delivering solid sound quality. Granted, the competition is heating up tremendously in this field, but Akai has done its homework: the DPS12 combines ease of use, cost-effectiveness, and quality sound in a compact package at a very reasonable price. EQ

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CIRCLE 71 ON FREE INFO CARD

AKG Acoustics Solidtube Microphone

AKG continues its "solid" history of producing tube microphones with this new entry

BY STEVE LA CERRA

These days it seems that no studio microphone manufacturer's product line is complete unless it offers at least one tube model. Few of those manufacturers, though, have AKG Acoustics' extensive history of making such mics. So, AKG's recent introduction of its Solidtube microphone was of special interest to *EQ*.

The Solidtube is a cardioid-only condenser employing a 12AX7a tube in its preamp section. It connects to an outboard power supply with a 6-pin XLR connector, and may either be directly mounted to a [generic] stand via threads in the base of the mic or mounted in AKG's H-Solid shock mount. Securing the mic in the plastic mount simply requires that you insert the base of the mic and gently twist the body. Though we don't like plastic shock mounts, this one appears to be built for the duration, and AKG has cleverly built a cable strain-relief clip into the mount to reduce stand-transmitted noises. The mic is supplied with a foam pop filter, but we performed our tests without it because we find that this type of filter generally colors a mic's response.

Our test unit had been delivered with the power supply's voltage selector set to 230 volts. This setting caused the mic's output to be low, and produced an annoying hum in the area of 120 cycles. Switching the voltage to 120 volts instantly remedied these problems.

One of our sessions with the Solidtube was a recording of an alto male vocal, roughly in the style of Jon Anderson from Yes. On this session, the character of the Solidtube was immediately evident — rich and silky smooth, with what sounded like a bump of a few dB in the range of about 250 Hz. It definitely worked for this vocalist, who can otherwise sound thin and wimpy. Although we did not patch a compressor on the Solidtube (it went straight to tape from a Demeter VTMP-2a mic pre), the Solidtube gave us the impression that a compressor was in line; that is, the vocal had a distinct sense of immediacy and a lot of presence without being edgy. It was a colored response, but, in this instance, most definitely flattering. When the vocalist moved in close and let it rip, the Solidtube had no trouble with the SPL

MANUFACTURER'S SPECIFICATIONS

Tube: 12AX7a
Polar Pattern: cardioid
Frequency Response: 20 Hz-20,000 Hz, ±2.5 dB
Sensitivity: 20 mV/Pa = –34 dBV re. 1V/Pa (@ 1000 Hz)
Impedance: 200 ohms, ±25%
LF Cut: 12 dB per octave @ 100 Hz
Signal-To-Noise Ratio: 74 dB (A-weighted, ref. 1 Pa)

(the pad was switched off).

Tapping on the microphone stand, shock mount, and the mic body itself produced some definite "thumping," all of which pretty much disappeared when the low cut was switched in. Interestingly, the "timbre" of the mic barely changed regardless of the whether the low cut was switched in or out. The exception was acoustic guitar: without the low cut, the bottom was a tad sloppy. Switching in the low cut effectively tightened up the bottom. The Solidtube generally proved to be a good choice for acoustic guitar, revealing the subtle differences of hand position when the guitarist picked near the bridge, sound hole, or fretboard.

After trying out the Solidtube on percussion — where it softened transients on instruments like maracas — we decided it was time to push the mic with some high SPLs. We started by using it overhead on a drum kit, about 6 inches above the rack toms. The toms sounded excellent, with a distinct *doooooowwmm* in the low end and a nice smack from the stick. We were able to record in this position without the pad, but getting any closer overloaded the mic and required the pad. Next, we placed it about 5 inches in front of a kick drum, where the pad was needed to avoid distortion. The best way to describe the sound on kick was a pile driver in the studio. The bottom end slammed and pushed plenty of air, but we still heard the knock and floppiness of the heads — which had been tuned very loose. Very rude, and we loved it.

AKG's Solidtube is a solid and versatile performer. We'd compare the mic to a unique color of paint: it's not great for everything, but often produces striking results. AKG's main competition for the Solidtube is the Peavey PVM-T9000 tube mic, which is similarly outfitted and priced. The big difference between them is that the Solidtube possesses a more distinct character as a color on the studio palette. If you're looking for a tube mic in this price range, you should definitely try the Solidtube.

MANUFACTURER: AKG Acoustics, US, 1449 Donelson Pike, Nashville, TN 37217. Tel: 615-360-0499. Web: www.akg-acoustics.com.
PRICE: $1500, including case, power supply, shock mount, cables and a pop filter.
EQ FREE LIT. #: 105

APOSTOLIC STUDIOS

continued from page 60

song become a national favorite, with Wall Street backers for a public offering. As a studio, we continued to keep up, providing clients not only with every kind of strange international/historical instrument they might want to play (want a kan, a del ruba, a viola d'amore, a rauschpfeife?...no problem), they even got the free services of world-class astrologer Al Morrison, who shared one of our floors. The latter was a good thing, as Al provided me with a second career when the majors ate us up at the beginning of the '70s and I found myself a quite useless music biz innovator alone on the street.

But although we were doomed to meet our demise at the hands of over-expansion, competition, and a business that utterly coopted our concepts, the studio retains memories that are unique to its origins. The day the brother of a *very* famous blues/rock guitarist took microgram-inspired wings from our window, falling face-down to the roof two floors below — leaving a tar-paper "angel" on the roof. He proceeded unscathed two more floors to the back garden. The boa constrictor entwined in the bidet (and afterwards, the water cooler) that could not be smoked out (we succumbed before it did...). Generations of Mothers trooping in and out for "Lumpy Gravy" and "Uncle Meat" sessions...and in the process creating the "flange" effect, first done by Zappa and our engineer at Apostolic by using a reverse-phased, slightly trailing variable-speed controlled 2-track Scully. A lot of firsts and, mainly, a family where creativity and technology finally worked hand-in-hand.

Now it all seems like old hat. These days, you can get lots more than this, by a landslide, in any pro recording studio and many home installations. But, hard as it is to believe, it wasn't always that way — there was a time when the design of the recording studio was utterly business-driven, not musician-driven, when creation in front of a mic was not exactly natural childbirth. In 1967, all that changed, forever — and I am proud and thankful to be able to say that Apostolic and its engineers, producers, and musicians — achieved that in, essentially, one take.

John Townley is performer/producer of over a dozen collections of pop, folk, and historical recordings for a variety of labels. He is also an internationally-known writer and astrologer with seven books in seven languages to date.

EMERSON LIVE

continued from page 113

It's hard to customize digital signal processors these days. It's not like the old days where I used to do a lot of analog modifications. The only thing that's been modified is our Voce module, which has more gain coming out of it.

Could you describe your role in this team when ELP is in the studio?

I'm involved in sound design and sequencing. I'm looked upon as an archivist of what they've done in the past. Because I know the music so well, if they come to me and say, "There is a sound we used on..." I can pretty much re-create the sound. Whether it came from the (Yamaha) GX-1 or the Moog or a Moog prototype.

There was a Moog polyphonic prototype that was used on the *Brain Salad Surgery* album. Did that give birth to the Polymoog that we all know?

There were two Moog prototypes that were fitted together as one unit on rollers. There was a Polyphonic Ensemble that was the prototype for the Polymoog and then on top of that was an instrument that was referred to as Apollo — which we still have. This was sort of a super miniMoog. It had an extensive modulation section and it had pressure sensitivity on the keyboard.

Sort of a prototype to the MultiMoog?

Yes, but this was definitely a miniMoog. It's all hand wired.

And Keith still owns the Apollo?

Yes. There is some confusion as to what synth was what. There was the Lyra and the Constellation — these were polyMoog prototypes — and there was the Apollo. Then they all became the Moog Polyphonic ensemble. That's what he used on Cal Jam or if you've seen any of the *Brain Salad* shows. Keith played "Benny The Bouncer" on it and he also used it on "Third Impression" for the horn lines. So the Polyphonic Ensemble was really two units. The top was the Apollo, sort of a miniMoog on steroids, and then there was the Constellation or the Lyra on the bottom.

Tony Di Lorenzo has been involved with synthesizers and keyboards for the last 20 years. He has produced his own CD-ROM for the Kurzweil K2000 and K2500 called "Producer Series Vol.1." He runs his own company called Front Room Productions. You can visit Front Room's Web page at www.interport.net/~thefront/index.html. You can also e-mail Tony at thefront@interport.net.

EVENT GINA

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tracks and simultaneously record 2 (±1) tracks.

Buoyed by such clear, useful and positive information, I tested a 100 MB Zip drive. I wasn't expecting much, but was surprised by the results. The tests determined the Zip could play back 6 (±1) simultaneous tracks, or record 7 (±2) simultaneous tracks, or play back 4 (±1) tracks while simultaneously recording 2 (±1) tracks. In practice, I found these figures to be quite reliable, if a bit conservative, as both the IDE and Zip drives slightly outperformed them (at least in playback; I only had two inputs for recording). The software also generated an easy-to-read, conversational version of the report (Fig Report 2) that verified the test results and put them in context of typical recording needs. Of course, actual performance depends on lot of factors, including the multitracking application you use.

Speaking of multitrack applications, Gina also bundles a special version of Syntrillium's CoolEdit. This version has only ten stereo tracks, limited DSP, and only an online manual, which I found somewhat frustrating to use because, for example, finding topics was less than easy. Gina owners can upgrade to the full version that offers printed documentation.

PERFORMANCE

Installation was a breeze, thanks largely to Echo Report. After that, Gina worked like a charm in both Sound Forge and CoolEdit. It creates a very quiet operating environment and it's easy to believe the 24-bit internal specs because the sound is clean, undistorted, and easy on the ears. EQ

SOUND EFFECTS

continued from page 88

remixing parts of the sound back together can solve the problem; some quick fades (in and out) might be necessary to smooth abrupt beginnings or endings. Finally, you can preview the video file with the audio to check the final result. All these operations, and more, are available in good audio editors.

There are no rules about creating sounds, but the stone door example demonstrated several approaches. It involved an original sample and one from a sound effects CD, which were pitch-shifted, layered, filtered, time-stretched, cut, and mixed back together. Given that there could be a couple of hundred sound effects in a game, that's quite a lot of work for just one sound — but that's why game companies and other multimedia producers need sound designers. EQ

ACROSS THE BOARD

continued from page 162

start at sample number zero, then after the drums have played for 100 seconds, the guitar will play and line up exactly. When a hard-disk recorder chases a SMPTE source, the sample number is mathematically calculated from the SMPTE location. When the values match, the digital audio starts, thus the "trigger point." There is a method by which digital hard disk sources can vari-speed, but we will skip that part for now.

IT'S GETTING CHUNKY IN HERE

With digital gear, all pieces in the puzzle must be locked to the same sample, or word clock, to operate without clicks and pops. In some instances I have seen two DAT machines act like they were synchronized, but there would be a slight click every few seconds. When the music was playing at a healthy level, it covered up the clicks, but when a piano chord was ringing out, you could hear the clicks easily. The reason it worked as well as it did is because accidentally the crystals of both machines were close enough together to only drift off one sample every few seconds, at which time one machine would snap to the next sample chunk like a chain skipping a tooth on a clogged wheel.

When you make a digital copy from DAT to DAT, or ADAT to ADAT, it works because the recording machine is synchronized to the playback machine by a digital clock. In this case, the clock is hidden in the digital audio traveling from the play machine to the record machine. It would work just as well if word clock out of the play machine was connected to word clock in of the record machine. If you had a third DAT machine connected to the digital audio out of the second machine, the clock information from the first playback machine would pass through the number two machine on to the number three machine. In some studios I see multiple DAT copies made this way all of the time, but each machine in the chain increases the jitter in the signal and increases the possibility of errors being inserted in the chain.

The proper way to connect the machines would be to use a DAT machine with multiple outputs, a digital processing box with multiple digital outputs like the TC Finalizer, or some form of digital audio distribution device like a Z Systems 8x8 routing box or the Aardvark Word Clock Distribution Amp. Now you have a single digital audio source from the playback machine that is cloned and distributed in parallel to as many DAT machines as you like. Same clock source, minimum jitter — the only way to go.

Remember to try to synchronize all of your digital machines, processors, and digital consoles in parallel as close to the common source as possible, unless the source is video sync, then run one master device from video sync and feed word clock from the master device to all of the other digital audio devices. That way, only one device has to cope with the task of deriving a word clock signal from video.

There are some devices available that provide common clocks from a single source. Aardvark makes the Aardsync, Mark of the Unicorn makes the MIDI Time Piece AV and the Digital Time Piece, Digidesign makes the Universal Slave Driver, Opcode makes a new clock source called the Studio 64XTC, and N-Vision makes master clock.

UN-SYNCABLE

Well, now that I've got you all worked up, I get that sync'ing feeling that I have to leave until next month. I have just touched the tip of the iceberg on a subject of Titanic proportions. And I am sure there are not enough life boats to save all of us. Me, women, and children first. EQ

Up To Our *ASCII in Mail*

PHOTO BY STEVE JENNINGS

Letters, we get electronic letters...

BY JON LUINI & ALLEN WHITMAN

It's that late winter lull and days are noticeably longer here in the Northern Hemisphere. It's time to think about spring. It's time to think about spring cleaning. It's time to clean out the mailbox!

We've been remiss. We've — *gasp* — allowed no small amount of e-mail from all over North America and Europe to pile up — seen, but unanswered. Since many questions revolve around a couple of central topics, we're going to pick a few that represent a broad cross section within our self-imposed focus of Internet audio.

The FezGuys always welcome comments (as this column will prove), and we suggest that questions, thoughts, and rants be posted on the threaded discussion area (tda) of the FezGuys's Web site (www.fezguys.com). By posting questions on the tda, everyone can see them and everyone can read the replies. Anyone can (conceivably) answer anyone else's question. This way we learn from each other. This way we share the wealth. Suddenly we're all comrades with a five-year plan. Isn't that awesome?

In related news: It's official. We have more columns then can safely fit on a page. Even wearing hardhats and eye protection, the FezGuys's Web site is running out of bounds. That, and the increased traffic on the tda is nudging us to redesign. Any suggestions about how to improve the site and what topics deserve expanded coverage are gratefully accepted. Do you think we need a search function for the Web site? Do you want to know more about how watermarking works? Whatever you feel is relevant, please drop us a line.

Let's go to the queries!

LONG DISTANCE RECORDING

I know this bass player that I used to jam with when we both lived in Hollywood, now he is back in England and I'm back in Cleveland. Is anyone multitracking/sync'ing via the 'Net? I would just love to cut some tracks with this guy. Is this possible? I have a Roland VS-880 and an ADAT plus a computer, obviously. Many thanks! —Wesley

It's feasible to use the Internet as an element of the song creation process. The key question here is *how* your bassist-friend is using the 'Net to record. He may have an ISDN line to send music "point to point" (i.e., a dedicated telephone line to send data from one physical location to another directly, bypassing the Internet entirely). If this is the case, you can order up your own ISDN line from your local telephone company and talk directly with your friend (with the appropriate hardware). For some useful information about ISDN go to: http://isdn.state.ut.us/.

If, however, he is using the Internet for real-time recording, he's probably not achieving a high-quality (44.1 kHz, uncompressed stereo) digital result. Let's face it: the Internet is not yet reliable enough for a musician to capture precise high fidelity. There are, however, several options available to communicate musically in "sort of" real-time with those unable (or unwilling) to show up at rehearsal. There is no substitute for the physical proximity of a live, ensemble performance, but there are several "next best things":

1. **Share musical ideas.** Attach a soundfile to an e-mail or place that soundfile on your site and make it available for downloading. Use your favorite audio compression tools (RealAudio, MP3, Xing, Audioactive, Liquid Audio, etc.) to reduce the transfer time. Make sure the person you are sending it to has the appropriate player.

2. **Send full RMF (MIDI) files.** Beatnik can do this (www.beatnik.com). You and your recipient use the Beatnik tools for encoding your audio as a MIDI file; thereby reducing the contents of your musical communiqué to a small string of digital commands that, when opened at the other end, translates exactly into the music you wrote. Since there is no loss in audio quality, you can transfer song elements that can be used in the final mix.

3. **Live jamming.** The ResRocket Surfer tools can do this www.resrocket.com. Download the player from their site and, again using MIDI, layer patterns together in real time. This is live jamming, and you know what that means...

Find out what the other player is doing, and with what equipment. From there you can duplicate their methodology and hook up with them!

THE RIGHTS STUFF

Let's go to another question, this time from Marten in Sweden.

I teach audio engineering at high school level, and we're just getting into the intricacies of Internet audio. We have a network of a couple of hundred computers at our school. The question is: Is it OK to download RealAudio and install it on the network server? —Marten

The RealAudio Player (not the "Player Plus") is free to use. There should be no problem downloading a copy for multiple users to access. Real Networks wants to be able to count each individual user in their install base. Recent revisions of their program include auto-registration (which includes notification of new versions). If you feel motivated to do the "right thing," drop them an e-mail and explain what you'd like to do and ask if they have any suggestions.

THINGS THAT ARE COOL

Now here's some people that are really doing something for the revolution. Straight outta North Carolina is the CDART site (**www.cdart.com**). They encode your video using Real Player streams and place it on their site, asking browsers to vote for their favorite. Depending on whether or not you go over well the video will be elevated from its starting point in the "Garage," to the "Club," to the "Music Hall," to the "Arena" and finally, if you've got what it takes, slaying all comers in the "Stadium." The site is a great example of Web-based musical community and, in keeping with what's right and proper, it's free. No cookies and a philosophy summed up in two words: "Be nice." In addition to being a cool and tongue-in-cheek online voting system, CDART also has a full service production studio for video and audio. It's an intriguing hybrid of traditional and multimedia methods and, as we intoned several months ago, is becoming more common.

This is interesting. Let's look at the facts. You're in Sweden! It would be great to have your students exchanging music with other students (think: audio penpals) using any number of streaming audio technologies. Let's not limit ourselves to just one codec. It would stand to reason that various technology companies would welcome your participation in the shared resource of a large academic user base. This is about education after all. Check out all the players and encoders. Contact the companies and explain your desire. The opportunities for students (which really means all of us) to use and, by using, increase our aggregate expertise in global audio communication is encouraged. It's been said before but it bears repeating: Knowledge is Power.

SAVE A STREAMING FILE

Here's another: *I want to save a streaming RealAudio file (while it's playing). How can I do this? —Brad*

The not-for-free version of the RealPlayer (RealPlayer Plus) has an option to record incoming streams to a file on your desktop. There is a catch — the people creating the stream must allow you to do so (no one really has, so it may be dropped from the product).

There are other options. The most obvious is to take an audio output from your computer and into your familiar tape deck. The quality may not be great but it is possible.

Another option is to download (instead of streaming) the RealAudio file to your computer. In the location window of your browser note the URL that is displayed while the file is streaming. At the end of that URL will likely be a ".ram" denoting the file type (in this case: a RealAudio streaming file). Try removing the "m" of the ".ram" so that it ends with ".ra." Hit return and see if the file will download. Files are saved in two ways when placing RealAudio encoded music on a server. When you remove the "m," you are entering (ostensibly) a likely location of the stored file from which the streamed audio originates. Be aware that in doing so, you may not be adhering to the author's wish — we recommend you ask the webmaster.

MPEG NEWS

I've been clickin' around and find that the MPEG format is my favorite. What is needed to convert soundfiles to this for-

continued on page 160


She only reads One Project Recording Magazine.
She's serious about her recording. And like over 110,000* project recordists who read EQ every month, she's invested a lot of time, creativity and money in her room. Project studio owner/operators own an average of $151,000 in equipment and are planning on buying an additional $32,900 worth of gear this year.* And nobody gets in the way of their buying decisions — project studio owner/operators are the decisionmakers*. No wonder project studio owner/operators like her are today's most powerful buying force. They are passionate about gear. They have the money to buy. And they get all their information from ads and articles in EQ. EQ magazine — it's the only recording and sound magazine that talks their language.
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The Art of Dynamics

From a purr to a roar — a look inside dynamics processors

BY EDDIE CILETTI

Studies of ancient cave paintings reveal that dynamics processing existed even in prehistoric times. One series of images begins with a man and a child in close proximity to a lion. With hand stretched out, the child is ready to apply affection, but, just before it can say "nice kitty," the parent quickly plucks the toddler from danger. After the chase, our ancestors are seen in a physically safe place, albeit still within earshot of the great cat. As the lion roars, both man and child have hands securely placed over their ears, equal to a brickwall limiter, with a ratio of infinite-to-one and the threshold set to 50 dB SPL. Tests are now underway to determine the unprocessed dynamic range of the lion — from a purr to a roar.

In more recent times, rodeo-style gain riding began with the ear as detector, the hand as control device, and the volume control as attenuator...

THE CLASSICS

This month's exploration into the blood and guts of dynamics processors begins with two classic cousins — the Teletronics LA2A (fig. 1) and the Universal Audio LA3A. Since both of these "Leveling Amplifiers" rely on the same optical attenuator, their response to dynamic challenges is nearly identical. Some of the sonic differences, however, can be attributed to the implementation: tubes in the LA2, transistors in the LA3.

Of course, there are many ways to dynamically skin a cat (any connection with the opening paragraph is purr coincidence). Classics like the UREI 1176 (fig. 2) and the Audio Designs Compex-Limiter, along with the more modern Empirical Labs Distressor, all use the Field Effect Transistor (FET). In the early '70s, dbx designed and built a transistorized Voltage Controlled Amplifier (VCA) as the gain manipulating device for their compressor/limiters. It also played a key role in their noise-reduction system.

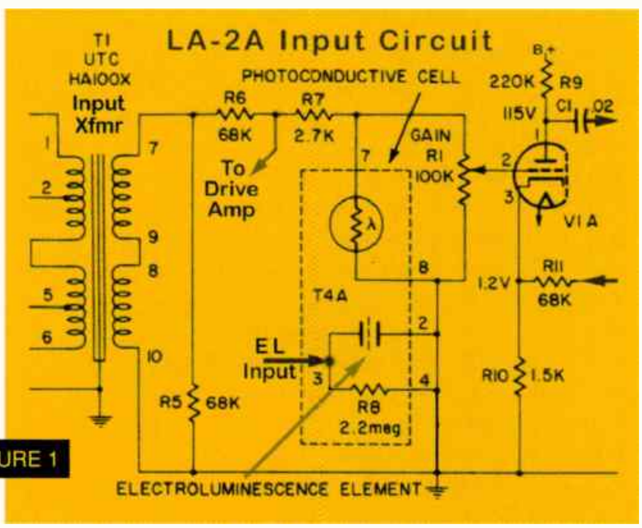

FIGURE 1

An optical attenuator is essentially a light-sensitive volume control. In the mechanical version, the knob (a.k.a. the control device) is connected to a wiper that "divides" one resistor into two parts — two resistors of equal value yield an output voltage half that of the input. Are ya ready? Let's go into the light...

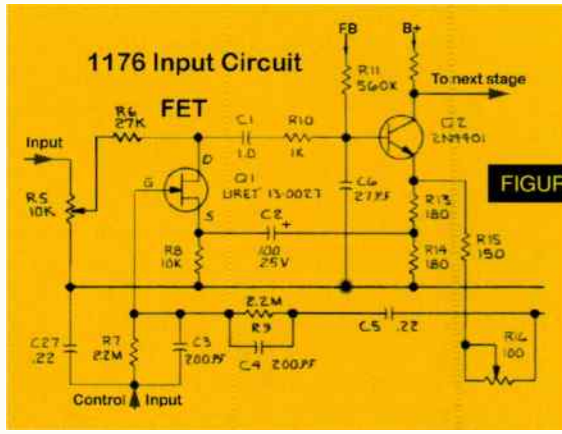

FIGURE 2

A photo-resistor changes its value with light: high when dark, low when bright. The filament in an incandescent lamp is too slow for transients, with a delayed "release" that's too long to be useful. Photo-resistor response varies, but most have a "natural" medium-fast attack and a nonlinear release that's initially fast, then slow. Achieving the desired characteristics requires testing and grading.

A LIGHT LUNCH

For speed, an LED light source might be the choice now, but the early '60s predates their birth. Neon bulbs have existed for years, but these devices are not linear. The answer came from technology employed by the aviation industry called "The Electro-Luminescent Panel." A high-impedance device, the EL consists of a capacitor sandwich sealed in a clear, flexible plastic (hold the mayo).

EL panels can be made into all sorts of shapes, a trait particularly well-suited for illuminating aircraft instrument panels — especially Warning indicators on planes — because they don't burn out. Available in a variety of colors (photo-resistors are color sensitive), ELs are currently used to backlight the LCD displays used in laptops and other fine electronic toys.

OK, LET'S SPLIT

After a signal enters any dynamics processor, it splits in two. One half is routed to the gain manipulating device while the other heads off to the detection circuitry — each with its own volume control. In the optical case, an amplifier drives the EL panel. Louder equals brighter equals more attenuation.

Note the variation in drive circuitry between the LA2 (fig. 3) and the LA3 (fig. 4). Both are simple amplifiers, but the transistor circuit uses a step-up transformer to generate the necessary drive voltage for the EL panel. General Electric used EL panels in some of its night lights because it takes 120 volts AC to get 'em going!

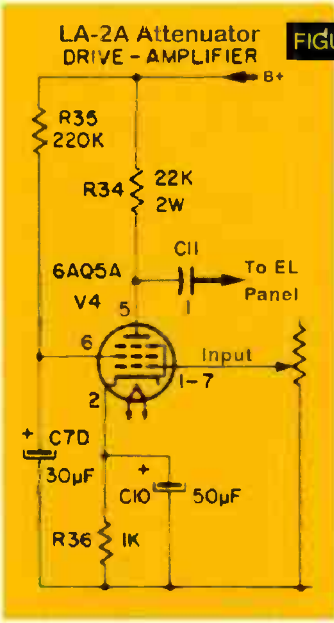

FIGURE 3

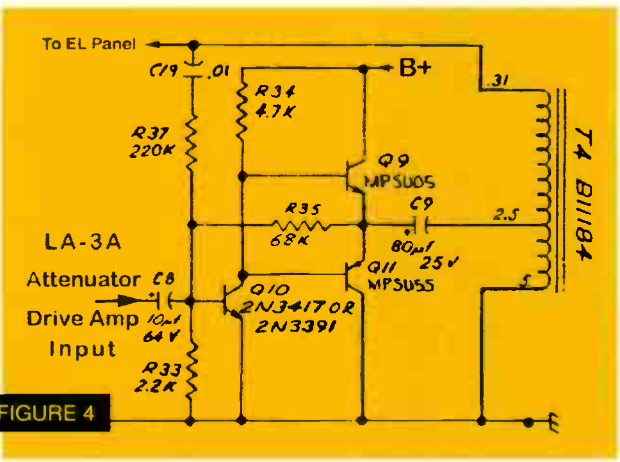

FIGURE 4

Unlike the optical attenuator, both an FET and a VCA require a more sophisticated detector circuit. The audio signal (AC) must be converted into a corresponding DC voltage and then further manipulated by the more familiar, attack, release, ratio, and threshold circuitry. Drums and vocals are diverse sonic challenges that require additional detector sophistication. Features like Peak and RMS detection make the processor more versatile.

With any dynamics module, the great question is whether the metering accurately reflects the processing being done. You may have noticed that the optical attenuator has at least two photo-resistors, one for gain reduction and one for metering that gain reduction. LA2A's particularly become inconsistent over time because their photo-resistors will age differently. In addition, when the meter's 0 VU setting wanders, check the "voltage regulator," which for this circuit is a *neon bulb*! Remember I mentioned that the neon bulb is nonlinear? Well, once "fired," it acts like a zener diode, clamping at 65 volts. Neon also degrades with age and should be replaced by a zener diode. See fig. 5.

How many engineers does it take to change an optical module? Just one! For a new part — called the T4B — all it takes is a phone call to JBL (818-895-3417) and $160. An alternate source for new ($145) and rebuilt ($60) modules is ADL (914-256-0032). EQ

Aimlessly wandering the Web? Drop by www.tangible-technology.com or send e-mail to: edaudio@interport.net.

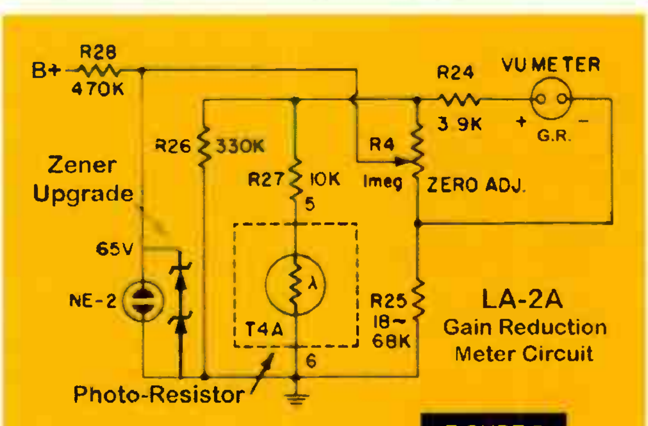

FIGURE 5

To Tax But To Dream

The Powers That Be scheme to get more money out of you and your project studio

BY MARTIN POLON

These days, it seems, that the two most dangerous occupations appear to be President of the United States and the owner/operator of a project recording studio. In both cases, there are interferences to an uneventful life that seem to come "out of the blue." There is little any of us can do to ease the burdens of the Presidency, but there are usually some things that can be done to keep project studios on the proverbial "straight and narrow!"

We now have a new kind of local and/or regional taxation plan that is sweeping the nation and that encompasses the project recording studio within its jurisdiction. It also fundamentally changes the relationship between these studios, the local and/or regional authorities, and zoning restrictions involving home businesses, including recording studios.

Many owners of such studios frequently talk of a taxation "conspiracy" while deep in their cups of single malt whiskey at a hotel bar during an audio trade show or professional meeting. In fact, they may be at least in some small part correct. Taxation officials at all levels meet at professional meetings devoted to taxing issues (no pun intended or offered) in exactly the same way that audio professionals travel to AES, NAMM, NAB, NSCA, etc.

The tax people exchange ideas and concepts with their local, regional, state and Federal counterparts and with international colleagues. So at these tax trade meetings, one will find representatives from Tacoma, Washington, Tampa, Florida, and especially (and most aggressively) Los Angeles, California, finding new ways to tax all home businesses but especially audio, video, and multimedia activities in the home. Representatives of scores of other municipalities are deep in discussion with those who are now taxing and wish to find out how they can begin such taxing — not regulating — of all in-home businesses in lieu of previous prohibitive zoning practices.

They acknowledge the fact that one business out of five is conducted from the home and that from 15 percent to over 50 percent of all new business licenses are for in-home activities. For those tax authorities using the new liens, there is the taste of new and relatively unlimited revenue. Listening to all of this as well are representatives from the UK's Inland Revenue, various European Community taxation authorities, the former Soviet Union's CIS, and even Middle Eastern powers such as Israel and the Independent Palestine Authority. Politics may be politics, but taxes are taxes and they come first.

> *The total for all of these taxes can reach as high as the thousands of dollars level, depending upon the activity at the facility and the jurisdiction.*

The practice of taxing is slightly slanted by the estimation in the minds of tax authorities that anyone doing audio, video, or multimedia software or project authoring, development, production, or postproduction is making "big" money and should be taxed for it. The new practices seem to operate as follows:

1. Existing zoning restrictions are suspended since to tax and to zone simultaneously raises questions of double jeopardy.

2. The first tax levied is a basic business class tax applied to all businesses in the locale in question, regardless of the location.

3. The second tax levied is a home business tax, which is considered a license to do business in the home.

4. There are numerous jurisdictions that have applied or are in the process of applying a gross revenue tax for home businesses.

5. Similarly, some jurisdictions are seeking to put home businesses under the provisions of yearly business inventory taxes.

The total for all of these taxes can reach as high as the thousands of dollars level, depending upon the activity at the facility and the jurisdiction. The argument that no commercial activity takes place or that no commercial activity takes place until a project is sold does not protect the project studio because the concept of taxation is in most cases a license to operate first what resembles a business, then secondly a business-like activity in the home.

That all of this will change the basic status of project studios remains to be seen, and that the studios who remain hidden in the suburbs face a mounting schedule of back taxes is also an issue. We are already seeing some individuals nationwide moving their business activities to a second home or other location in a more benign locale. Hollywood screenwriters are challenging the Los Angeles statues in court to try and force the status of free speech and the basic right that a man's (woman's) home is his (her) castle. The outcome of these legal proceedings could change the way this new mode of taxation proceeds, but one thing is for sure — operation of a project studio will never be quite the same. EQ

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• TRS jack w/ level control for monitoring.
• Includes shoulder belt, AC adapter, & battery.
HHb
PDR1000/PDR1000TC
The PORTADAT has fast become an industry standard location DAT machine. Popular for film and video production use, as well as ENG/EFP, the PDR1000 features a large backlit display, 4 motor transport and AES/EBU and SPDIF digital ins and outs. The POR1000TC adds the ability to record, generate and reference to timecode as well as jam sync to convert absolute time to timecode.
FEATURES-
• 4 head Direct Drive transport
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• L/R channel mic input attenuation selector(0dB/-30dB)
• 48V phantom power, limiter & internal speaker.
• Illuminated LCD display shows clock and counter, peak level metering, margin display, battery status, ID number, tape source status and machine status.
• Nickel Metal Hydride battery powers the PDR1000 for 2 hours, AC Adapter/charger included.
POR1000TC Additional Features–
• All standard SMPTE/EBU time codes are supported, including 24, 25, 29.97, 29.97DF, & 30 fps.
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NEW Options-
• MS1000 Master Sync module ensures drift will be no more than 1 frame every 10 hours.
• HM1000 Headphone Matrix provides a rotary switch for selection of Stereo, Mono Left, Mono Sum, & M/S (mid-side) Stereo modes.
SONY
TCD-D10 PROII
A reliable, high-performance DAT recorder, the portable TCD-D10 PROII is designed with rugged professional use in mind. It has many enhanced features including absolute time recording, allowing immediate use of the tapes as source material for Sony's PCM-7000 series DAT editing system.
FOSTEX PD-4 V.2
The second-generation portable, professional TC DAT recorder from Fostex. Version 2 software includes many features and functions for improved performance in the field.
FEATURES–
• 4-head design • Punch-In/out
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• 3 in X 2 output mixer with 3-position pan functions
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A great combination for any studio owner with an ear for the best. The PCM-80 delivers high quality multi-effects based on the legendary PCM 70, maintaining Lexicon's high standards for sonic clarity and extraordinary processing power. The PCM 90 is a digital reverb with its roots stemming from the studio standard 480L and 300L effects systems. Reverbs from telephone booths to the grand canyon, the PCM 90 is incredibly realistic, an excellent addition to any rack mount arsenal.
Buy a PCM-80 from B&H before December 25th, 1997 and receive a FREE Pitch FX Card & $100 - a $350 Value!
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lexicon MPX-1 Multi-Effects Processor
Lexicon's latest addition to their Digital effects family, the MPX-1 features top-quality effects in an easy to use, 1 rack space unit. With 56 Pitch, Chorus, EQ, Modulation, Delay, and world-class reverb effects accessible from the front panel, as well as TRS and XLR balanced I/O and complete MIDI implementation, the MPX-1 creates a new standard for cost and quality in a multi-effects device.
t.c.electronic
Wizard M2000 Studio Effects Processor
The M2000 features a "Dual Engine" architecture that permits multiple effects and 5 different routing modes making it a great choice for high-end studio effects processing.
FEATURES–
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• "Wizard" help menus, 16-bit dithering tools,
• Tap and MIDI tempo modes.
• Single page parameter editing, 1 rack space.
SONY
DPS-V77 2 Ch. Master Effects Processor
Sony's latest effects processor, the DPS-V77 yields excellent sonic quality combined with realtime control, a digital I/O and many more features that will put a smile on the face of any discerning studio engineer.
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• Use the AES/EBU & SPDIF digital I/O to link multiple V-77s together & when working with digital mixers
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FEATURES–
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K2500 Series Music Workstation

Building on the same features that made the 2000 series popular, the 2500 utilizes the acclaimed V.A.S.T. technology for top-quality professional sound. Available in 76-key (K2500), 88 weighted key (K2500X) and rackmount (K2500R) configurations, the K2500 combines ROM based samples chosen from the best of Kurzweil's collection, on-board effects, and full sampling capabilities on some models (K2500S, K2500RS & K2500XS).

FEATURES–
- True 48-voice polyphony
- Fluorescent 64 x 240 backlit display
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SAMPLING

E-mu Systems, Inc. e-6400 Emulator

The e-6400 offers the power of E-mu Systems' renowned Emulator Operating System (EOS) and superb audio quality in a package perfect for the budget-minded professional. The e-6400 comes with stereo sampling, 4MB of RAM and is fully upgradeable to E-mu's top of the line Emulator sampling synthesizers, the E4X, and E4XTurbo.

FEATURES–
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MIDI HARDWARE

Mark of the Unicorn — MIDI Time Piece™ AV
8x8 Mac/PC MIDI Interface

The MTP AV takes the world renowned MTP II and adds synchronization that you really need like video genlock, ADAT sync, and word clock sync, even Digidesign superclock'

FEATURES–
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Mark of the Unicorn — Digital Time Piece™
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Think of it as the digital synchronization hub for your recording studio. The Digital Timepiece provides stable, centralized sync for most analog, digital audio, and video equipment. Lock together ADATs, DA-88's, ProTools, word clock, S/PDIF, video, SMPTE, and MMC computers and devices flawlessly. It ships with "Clockworks" software which gives you access to its many advanced features and remote control of some equipment settings such as record arm.

OPCODE Studio 64X
Cross Platform Interface

The Studio 64X is part of Opcode's new Studio X Series. This 64 channel MIDI interface is compatible with both Macintosh and Windows and features SMPTE synchronization as well as OMS (Open Music System) compatibility for total integration with any OMS compatible application.

FEATURES–
- Cross-platform Mac and Windows compatibility
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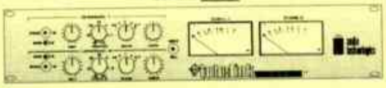

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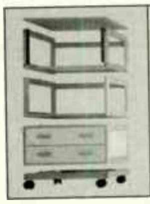

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FEZGUYS

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mat? Thanks, Sev

There's tons of information about MPEG audio. For MPEG Layer 3 (MP3) specific information, go to: **www.mpeg3. org**. For general MPEG information, go to: **www.powerweb. de/mpeg/**. Also: Shockwave and Audioactive use the MP3 codec as the basis of their encoder/player "suite of tools." When Xing's v3.0 is released upon an unsuspecting world it will be adding MP3 to its already existing MPEG support. Of course, there are many ways to skin the proverbial Internet audio cat. We recommend messing around with all of these formats and see which one you like the best. (Audioactive, **www.audioactive.com**; Liquid Audio, **www.liquidaudio.com**; RealAudio, **www.realaudio.com**; Shockwave Audio, **www.macromedia.com**; Xing Streamworks, **www.xingtech.com**.

FEZ FACTS

And the last victim (for now): *I just have a question about your nomenclature: Where does the "fez" name come from? Did you two first meet at a Red Fez Shriners meeting? Is "fez" code for an old band joke? Or did you both happen to have red hats and thought there was some sort of destiny? (FezGuys...comes with everything you see here, action figures sold separately.) —Bobby*

That information is classified. If we told you, we'd have to kill you.

In keeping with the rather sun-drenched tone of this particular column, we include a sampling of some of the more interesting ways our community signs off: "making things rounder and redder," "becomes rounder and redder...," "my signal sounds like it's been through a Cuisinart...any ideas?" "I'm looking for a fez," "let your fez flag fly," "go fezguys, go!" "what does MPEG stand for anyway?" "...how's the rutabaga hanging?" "in fez we trust," "blah blah thanks again," "may the fez be with you," "some people were born to wear a fez," "Shriners all over the world are turning in their graves," "you guys are sincerely whacked," "I plan on actually using this info once I pull my fez out of my ass."

What is up with the fez on Mr. Hankey? We welcome your comments. Duh! Visit us online at www.fezguys.com.

EQ&A

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ever, being capable of sending eight channels per each of its card slots, outputs 24-bit signals this way: the first signal is split between tracks 1 & 5, the second to tracks 2 & 6, the third to tracks 3 & 7, and the fourth to tracks 4 & 8. Therefore, 24-bit tracks recorded using the PaqRat must be played back through the PaqRat, and 24-bit tracks created using the 02R and must be played back using the 02R.

3. Your request for 24-bit converters on the 02R. The 02R's 20-bit converters already generate usable dynamic range that is significantly greater than what the CD can offer. For those who would like to hear the 02R's converters in action, listen to James Taylor's *Hourglass* CD, recorded on the 02R by Frank Filipetti. *Hourglass* has been nominated for multiple Grammys, including Best Engineered Pop Album. [Also check out the story in the November '96 issue of *EQ*.] As 24-bit converters begin to appear, they can be connected to the 02R using AES/EBU format.

Peter Chaikin
Product Manager
Yamaha Corporation of America

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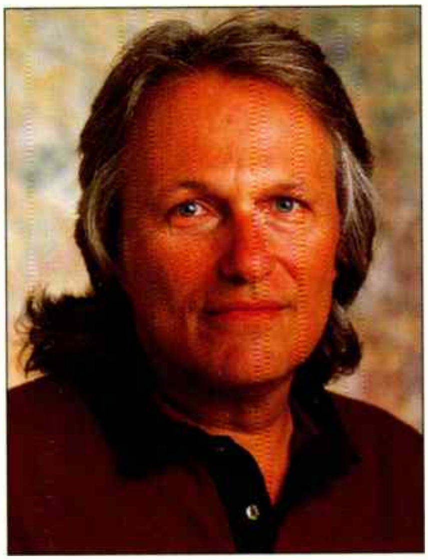

Do you know synchronicity?

BY ROGER NICHOLS

From time to time I have discussed the problem of synchronizing multiple audio devices. More and more often, these audio devices are becoming digital. The fact is, that there are fewer and fewer companies even manufacturing analog tape machines. Soon you won't be able to purchase a new analog machine from anyone.

The two principal ingredients in synchronizing multiple devices is speed and time. There must be some way to tell where you are and how fast you are going. With analog audio machines and video machines, the answer is easy: video sync and SMPTE timecode. Video sync provides a speed reference and SMPTE provides time and speed reference information. With the advent of digital audio recorders and hard-disk recording, two new parameters enter the game: word clock and trigger point.

SMPTE comes in several flavors, including 30 fps, 29.97 fps, 25 fps, and 24 fps (fps = frames per second). In the visual world, these refer to how many video or film frames are shown each second. Audio on analog tape does not occur in discrete chunks. The audio can be recorded at any speed from 1 7/8 inches per second up to 30 inches per second, and anywhere in between with the use of variable speed tape machines. If you only want to synchronize analog audio machines, you can use any rate of SMPTE that you desire. If you plan to use inexpensive synchronizers, the code must be the same on all machines, but more expensive synchronizers allow you to use different frame rates on different machines and still keep them locked together. You could have one machine running at 7 1/2 ips and another machine running with the VSO cranked wide open to 45 ips and they would stay synchronized just fine...Well, almost.

> *With digital gear, all pieces in the puzzle must be locked to the same sample, or word clock, to operate without clicks and pops.*

ANY WEEK NOW

Sometimes it is cumbersome to keep two machines running together throughout a project. Sometimes two machines won't be available, and sometimes you just don't want to wait for the machines to synchronize. Sometimes it takes 10 to 15 or more seconds for two analog machines to finally lock up. Most analog machines do not mute their audio while in the locking process, so you must endure the pitch changing and warbling of the slave machine until it is in sync. Sometimes this can be very distracting when you are trying to minimize run-up time while working on a guitar solo or vocal chorus. The answer? Slave tapes.

A slave tape is a tape that will be played on the secondary machine that chases the master. The slave tape can, however, be used on its own. If you intend on using a slave tape by itself, it must have enough information on it so you can hear all of the instrumental parts you need to do the overdubs. Since a lot of the instruments were recorded on the master tape, you need to make a rough mix of these tracks and copy them over to a minimum number of tracks on the slave machine. At some point, the slave tape will no longer need to be used alone, and the reference tracks can be erased and the slave tape sync'd back up with the master.

If you synchronize the master and slave tapes before the reference tracks have been erased, you will hear phasing between the original tracks that exist on the master and the reference track of the same instrument that resides on the slave. This is because of the resolution of the synchronizer, the wow and flutter of the analog tape machines, and the accuracy of the SMPTE timecode recorded on the tape. You can not get rid of the phasing completely, so everybody has secretly promised not to record multi-microphone images such as drums or stereo instruments with tracks that span machines. That way you will not generally notice the discrepancies.

OK, CAN I TALK ABOUT DIGITAL NOW?

Here is where word clock comes in. Digital audio is recorded in chunks. CDs have 44,100 chunks per second. Digital audio on video tape has 48,000 chunks per second. When synchronizing two digital machines they must have the same number of chunks per second. (There are exceptions, but it gets tricky.)

Digital audio should be synchronized speedwise by using the same word clock for both the master and slave. There is no phasing when you listen to a stereo track split between the two machines because all of the chunks line up exactly. All you have to do is get them to the same spot and they will stay together forever. SMPTE can be used for the purpose of getting to the right spot — just like it is with analog audio.

Hard-disk recorders keep tracks synchronized by keeping track of sample numbers. If the guitar part starts at sample number 4,800,000 and the drums

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