CREATIVE SAMPLING TECHNIQUES

EQ

THE PROJECT RECORDING & SOUND MAGAZINE

SEPTEMBER 1996

IN REVIEW
SABINE
SYMETRIX
DIGITECH
NIGHT TECH
BELLARI
HORIZON
TL AUDIO

YAMAHA 02R RECORDING TIPS BY PHIL RAMONE

TOP MIXER DISC PICKS: LIVE REFERENCE CDS

ROOTS OF ROCKABILLY RECORDING BY CARL PERKINS & DAVE MCGEE

CYNDI LAUPER

STUDIO IN SECLUSION

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VOLUME 7, ISSUE 9
SEPTEMBER 1996

FEATURES

EQ LIVE

TECHNIQUES/WORKSHOPS

COLUMNS/DEPARTMENTS

ON THE COVER: *Cyndi Lauper at her upstate New York project studio. Photo by Julian Jaime.*

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EQ

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Cover Your AES

BY MARTIN PORTER, EXECUTIVE EDITOR

Fortunately for those of us who make a living in recording, there is actually someone out there who understands what "Isochronous Data Transmission and Management for the IEEE 1394-1995 High Performance Serial Bus" really means. It's probably the same person who also can appreciate the significance of "Audibility Analysis of Processing Noise in Digital Filter Morphing Schema."

The brainiacs who make (and attend) presentations such as these at the AES are the scientific backbone of our business and art. Without them we would still be scratching our songs out of wax cylinders.

The AES, otherwise known as the Audio Engineering Society, is the world's preeminent association of audio geniuses. It's where all the basic technology that makes its way into home theaters, car stereos, HDTVs, DVDs — not to mention home studios and concert PAs — begins. It's where several thousand extremely smart scientists get together twice a year to hash out the circuitry that ultimately creates the next great signal processor that ends up inside that $99 black box that you'll buy at your local dealer.

Once each year, the AES holds a convention in the U.S. For three days it's a gear slut's holiday, with over 300 pro-audio manufacturers exhibiting their wares. And for those of us who really don't care where the basic technology comes from (as long as our ADATs don't cough up our tapes), the AES Convention is "the" place to see what's new and find out what's happening.

Yes, recording musicians, engineers, and creative production yahoos are finally discovering the importance of the AES Convention and, while we're all probably giving heartburn to the scientists who once thought it was their exclusive club, AES has become the ultimate place to shop and talk shop. It's where someone will finally tell you how to get the ultimate snare drum sound. It's where you'll see Rupert Neve hanging out with Nick Franks (both in the flesh). It's where you can finger a fader on the new SSL 9000 without paying $250 per hour to get the privilege. And for the liner-note groupies among us, it's where you'll see Bob Clearmountain mixing, George Massenburg listening, George Martin being George Martin, and Phil Ramone simply holding court (Roger Nichols is there, too — but, please, no autographs).

If you own a studio, work in a studio, or care about studios, you can't miss the AES this fall (L.A. Convention Center, Nov. 8-11). You have to preregister, so go ahead, drive those receptionists crazy at AES in NYC (212-661-8528). Tell 'em *EQ* invited you.

This year you'll see the next great digital project console from Yamaha before your dealer gets them in stock. And you'll finally be able to corner that tech in person who you've been driving bonkers over the phone at Digidesign. Besides, now that AES has been infiltrated by redblooded end-users like ourselves, you'll even be able to attend four-days' worth of the finest recording workshops on the planet. (*EQ* recommends Craig Anderton's workshop on digital recording techniques and Russ Berger's session on project studio design).

And who knows? You may even decide to sneak into one of the technical sessions after all. (If you figure out what they mean by "Transform-Domain Weighted Interleave Vector Quantization," please let me know). Just before you leave the Los Angeles Convention Center make sure to pick up your AES membership application.

When *EQ* readers start becoming members, that's when AES will start getting especially cool. EQ

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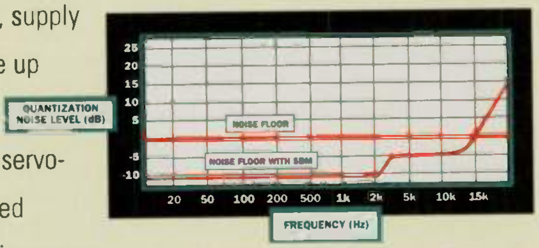

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SONY

REVIEWING THE REVIEW

JBL Professional would like to thank Robert Scovill for taking the time to audition our new TR and MR900 Series of sound-reinforcement products [see *EQ* August '96]. We are certain that he is used to listening to "world class" sound-reinforcement systems. We are also sure that most of those systems are JBL. That notwithstanding, his interest in our Musician Market product shows his belief that JBL Professional products are worth a look at any price point.

In his review, he hits the nail on the head when he concludes that: "JBL Professional is apparently focusing a lot of its R&D energy towards a broader audience. The MR and TR series are certainly another step toward reaching the users for this legendary company."

He is absolutely right. A significant part of our new product R&D focuses on what the end users in every segment of the professional sound-reinforcement market have told us they need. Whether it is the FOH engineer who needs a world class system or the working musician who needs the same power, clarity, intelligibility, and portability in a more affordable package.

Consequently, what Robert pinpoints in his audition of the MR900 and TR Series is exactly what the end users for this type of product asked us for. It also pinpoints why the TR Series only has four models and MR900 has eight offerings in the line. The TR Series is a simpler line purposefully, so that their application can be addressed more easily. MR900 offers more specific choices for the more experienced user who wants to choose the right speaker for a specific application. These applications are more likely to include the use of a subwoofer with a multi-amp setup.

The MR900 Series is an improvement on the original MR800 Series, one of the most commonly used sound-reinforcement boxes in the world. This improvement dramatically benefits working musicians who want exactly what Robert described in his review of the MR925 [when he said,] "...vocals sounded the best." The MR Series is at its best when it is supporting vocals over a band that is grooving away.

On the other hand, the TR Series was developed to address a segment of the musician market that had not been addressed by JBL Professional before. JBL Professional has always been an aspirational brand for up-and-coming musicians as well as the huge DJ and Karaoke segments of the worldwide market. Their needs are different from the user who has been at it for a while and has begun to generate income from their involvement. They need quality choices at this end of the price spectrum. These guys go to high-level concerts and clubs and see professionals like Robert using JBLs and say to themselves, "Someday, I'll have JBLs too." With TR Series, "someday" has come!

As I wrote earlier, one of the most important concerns that our end users expressed was price. For most speaker manufacturers, this would mean compromise. Instead, JBL Professional decided to give them more. For instance, the horn material, even though it is not the composite material of the MR horn, is made of an ABS that is akin to what football helmets are made from. The 90° x 45° "Directivity" (which Robert refers to as "Q") of the TR Series horn was purposely narrower than the 100° x 80° of the MR900's flat-front Bi-Radial®. Robert was very astute to notice that it sounded better to him with the prerecorded music since a narrower coverage pattern creates a nicer "sweet spot" like what your home Hi-Fi might produce. The wider directivity of the MR flat-front Bi-Radial horn (made of a more durable, albeit more expensive, nonresonant composite material) is needed for the working musician who has experienced not knowing what the room conditions will be like when they show up and may need the added controlled dispersion. An added plus is that the lower directivity of the TR horn can also help the less experienced user with unwanted feedback.

JBL Professional believes that with its lineup of TR, MR, SR, and EON end users now have quality choices to match any application.

Philip Manor & Mark Gander
JBL Professional

REVIEWING THE REVIEW: JBL TR Series Speakers

LIVE WIRE

Thank you for the extended live section in the July issue, especially the Greg Price and Eddie Ciletti articles. I own a small sound-reinforcement company in Memphis, TN and I am constantly amazed at how many "live engineers" don't know the first thing about gain structure and how important it is in building the overall mix. The two different schools of thought in this area are: "Everything needs to be at unity gain man!" and "This console doesn't sound good unless you snap a couple of VU meters in half."

The "unity gain" theory is usually believed by the guys that have just enough knowledge to be dangerous. They can't figure out why it sounds like crap when they have every gain knob in the signal chain set at 12:00 or 0 VU. I do believe that certain pieces of gear (EQs, crossovers, compressors) sound better when they are bumped up a little, especially some of the cheaper, unbalanced gear (check those –10, +4 switches).

The bottom line is that you want to give yourself the most headroom possible with the equipment that you are using. If you can turn up the EQs gain and increase your output gain by 50 percent without increasing the noise floor or clipping the input to the next piece of gear, then do it! It will only allow you to mix easier.

I think Mr. Price's plumbing theory is the best way to explain it. If you picture the signal chain as a series of different size pipes connected with

variable valves or faucets, then it should start to become clear. If you have one faucet on full blast shoving water into the next pipe and its faucet is barely open, then what you usually get is a burst pipe, or distortion. Have all of your faucets set so that water can flow as easily and as much as possible, and instead of drowning, you should have plenty of usable headroom.

William J. Floyd
Profound Sound
via Internet

COMPUTER FIX

In reference to your PC-based Q & A on Russ Diamond's request in the July issue of *EQ*, thank you for pointing out that there is plenty going on in the PC field in the software-only category. However, there are also several dedicated systems that more directly compete with Pro Tools on the PC platform, including SADiE (my personal favorite), Spectral Synthesis, Emerging Technologies, Micro Technologies, and the tripleDAT system that you mentioned.

Also, the "A/V" designation has little to do with ATA-2 (or Fast IDE) compatibility. "A/V" generally refers to a SCSI drive's capability to negate or defer Thermal calibration, which can interrupt data flow briefly. This is not a hassle on data when making a disk copy, but can be crippling on systems handling audio — particularly when burning Red Book CDs, which insist on being written in disk-at-once mode. In other words, if you interrupt that data, you just made another coaster. As drives and software continue to improve, this will be less of an issue.

Finally, use the tools you are comfortable using. The old software adage goes "choose the software, then the platform." So if you love the way Pro Tools runs, get a Mac, but try out several things before you buy. You may find something you like even better on a platform you already own.

Jeff Mac
SADiE Inc.

CORRECTIONS

Due to a printing error, several paragraphs were cut from Frank Serafine's article entitled "Lights! Camera! Audio!" in the August '95 issue of *EQ*. Here are the missing words in cut-and-paste form (we mean with scissors, computer heads):

Place this set at the bottom of the first column on page 70:

I also create many sound effects with electronic musical instruments, such as the Yamaha VL-1 interfaced to a WX11

Place this set at the bottom of the third column:

We also use the KYMA, a sound design tool from Symbolic Sound, which I refer to as the morphing synthesizer. It actually allows you to take two different

We apologize for causing any confusion.

WRITE TO US

We haven't heard from you lately. Was it something we said? Send your cards, letters, or postcards to:
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EQ&A

HOT WIRED

Q *I have wired my project studio with eight XLR balanced lines from my studio to control room. All eight of my channels have a noisy hiss on them along with periodic crackles. I am using an AKG 414 and an AT 4033, which both require phantom power. Does phantom power require a different wiring scheme on the XLR?*

Bryan Carey
Visual Reality
via Internet

A Phantom power requires no special wiring variations assuming you have made the following connections at both the studio and the control room ends of the harness:

1. Shield on pin 1
2. Hot on pin 2
3. Low on pin 3

What you didn't provide are additional details that might help provide insight. That is,

a. Is the problem only with the mixer, and what type of mixer is it?

b. Does the problem also show up with (and have you tried) outboard mic preamps?

c. Are cable connectors at both ends?

d. Are connectors at either end mounted to a box or panel?

e. Is the panel conductive or insulated?

f. What type of cable was used?

g. Is there a possibility that your mic panel (if applicable) has come in contact with the electrical power ground?

Measure the voltage from pin 1 to both pin 2 and pin 3. There should be about 48 volts each time. If not, check the voltage at the mixer. Another test would be to plug the microphones directly into the mixer. If the problems go away, something is up with your wiring. Take a known, properly wired cable that is long enough to reach from the studio to the control room. Use a VOM and check continuity (using the "ohms" setting) to make sure you didn't accidentally flip the wires on pins 1 and 3.

Eddie Ciletti
Contributing Editor
EQ Magazine

MIDI ON MY MIND

Q *I want to control the Yamaha 02R's talkback feature, which responds to MIDI note middle C (note number 64). The simplest way I can think of to control this feature is to buy a cheap MIDI controller to send the note, but this seems like overkill. Is there a less expensive solution?*

Francis
via Internet

A MIDI Solutions Inc. makes a product called the Footswitch Controller; it has MIDI in, MIDI out, and a 1/4-inch phone jack that can connect to a footswitch or just about any other type of switch. You can program the unit to transmit a MIDI note upon contact closure, including sending a middle C to the 02R. For more information, contact MIDI Solutions Inc., 816-810 W. Broadway, Vancouver, BC Canada V5Z 4C9. Tel: 604-794-3013; fax: 604-794-3396; e-mail: info@midisolutions.com.

John Fast
MIDI Solutions

A LITTLE LOOPY

Q *Mr. Stosich, in his letter [July '96] complimenting Eddie Ciletti's column, mentions that when his Sony PCM-2300 was "...ejecting a DAT, it happened to pull a little loop out of the shell."*

I am having similar problems with a Sony DCT-A7, as it also is pulling a loop out of the shell when ejecting the tape.

Has Eddie Ciletti heard about this problem on Sony DATs? Is it happening to others? Is there a design flaw in the Sony DAT? Is there a prior column on the maintenance of the Sony DAT that perhaps addresses this problem? If so, please let me know which one.

I do live location recording on a Sony PCM-2300 and playback on a Sony DCT-A7, so I get a little apprehensive when I hear of a related problem.

Paul Butterfield
Overland Media
via Internet

A Whenever a machine mishandles a tape, it's time for service. Don't wait for it to happen a second time — especially to an important tape. Do test the machine several times using a noncritical tape.

Both the PCM-2300 and the DTC-A7 use the same transport. (Both

models have been discontinued.) These transports are known to have problems. I have seen press-fit reel tables, which are spring loaded, expand from pressure and cause major tension headaches such as no take-up or slow wind (non-play) speed.

Each reel table should have a small amount of vertical play. If not, it's time for a trip to the mothership. Sony charges $150 for adjustments and $400 for a transport swap.

Eddie Ciletti
Manhattan Sound Technicians
NYC, NY

PINCH ME

Q *My ADAT is chasing to Pro Tools Transport, but I don't get any metering in the software nor on the front-panel display on the ADAT.*

Amos Vultaggio
Fairfax, VA

A Pinched or stretched optical cables are the leading cause of this failure. Fiber-optic cables are extremely delicate and prone to failure, especially when stressed with the weight of other cables or stretched beyond their capacity and tolerance. Check those cables!

Tom Cockrell
Technical Support Manager
Digidesign

ANCHORS AWEIGH

Q *I own a Sony PCM-2500 (yes, I still use the old boat anchor, but I paid full price for it when it first came out, and I can't afford a new one as of yet...ouch!).*

I would like to be able to use the S/PDIF inputs for 44.1 Hz direct recording from my PC using the DAL digital-only card. However, Sony added a copy-protect feature preventing me from using this function. I heard somewhere that there is a way to bypass this. If anyone would know this, it would have to be Eddie. Please help!!

Robert Sampler
Tempe, AZ

A Several informed sources tell me that copy protection and the Sony PCM-2500 are murky waters. The current version of SCMS, the copy-protection scheme, looks for protection flags coming in through S/PDIF ports, but not through AES ports.

The PCM-2500 is the professional version of the PCM-1000, a consumer product that recorded only at 48 kHz. There is a mod for the 1000 that allows it to record at 44.1 kHz. The gray area on the PCM-2500 is that both the S/PDIF and AES ports see copy-protection flags. While a modification may be possible, there is no "official" documentation from Sony, and so the research alone makes the task cost prohibitive.

Mechanically, the PCM-2500 has a great four-motor transport, but its lack of absolute timecode and incredible bulk do make it an anchor of sorts. It does, though, make a great playback machine. You can even use the RCA jacks on the top section without connecting the lower portion (the digital and analog I/O) for playback only.

Both Digital Domain and Z-Systems make boxes that change the status bits (which include the copy flags) for about what you'd pay for the mod. Perhaps those companies would let you try their product first to make sure it answers your needs.

Eddie Ciletti
Manhattan Sound Technicians
NYC, NY

WHERE ARE Q?

Q *I've heard good things about the "QSYS" TDM Plug-In Localization Processor, but can't seem to get solid information on it. Can you help?*

Mike Rogers
via Internet

A The QSYS Plug In is a software-based localization processor that permits the panning of mix components into an expanded stereo field. The QSYS 4-channel system is one of the products in the QSound Labs line of soundfield expanders. Contact them directly at: QSound Labs, Inc., 2748 37 Avenue N.E., Calgary, Alberta T1Y 5L3 Canada. Tel. 403-291-2492. WWW: www.qsound.ca

Hector G. La Torre
Executive Director
EQ Magazine

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The FR Series Development Team

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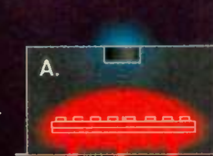

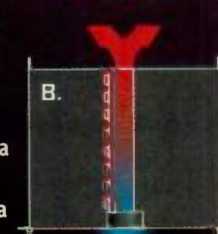

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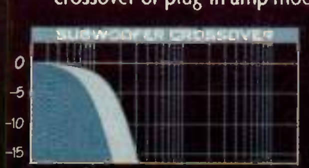
80° F (and potential failure).

The M•1200 uses a T-design that cuts tunnel distance in half. All power transistors are flooded with cool air concentrated through an oversize front manifold (Fig. C) that keeps airborn spooge away from internal electronics. A variable-speed fan controls air flow based on the cooling demands of the amplifier. The result is a far more constant temperature gradient and vastly increased reliability. In fact, the M•1200 will run all day into 2-ohm loads at ambient air temperatures as high as 113° F!

BUILT-IN FEATURES INSTEAD OF EXPENSIVE ADD-ON MODULES.

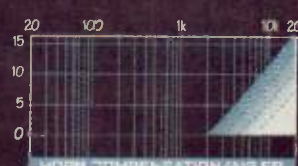
Switchable low pass subwoofer crossover. Want more bass in your PA system? Just buy an M•1200 and a subwoofer. There's no need for an external electronic crossover or plug-in amp module... because the M•1200 has a built-in 18dB/octave, linear-phase, uniform-time-delay filter with selectable 63 and 120Hz roll-off frequencies.



Variable low cut filter. Feeding a speaker system frequencies below its tuned bass cut-off point, results in terrible sound and potential woofer damage. Our variable low-cut filter lets you dial in the right transition frequency for your speaker cabinets or stage monitors — anywhere up to 170Hz.

No extra plug-in cards. No unsuccessful fiddling with a graphic equalizer. Instead, precise control.

Variable high frequency compensation. All compression drivers mounted on Constant Directivity horns require compensation somewhere between 2.5kHz and 5kHz. But until now you had to rely on hard-to-find, harder-to-adjust crossover modules (or resort to tweaking a graphic equalizer, which works poorly if at all). The M•1200 has variable compensation that's sweepable from 2kHz to 6kHz (we extended the high end boost another 1K so that you can also use the control to add "air" equalization).



Subsonic stabilization. Visible, random woofer cone movement is a symptom of subsonics. Caused by the extended low-end bandwidth of modern condenser mics and exaggerated by stage vibration, subsonics rob amp power and cause intermodulation distortion. The M•1200's input stage includes a circuit that eliminates subsonics. Woofer cones stay rock steady, centered in their voice coil gaps, ready to reproduce only the frequencies that you can hear.

FR SERIES FAST RECOVERY™

WE HAVE BOTH ENDS COVERED.

Compare the M•1200's front LED displays with comparably-priced amps. We not only provide Signal Present, Standby and 5-step level displays for each channel...we also include Cold and Hot temperature indicators and an industry first — a Short Circuit LED that warns you in advance of a short circuit during set-up or operation. Multi-step detented Level controls are calibrated in both dB and volts for accurate system set up and adjustment.

The back panel is equally complete. Instead of just 1/4" input jacks found on "stripped down" amp models, we've also included a balanced female/male XLR set to make signal pass-through and signal splitting easy. Outputs include extra-heavy-duty binding posts spaced on 3/4" centers for bridged operation.

Plus there are a lot more features including elaborate short circuit, overload and thermal protection, automatic turn-on delay, lighted rocker power switch, double-sided thru-hole-plated fiberglass circuit boards, up-front center of gravity and rear rails for extra stability in road racks.

We could go on and on. And we *will* if you call us toll-free for more detailed information.

Better yet, visit your nearest Mackie Designs dealer and get face to faceplate with an FR Series M•1200 High-Current Power Amplifier.

In terms of specs, features and durability, it's a "spare-no-expense" amp. Yet in terms of watts per dollar, it's a far better value than any comparably-priced model.

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CIRCLE 49 ON FREE INFO CARD

EQ PRODUCT VIEWS

WHO'S ZOOMIN' WHO?

Measuring only 5.7 inches wide x 6.1 inches high x 1.8 inches deep and powered by the new Zoom ZFX-2 digital processor, the Zoom 505 guitar processor features a total of 24 effects. Nine of these effects can be used simultaneously. The 505 boasts Zoom hall and room reverbs, delays, eight analog distortions, chorus, flanger, doubler, compressor, limiter, auto wah, pedal wah, acoustic guitar, 4-band EQ, phase shift, and pitch shift parameters. Key-touch operation allows programming with up to 24 user-programmable patch memory locations. Other features include: a stereo line/headphone output with master level control, an amp simulator for direct recording, DC (9 V) battery powering or optional AC power, and noise reduction. Retail price is $139. For more details contact Samson Technologies Corporation, P.O. Box 9031, Syosset, NY 11791-9031. Tel: 516-364-2244. Circle EQ free lit. #101.

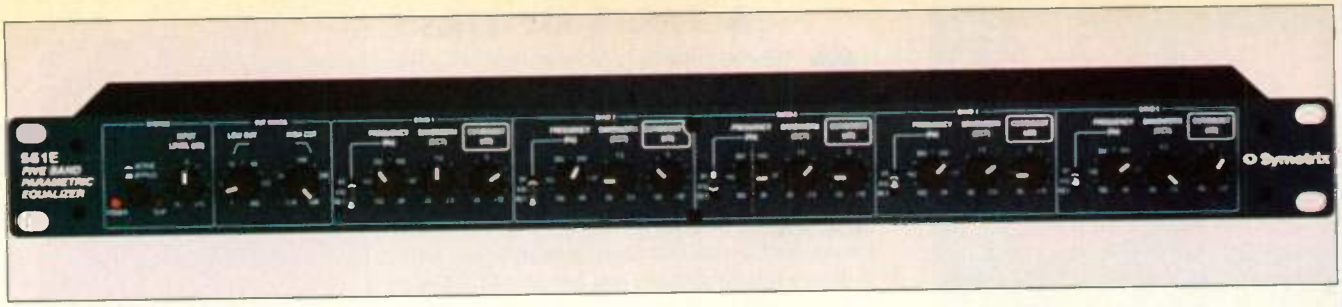
A WEISS DECISION

The new EQ1 digital parametric equalizer from Weiss features a large, backlit LCD that displays the overall frequency response (calculated in real time) and the detailed parameter values in dB, Hz, and Q. The EQ1's seven identical parametric bands cover the entire audio frequency range with boost/cut, frequency, and Q/slope controls. Each band operates in any of the following modes: high shelving, low shelving, peaking, high cut, low cut, and bypass. The unit also has dedicated switches for overall bypass, A/B memory comparison, CH1 only, CH2 only, CH1/CH2 ganged, snapshot store, snapshot recall, and copy functions between A and B. Weiss's EQ1 Digital Parametric EQ has a retail price under $4500. For further details, contact G Prime Limited, 1790 Broadway, Suite 402, New York, NY 10019. Tel: 212-765-3415. Circle EQ free lit. #102.

GUITARISTS GET DIGITAL

Guitarists will soon be able to take full advantage of digital music equipment with the introduction of the Yamaha G50 guitar MIDI converter. The Yamaha G50 allows guitarists to play MIDI tone generators and to interface with computers by taking the information from the pickup. From there the G50 converts the information into MIDI data by detecting pitch, picking position, and envelope. The G50 features four tone-generator modes with automatic setups in ROM for the Yamaha MU50, MU80, VL70-m, and VL1-m tone generators. The Mono Synth mode, which does not require a special pickup, allows

the G50 to become a pitch-to-MIDI converter, enabling virtually any audio signal to become MIDI information. The Yamaha G50 carries a suggested retail price of $749.95 and the G1D Hex guitar pick-up has a suggested retail price of $199.95. For further details, contact Yamaha Corporation of America, Audio, Guitar, and Synthesizer Division, P.O. Box 6600, Buena Park, CA 90622. Tel: 714-522-9011. Circle EQ free lit. #103.



GIMME FIVE

The 551E five-band parametric EQ from Symetrix features a new approach to equalization circuitry, known as UltraQ. The 551E incorporates five fully overlapping bands, each allowing for a range of 10 Hz to 20 kHz and independent knobs for adjusting frequency, bandwidth, and boost/cut. The 551E also features a THD+Noise of <0.002 percent. Other features include low- and high-cut filters as well as both XLR and 1/4-inch connectors. The Symetrix 551E carries both UL and CE approval and retails at $449. For more information, contact Symetrix, 14926 35th Avenue West, Lynnwood, WA 98037. Tel: 206-787-3222. Circle EQ free lit. #104.

THE SKINNY ON MINI

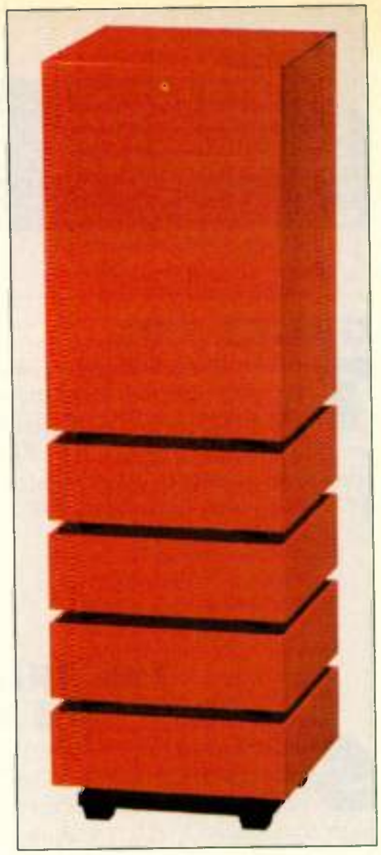
Sony has introduced the MDM-X4 MiniDisc multitrack recorder. The MDM-X4 utilizes the MD-DATA format, as well as Sony's advanced ATRAC technology to offer 4-channel recording and playback (up to 37 minutes per track). Editing features include song edit, which allows for full song-based editing, and track edit, which allows for various digital editing between tracks or part of a track. The unit also features a jog/shuttle feature and ±8.8 percent varispeed control capability. The availability of MMC (MIDI Machine Control) and MTC (MIDI Time Control) allows the MDM-X4 to be synchronized to other MIDI devices, and external control of the unit is made possible by an assignable pedal. The MDM-X4 also features ten inputs: four mic/line, one stereo, two aux returns, and two aux sends. Sony will begin shipping the MDM-X4 in November at a suggested list price of $1250. For more information, contact Sony Electronics, 1 Sony Drive, Park Ridge, NJ 07656. Tel: 1-800-635-SONY. Circle EQ free lit. #105.

KEY TOOL

Featuring a new design and a metallic two-tone, titanium finish, the 76-key N264 and 61-key N364 N-series music workstations from Korg feature 64-voice polyphony, real-time pattern play and record (RPPR), and a four-octave arpeggiator. Both models utilize Korg's AI² Synthesis System, which includes an expanded 8 MB of PCM waveform memory featuring 430 multisounds and 215 drum sounds yielding 936 programs and combinations. The N264 and N364 both feature 16-track, 32,000 event sequencers that support the standard MIDI file format. In addition, both music workstations feature two independent, fully programmable, stereo digital multieffects processors with 47 different effects types. For more details, contact Korg USA, Inc., 316 South Service Road, Melville, NY 11747-3201. Tel: 516-333-9100. Circle EQ free lit. #106.

EQ PRODUCT VIEWS

A TASTE OF ITALY



Acoustic Sciences Corporation's Mobilio line is a designer series of acoustical furnishings with two models of art deco bass traps. The Zavorra is a three-tier end table with outline dimensions of 24" x 24" x 21". The Gavitello is a bust table composed of five tiers, the top most being much thicker than the others, with overall dimension of 16" x 16" x 48". Both are finished in a matte lacquer finish with color or wood options. The bass absorption mechanism of the Mobilio series remains rooted in the RC-Time Constant technique pioneered by ASC's original Tube Trap, but the sequence of the acoustical elements that get the work done has been altered by extending the volume expansion methods used in the Super Trap. Both of the Mobilio traps have the same bass absorbing surface, volume, and performance as ASC's 16- x 4-foot Tube Trap. Price is $750 each. For more information, contact Acoustic Sciences Corporation, P.O. Box 1189, Eugene, OR 97440. Tel: 541-343-9727. WWW: tubetrap.com. Circle EQ free lit. #107.

DADDY NEVER SLEEPS AT NIGHT

The Squeeze Box from Retrospec is an electro-optically-controlled tube compressor/limiter housed in a rugged "stomp box." Not only does the unit accommodate musical instrument levels, but it also was designed to handle moderate (–10) line levels, making it usable at semi-pro unbalanced tape machine and console insert points. A balanced XLR output allows connection to mic level inputs, and Retrospec has included controls for EQ contour and attenuation ratio. There are no input or output transformers, thus minimizing problems associated with frequency and transient response as well as low-end loss. The Squeeze Box carries a list price of $495. For more details, contact Retrospec, Inc., P.O. Box 466, Phoenicia, NY 12464. Tel: 914-688-7329. Circle EQ free lit. #108.

ABSOLUTELY FABULOUS

Sporting eye-catching red drivers, Spirit's Absolute Zero is a compact nearfield monitor that utilizes a woofer with a deep 30 mm voice coil and a soft-dome tweeter driven by a 25 mm ferrofluid-cooled voice coil mounted on a specially shaped waveguide. The waveguide allows for increased power handling and controlled dispersion of high-frequency sounds. The Absolute Zero also features flat on- and off-axis response, a 75-watt RMS power rating, and a vented cabinet with a large port for extended LF response even at high levels. For more information, contact Spirit by Soundcraft, Inc., 11820 Kemper Road, Auburn, CA 95603. Tel: 916-888-0488. Circle EQ free lit. #109.

Photo by Julian Jaime

Neumann TLM 193 Special

A satin finish makes this sparsely produced mic a special classic

MIC NAME: Neumann TLM 193 Special
PRICE: $1495; shock mount: $210
TYPE OF MIC: Pressure gradient condenser
POLAR PATTERN: Cardioid
FREQUENCY RESPONSE: 20 Hz to 20 kHz
SENSITIVITY: 11 millivolts/Pa (@ 1 kHz into a 1 kohm load)
RATED LOAD IMPEDANCE: 1000 ohms
SIGNAL-TO-NOISE RATIO: 73 dB (CCIR); 83 dB (DIN/IEC)
DYNAMIC RANGE: 130 dB
MAXIMUM INPUT SPL: 140 dB for 0.5% THD
POWER REQUIREMENTS: 49 V dc, ±4 V
CURRENT CONSUMPTION: 2.4 milliamps
DIMENSIONS: 175 mm (length) x 49 mm (maximum diameter)
WEIGHT: 480 grams
MIC NOTES: Look twice at this microphone because it is not a Neumann U89i. The TLM 193 Special is in fact, a TLM 193 with a satin-nickel finish. This microphone has the same specs, circuitry, and distinctive sound of the original matte black 193 with the classic finish (Neumann has also manufactured a matching shock mount, the EA 193). This silver version uses the same dual-diaphragm capsule as the now-famous "regular" TLM 193, TLM 170R, and U89i microphones.

Like its matte-black brother, the TLM 193 Special is hard-wired for cardioid operation and features a transformerless output. Only 50 of the 200 TLM 193 Special microphones manufactured will be imported into the U.S.
USER TIPS: The TLM 193 Special excels at studio vocals, acoustic guitar, and drum overheads. Guitarist Ed Gerhard uses the TLM 193 in conjunction with a pair of Neumann KM 140's to record his acoustic guitar, aiming the 193 at the sound hole and placing it about eight to ten inches away from the instrument. Due to the mic's low noise factor and transparency of sound, classical engineers will also find the 193 Special a natural choice for orchestral recording. EQ

Photo by Edward Colver

MEMO

FROM: Mega Records

TO: Chief Engineer

RE: Demo

Problem

Vocals are lacking something - how about adding some harmonies and thickening?

Solution

the DigiTech Vocalist Workstation

WITH REVERB ON-BOARD

2-5 PART HARMONIES
triggered by one voice.
Uses: Live and studio

PITCH CORRECTION
Save an otherwise great recorded vocal take by fixing a bad note

BIG, BRIGHT DISPLAY
Easy to read, easy to edit. Soft keys let you zoom in on editing parameters quickly and simply

GENDER BENDER™
Apply male or female tonalities to your harmonies. Make your voice sound skinny or fat, more human!

BUILT IN KEYBOARD
for quick and easy commands.
Also includes a tone generator for cueing singers

SEQUENCE YOUR HARMONIES
If you are using virtual tracks, you can build your harmonies and edit them like any other sequence data

LUSH VOCAL THICKENING
Add 4-voice detuning to give your lead vocal that big double-tracking sound for live or recorded use

100 HARMONY PRESETS
Great for any style of music

FOOTSWITCH
For live use - programs up/down/bypass

NO PROBLEMS, ONLY SOLUTIONS

From the studio to the gig. On the road or at home. The Vocalist Workstation is the hottest new vocal processor to come along since the invention of the microphone! The Vocalist Workstation is an easy to use vocal workstation at your fingertips. *And best of all, it's affordable!*

Imagine, 2 to 5 part vocal harmonies triggered by one input voice with built-in reverb dedicated specifically to your vocal. The sleek new notebook design makes live or studio operation a snap.

You'll find loads of presets for any type of music. The Vocalist Workstation has 50 Factory Presets or you can customize your own styles and save them into 50 User Presets and each preset can have up to 8 variations. You can produce great harmonies for any style of music.

Control your mix in real time with easy to use control panel faders. Utilize the Vocalist Workstation in live performance with the Digitech FS-300 footswitch. Apply Gender Bender™ to the harmony voices to make them sound like different people. Try the pitch correct feature to fix that one bad note in an otherwise great recorded take.

New, The Vocalist Workstation, another Vocal Solution from Digitech.

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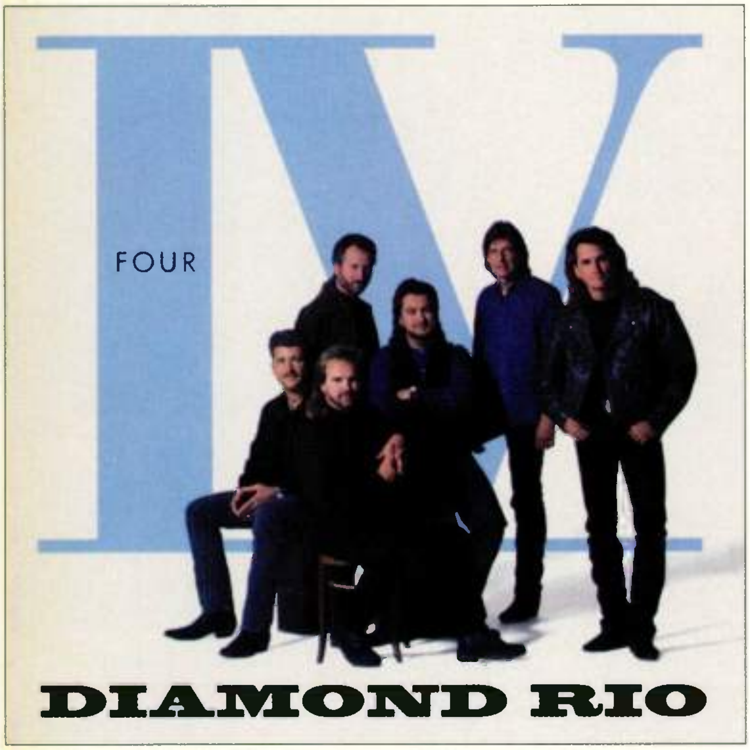
Diamond Rio "It's All in Your Head"

EQ speaks to producer/engineer Mike Clute about recording the country act direct to hard disk

BY STEVE LA CERRA



BRAVO RIO: This hot country group has been nominated for a Grammy.

EQ: What prompted your decision to record this project to hard disk as opposed to tape?

Mike Clute: Sonically, I feel that the Fairlight MFX-3 is the best sounding digital format that I have heard, and that is my first prerequisite. Then there's the flexibility that it leads to with a group recording situation. We had done overdubs on several previous projects with the Fairlight MFX-2 system, but it was, I believe, a 4-input, 16-output system.

So with four inputs it really didn't lend itself to tracking a band like Diamond Rio.

No, but it was a good way to get them acquainted with hard-disk recording. The system we used for "It's All In Your Head" has 24 inputs, but you have 999 layers deep per track — so you are only limited by the amount of hard-drive space. I used about 12 GB of space for the project. The band could play as many takes of a song as they wanted. I could go back and grab intros, sections, or lines that they played on any of the tracks and put them together into one master take. It allowed the band to do entire performances of a song without the worry of erasing tracks. It allows, in the band-sense, for more performance-oriented records.

So you composited the finished master rather than punching tracks in and out?

I don't know that I ever even punched anything. It was a lot more of comping. The way Diamond Rio tracks is that everyone is in the studio playing live — essentially drums, bass, piano, and about half of the acoustic parts were kept. Some of the acoustic parts would be overdubbed, but with the basic track, everybody in the band would play. It gives them a sense of how the arrangement works and allows them the ability to hone their performances. Generally we end up doing the electric and acoustic guitars and mandolins as overdubs.

Did you find that you might have, for example, used a section of piano from take five of the song and flown it into take three of the rest of the band?

Oh definitely, and that is the performance aspect of it. The guys could just go full-tilt for their track and not have to worry about having to be safe or blowing a take for the rest of the band. There's a lot more freedom for them to play on the edge. It saves them time and energy, but it does take me a bit more time because I am using these tools and recording about six or eight takes. If a pass went by and I felt that there was something special then I'd hang on to it, but if not then I might let it go.

Would you then overdub onto that composite rhythm track?

Yes.

What was the front end for the recording?

Most of the critical recording was through outboard mic pres. For the lead vocal, we used what started out as a Neumann U87, but it was custom-modified by Fred Cameron. It's a tube modification. He essentially guts an '87 and builds a tube preamp circuit for it. It is sort of like a '67 but is a lot more open at both extremes of the frequency range. It's one of my favorite mics. That was run through a Telefunken V76 mic pre, generally to a Valley People 440 for compression and then direct to disk through the Fairlight converters with no EQ. The console we used was an Otari Series 54 modified by Kevin Anderson.

How was the project mixed?

I used the AT&T Disc system at Masterfonics in Nashville, so I was able to keep the project in the digital domain almost all the way. As most people have already discovered, the colorations or changes in a digital recording occur in the conversion. I feel that once the signal has been converted, you are better off staying there and that is what I was able to do. We mastered 24-bit to a Nagra D machine, and the higher bit rate definitely had a positive effect on the dynamics. Then we went down to 16-bit with an Apogee UV22.

Did you use any of the Fairlight's DSP functions?

One of the things I am doing with the Fairlight is using the DSP gate function on the drums. I like adding other sounds into the acoustic kit, and I tried using sequencers and locking them to the multitrack, but I finally discovered that there is a certain amount of play in using SMPTE no matter how tight the lock is. It's reading and correcting constantly, so if you use two drum sounds — as opposed to simply *replacing* a drum sound — that amount of drift changes the phase relationship of the two sounds. It would sonically change every hit on, let's say, a kick drum, enough to really bug me.

On the Fairlight, there is a gating function for each track. So I copy the kick track, set a threshold, and gate the entire track just like an analog gate. But the Fairlight can give me a one frame "handle" or space in front of each kick drum that "pre-opens" the gate so you don't lose the transient. The Fairlight will then go through the track, gate the drum, and cut the kick drum track into a series of separate slices or clips. Each box has a kick drum inside and a one-frame space in front. I make another clip with whatever sample I want to use and I match the transient and the length of the sample with one of the kicks from the gated kick drum track, until it sounds just the way I want. Then I use a paste function to paste this new sample clip into the boxes of the kick drum copy track. Now I go back to my original kick drum, which is not gated (I just copied it for this purpose) and combine the two sounds to change the overall sound. I get all the tightness and distinction of the sample, but still have all the ring and the acoustics of a normal kick sound.

I haven't been able to find anything else that worked quite the same way. It is a real important tool to me and I'm able to make the sound consistent without the sterility of replacing the entire sound. I record good acoustic sounds in the first place, and I don't want to remove them; I want to supplement them. This helps me get the best of both worlds and I'm not messing with triggers. Sometimes the sonic decision comes down to the mix and I might want the kick to sound a little more thunderous or put a bit more crack on the snare. I can put a rimshot in with the snare and it becomes another beast, or maybe put some strainer sound on, but it will still have acoustic integrity. You just can't do that anywhere else. EQ

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"...You can't beat the combination of features and quality offered here." "Clean and punchy... the RM800 does its brand name proud..." **Recording Magazine**

"Sound...is exceptional (particularly the low noise and sensitive eq) for a board in this class." "Inexpensive, great sound, very quiet, versatile." **EQ Magazine**

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Mackie FR Series Power Amps

The company moves to diversify with a new line of power amps

BY STEVE LA CERRA

FR SURE: Mackie is looking to expand into other project studio avenues, starting with a new line of amps.

Mackie has been able to accomplish a pretty amazing feat for the recording console market: they have brought unheard-of quality down to a price point that was previously a dream. Now Mackie is ready to address other segments of the audio market including studio monitors (look sharp at the '96 AES in Los Angeles) and power amps.

In a move that raised eyebrows at Summer NAMM, Mackie introduced a new power amplifier: the FR (Fast Recovery) Series M-1200. Mackie's goal with the M-1200 is to bring you a serious power amp — one that can compete with the "money-no-object" amps — at a list price of only $599.

From a technical standpoint, Mackie has endowed its first amp with a number of strengths. For starters, the M-1200 implements a Fast Recovery (FR) design that allows the unit to remain stable when powering hard-to-drive reactive loads. Circuit design is fully discrete (as opposed to using integrated circuits), and the output stage of the M-1200 is capable of delivering in excess of 60 amps (!) of current. Since that kind of power normally generates serious heat, Mackie's engineering team has cleverly come up with a "T-Design" cooling tunnel to reduce heat and thus increase reliability. The problem with traditional cooling tunnels is that transistors at the opposite end from the fan wind up with warm air blowing across them — air that has already been heated on its way down the tunnel. Mackie's T-Design places a variable-speed fan in the middle of the tunnel, forcing cool air across the output devices along both sides of the tunnel. And since the fan is not sucking air into the amp's chassis, the dust and cat fur factor inside the amp should be at a minimum.

As far as using the M-1200 is concerned, a look at the rear panel tells quite a story. Input connectors are both XLR balanced and 1/4-inch balanced or unbalanced jacks. Mackie has kindly provided an XLR throughput for those of us who run multiple amps on stacked PA arrays. Speaker outputs are on both 1/4-inch connectors and banana connectors spaced to accommodate the dual-type regardless of which mode you use the amplifier in. Keep looking and you'll find a low-cut filter, variable from "off" up to 170 Hz. Mackie engineers have marked the filter frequency pot at 35 Hz ("typical") and 100 Hz ("stage monitor") for suggested uses.

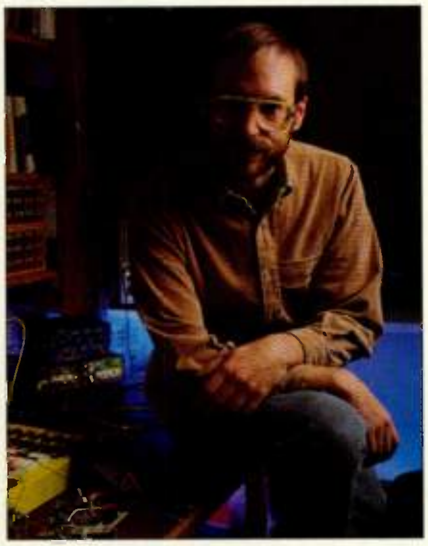
You will also find a high-frequency compensation pot (with On/Off switch) variable from 2 kHz to 6 kHz, designed to EQ constant-directivity horns. These functions are separately switchable for the left and the right channels. Two more "global" controls affect both channels: the first pertains to the amp's output application and can be set to full range (limiter off), full range (limiter on), and subwoofer. In the subwoofer position, the M-1200 gives you a choice of either 62 or 125 Hz turnover frequencies, so you won't need to buy an external electronic crossover when you're ready to add a sub.

Finally, there is a three-position switch for choosing the M-1200's output mode: stereo, mono, or bridged mono. With two channels driven (i.e., in either stereo or mono mode) the amplifier is capable of delivering 225, 400, or 600 watts per channel into 8-, 4-, or 2-ohm loads, respectively. In bridged mode, the amp can pump 800 watts into an 8-ohm load and 1200 watts into a 4-ohm load.

Mackie has also designed in a number of front-panel features that will make the M-1200 easy to use. Each channel has LEDs showing protect status and something that's really useful: "short." This indicates presence of a short circuit *before* you get the amp cranking. Two more LEDs show "cold" and "hot" temperature status of the amp, allowing you to keep a close eye on what is happening inside before a disaster occurs. Input level controls are calibrated in both dB and volts.

Price is $599. For more information, contact Mackie at 16220 Wood-Red Rd., Woodinville, WA 98072. Tel: 800-898-3211. Circle EQ free lit. #118.

Understanding Sampler Features



Finding your way through the myriad of sampler options available today

BY CRAIG ANDERTON

The digital sampler has evolved dramatically in recent years. Borrowing from synthesis, digital recording, computers, and digital signal processing, today's sampler is essentially a recording studio in a box.

Furthermore, synthesizers have copied samplers by not just including sampled sounds, but the ability to load samples and communicate with other devices. Several manufacturers offer sampling add-ons to synths, while others add onboard ROM sounds (just like a synth!) to samplers. Peavey's DPM series started the trend by adding sample RAM and sample editing to a basic sample-playback synth, along with an optional rack box for sampling into the instrument: Korg's T-series synths and Yamaha's SY family included similar features shortly thereafter.

Now synthesizers like the Korg Trinity offer optional "flash" RAM for storing samples, the ability to read Akai S1000 samples, computer interface, ADAT digital interface, and even hard-disk recording. So is it a synth, a sampler, or a digital audio workstation? And what about Roland's DJ-70mkII — a version of its flagship S-series samplers optimized specifically for DJ applications with automatic looping, load-while-play, and even a scratch dial? The boundaries are indeed blurring. In fact, Digidesign's SampleCell II sample player isn't even a rack unit or keyboard — it comes on a Mac or PC-compatible NuBus or ISA card (a PCI version is coming this fall).

So let's ditch the definitions and look at some of the most important features in today's samplers/synthesizers. We'll make sense of the sometimes bewildering options, so you can determine which are most important to your needs. Just remember that most of today's keyboards are so expandable that if you end up needing some feature further down the line, the odds are good it will be available as an add-on.

FILE COMPATIBILITY

Early samplers could read only their own proprietary format, but more samplers can now read and translate samples from multiple formats. The most common file translations are for Akai and Roland format (although newer Akai samplers read E-mu EIII format, and Kurzweil reads Ensoniq), which allows access to a huge library of quality sounds. Some samplers can read other formats only from a CD-ROM connected to the SCSI port (see next section), while others can translate from floppy disks as well.

More and more samplers can read WAV files (Windows' native audio format) or AIFF (Mac format). For example, Kurzweil's K2500R can read Roland, Akai, and Ensoniq EPS/ASR files and keymaps via SCSI, and Ensoniq and Akai floppies. It can also read/write WAV and AIFF files from/to disk or a SCSI device.

One important point: transferring raw audio is no big deal, but translating the associated synth-like parameters is more difficult — if a sampler processes a sound with resonant filters and you transfer the raw sample to a sampler lacking resonant filters, the final sound will be very different. Translation should convey as much data as possible; for example, Peavey's SP Plus loads complete presets from Akai S-1000 CD-ROMs, including samples, keymaps, and all synthesis parameters (filters, envelopes, etc.; of course, it also reads Peavey SP format samples as well).

Many samplers still support the MIDI Sample Dump Standard (SDS) protocol for transferring digital audio via MIDI, but it's a glacially slow process (SDS dates from when samples were typically a few hundred KB of memory, not several MB). SMDI (SCSI Musical Data Interchange) is a newer protocol for transferring samples over SCSI instead of MIDI, and runs about 50 times faster than SDS.

While we are discussing file compatibility, if the sampler includes a sequencer, it's helpful to be able to import and export Standard MIDI Files. This ability lets you create and edit your sequence on a spiffy computer-based program rather than using the sampler's limited graphic interface, then load the completed file into the sampler.

SCSI

SCSI (Small Computer System Interface) is a hardware/software protocol for transferring vast amounts of data quickly. If you're serious about sampling, SCSI ports are a must since you can connect hard drives, CD-ROMs, optical drives, and computers to the sampler. With 1 GB hard drives listing for $300 and less, SCSI gives access to a lot of sounds for very little bucks. SCSI is also required for using SMDI.

RAM EXPANSION

More memory allows loading more and/or longer samples. At 5 MB/minute for 44.1 kHz mono digital audio, 16 MB suffices to fly in a 3-minute vocal, while 128 MB allows almost 13 minutes of stereo sampling. The king of expansion has to be SampleCell II; although each card holds "only" 32 MB of RAM,

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WRH

an optional Digidesign Expansion Chassis can cascade up to 10 cards for 320 MB total.

Most RAM is *volatile*, meaning that the samples have to be reloaded after a power-down. Some samplers provide for battery-backed RAM or EEPROMs ("flash" ROM) so that samples are accessible at all times. Battery-backed and flash RAM tends to be more costly than standard RAM, so a typical scenario is to keep your "greatest hits" in battery-backed RAM, and load other samples into standard RAM. Akai's S3000XL and S3200XL both accommodate up to 16 MB of flash ROM.

POLYPHONY

With more voices, sustaining notes won't cut off when you play new notes, more notes, and sounds are available for multitimbral setups, and cross-fades between samples are more realistic because voices aren't "stolen" when you run out of polyphony. At the low end, 16 voices is common, but these days, 32 to 64 voices is just about standard (and E-mu's E4K can be expanded from 64 voices to a whopping 128 voices for an extra $945).

DIGITAL I/O

It's a digital audio world, and in addition to SCSI, many samplers now include built-in or optional AES/EBU, S/PDIF, or ADAT "light pipe" interfacing. This has several applications: blast samples from audio CDs directly into a sampler from a CD player with S/PDIF, take tracks recorded on digital tape or hard disk and move them into the sampler (ideal for flying in vocals or adding background voices), and transfer samples between similar-equipped samplers.

The TDM version of SampleCell integrates with a TDM-equipped Pro Tools system, so the card's 8 outputs appear as direct digital inputs to the Pro Tools Mix Window. This provides for automation, digital equalization, DSP processing, balanced +4 dBv analog outputs, and both AES/EBU and S/PDIF digital outputs. Furthermore, audio recorded in Pro Tools can be loaded into SampleCell and played back from RAM, thus freeing up Pro Tools voices.

RESONANT FILTERS

Resonant filters are difficult to implement digitally, but many classic analog synth sounds (essential for today's cutting-edge dance music) rely on resonance. Most newer samplers include resonant filters; one exception is the ASR family, which trades off resonant filters for "transwave" technology. This can create not only resonant filter effects, but pulse-width modulation, vocalizations, and other swept effects that have more in common with synths than samplers. Also noteworthy: the E4K's Z-plane filters, which are extremely flexible.

GRAPHIC INTERFACE

Although the graphic interfaces on samplers continue to improve (many have oversized LCDs that let you see the waveform, zoom in on loop points, and the like), nothing beats a good computer-based editor such as Alchemy or Peak for the Mac, or Sound Forge or

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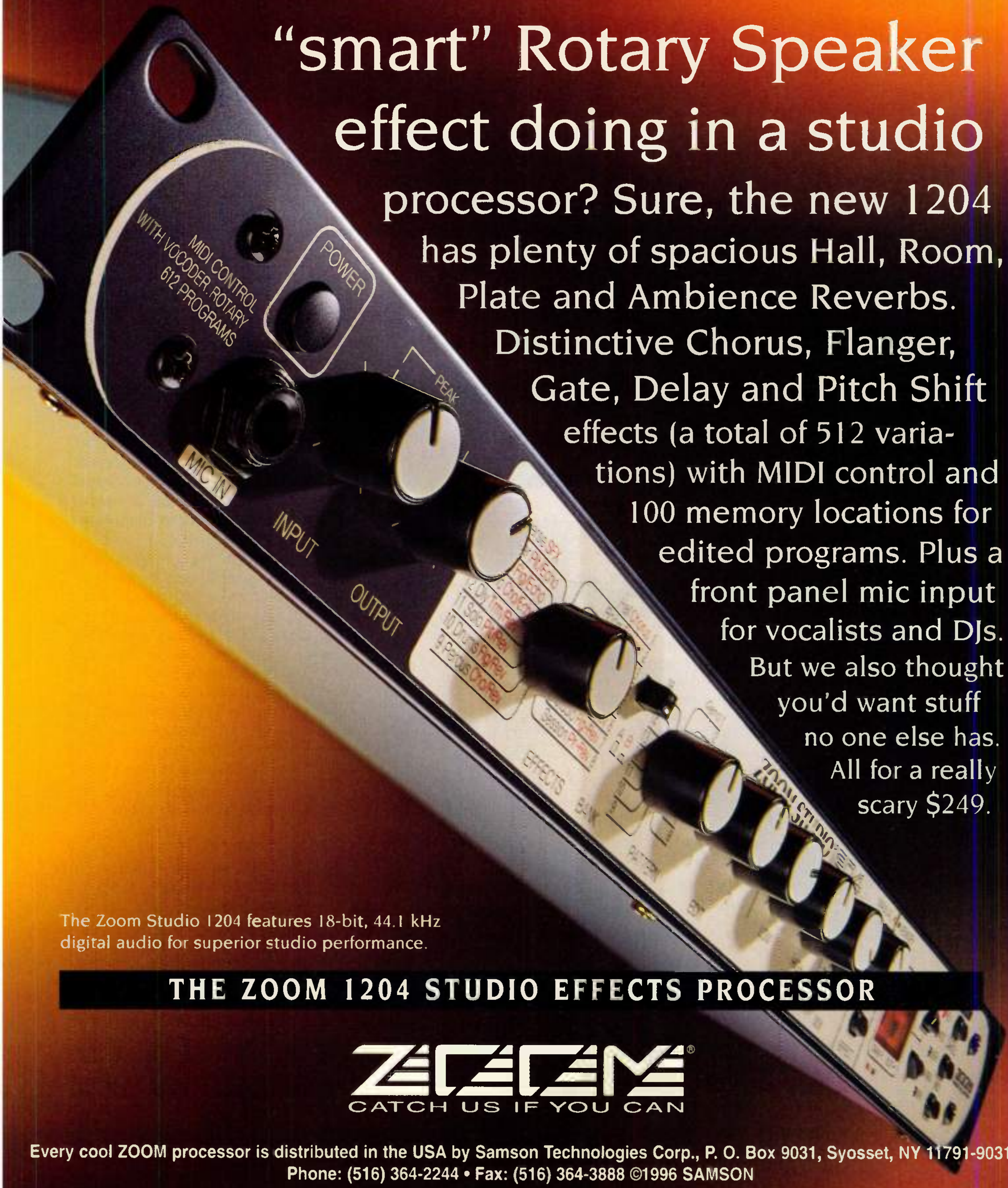

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SampleVision for the PC. If you're into editing, make sure there's a program available that's compatible with both your sampler and computer. Of course, with SampleCell II, since it's part of the computer environment it has no problem using the computer's graphics resources. Akai gets around the problem of compatible programs by providing free Mac graphic editing software to registered owners of the S2000, S3000XL, and S3200XL.

Roland goes one step further by offering the OP-760-1 option board for its S-760 sampler, which eliminates the need for a separate computer and editing program. This board includes a digital RGB output for connecting to a computer monitor, and both composite video out and S-video out for connecting to a standard TV (NTSC or PAL). There's also a mouse interface for point-and-click editing.

LOTSA DISCRETE OUTPUTS

More outputs simplify life in the studio, since you can send different instrument sounds to different outputs for external processing and/or mixing. As examples, the OP-760-1 board mentioned above adds two stereo outputs to the S-760's four outputs for a total of eight outs, the E-mu E4K and Digidesign's SampleCell II come with eight outs, while the Kurzweil K2vx and Peavey SP Plus have four outs. Ensoniq's ASR samplers have only two outs; however, an optional output expander can bring the total up to eight. Akai takes the same approach with their S2000; to keep costs down there are only two outs, but $299 buys you eight additional audio outs and S/PDIF I/O. Finally, the Akai S3200X is the heavyweight of outputs, with two XLR and ten 1/4-inch unbalanced.

SUPPORT SOFTWARE

Check for utility programs that make your job easier. For example, Alesis includes Sound Bridge with their synths. This program can transfer sounds from your computer to PCMCIA cards. Stuffing a card into the synth is an incredibly fast way to load new sounds and avoids the hassles of SCSI and other types of transfers. SampleCell II comes bundled with two CD-ROMs of samples ready for loading in, and the Mac version comes with a special version of Sound Designer II for editing.

CD-ROMs also come under the

continued on page 142

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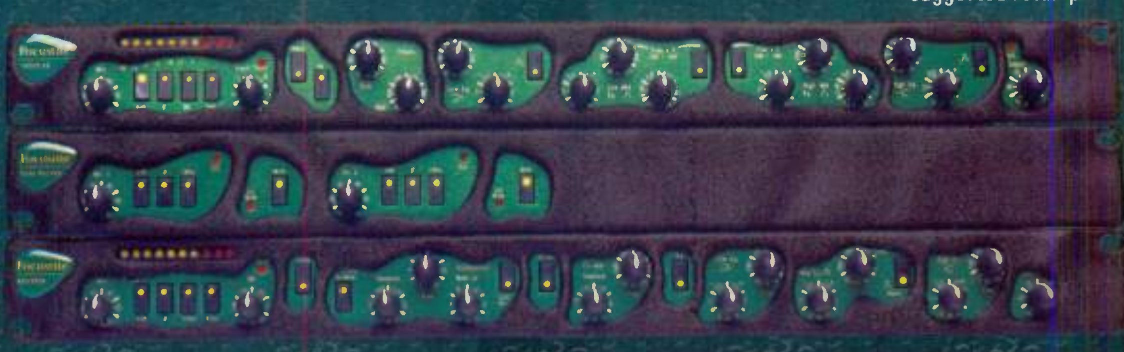

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Pure Digital

Sticking with ones and zeros from start to finish

BY PHIL RAMONE

WHO NEEDS ANALOG?: Phil Ramone, Fran Lucci, and John Patterson at the Yamaha 02R.

Recently, Phil Ramone went into Capitol's Studio A to record the song "Mistakes I Made" with artist Fran Lucci. The track was completely recorded and mixed on a Yamaha 02R digital recording console using no external effects processors. Phil describes how this unique session progressed to completion.

While a lot of recordings claim to be "digital recordings," the term is often misused. Recently my engineer, John Patterson, and I completed a project with singer/songwriter Fran Lucci that truly was a "digital recording." We used a Yamaha 02R digital console and recorded the audio directly onto TASCAM DA-88's. Throughout the process, the audio signals never returned to the analog domain (except for monitoring the session) until after the final mix was put onto DAT. We even used only the digital reverbs found in the board.

When Peter Chaikin, pro audio recording products manager for Yamaha, first told me about the new Yamaha 02R digital console, I was intrigued by its possibilities. But to be really sure of what it could do, the board had to be tested in a demanding setting. With a group of musicians concentrating on getting the music right, the last thing a producer wants to hear is technical glitches. So, for me, recording a rhythm section cutting a new tune was the perfect situation for this test, because if the board was going to act up, we'd know it right away. I chose Capitol's Studio A because I am very familiar with how it sounds and because it would provide a nice, open acoustical environment.

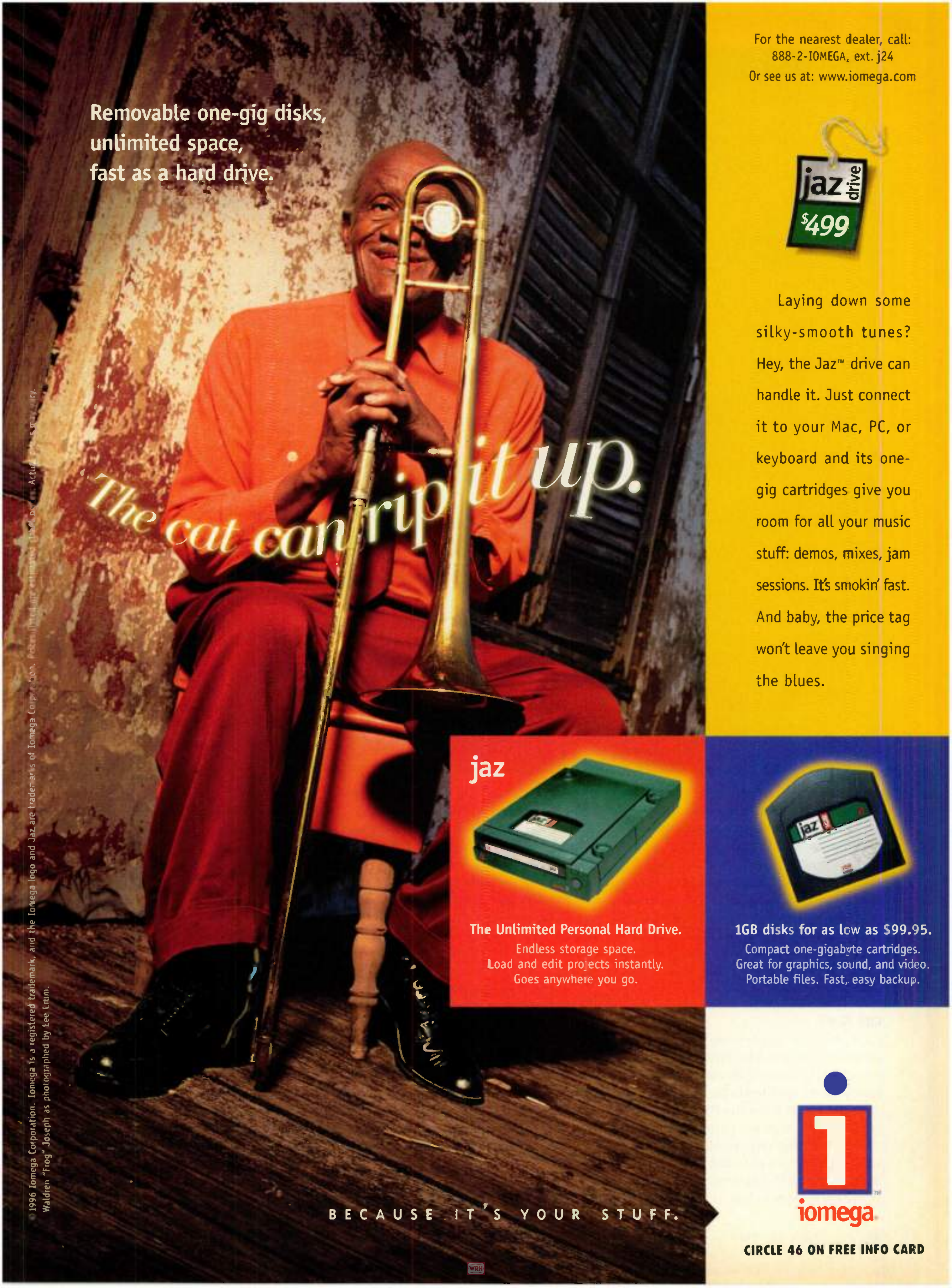
It was important to see if the 02R could faithfully reproduce this wonderful room. I also chose to work at the studio because of the great people there. Everyone was helpful and enthusiastic — from Michael Frondelli and Paula Salvatore on the administrative side to chief engineer Jeff Minnich and engineer Leslie Ann Jones.

The band was made up of Mark Portmann on keyboards, Steve Schaffer on drums, Paul Jackson Jr. on guitar, and Abe Laboriel on bass. Fran would sing a vocal with the band as a reference, but I'm always aware that there is a possibility that the performance could be a keeper. It turned out that quite a bit of what she sang with the band was used in the final mix.

You can see from the accompanying input and track-assignment list that the 02R was going to be pretty full. We wanted to do this with just one 02R, so we had to limit ourselves to just 16 tracks for the basics. It took a little bit of thinking to fit all of this stuff onto two machines; we grouped the four toms to a stereo pair of tracks and Mark gave us left and right outputs from his rack's mixer for his keyboard sounds (his sound for the basic track was a stereo Rhodes-type sound). He also sent us a second pair of inputs (19 and 20), which were playing a loop.

Though we brought the console with us, we used the studio's existing wiring. Microphone inputs came from the studio into the control room and then were patched into the 02R. Though only the first eight inputs have XLR connections, all of inputs 1 through 16 have the same controls, including a variable gain control and a 20 dB pad (there is no mic/line switch). Inputs 1 through 8 also give you phantom power and the kind of insert typically found on many project boards: one 1/4-inch connector with the send at the tip and the return on the ring. Since channels 1 through 8 have the two types of connectors, they also have a switch that toggles between input A (the XLR) and input B (the 1/4-inch). The external phantom power supplied for the condenser mics on the toms (inputs 9 through 12) and guitar (inputs 15 & 16) was our only use of any external gear for the session.

Although 9 through 16 are on 1/4-inch connectors, those channels still have enough gain to accept a mic level



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Waldren "Frog" Joseph as photographed by Lee Crum.

MAINTAINING DIGITAL GAIN STRUCTURE

By John Patterson

Towards the end of the recording session at Capitol Studios, I had one experience that is definitely worth relating. Upon hearing some intermittent distortion in the stereo mix (hey, it happens), we realized that the meters on the stereo output bus were showing "overs." Normally to cure something like this in the heat of battle, you might think of just pulling back the master fader and then the output would be fine. Maybe that works in the analog world, but that doesn't work on a digital console because it doesn't address the fact that the overs are occurring before the master fader. It's not like an analog board where you have headroom above a nominal 0 level. For example, say you have 20 dB of headroom on the stereo bus of the board you're using. Pulling down the master to trim your mix isn't a problem unless you have already trimmed it 20 dB and are still over zero on your meters. When you are over on a digital board, you are over, and while pulling the master fader down might make the output level lower, it does nothing to change the internal gain structure of the console, causing the digital distortion.

To solve this problem quickly without interrupting the session, I used the attenuators on each channel of the 02R to trim all of the inputs down by about 2 or 3 dB. This gave me back a bit of headroom so that there wouldn't be this problem. If the gain structure coming into the board is fine, then you can use a different method of grouping all of the monitor sources (faders or pots) together and pulling them all down simultaneously.

The 02R's View page helps with this since there is a readout in dB next to the diagram of the fader. This eliminates the guesswork on how much you are trimming. (Remember the all-too-familiar step of patching an oscillator into a channel to establish a fader trim reference?)

You can also run into this sort of gain structure problem on the input side of things. Let's say you have a signal coming into a digital board and are using an on-board compressor to give you a nice "0" output. To get this nice 0 output, you have attenuated the compressor's output by 2 dB. Now let's say that in some spots where you have this nice 0 output, the compressor is compressing 6 dB. Not nice! The input to the compressor is distorting, and I'll leave it to you to do the math! You need to be especially cognizant about gain structure on a digital board. You simply cannot think of using a digital console as you would an analog one.

signal. During some initial testing at Capitol, the gain structure on these inputs seemed different, and we thought that there would not be enough gain for a mic signal in channels 9 through 16. But it turned out that we were using a snake with some military-style connectors, and we found that these plugs will not work with the 02R. The 02R wants to see consumer-type 1/4-inch TRS plugs like you'd find at the end of a headphone cable.

CUE IT UP

Capitol has a headphone system with several 8-channel mix stations that allow musicians to adjust their own headphone mixes, so we decided to feed that system with the sends from the 02R, which are easily linked in stereo pairs. This makes them great for cue sends. We linked sends 1 and 2 as a pair and used them to make a keyboard-heavy mix of all the instruments. We also linked sends 3 and 4 and made that a drum-heavy mix. Send 5 was guitar-only, and send 6 was vocal-only. The click, the bass, and the stereo bus were patched directly from tape into the Capitol system. The last two sends of the 02R, 7 and 8, are dedicated for use with the 02R's two internal effect processors. John set send 7 to feed a short-to-medium-sized room reverb (for the instruments) and 8 a longer plate reverb (for the vocals and strings).

The 02R has a great feature that allows you to see quickly and easily what channels are being sent to which effect send. For example, let's say you are wondering, "Did I send the guitar to send 7, which is my plate reverb?" If you press the master button labeled "send 7," the channel faders snap to show the level of each channel going to send 7. The channel faders act as send controls normally would, so you can make any changes right there. If you press the master "send 1" button, the faders snap to show you which channels are turned up on send 1. If you hit "view," then the faders snap back to represent the tape return level in the mix.

After we got a basic track that the musicians and I felt was good, we gathered in the control room for a playback. Musicians are very quick to notice if anything is messing with the sound of their instrument. But without question, everyone was excited by what they were hearing through the speakers. So with the basic track out of the way, we were ready to move on and do a few overdubs. There would be some synth string sweetening, a quick drum overdub, and a few vocal fixes. Paul wanted to shoot another guitar track as well. Though the entire console was full, I still needed to add a third DA-88 for the overdubs. But I knew that interfacing this third machine was going to be a piece of cake. We had put a third TDIF (TASCAM Digital Interface) card in the slot of the 02R, which corresponds to channel faders 9 through 16. In the digital I/O page of the 02R, you can change these faders to accept digital input instead of analog sources, so they can become tape monitor inputs. Once we knew that we had the basic track down and didn't need those analog inputs anymore, we made the change. No patching was needed.

In addition, before we actually started overdubbing, we used the 02R's scene memory to save the tracking setup so that we would be able to recall bussing, EQ, send levels, panning, and dynamics for every channel, just in case we needed to come back to that setup.

Anyway, the first overdub we did was a mallet-cymbal track. We made the switch so we could hear all three DA-88's. The cymbal mics were on inputs 7 and 8, so we didn't have to do anything special with them. Once this overdub was done, we saved the overdub scene to memory and then went back to the tracking setup to redo the guitar. Since Paul didn't really need to hear the cymbal overdub to do his part, it was easier to do this switch back instead of repatching his guitar inputs down to a pair of inputs between 1 through 8. Though we still had to do the software toggle to switch channels 9 through 16 between analog and digital inputs, the scene memory recalled everything else we had going. It was literally done in a matter of seconds. When Paul was happy with his take, we recalled the overdub

The Dynamic Duo

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Peter Horvath,
Professional Sound Magazine

There was something else that really appealed to me with these babies: the low end. I mean real honest to God low frequencies. PBM 8 LM's are absolutely perfect for the major project studio that can afford only one set of monitors.

Bobby Owsinski – EQ Magazine

Tannoy's PBM 8 LM's offer superior balance and frequency response at all monitoring levels, amazing low end, and excellent stereo imaging. The dark Tyner-esque midrange from the acoustic piano was so accurately reproduced and the highs from the cymbals and vibes were so clear that I was almost convinced that the "quartet" was performing in my home studio.

Steve Wilke,
Electronic Musician

Tannoy is one of the most respected manufacturers of studio monitors. After reviewing the PBM 8 LM nearfield monitor I can see why. The monitor is remarkably neutral and has a wide range, and its resolution of time and space is superb. The Tannoy PBM 8 LM is a winner as a professional monitor. With its first rate amp and neutral response, you can trust it to tell you the truth about your mixes.

Bruce and Jenny Bartlett,
Pro Audio Review

Though the Tannoy's bass response was very beefy all the way down to 50Hz, it was not overbearing in musical contexts. Its sound was very smooth across the entire spectrum. The midrange exhibited fantastic detail. We felt the Tannoys were the best of any speakers at reproducing solo piano, as a result of this definition in the midrange. Listening to these monitors gave us a feeling of being in the same room as the musicians.

Keyboard Magazine

I found the horizontal dispersion to be quite a bit wider than most. On and off axis imaging is well above average. This is where the PBM 8 LM's shine – they're absolutely trustworthy at all volume levels, and certainly among the best speakers we've ever heard in their price range.

Nick Batzdorf – Recording

One advantage of the Limpet's monoblock approach is that there are no shared electronics. This really pays off in terms of stereo imaging, punchy, clear, bright, with well defined bass and just the right amount of midrange. Tannoy Limpets are an excellent choice, offering a formidable combination of flat, wide ranging response in a compact high power package.

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scene and all of those parameters came back for us to do the string sweetening.

We all know how important getting that killer lead vocal is in pop music. Any distraction on the technical side can blow the vibe and make it difficult for the artist to keep their concentration level up. As we switched gears and went to record Fran's vocal, the scene memory feature and the 02R's "libraries" made it easy for me to make the switch without missing a beat. The lead vocal mic (a Neumann tube U47) had to be moved from channel 13 for overdubbing since channel 13 was now being used to monitor track 5 from the third DA-88. It was very easy to match the gain structure and settings of channel 13 and use them for this new input channel. Although the channel's gain pot setting is not saved in scene memory (the pot is a conventional audio potentiometer), the control is detented. It's easy to count the number of clicks from "zero" and reset the pot. For the basic tracking, John had used a compressor preset from the 02R's library and modified it for Fran. He also saved this new compressor into the library. When we switched the vocal input, we went into the library and assigned Fran's compressor setting and EQ to the new channel. Again, all this was done in a matter of seconds without disrupting the flow of the session.

continued on page 140

INPUT LIST

Input Section

1	click	direct
2	bass DI	direct
3	snare bottom ø	5
4	kick	direct
5	snare top	5
6	hihat	direct
7	OH Left	direct
8	OH right	direct
9	tom	1/2
10	tom	1/2
11	tom	1/2
12	tom	1/2
13	vocal	direct
14	ac. gtr	direct
15	el. gtr	direct
16	el. gtr	direct
17	keys left	3
18	keys right	4
19	keys/loop	6
20	keys/loop	6
21	gtr efx (unused)	7
22	gtr efx (unused)	8
23	talkback	stereo bus only
24	talkback	stereo bus only

Monitor section:

1	click
2	bass
3	
4	kick
5	snare
6	hihat
7	OH left
8	OH right
9	toms left
10	toms right
11	keys left
12	keys right
13	vocal
14	loop
15	el. guitar left
16	el. guitar right

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CIRCLE 69 ON FREE INFO CARD

Made in the Suede

Some technical tidbits from the legendary Carl Perkins

BY DAVID McGEE AND CARL PERKINS

The Sun Studio was slightly larger than those Carl had been in at Jackson radio stations, but still comfortably snug. In the control booth at the far end, where Phillips prepared to listen to the trio [Carl, Jay, and Clayton Perkins], Carl could see two tape recorders, Ampex 350 models. The out-of-view recording console was a rudimentary Presto 5-input mixer with four microphone ports and one switch-operated multiselector port for microphone or playback. As Carl would learn, the echo he had heard on Elvis's recorded voice — an echo reproduced live by Scotty Moore's amplifier — had been the product of Phillips manipulating the signal from one recorder to the other to create a split-second delay. Primitive but effective, it had given Sun Records a signature sound on the most meager of budgets.

THE MILLION DOLLAR QUARTET: Jerry Lee Lewis, Perkins, Johnny Cash, and Elvis Presley at Sun Studios.

Photo courtesy of Michael Ochs Archives

The following passages are excerpts from Go, Cat, Go!, *a new book by Carl Perkins and David McGee that chronicles the life and times of Rockabilly king Perkins. The sections here tell how the technology of the mid-'50s was used to capture Carl's distinct sound. This first passage reveals Perkins's impressions during his first visit to Sam Phillips's Sun Studio in October of 1954.*

GUITAR HERO

Since 1948, [Carl] had been listening to one of the most intriguing stylists he had ever heard in Les Paul, whose recordings with his wife Mary Ford demonstrated a visionary's command of studio technology. Paul, born near Milwaukee in 1915, had, as a teenager, devised a way to obtain a primitive stereo effect by wiring his guitar into a pair of radios placed on either side of him onstage. He also began experimenting with homegrown recording technology: At age 14 he recorded himself on a homemade device fashioned from a weighted cutting lathe, a turntable made from a Cadillac flywheel, driven by a small motor and dental belts. At age 19, while tinkering with the idea of disc-to-disc multiple recordings, he found he could record duets with himself, or cut himself playing each of the instruments in the band.

The same year he cut his first disc multiples he also designed a solid-body electric guitar with two pickups, the latter a revolutionary innovation. Over the next few years he continued modifying and redesigning solid-bodies, and in 1952 signed a deal with Gibson for production of the Gibson Les Paul model, or the Les Paul Standard, with a gold top, two single-coil pickups, and what was called a "trapeze" tailpiece bar for muting the strings, a technique critical to Paul's unique sound, one he has steadfastly refused to dissect in public, save to describe it as "that big, fat, round, ballsy sound with the bright high end — nobody else has it."

This was the sound Carl was hearing on Paul and Ford's late '40s-early '50s recordings for Capitol — "Lover," What Is This Thing Called Love?" (a number-one single in '48), "Nola," "How High the Moon" (another number-one from '51), and others. What Carl didn't know was that Paul was employing his entire battalion of forward-thinking techniques — "delay, echo, reverb, phasing, flanging, sped-up sounds, muted picking, and everything else," according to Paul — using multiple recording. Those impossible cascades of notes, so delicate and so dexterously executed, from one guitar! Carl was thunderstruck by the man's gift and,

Guerrilla Troubleshooting

Getting by with a little help from your technically proficient friends

BY BENNET SPIELVOGEL

It was Friday. My three ADATs had to be in the groove and on the move for a 10 AM Saturday downbeat. Those machines had 3500-plus hours on them, so a bit of general cleaning and new rubber was scheduled. The client, facing a monster drive, would not be in the mood for any downtime. The decks were swabbed and spit polished, but on power up, shatter my crank if trouble ain't afoot. Machine #3 stared back at me dark and dumb. I'd seen this problem before, it was the power supply.

I popped open #3 (again) and a soft, slow periodic *click, click, click* meant that the switching power was slacking between pulses. As a member of the ADAT power-supply-of-the-month club, I knew that sound. I had even purchased two replacement supplies in the past. Unfortunately, those bad replacements were holed up in a dark, dank basement waiting for the Wizard — the person who gets my malfunctioning equipment breathing again. You can't push the Wizard. He repairs to his own beat.

SPARES CHANGING HANDS

With the spares out of the picture and a tough client and studio full of players due in less than 12 hours, I had to goose the reverb and come up with a plan. The first thing to do was panic. OK, got that out of the way. I then called Frank at Bismeaux Studio here in Austin, TX. (Bismeaux rents ADATs and Frank's a graduate of the ADAT school.) But it was 10 PM on a Friday night in Austin, a place where every day's a holiday. I dialed expecting the worst.

Frank picks up and I quickly explain the situation. He arranges to be open at 9 AM on Saturday. Frank is a saint. Next morning, screwdriver in hand, I install the supply in my machine and engineer the weekend session. Things couldn't have gone better.

On Sunday night, after everyone has left, I still have a power supply problem to deal with. The loaner was mine for five days. My options were to buy another power supply or figure out a way to repair my three dead supplies.

GOOD PUNCH, BAD PUNCH

Option #1: I could call and order a power supply from Alesis. Cost about $175 with shipping. But then I recall that one time, after being on hold for a while, when I finally got connected to the power supply department, it was the wall-wart division. Bad punch.

Option #2: I could rattle the Wizard's cage in New York. The cosmos is not yet aligned. Bad punch.

Option #3: I ask my friend Ed, a guitarist with a home studio, if he's interested in a challenging circuit. Good punch!

Ed had a ton of his own work to do, but agreed to have a look. I scored a schematic (if I told you how I got it, I'd have to kill you...) and, before going over to Ed's, I looked the circuit board over. I noticed some rework on the board. The main FET (Q101, if you're following along at home) and the little opto-isolator (U102) had been replaced. Maybe they had failed again.

After a short Monday session, I ran down to the local parts depot and scored the opto, but no luck on the FET. Nobody's got it. After dropping off the supply at Ed's that night. I flipped through the MCM catalog and, bingo, a 2SK1118 with my name on it. I ordered a few and sprang for the next-day air.

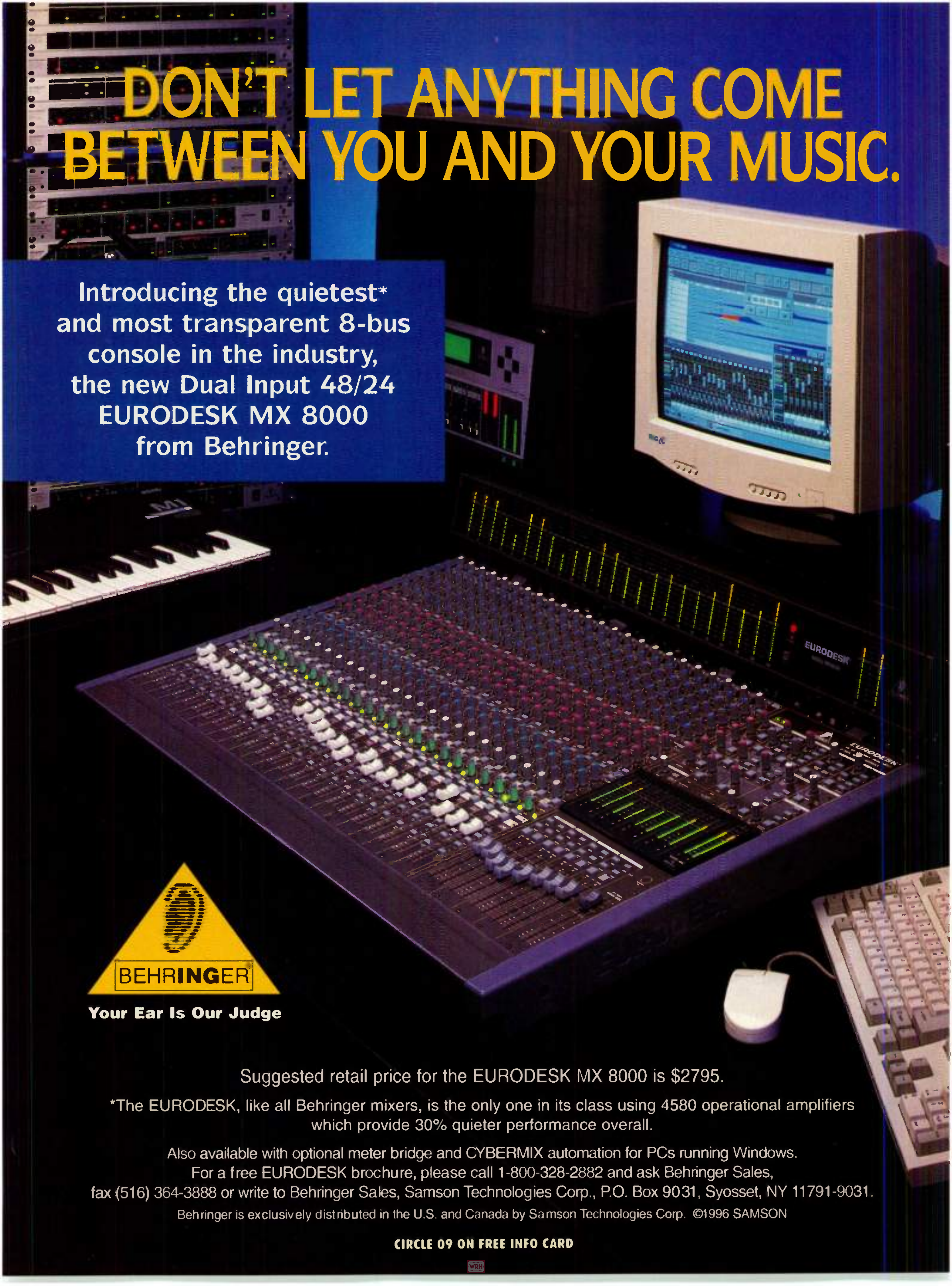
Note: Of the four ADAT supplies, the component used for Q101 varies. In addition to the 2SK1118, there may also be a 2SK792 and an SSP6N60. (The schematic lists the part as 6N60.)

CONFUSING DAZE & NIGHTS

Ed's a super busy guy who works a regular day gig, but he manages to get to the supply by Wednesday. He calls with the bad news, "Your old FET is good and so is the opto." So much for the easy fix. We spent three hours Thursday night 'scoping and measuring, without much change in the patient.

By Friday morning, the loan window on my borrowed working power supply

Illustration by Mike Browne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CIRCLE 09 ON FREE INFO CARD

was starting to close. I called Paul at Pro Digital in Pennsylvania. Paul also repairs ADATs and their supplies (some folk don't do switching supplies). His recommendation was to upgrade C109 from 47 µF/50 V to 100 µF/50 V and specified a high-temperature, high-frequency capacitor. In addition, R107 should be 24 ohms, though it's spec'd as 22 ohms in the schematic.

I thanked Paul, checked R107 and changed C109 even though it already was 100 µF. The ADAT didn't exactly spring to life, but the sound of the clicking changed. It was now oscillating like it really wanted to start, but I still had no front panel. I called Paul to set up a Red Label supply shipment. Afterward, I went back to the supply, fired up the machine for about thirty seconds, and pulled the AC cord. I felt the various components on the board, then *ouch!* Q102, the SCR, is hot.

After another run to the electronics "palace" for an SCR (NTE5437), I power up and still no front panel. More probing around led me to D106, a diode near the

continued on page 140

THE WIZ SPEAKS

Reading this story will give you a clue that $175 bucks for a new supply is not outrageous. In this case, the power supply differs from the type normally used in audio gear.

Traditional linear power supplies (LPS) use a transformer that serves two purposes. The very nature of transformers is to isolate the circuitry from the power line for safety purposes. They also "transform the juice" from a high voltage to a low voltage that is then rectified, filtered, and regulated. LPS are not always the best solution for gear that requires lots of current in tight quarters because a transformer does not make efficient use of space.

Switching power supplies (SPS) rectify the juice right off the AC line and can deliver their rated output even when the input varies from 90 volts to 220 volts. This is why their primary filter capacitors are rated at such high voltages (typically 400 volts). Obviously, SPS repair is not for those retrofitted with a pacemaker and/or with both feet in the water. Though a transformer is used, there is a high-voltage section on the primary side that is not isolated from the earth. The filter cap may also stay charged even when the supply is unplugged.

Note: Use the same caution for an SPS as you would for a vacuum tube power amp!

The ADAT eats bipolar 5 and 12 volts. For the circuits that require high current, an LPS transformer would be too large to fit into the space provided. An SPS, however, replaces the 60 Hz line signal with one that is typically above the range of human hearing. That signal is fed to a switching FET (Field Effect Transistor) that then modulates the raw juice into the primary of a transformer.

The high frequency at the various secondary windings can be half-wave rectified and filtered without fear of high frequency "hum" getting into the audio. Regulation is also approached in a different manner. A portion of the low-voltage DC signal is sent to an opto-isolator — an LED and phototransistor on a typically 6-pin IC. Remember that the transformer still provides isolation from the line, as does the opto. The transistor side of the opto is used to control the high-frequency signal sent to the FET, which simultaneously regulates all of the output voltages. This, as well as the half-wave rectification, reduces the component count.

Finally, because the SPS lives at high frequencies, special components are used. Capacitors and diodes must all be rated to operate at high frequencies. Traditional caps would not do the job. *—Eddie Ciletti*

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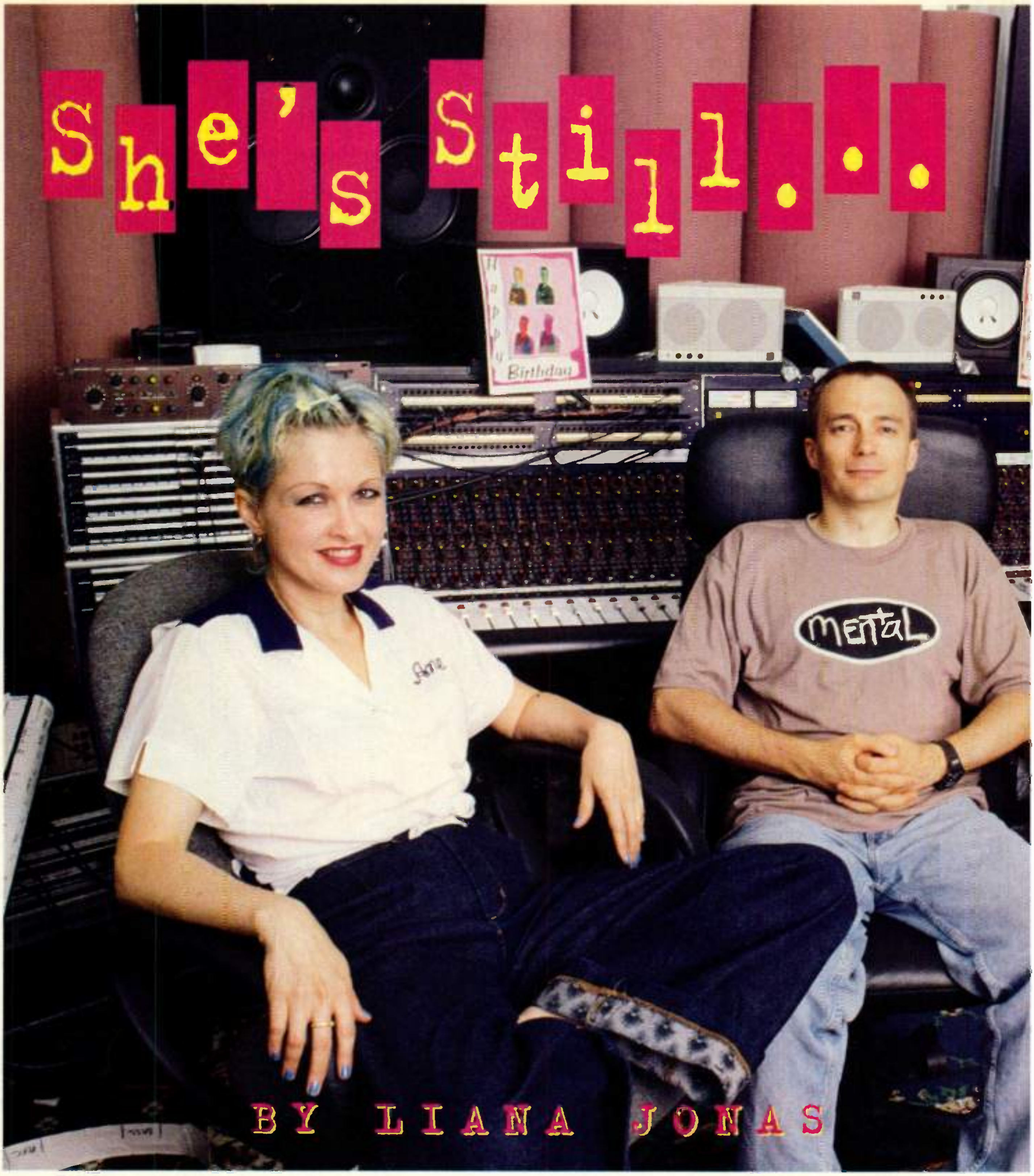
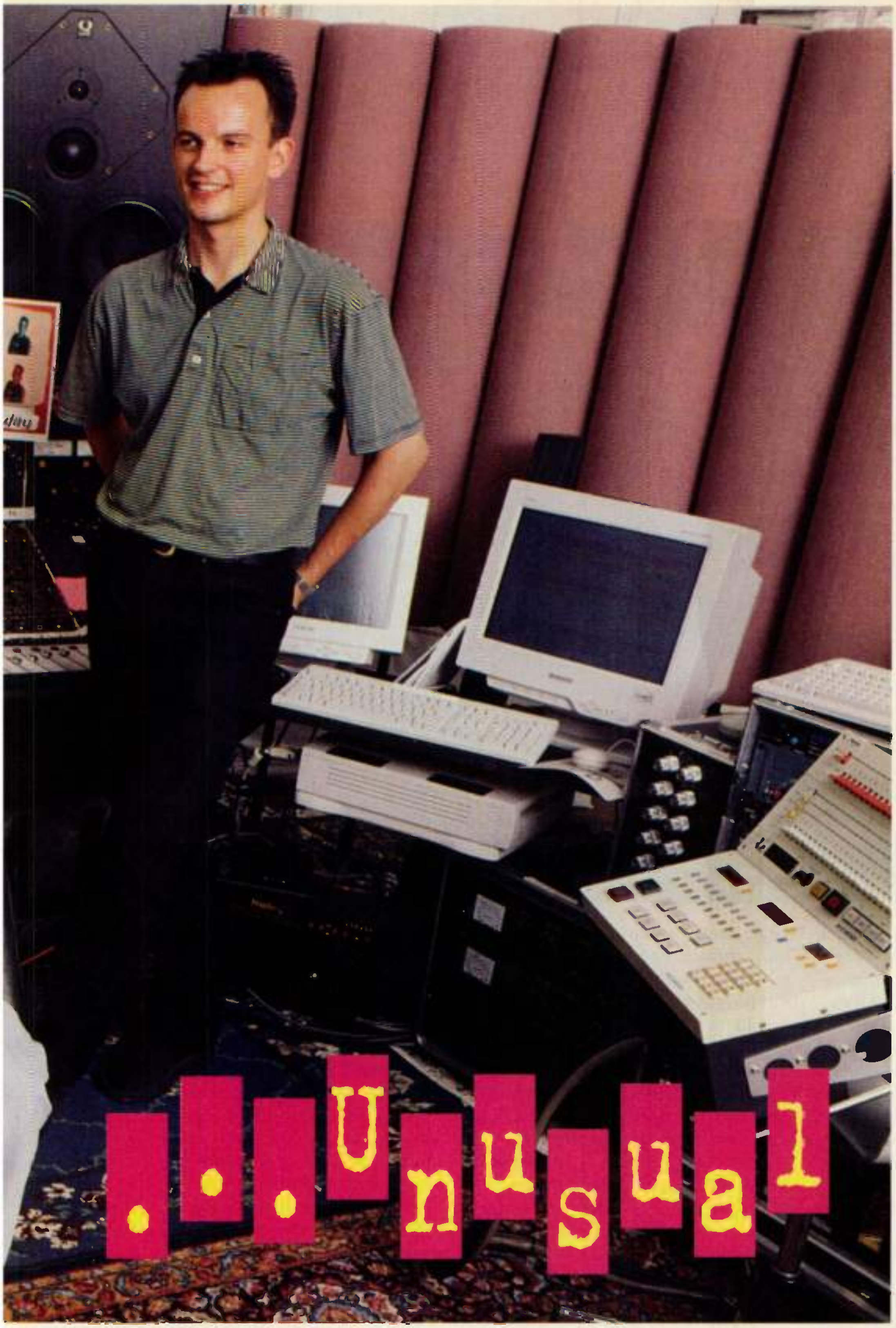


She's Still...

BY LIANA JONAS

With an opulent project studio made up of rare vintage gear and state-of-the-art equipment, Cyndi Lauper turns a suburban mansion into her personal studio



...Unusual

Photos by Julian Jaime

Tuxedo, NY is an out-of-the-way town just about 60 miles from New York City. Nestled deep into its woods in a stately manor are Cyndi Lauper and her crew, recording and mixing her next disc, *Sisters of Avalon*. It was a rare opportunity to get a tour of her project studio (situated in a very picturesque country-like estate) and to sit and talk with the group while they were still in work mode; specifically in mixdown. Concerns about meeting deadlines, completing mixdown sessions, and other final details were still very present, and it added an energy that can't be felt after the disc is pressed and already on the shelf.

The crew consisted of Cyndi, engineer/coproducer Mark Saunders, and studio designer Geoff Daking. Matt Gregory, another engineer, came in at the tail end of the interview. Jan Pulseford, Cyndi's musical partner and album coproducer, was out of town.

THE TIE THAT BINDS

Forming the production team involved bringing in something old and something new. New was Mark Saunders who has recorded the likes of Neneh Cherry, Cathy Dennis, Marcella Detroit of the Shakespeare Sisters, John Lydon of the Sex Pistols, and the Cure. But it was his work with recording artist Tricky that caught Cyndi's ear. Old (but not in age!) was Geoff Daking (of Geoffrey Daking and Co.). Geoff and Cyndi go back a ways; she recorded her *Hat Full of Stars* album at his former studio in Messina, CA. The decision to sign on Geoff to put the studio together was an easy one for Cyndi to make. What wasn't so easy was the fact that she had only given him a little more than a month to build an up-and-running studio for a May 1 start.

"I wanted to create an environment where we could feel really magical and I thought the land and the view here supported that," states Lauper. "Mark and I run five miles a day out here; you couldn't do that anywhere else and I think it's really helped me (vocally)." When asked if she preferred recording this way as opposed to in a commercial studio, Lauper replied with a strong, "Absolutely."

FOR THE RECORD

Lauper's "wish list" as far as studio equipment was concerned included vintage gear and a "fat" sound. To support this, there are racks of vintage and tube gear that sport legendary brand names such as Pye, Fairchild, Pul-

Cyndi

TUXEDO TALES: Cyndi Lauper and studio designer Geoff Daking in her upstate New York hideaway.

tec, Aphex, and dbx; an analog 16-track 2-inch Otari MTR-90 operating at 15 ips; and an old Cadac console from 1974 (more on this board later). All of the microphone preamplifiers used — Cadac, Daking, and Neve — have discrete transistors with transformer inputs. Adding '90s technology to the studio are 16 tracks of Pro Tools coupled with Emagic's Logic Audio recording software. The album was recorded (not simultaneously) on these two different mediums. Backup was handled by 30–40 Iomega Jaz disks.

Saunders attributes the quality of the A/D and D/A converters in getting a good blend between analog and digital mediums. He used the Symetrix 620 20-bit A/D converter and a Yamaha DA202 D/A converter, opting not to use the onboard Pro Tools converters. He notes that when working with 16 tracks on a hard disk, on-board converters run the risk of losing sound quality and becoming "brittle-sounding."

"I'm really hooked on Emagic Logic Audio's recording, it's so flexible," comments Saunders. "When you punch in though, all the audio cuts out for about a half second. This would be okay if we were dealing with MIDI tracks, but we're in most cases dealing with 16 tracks of audio. When Cyndi was trying to do a vocal and I dropped in, everything cut out. This made overdubs really tough; you couldn't really punch-in in a conventional way. We learned to deal with it, of course, but it was disconcerting for Cyndi. I probably wouldn't have had the same problem had I been using Pro Tools software, but I'm so hooked on having control over MIDI and audio on the same page with Logic. As it turned out, we actually worked more on the hard disks and then put it onto the tape for the mixes."

TALES TO TULL

As previously mentioned, at the center of the control room is an old discrete, Class A 32-channel Cadac console from 1974. And talk about history, this board was on the road with Jethro Tull in 1974 — it weighs close to 600 pounds. "The band is probably laying in the bottom of it," jokes Saunders.

The console was not left in its original form as Daking had performed some modifications to it for added flexibility. For one thing, all four aux sends on each channel of the board were rewired to operate in post-fader; prior to the mod, half of the auxs were prefader. Audiomate flying fader automation by Selmark was also installed in the Cadac.

Surprisingly, Daking didn't experience any difficulty installing the motorized faders in the Cadac because the board had no bottom to deal with. This left plenty of room for the strips to drop right in.

Further supporting a vintage theme is Saunders's personal Pye stereo compressor/limiter, which hails from the '60s. The unit definitely looks its age, yet it's Saunders's secret

Cyndi

ment eliminated the need for 100-foot-long guitar cables running downstairs, thus eliminating signal loss. Daking explains, "The Juice Box has got a very low impedance output. So you can drive a very long line to the amplifier and not experience any loss." The guitar amps (there are some vintage Fenders and Voxs coupled with Marshall cabinets) were then miked and run to the Juice Box.

Providing even more isolation was a self-contained bass cabinet with a microphone integrated on its inside, which avoided picking up surrounding sounds. The cabinet was designed by Daking and permitted the bass and drums to be recorded in a hall just about twenty feet from the library. "This was really nice because when we did do group recording everyone could be next to one another," comments Saunders.

THE PROOF IS IN THE SOUND

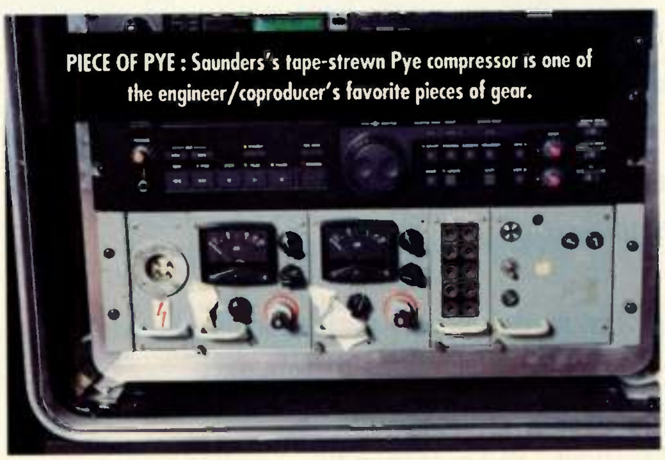
The only area of the home that was acoustically treated was in the control room. Positioned behind the Cadac in a semi-circular wall was an army of ASC (Acoustic Science Corporation) Tube Traps. They provide a dead space for the mix position. On the wall just opposite the Tube Traps was a large diffusor made from molded plastic from Systems Development. The diffusor directly covers a mirror, which was pretty much equal in size. Beneath the mirror was a fireplace that was stuffed with foam. "Doggie pads" — which were these 3- by 3-foot [approximately] vinyl-covered pieces of white foam — were positioned to the fireplace's left and right. Saunders notes that prior to treatment, this room sounded like a "movie theater" in that everything was so loud and resonant.

PINK SLIPS

So where'd all this stuff come from anyway? The answer is a combination of rental companies and personal gear from Geoff Daking, Mark Saunders, Jan Pulseford, and Cyndi. The rental equipment was mostly comprised of the digital components. The vintage and abstract gear all have a place to call home with some of Daking's and Saunders's producer friends. Cyndi owns the Mackies and the ASC Tubes and she says that the next time she sets up shop to record an album it will be less costly and chaotic because she will already have a head start. She also expressed that as a professional musician she *should* own some gear and have microphones that she knows work well on her voice. Lauper explains, "When your disc is done, you're paying off the studio time for almost two years and all you've got to show for it is a disc. That's just not right."

The cost for this project came out to be just as much as commercial studio time, but this time around Cyndi will have a lot more than a disc to show for it. She'll carry along with her some great memories, sharp musical *and* technical skills, and some pretty cool gear that will provide years of service. The studio is slated to remain intact for about a week into the mastering procedure in case any touch-ups need be done. After that it will come down as quickly as it went up (well, maybe not that fast!).

Sisters of Avalon is scheduled to be released this fall "come hell or high water," as Cyndi said. Having seen first hand what went into this album, it seems the chances are good that Ms. Lauper will have a hit on her hands. EQ



PIECE OF PYE: Saunders's tape-strewn Pye compressor is one of the engineer/coproducer's favorite pieces of gear.

Free

Samples have become a major creative force in composing and producing music. Still, they offer an even broader range if used properly. This section gives you the info you need to max out your samples.

Sample

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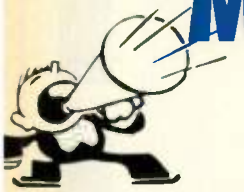
TURNING SAMPLE CDS INTO MUSIC



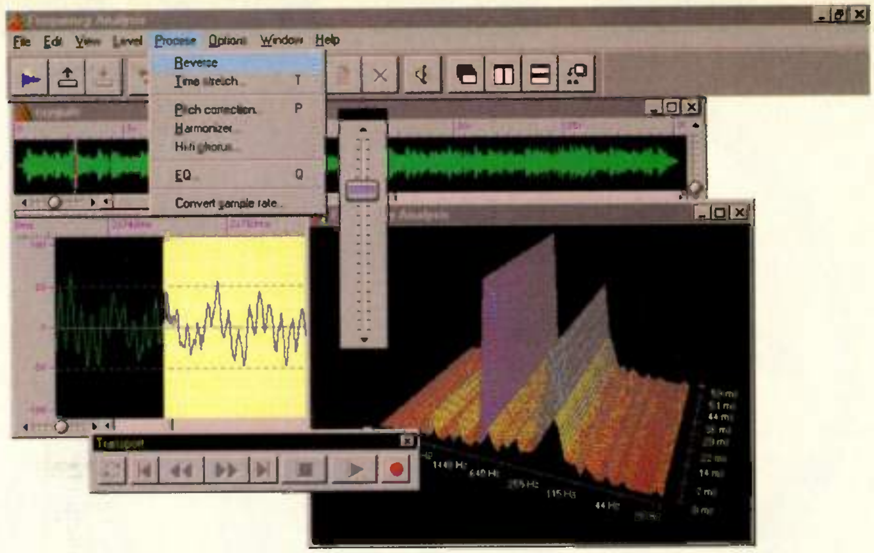


FIGURE 2: Steinberg's Wavelab for the PC reads WAV/AIFF files and offers several DSP functions (as shown on the drop-down menu).

No longer content to be used for simple sounds, today's sample CDs are invaluable production tools

BY CRAIG ANDERTON

A lot of people still don't quite understand the concept behind sample CDs. If you can already play an instrument, why do you need other people's licks and/or sounds?

What makes sample CDs wonderful is not that they can generate a canned composition "out of the box," although that is one attraction. More importantly, just as a soulful guitar sound, clever lyric, or innovative synth patch can be an inspiration, so can a sample CD.

It's a short step from hearing a hot drum loop and wanting to shake your body, to wanting to stuff the loop in a sampler and add sounds on top of it. In the process, you can often create a tune during the initial rush of inspiration because at the very least, you won't need to program or record a drum part...just loop one.

Although today's samplers are popular tools for assembling bits of audio into the equivalent of an audio collage (a task also tackled with hard-disk recorders), their other main function is emulative synthesis. Sample CDs can provide orchestras, drum sets, vintage synths, ethnic instruments, and other hard-to-record sounds. But lifting loops and sounds is only the start — the true test is how you combine these to make music.

Some sample CDs are audio, some are CD-ROMs for loading directly into your sampler from a CD-ROM drive, and some are mixed-mode. Audio versions are less expensive (see sidebar) and easier to audition, but a CD-ROM saves you hours — maybe even days or weeks — of programming time, and is very "plug and play."

So how do you get started with sampling and sample CDs? Here's how.

FIGURE 1: Opcode's AudioShop is a great way to audition audio CDs on the Mac, and it even does some editing.

AUDITIONING THE CD

Sample CDs are organized in different ways. Some are "construction kits" of related, easy-to-loop riffs that you mix and match to create a composition; others have phrases that are not necessarily loops, but can be combined with other phrases to make ambient washes, or overlaid as "solos" on top of more loop-oriented material. Still others contain individual samples of specific instruments (e.g., piano, drums, gamelan, rock guitar, etc.), while many CDs combine looped riffs and the individual samples that make up the loops. This makes it easy to customize the loops by adding additional elements.

Begin by auditioning the CD to choose a collection of possible sounds, and take notes. Computer programs that play audio CDs from your CD-ROM drive (fig. 1) can usually build playlists of particular tracks you want to sample so you can audition and record them in the desired order.

DIGITAL RAZOR BLADES

Unless you record directly into your sampler (or use a compatible CD-ROM), you'll need to transfer the desired samples to a computer for editing, then send them to the sampler.

The standard Mac audio format is AIFF and the PC,

WAV (however, OS/2 Warp can play AIFF files using the media player — just click on the file's icon). CDs using a native file format are easiest to use: pop 'em in the CD-ROM drive, drag the files over, and import them into your sample editing program. To convert formats on the Mac, Sound Designer, Alchemy, WaveConvert, and Peak can all import and export WAV and AIFF files. For the PC, Sound Forge and Wavelab can import AIFF (although when exporting, change the file type to AIFF at the Mac using ResEdit or Disktop). PC shareware translation programs include SOX: V1.0 Sound Exchange and Wave-To for Windows.

To record from audio CDs with a Mac, check out utilities such as CD Studio or SoundEdit 16 (V2), which can grab data from an audio CD and perform file conversion. For the PC, most sound cards have bundled software that can record from CDs, but beware — traditional sound cards don't record digitally from the CD so expect some noise, distortion, and DC offset. If you're a cross-platform kinda samplist, Disc-to-Disk is an audio-grabbing utility available in both Mac and PC versions.

If your computer has a digital I/O card such as CardD Plus or MultiWav Pro, a CD player with S/PDIF digital out (e.g., Denon DCD-1290) can squirt audio directly into typical digital audio editing programs.

SAMPLE MASSAGE THERAPY

Once the samples are in the computer, you can customize them before transmitting them to the sampler — perhaps EQ, truncation, or time compression/expansion to match tempo with a different loop. Digital audio editors (fig. 2) and sample editors provide a graphic window on digital audio; most digital audio editors read WAV or AIFF as their main formats and save to and from the computer only, whereas sample editors can also communicate with samplers hooked up through SCSI or MIDI.

For the Mac, older versions of Sound Designer support older samplers, but the current version communicates only with SampleCell. However, its sibling program, Turbosynth, is a cross between a sampler and synthesizer — it takes your sample and lets you twist it in fiendish ways (although you still need a way to get the sound to your sampler).

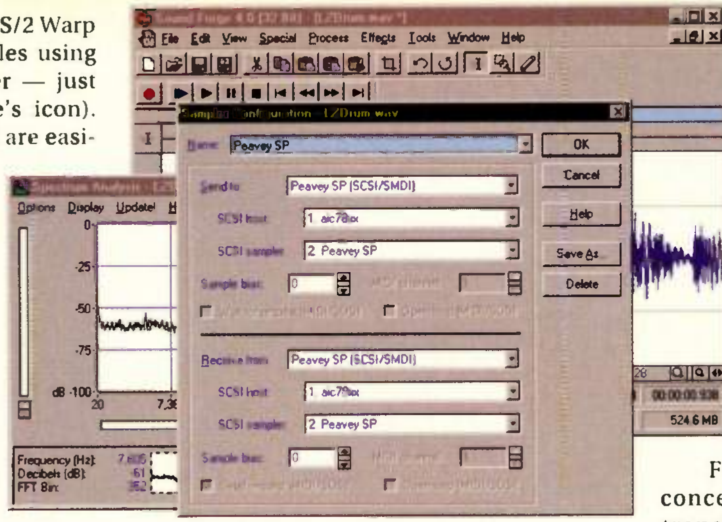


FIGURE 3: Sound Forge's Sampler Tool can be configured for a variety of different samplers (e.g., Peavey SP).

For sound editing, Peak runs in PowerPC native mode (it's also compatible with many older Macs) and is very fast. It supports SMDI (a SCSI-based file transfer protocol) and third-party DSP plug-ins from Adobe Premiere, CyberSound FX, and WAVES. It can also copy loop points and move them in real time — wonderful for dance mixes.

Alchemy, an industry standard for years, limits sample size to available RAM, and the DSP is primitive. However, there's extensive sampler support, easy navigation, SMDI, and a unique window for adjusting the amplitude of every harmonic in a short sample.

For looping, Infinity specializes in looping just about any AIFF file you throw at it. It sounds too good to be true, but it works extremely well. For Mac shareware, Tom Erbe's SoundHack (commercial version also available) offers unusual DSP functions (phase vocoding, binaural filtering, ring modulation, "spectral mutation," and the like).

The PC has lagged behind the Mac, but is catching up. SampleVision supports SDS, SMDI, Ensoniq EPS/ASR, and Akai S900/950. DSP options are extensive: distortion, flanging, delay, 4-band parametric EQ, time compression/expansion, frequency analysis, and so on.

Sound Forge, while designed for 2-track digital audio editing, supports SDS, SMDI, and SampleCell II (fig. 3). It boasts very good DSP (noise reduction, vinyl restoration, reverb, and other goodies), file translation, and batch processing (with an optional plug-in). QSound is also available as a plug-in.

Wavelab has no sampler support, but is a fast audio editor with decent time compression/expansion options, pitch correction, and parametric EQ. It also builds up an audio database of samples — very convenient — and handles AIFF and WAV files interchangeably.

Even some hard-disk recording programs are in on the act: Samplitude Pro and Samplitude Studio allow for SDS transfers (no SMDI, though) and offer looping, EQ, a variety of DSP functions, spectrum analysis, etc.

For shareware, Wave-To concentrates on file format translation, but also receives/transmits SDS and includes tools such as cut/copy/paste, loop point adjustment, and resampling. Syntrillium Software's Cool Edit (currently shareware, but a commercial version is due soon) is a full-function editor that's heavy on the DSP. It offers real-time previews of many processes.

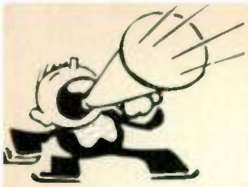
Don't overlook other possibilities. PC programs Fast Eddie and EdDitor (Digital Audio Labs) can handle basic WAV file processing (cut, paste, EQ, change pitch). On the Mac, StudioVision Pro can convert audio into MIDI for editing then back into audio again, while Digital Performer's Hi-Fi pitch-shifting algorithms are outstanding in their ability to minimize the artifacts inherent in this process.

Once the samples have been suitably customized, it's time to send them to the sampler. Before we do, though, there's more to sampler software than audio. SP Remote provides a software front panel for the SP, and sEdit does the same for Akai S1000 series samplers. The Waveboy series of EPS/ASR disks provide extra functionality such as parallel effects, pseudo-vocoding, and resonant filtering. Finally, ReCycle (Mac or PC) is a specialized tool for rhythmic loop fans that can stretch or squeeze a sampled groove (without altering pitch) to fit a particular tempo, alter the groove's timing, change pitch without changing tempo, and other useful remix tricks.

COMPUTER-TO-SAMPLER TOUGH CASES

What if you want to take one format, such as Roland, and translate it into something that your Kurzweil or Ensoniq can read? No problem (well, not much of a problem). First off, many samplers can now read more than one format anyway; but if you do encounter problems, here are some possible solutions.

TransferStation imports samples



(8/16-bit, mono/stereo) into your Mac from CDs/hard disks/SyQuests in Akai, Roland S700, or audio format, saves them as AIFF or Sound Designer II files, then transmits via SCSI to Akai S1000/2000/3000-series devices, E-mu E-IV/E4K/e64/ESI-32, Kurzweil K2000/K2500, Peavey SXII and DPM-SP, and Roland S-760. You can even load "foreign" discs into the Mac CD-ROM drive without having them ejected.

The Sampler Pak plug-in for Peak adds send/receive support and translation for Ensoniq/ASR family, SMDI, and E-mu's ESI-32/e64/E-IV samplers. However, to just shoot AIFF samples out via SCSI to an Ensoniq EPS/ASR, try epSCSI by Steve Berkley (Peak's author) or the ASR transfer utility included with Infinity. They work, but be forewarned: SCSI, under the best of circumstances, can be touchy. Add a sampler to the equation, and things can get downright confusing. (For example, Sound Forge can talk to a Peavey SP with no problems, but is not compatible with the ASR.)

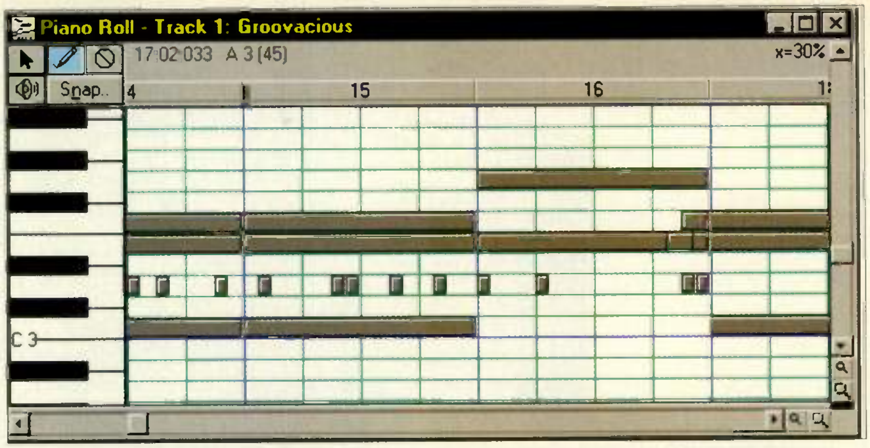


FIGURE 4: A Cakewalk sequence where each note triggers a different loop. Note the short notes; they retrigger loops to add percussive effects. In this case the loop has a decay time, so the effect is very dense and complex as new loops play over decaying loops.

An easier ASR solution is to save samples in the computer to disks the ASR can read, then load them in like normal instruments. Although a PC will normally not read EPS/ASR disks (let alone write files to them), Markus Jönsson's Awave (a great shareware Windows 95 program) can convert just about anything to anything, including WAV to Ensoniq family EFE format. Michael Chen's EPS Disk takes over from there — it can copy the EFE files directly to ASR disks formatted using the ASR's Mac/PC-compatible "computer" mode. On the commercial front, Giebler Enterprise's Ensoniq Diskette Manager lets you read, write, format, and copy Ensoniq disks on your PC (a shareware reader is also available).

Someday, utilities such as these will not be necessary; many newer samplers can now read WAV and/or AIFF files directly from DOS or Mac disks, and this trend is increasing. Another option is one already embraced by Alesis — build a PCMCIA card slot into a synth and transfer samples from your computer to a card (they bundle Sound Bridge software with their synths to do this). Unplug the card from the computer, plug it into the sampler, and load megabytes of samples in a few milliseconds.

MAKING THE MUSIC

The hot sampling topic for music is "pattern looping." For the uninitiated, you map long, looped samples — e.g., a two-measure drum pattern — into your sampler along with bass riffs, drum variations, and solo samples (vocal sounds, guitar licks, FX, etc.). You then bring different loops and solo sounds in and out with the keyboard to create a finished composition. Sound

SAMPLE RESOURCES

- AdB International (MultiWav Pro): 770-623-1410
- Adobe (Premiere plug-ins): 800-521-1976
- Alesis (Sound Bridge): 310-558-4530
- Antares Systems (Infinity, ASR transfer utility): 916-878-6666/800-446-2356
- Beatboy (Artist Signature Series SMFs): 800-838-BEAT
- Bias Systems (Peak): 415-331-2446
- Cakewalk Music Software (Cakewalk): 800-234-1171
- Digidesign (Sound Designer, Turbosynth, SampleCell): 800-333-2137, xt 251
- Digital Audio Labs (CardD, Fast Eddie, EdDitor Plus): 612-559-9098
- Giebler Enterprises (Ensoniq Disk Manager): 610-933-0332; e-mail: giebler@aol.com; web: http://users.aol.com/giebler
- Interval Music Systems (SoundHack, SP Remote, sEDIT, Transfer Station): 310-478-3956; e-mail: interval@netcom.com; web: www.imuse.com
- InVision (CyberSound FX, CD Studio): 415-812-7380/800-468-5530
- KeyFax (Twiddly.Bits SMFs): 408-688-4505
- Macromedia (SoundEdit 16): 415-252-2000
- Mark of the Unicorn (Digital Performer): 617-576-2760
- Opcode Systems (StudioVision, Audioshop): 415-856-3333
- Passport Designs (Alchemy): 415-726-0280
- Qup Arts (Disc-to-Disk): 801-944-9607/800-454-4563
- Sek'd (Samplitude): U.S. Distributor: Soundspiration Systems, 214-298-3472
- Sonic Foundry (Sound Forge): 608-256-3133/800-577-6642
- Steinberg (Wavelab, Recycle): 818-993-4091
- Turtle Beach (SampleVision): 510-624-6200
- Waveboy (EPS/ASR Accessories): 610-257-9562
- WAVES (plug-ins: WaveConvert, C1): 423-588-9307

Online Resources

- Download Awave from: http://www.nada.kth.se/~f93-maj/fmjsoft.html
- EPS Disk, and a bunch of other utilities, are available at a site well worth visiting by any Ensoniq user: http://oak.oakland.edu/pub/eps
- Download AIFF/WAV samples and shareware programs from "Craig Anderton's Sound, Studio, and Stage" on AOL (keyword SSS) and other MIDI forums on AOL, CompuServe, Prodigy, and GEnie.

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Looping the sample itself is problematic because even the slightest timing difference between loops will cause them to lose sync eventually. Generally loops are triggered by a sequencer (fig. 4). For example, to extend a 2-bar loop to 8 bars, you simply send a new MIDI note every 2 bars so that as soon as the loop stops playing, it's retriggered. Most sample CD documentation correlates sample length to tempo so that you can set the sequencer to the proper tempo.

Sometimes you'll want to use loops at a different tempo than the original recording; there are several potential workarounds. For example, the *Def House* CD-ROM (dist. by Ilio) assigns each loop so that playing G1 gives a tempo of 110 BPM, A1 gives 111, and so on in 1 BPM increments up to B5 (140 BPM). On the other hand, discs by E-Lab (e.g., *X-Static Goldmine Series*) record loops at tempos that are exactly 1/2 step apart (112, 119, 126, etc., BPM). Transposing the 112 BPM loop up a semitone changes its tempo to 119; a whole tone takes it to 126 BPM. This lets you mix and match otherwise incompatible loops, although the pitch changes a bit (you can't have everything).

Mapping samples on the keyboard is a personal matter. For "live" mixes, I prefer to layer all the sounds across the keyboard on one channel, assign each loop to its own note, and create a mix just by playing. If you're writing at a sequencer, consider placing each loop on its own channel, and track for more flexible arranging.

When importing files into a sampler, you may not be able to multisample in the conventional way. For example, with the ASR, since each translated sample on the disk looks like a complete instrument (not just a wavesample), loading in a new "instrument" to the same slot deletes the previously loaded instrument. To get around this, create a second instrument as a scratchpad memory. Load files into the scratch instrument, then copy over each wavesample to the main instrument and place them in the appropriate layer(s).

There's also plenty of tweaking you can do with a sampler. For really dense sounds, set up the envelope for each sample to a "repeat" or "finish" mode (also called "loop in release") and set the sample to keep repeating (or copy it to itself to double the length) so that you can just touch a key and get a long, decaying phrase. Every time you send it another MIDI note-on, another layer gets added to what is already playing.

Don't forget the sampler's onboard effects. Distortion is pretty amazing on drum parts, and transposing slow loops up in pitch can not only speed them up, but change the timbre. Perhaps the most valuable process is time/compression expansion. This can change the loop length without changing pitch, and, often, change pitch without changing length. If you want to use a 137 BPM loop with a 140 BPM loop, you'll really appreciate this feature. The sampler's normalization function can increase a signal's level, and most samplers can vary the volume for individual MIDI channels.

EMULATIVE SAMPLING

Sometimes you'll use the sampler to create dynamite instrument sounds — a rich sampled grand, perhaps, or a string section. Adding realism is a whole other topic, but here are a few tips.

• Don't just change dynamics with level; also change the filter for a brighter sound with higher velocities. Modulating the sample start point can be very effective, too — at low velocities, start further into the sample to bypass the attack, then program higher velocities to modulate the start point negatively so it plays through more of the attack. Try this with any percussive instrument, including melodic ones like guitar.

continued on page 160

SAMPLE POWER: Use these 12 tips to get more out of your sampler than you ever dreamed possible. Shown here: the Akai S3000XL.

STATE OF THE SAMPLER

A sampling tips 12-pack

BY CRAIG ANDERTON

Samplers have a reputation of being underutilized — lots of musicians just pop in a piano or strings disk and leave it at that. And that's a shame, because a sampler is much more than a keyboard instrument: it's a bunch of little digital recorders, each turned on and off by a switch (key). Following are 12 tips designed to inspire you to get just a bit more out of your sampler.

CUT & PASTE SOLOS

Composite solos, where you record multiple takes and mix down the best bits from each take to create an idealized solo, are a natural for samplers. Record multiple takes on tape (analog or digital), then sample the best sections. Sync the sampler's onboard sequencer to tape, and play back the resulting mix into an open tape track. Granted, assembling a solo this way can be a bit more tedious than with a hard-disk system, but you also gain options like envelopes, filters, LFOs, etc.

FIXING OUT-OF-TUNE NOTES

Samplers were born to bend, or they wouldn't have pitch-bend wheels. If a vocal performance is fantastic except for a couple of slightly flat or sharp notes, sample the phrase into the sampler.

For easy fixes, move the pitch-bend wheel manually, in real time, as you bounce over to another track. For more complicated pitch changes, sync a sequencer to the multitrack and "draw in" the pitch-bend messages, then edit as needed. This also works for generating nifty special effects, like slides.

If you add a lot of pitch bend, the length of the sample might change. Breaking a vocal down into phrases and editing each one individually generally solves the problem.

YES, YOU CAN HIT HIGH C!

Well, maybe not quite. But if a note is just a couple steps above or below your range, sample the nearest note you can sing correctly and transpose it up or down as needed. Drastic pitch shifts will cause unnatural vocals, but if you're dealing with harmony parts that aren't too far from the leads and mixed somewhat in the background, no problem.

THE ULTIMATE VIBRATO

To add vibrato to a signal, sample it, then modulate the pitch with a low-frequency triangle wave (bring it in and out with the mod wheel for expressiveness). Because the LFO waveform is symmetrical, it shortens and lengthens the note by equal and opposite amounts in the process of changing pitch, so the total note length remains unchanged.

THE ULTIMATE DDL

Sample the phrase you want to echo and set the sampler keyboard to play all the same pitch (this mode is used for sound effects and some percussion; refer to your sampler's manual for how to do this). Also set the amplitude envelope for a long release time, and make sure the overall amplitude responds to velocity.

Hitting any key will trigger the phrase, so whenever you want another echo, just hit a key. And no law says you have to emulate a traditional echo unit — try polyrhythmic echoes or changing the volume levels of different echoes. Since samplers generally let you place different keyranges at different locations in the stereo field, this also works for cross-channel effects.

MAKING BIDIRECTIONAL LOOPS WORK

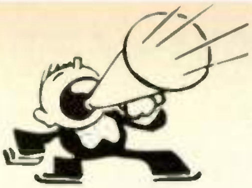
When using bidirectional (forward/ backward) looping, remember to turn off any "autofind" function that puts the loop points on zero crossings. If the signal reverses at a zero crossing, it will probably produce a discontinuity in the waveform (upper signal in fig. 1; the gray wave shows what the reversed signal would look like if played right after the original signal). Setting the loop point at a peak, as shown in the lower signal, will create a smoother loop.

FUN WITH SAMPLE START MODULATION

A sample is "freeze-dried" sound that lacks the nuances that occur over time with acoustic instruments, but modulation can help. One trick is to use velocity to increase brightness, but sample start point modulation — a feature that affects where in the sample playback begins — can be equally, if not more, effective. By starting a sample just after the attack and adding negative velocity modulation, the harder you play, the closer the start point moves toward the beginning, thus picking up more of the attack. Try this with acoustic guitar to make for much more dynamic picking effects. Note that most envelopes can be modulated by velocity as well.

Another use for sample start point alteration is to create stereo from mono. Copy the mono sample, pan it to the opposite channel compared to the original signal, and change the start point of one of the samples until you hear a distinct stereo spread. *Caution:* Also play this back in mono to check for cancellations; if the sound gets thinner, increase or reduce the start time difference until it sounds right.

Our final sample start tip assumes you're recording a vocalist into a multitrack, and, unfortunately, there's a "pop" at the beginning of an otherwise flawless phrase. Sample the phrase (sync the sampler to the multitrack so you don't have to worry about punching



in at the right time later on) and use the truncation parameter to move the start point a bit further into the sample, past the pop. Lay the sampled track over the original track, and the problem is solved. (An amplitude envelope change could also work for this.)

PRACTICE MAKES PERFECT

If you want to loop part of a song (e.g., solo, chorus) for practicing, record that section into a sampler (use a low sample rate to provide more available recording time) and loop away. If you're using digital or analog tape, this saves a lot of head wear compared to using the recorder's "block repeat" mode.

OUT OF BOUNDS

Transposing samples out of their normal ranges can create an entirely new sound. Transpose slap bass up a couple octaves for a meaty clavinet sound (better yet, layer it with a real clavinet), or transpose a snare way down (and add lots of filtering) for a distant cannon or thunder sound. Transposing a sample to the maximum highest pitch will often provide ring modulator-like effects due to aliasing.

TRIMMING RHYTHMIC LOOPS

Using rhythmic loops from CDs can be fun, but they aren't always at exactly the right tempo to sync up with other loops you may have sitting around. However, as long as the loop you want to change uses nonpitched material (e.g., drums), you can lengthen the loop by bending pitch down somewhat, or shorten the loop by bending pitch up. This will often give better-sounding loops than using computer-based time compression/expansion programs.

THE LOOPMEISTER

If you can translate audio files among different samplers, then you can use one sampler's capabilities to process samples created on a different machine, or modified with a particular program. For example, most sample editing programs (aside from Infinity) allow only for forward looping or crossfade looping, but Ensoniq's samplers offer a variety of loop processing options (backwards/forwards, reverse "bow tie," etc.) that process the samples to make the loop points far less obvious. As one example of how to use inter-machine transfers, I was trying to loop a string section on a

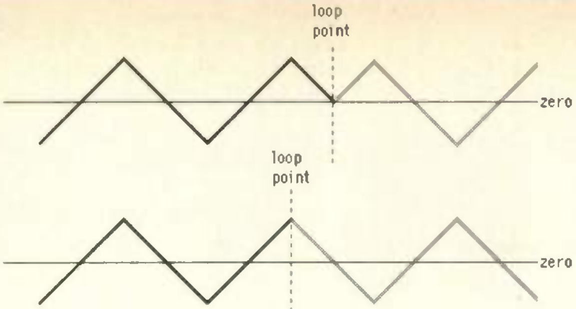


FIGURE 1

Peavey DPM3. Alchemy did a pretty good job, but the loop point was still fairly obvious. So I sent the sample to an EPS 16 Plus, used its looping capabilities to create a perfect loop, bounced the sample back to Alchemy, then exported it to the DPM3. Of course, because all this was in the digital domain, there was no audible deterioration.

CATCHING THE BUS

If you have a spare aux send, route it to your sampler's input. You never know when you might want to sample something, and if the bus is already routed, all you have to do is turn up the appropriate aux send control and set levels.

Happy sampling, and be creative — machines love it when you do that. EQ

THE STANDARD MIDI FILE/SAMPLE CD CONNECTION

If you're into sample CDs, Standard MIDI Files of licks (marketed by companies such as Beatboy and KeyFax) can be a useful complement. Unlike a CD, which contains audio, the SMF contains MIDI data for drum loops and other riffs. SMF advantages include easy transposition, tempo changes that don't shift pitch, and simpler editing. The downside is that you have to supply the sounds that the SMF triggers, and it can take some time to set up something like a complete MIDI drum sound kit.

Some companies now package SMFs and CDs together, where the CD supplies the sounds and the SMF contains MIDI versions of audio loops on CD. Thus to modify the loop, you would import the SMF into a sequencer, do your editing, drive a sampled drum kit, and, optionally, re-record the tweaked loop as digital audio.

Peter Siedlaczek

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the samples in your Keymap. After you've truncated all your samples, exit out to the "ProgramMode" page once again and you'll notice that the samples are still arranged in the same program. Save this to disk.

We will now use the "VolRamp" function on a sample that decays naturally over time like our "TIPS BASS" sound. When sampling these types of sounds from some older synths you may notice some noise as the sound decays. This noise is present at the initial attack of the sound, however, you can't hear it because the sound is louder than the noise itself. The noise stays at a constant level, but the sound itself is decaying and becoming lower in volume. As the sound fades and becomes lower in volume, the noise becomes more apparent. When you're faced with this type of noise you can use the "VolRamp" function in the DSP section to fade the sample and the noise at the same time.

Cursor over to the "EndLvl:" section of the display on the "VolRamp" page. Set the "EndLvl:" to –96 dB and hit GO. This will fade the sample along with any slight noise you may be hearing. When the Kurzweil has finished this fade it will ask "Keep this change?" Play the sound. If you don't like the new version of the sound, answer NO. Remember, if you do answer NO, it will bring you back to the "VolRamp" page. At this point you can experiment with other settings or change the type of curve. I find that setting the curve to "LIN" works fine for this application. After you've used the "VolRamp" function on all your samples, exit out to the "ProgramMode" page. Again you'll notice that the samples are still arranged in the same program. Save this to disk.

The "Resample" function could be very useful to musicians who would like to conserve memory. If you've been sampling in stereo and at 44.1 kHz, this will certainly give you quality samples, but it may be wasting memory. Certain sounds do not necessarily need to be sampled at 44.1 kHz. You may find that sampling at 32 kHz or 29.4 kHz is acceptable for what you are doing.

From the "SampleRecordSamples:" page select the sample you need to resample. Press EDIT and then DSP. Dial to the DSP page called "Resample." If your original sample was taken at 44.1 kHz, try setting the "NewRate:" to 29400 Hz. On this page you'll also notice a choice called "Quick." Set Quick to "0." The "0" setting gives a quick preview of what the new sample will sound like. Press GO. When the Kurzweil is done resampling it will ask "Keep this change?" Listen to the difference. If you like what you hear press NO (yes, I did say NO). This will take you out to the "Resample" page in the DSP section. Now you can go to the "Quick" parameter and select "1." Now press GO. This will take a little longer, but it will give you the final version of your resampled sound. When the Kurzweil is done, it will ask you "Keep this change?" Play the sound. If you like the new sample, answer YES. If not, answer NO and the Kurzweil will take you back to "Resample" page. Here you can experiment with other sample rates till the sample sounds like you want it to.

Until next time...stay well.

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 http://www.interport.net/~thefront/index.html You can also e-mail Tony at thefront@interport.net


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AUDIO SAMPLE CD REVIEWS

A sampling of currently available sample CDs

BY CRAIG ANDERTON

[*Note: Companies listed are distributors of the CDs, not necessarily the creators.*]

DRUM LOOPS 3/ BIG FISH AUDIO

Musical Style: Rap, house, old school, techno. **Format:** Drum loops, identified by style and BPM, plus individual drum hits. **Best Features:** Great selection and variety. **Limitations:** No "construction kits" for putting different sounds together; stereo field is underexploited. **Bottom Line:** Cost-effective and gets the job done fast. 640 MB. $79.95.

LOOPZILLA 2/ BIG FISH AUDIO

Musical Style: Rap/hip-hop. **Format:** 91 "construction kits" (70–145 BPM) with typically half a dozen or more samples per drum kit, plus individual solo samples (guitar loops, bass loops, drum hits, scratches, etc.). **Best Features:** Good playing, good feel, very genuine, and usable. **Limitations:** Technically unpolished (level variations, leakage in some solo samples). **Bottom Line:** The paradox is that the less-than-perfect polish is what makes this genuine and usable. 1290 MB. $99.95.

OUT OF AFRICA 1/ BIG FISH AUDIO

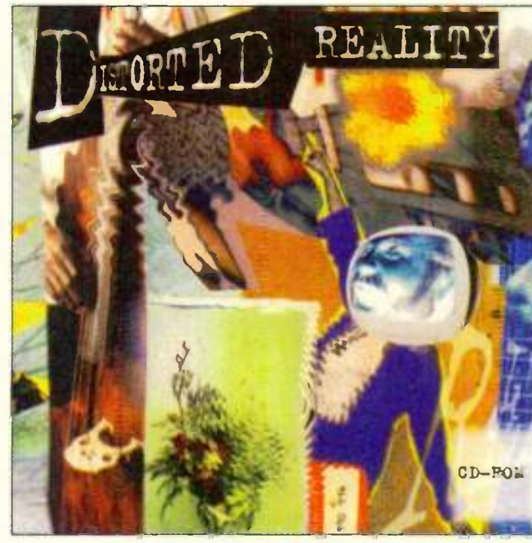
Musical Style: Traditional music from Ghana. **Format:** Fairly lengthy samples of mostly ensemble sounds, many usable as loops. **Best Features:** Authentic and interesting (cool xylophone loops!). **Limitations:** Reliance on ensemble sounds limits ability to integrate into many types of music. **Bottom Line:** More for soundtracks and ambience than creating grooves. 618 MB. $99.95.

PERCUSSION SLAM/ INVISION

Musical Style: Latin, dance, R&B, urban, etc. **Format:** Loops with associated versions that eliminate particular instruments, plus individual samples; standard MIDI files on accompanying disk. **Best Features:** Superb sound, SMFs are a useful accessory, superior organization, excellent stereo. **Limitations:** Not a lot of samples at different velocities. **Bottom Line:** A solid sample disc by any standards, and invaluable if you want great percussion grooves. Highly recommended. 577 MB. $99 (includes PC or Mac MIDI files); Akai CD-ROM, $199.

HIP HOP/ INVISION

Musical Style: Rap/hip-hop. **Format:** 48 "construction kits" (75–100 BPM) with a complete riff and constituent elements, plus additional drum loops, guitar and bass riffs, scratches, and 10 drum kits. **Best Features:** Big sound, very professional yet retains a suitably greasy feel. Lots of wah-wah guitar. **Limitations:** No major limitations. **Bottom Line:** The sound is excellent, the samples are there, and the price is right. 629 MB. $99; Akai CD-ROM, $199.



DISTORTED REALITY/ ILIO

Musical Style: Ambient, dance, soundtracks. **Format:** Individual samples, from dreamy ambient to intense distorted drums, and about 50 body-shaking loops. **Best Features:** Exemplary attitude; highly "organic" electronic sounds; reminds you that all this gear can be used creatively. **Limitations:** No multisampling (except on CD-ROM). **Bottom Line:** Get it quick before everyone else finds out and overexposes these wonderful sounds. 651 MB. $99; Akai/E-mu/ASR/Roland/Kurzweil/SampleCell CD-ROM, $199.

HEART OF AFRICA/ ILIO

Musical Style: Ethnic African music. For-

CONTACT LIST

Big Fish Audio, 11003 Penrose St. Suite C, Sun Valley, CA 91352. Tel: 800-717-3474; 818-768-6115. Circle EQ free lit. #119.

Ensoniq, 155 Great Valley Parkway, Malvern, PA 19355. Tel: 800-553-5151; 610-347-3930. Circle EQ free lit. #120.

East-West, 345 N. Maple Dr., Suite 277, Beverly Hills, CA 90210. Tel: 800-833-8339; 310-858-8797. Circle EQ free lit. #121.

Hollywood Edge, 7060 Hollywood Blvd., Suite 1120, Hollywood, CA 90028. Tel: 800-292-3755; 213-466-6723. Circle EQ free lit. #122.

Ilio Entertainments, Box 6211, Malibu, CA 90264. Tel: 800-747-4546; 818-707-7222. Circle EQ free lit. #123.

InVision, 2445 Faber Place, Suite 102, Palo Alto, CA 94303-3316. Tel: 800-468-5530; 415-812-7380. Circle EQ free lit. #124.

Q Up Arts, 1231 East Warnock Ave., Salt Lake City, UT 84106. Tel: 800-454-4563; 801-488-0062. Circle EQ free lit. #125.

mat: One disc of multisamples, one of phrases (not necessarily loops). **Best Features:** Excellent sound quality; great vocal samples; outstanding documentation and organization; all samples and phrases very usable; audio CD has easy-to-find loop points. **Limitations:** None. **Bottom Line:** For African sounds, you'll have a hard time finding anything better. 1201 MB. $129; Akai/E-mu/ASR/Roland/Kurzweil/SampleCell CD-ROM, $299.

HEART OF ASIA/ ILIO

Musical Style: Japanese, Indian, Malaysian, Thai, Tibetan. **Format:** One disc of mostly loops and one of mostly phrases. **Best Features:** Clean sound quality; good variety; very atmospheric. **Limitations:** No multisampling (except on CD-ROM). **Bottom Line:** An ideal companion to *Heart of Africa*, with equivalent quality. 1299 MB. $129; Akai/E-mu/ASR/Roland/Kurzweil/SampleCell CD-ROM, $499.

HEAVY GUITAR LIBRARY/ Q UP ARTS

Musical Style: Rock. **Format:** Individual samples (chords, riffs, finger noise, leads, mutes, effects, etc.; no rhythmic loops). **Best Features:** Samples of guitar sounds with effects are great. **Limitations:** Demo song is unimpressive; few complex samples like riffs; uses up only about 1/3 of the CD's capacity. **Bottom Line:** You still can't play guitar with a keyboard, but this is a decent attempt — and who says you always have to use guitar samples to sound like a guitar any-

ENSONIQ CD-ROM REVIEWS

By Tona Ohama

Sonic Arts/Ensoniq

Musical Style: Classic pop/rock. **Format:** Multisampled instruments, many with 2 or 3 cross-switch layers, some drum loops. **Best Features:** Vintage instruments sampled through vintage equipment with some of the clanks and grinds of the original instruments left in for a more realistic sound. **Limitations:** The only guitar included is a nylon-string acoustic; many samples are at 30 kHz. **Bottom Line:** A solid general purpose sampling of the sounds of the '60s and '70s (e.g., Rhodes 73, "Zeppelin" drums, "Superstitious" Clavinet). 311 MB. $199.95.

Psychic Horns/Ensoniq

Musical Style: Funk, dance, rock. **Format:** Small, medium, and large horn section samples of stabs, swells, falls, and funk riffs. **Best Features:** Great samples; excellent energy and feeling. **Limitations:** Despite the "Psychic" title, these horn sections are very traditional. **Bottom Line:** Well done! But you're going to want more. 383 MB. $199.95 (Also available in audio and Akai format from Q Up Arts.)

way? 176 MB. $99; E-mu CD-ROM, $199.

SYMPHONIC ADVENTURES/ EAST WEST

Musical Style: Orchestral-based soundtrack music. **Format:** Relatively long orchestral phrases, usually with variations (different arrangements, individual parts, endings, etc.). **Best Features:** Phrases from different tracks often work well together; very effective music; great concept. **Limitations:** No significant limitations. **Bottom Line:** Mostly the dark side of the orchestra (chases, suspense, FX, etc.), but you really could score an action-adventure-suspense flick with just this CD. (Techno/ambient fans take note: there's a gold mine here.) 481 MB. $99.95.

MASTERS OF MAYHEM/ EAST WEST

Musical Style: Industrial/hardcore techno. **Format:** Loopable arpeggiations, drum loops, phrases, individual samples, distorted guitar licks and bass lines, spoken words from old flicks, you name it. **Best Features:** Truly rude noises done very well; extreme variety; lots of value for the money. **Limitations:** Listening to the CD in its entirety can drive you insane. **Bottom Line:** If you're a noise band enthusiast, this is real in-your-face stuff that's easy to use, extremely analog-sounding, and capable of frightening small animals if applied correctly. Two thumbs up. 614 MB. $99.95.

SFX AND SFX 2/ HOLLYWOOD EDGE

Musical Style: 2-CD set of sound effects. **Format:** One CD has 300 sound effects including some musical phrases, available as audio (16 bit/44.1 kHz) and 4 WAV formats (8- or 16-bit stereo, 8- or 16-bit mono, 22.050 kHz). The other has 420 sound effects and 83 short pieces of music as 8- or 16-bit, 22.050 kHz, mono WAV files. **Best Features:** Includes database and search software for locating and auditioning particular sounds; well-rounded selection; multiple formats. **Limitations:** 44.1 kHz WAV files not available; search software is different for each CD and could use a better interface. **Bottom Line:** This is clearly targeted at multimedia developers who favor using a sampling rate of 22.050 kHz and need access to a large number of popular sound effects. 915 MB. $29.95 each. EQ

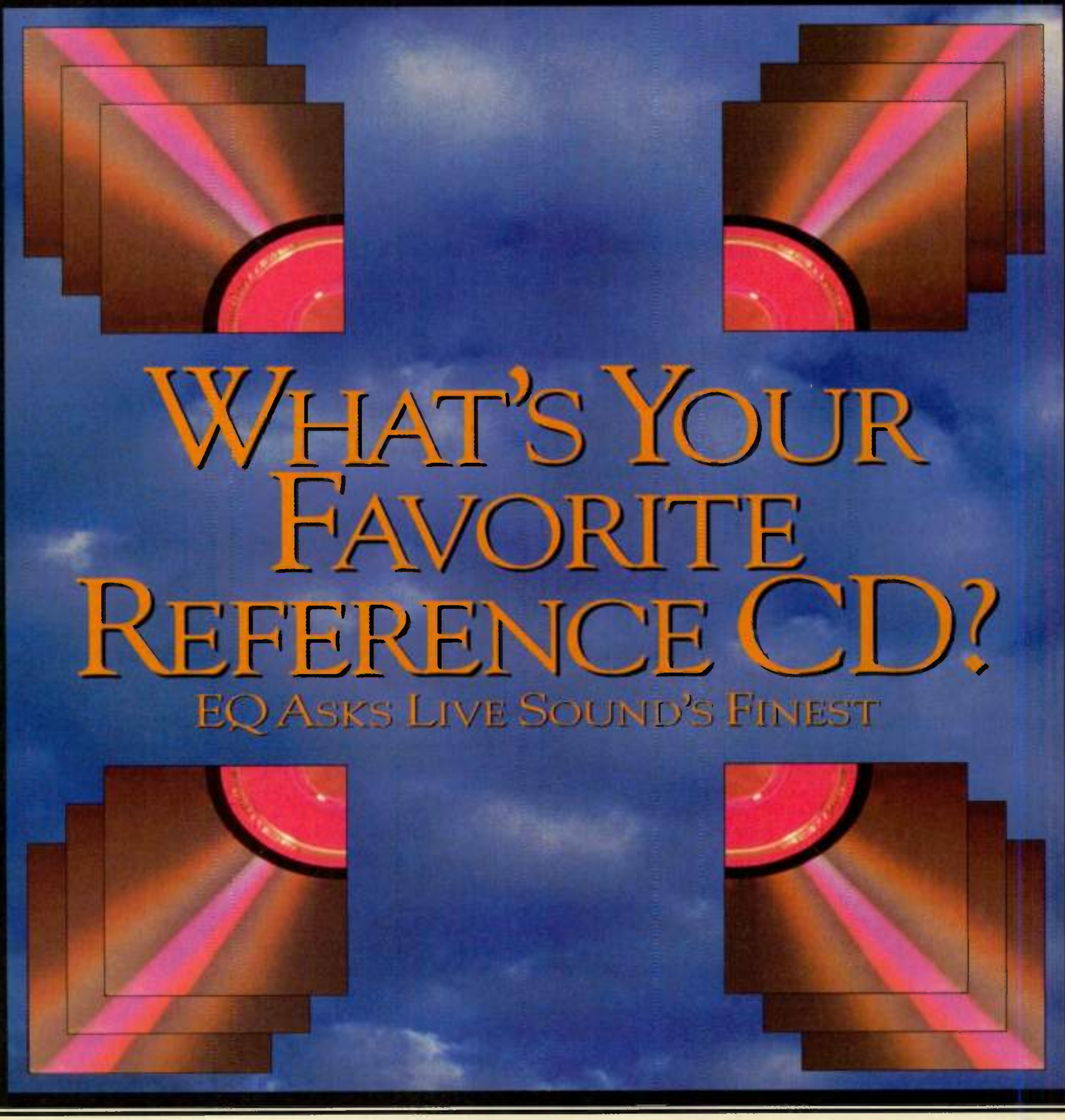
SO WHY DO CD-ROMS COST SO MUCH?

Audio sampling CDs are quite inexpensive, typically ranging anywhere from $10 to $100. CD-ROMs tend to be much more expensive, often hitting $200 to $300 and even more. Why?

It's the same principle as buying a steak at the supermarket compared to going into a four-star restaurant and ordering filet mignon. The raw samples require a certain level of expertise just to record them well, but to turn them into presets with proper mapping, envelopes, effects, and so on is extremely time-consuming. Basically, you're paying for someone's time and expertise. If you're willing to put in that time and expertise yourself, you can always buy the audio version and do the preset creation yourself.

—Craig Anderton

EQ LIVE



WHAT'S YOUR FAVORITE REFERENCE CD?

EQ ASKS LIVE SOUND'S FINEST

THE REFERENCE SECTION

SOUND-REINFORCEMENT'S TOP PEOPLE REVEAL THEIR FAVORITE TUNING TUNES

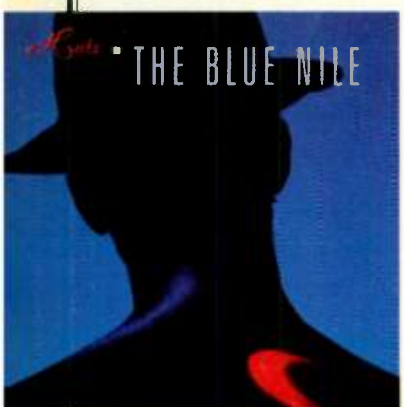
BY STEVE LA CERRA

EQ ASKED THE industry's top live mixing engineers what their favorite CD was for EQing and tuning PA systems. Their answers varied widely and some engineers even bucked the idea altogether of using a CD as a reference. Here's what they had to say...

STAN MILLER

Presently mixing house for Neil Diamond.

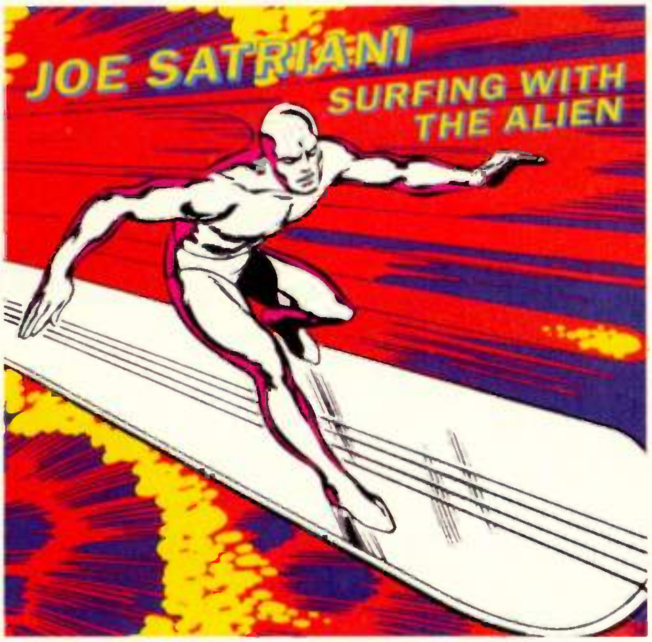
We use several different songs from Neil's latest CD *Under The Tennessee Moon*, in particular the songs "Blue Highway" and "Can Anybody



HATS OFF: Rob "Cubby" Colby likes The Blue Nile.

Hear Me." I am most interested in what Neil's voice sounds like in the PA. Whereas many other engineers will build their mix from the drums, I build my mix around Neil's voice — if his voice doesn't sound right, then the rest of the mix doesn't matter.

"Blue Highway" has only two instruments: Neil's voice and Chet Atkins's guitar, and there is very little processing



SURF DUDE: Bob Butler's choice for a reference CD.

on Neil's voice. This gives me a very good idea of what Neil's voice will sound like in the PA. The song "Can Anybody Hear Me" features the whole band and is part of the show, so it really gives me an idea of what things will sound like when the whole band starts playing.

GREG PRICE

Currently with Ozzy Osbourne; Greg has also mixed Steve Miller, Boston, and Huey Lewis and the News.

A lot of engineers use a Steely Dan CD or the Donald Fagen *Nightfly* CD, but for hard rock, the popular ones seem to be AC/DC's *Back In Black* or Motley Crüe's *Dr. Feelgood*. I think you need two CDs to check a PA system: one to test and one to tune. For an engineer on a club tour — who is dealing with a different room and a strange PA system every night — the first thing is to make sure that the circuit breakers are supplying enough voltage to the power amplifiers. For this you need something heavy with a lot of energy in the drums and bass (like the AC/DC or the Motley Crüe). Run the PA up to about 105 dB and see how the system handles it. Will the breakers pop? Are things hooked up correctly? Also try putting bursts of pink noise through the system one band at a time. This will let you find out if there are any blown drivers.

Once you are past the point of checking the PA's apparatus, you can move on to the tuning. I have been using "If I Ever Lose My Faith In You" from Sting's *Ten Summoner's Tales* for most of this Ozzy tour. After you settle on a CD, try to stick with it for a while. The rest of the crew will get sick of it, but if you keep changing CDs, you keep changing the reference point.

For me, the Sting CD is perfect. It has very good articulation in the vocal range and that is most important to me. I need to know how the vocals will read in the room. If I can get the vocals articulate, then the guitars and drums will follow. A lot of engineers focus on the bottom end too much. If I feel that I need a little more slam in the bottom end, I am more likely to turn up the low-frequency output of the crossover than to start messing with the EQ. If you turn up the bass frequencies on the EQ, you will be introducing distortion into the system. Right before our show starts, we play an Ozzy song and that gives me a chance to do some final tuning with the room full of people. So if you can, play a familiar song right before show time for final tweaking.

ROB "CUBBY" COLBY

Most recently with Bob Seger, Cubby also works with Genesis and Phil Collins.

I use a variety of CDs on a daily basis to listen to the PA system. Lately I have been using the title cut from Sting's *Mercury Falling*. This is a great production by Sting and Hugh Padgham. Somehow they have managed to achieve a very warm, soft sound in the upper midrange, which gives me a good reference. "Downtown Lights" from The Blue Nile's *Hats* CD is also a good reference. From Donald Fagen's *Nightfly*, I listen to about 35 to 40 seconds of the intros to a variety of songs. I'll listen to the way the snare drum reverberates in the room to get an idea of how the room sounds. I use "Driving While You Sleep" and "Fading Lights" from the Genesis CD *The Longs*. "Fading Lights" in particular has a lot of keyboard sounds that could get lost in a busy room if you're not careful. It starts with drum machine and then Phil (Collins) comes in with the live drums, so I can get an idea of the dynamics of the system. Sometimes I will also use songs from Phil's *Hello I Must Be Going* or *No Jacket Required* — both clean recordings with a lot of space.

DAVID NORMAN

Currently mixing house for the Neville Brothers Featuring Aaron Neville, he has also worked with Arrested Devel-

THIS IS ONLY A TEST

By Wade McGregor

I usually use a Techron TEF 20 to measure studio and sound-reinforcement loudspeakers. However, the subjective testing must be done before and after to ensure that the objective result of system adjustments is appropriate. With a little Sony Discman and three CDs, I can be confident that the loudspeaker system has been optimized for playback.

The first CD I use is the Bruel & Kjaer Pro Audio Promo CD (CD 4090). It is a collection of excellent recordings that also happen to use the B&K studio mics. I trust this CD to evaluate the system's musical qualities with excellent recordings by George Massenburg and Jack Renner.

The most telling test of a system is voice quality, and the SynAudCon Test CD includes familiar voices for judging speech intelligibility and tone. It also includes pink noise in a variety of bandwidths to check for frequency response changes and comb-filtering while walking across the coverage pattern of the loudspeaker system. There are also a series of shaped tone bursts that are useful in listening for transient response anomalies and image shifts (in studio monitors).

I use the Prosonus Studio Reference Disk for a test tone generator and the Acoustic Science Corp. — Music Articulation Test Tape (track 50) to evaluate the subjective time and frequency behavior of a loudspeaker. The MATT signal is a series of stepped tones (1/16th second on/off) that quickly makes resonances and rattles audible. The resonances noticeably change the pace of the stair-step tones as they progress from 20 Hz to 800 Hz and back to 20 Hz. This is also probably the easiest way to set the balance between the subwoofer and full-range loudspeaker in a control room. The CDs do not replace listening to someone speak through a microphone in a sound-reinforcement system, but they offer an excellent way to have a reliable source in any studio or venue.

CD Sources: Bruel & Kjaer Pro Audio Promo CD — CD 4090 (out of print!): TGI North America, 300 Gage Avenue, Kitchener, ON, Canada, N2M 2C8. Tel: 519-745-2364. SynAudCon Test CD: Synergetic Audio Concepts 8780 Rufing Road, Greenville, IN 47124. Tel: 812-923-0174. Prosonus Studio Reference Disk – SRD: Synergetic Audio Concepts 8780 Rufing Road, Greenville, IN 47124. Tel: 812-923-0174.

opment, Patti Austin, and Peabo Bryson.

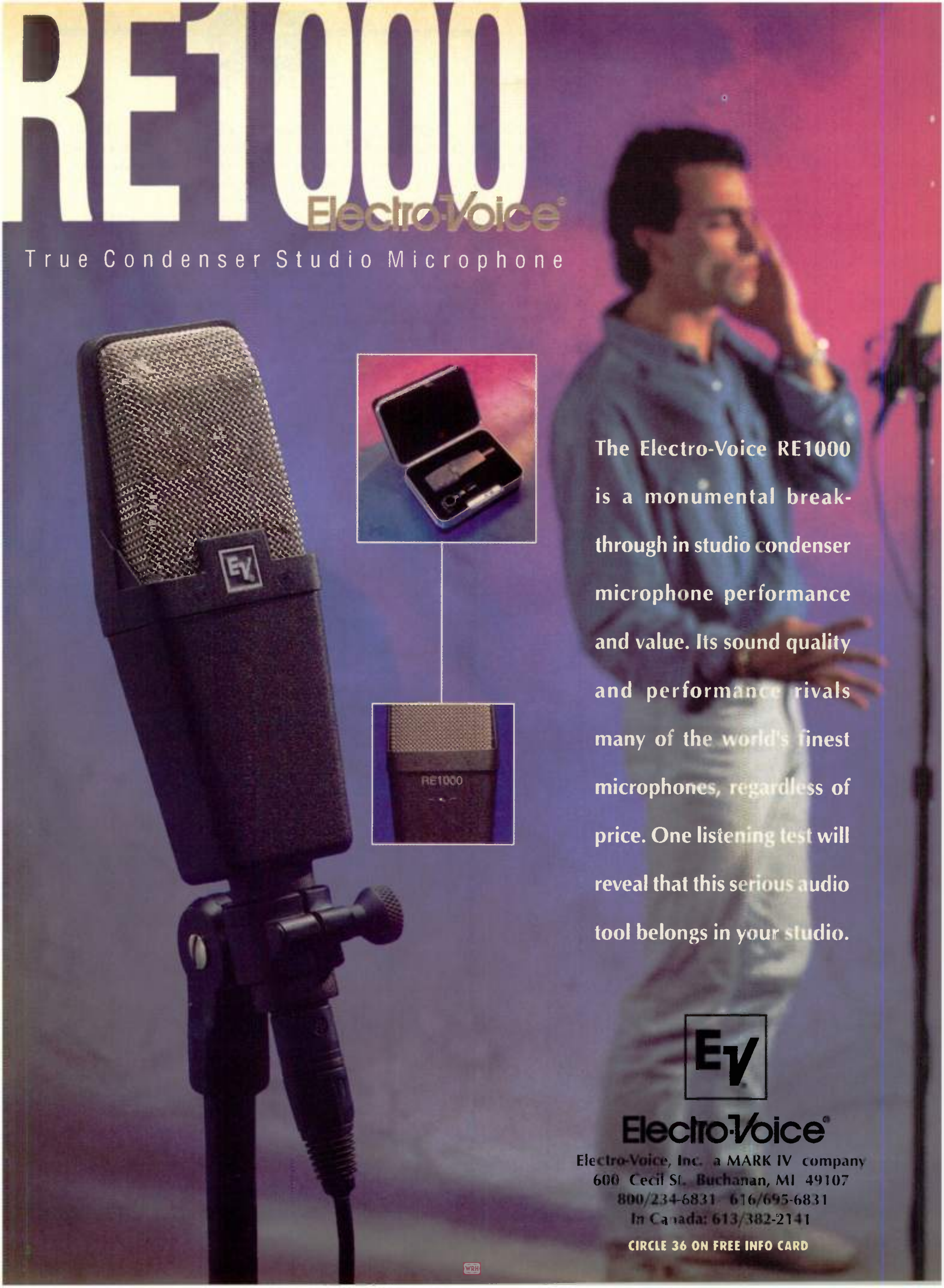
My reference CD for tuning a PA system is the title song from Sheila E.'s *Sex Cymbal*. The reason why I love this CD is because the kick drum and the vocals are very well compressed. I listen for the tightness of the low end — especially on the kick drum. Also, there is a lot of stereo panning going on, so that helps me make sure that the stereo imaging of the PA is right.

Two other favorites that I use to tune a PA is the new Annie Lennox CD, *Diva* and just about any song by Mary Chapin Carpenter. On the Annie Lennox CD, I listen to the clarity of the vocals, which is important in dealing with Aaron's high-pitched voice. The Mary Chapin Carpenter discs give me a really good idea of what the guitar, piano, and vocal should sound like through the PA.

BOB BUTLER

Presently mixing Brooks and Dunn; Bob has also manned faders for Waylon Jennings, Randy Travis, and the Oak Ridge Boys.

Before I tune the PA with music, I use pink noise and an analyzer to hear what the PA is doing. I don't EQ for flat response because when the PA is flat, it is usually so bright and brittle that it is offensive to the audience. I have a specific tonal character in mind that I want to hear with the pink noise. Then I will listen to some music. For the last three or four years I have been using two cuts in particular: "Bridge Of Sighs" from Louise Goffin's *Fifth Of July* and Joe Satriani's "Always With Me, Always With You" from *Surfing With The Alien*.



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EQ LIVE

I like the Louise Goffin cut because it is not a cluttered mix. A lot of well-recorded music doesn't translate well on a large PA because the mix is too busy. This cut has a lot of definition in the mid- and high-mid frequencies, which I need for getting the vocals right with Brooks and Dunn. It helps me pick out potential problems and make sure that any problems are with the room and not the source. The Satriani cut has a fuller, more extended bottom end and I use that primarily to make sure that my low end doesn't get swimmy. Familiarity with the song allows you to identify problems that exists with the system, as opposed to the sound source. I think I may have to change them soon, though, because I'm getting grief from my crew — they are getting a bit tired of them!

ROBERT SCOVILL

Presently mixing for Foreigner; Robert has also performed house mixing duties for Rush and Tom Petty.

First off, I must say that my favorite source for tuning the PA is the band. Something that I started doing on the Tom Petty tour was running ADATs and using an ADAT track per channel on the console. You insert an ADAT track pre-dynamics and pre-EQ and record the whole show. The next day when you walk in, you have the band sitting there on tape. You roll tape and it's as if the band is up there playing. You can work on any sound you want and really sort the sound out for the room. Not every console is going to allow you to do that just from the logistics of patching, but, to me, that is the most accurate way to do it.

When I'm EQing the PA, I think of it more as de-equalizing the room, and I'm not necessarily concerned with achieving flat response. For checking the frequency response, I use pink noise and an analyzer, but to check the energy curve, I'll use CD tracks. I have compiled a tuning tape on DAT with bits and pieces of songs that I need to hear from day to day. I have probably been using some of these songs for about 10 or 12 years. One is "Sixes and Sevens" from Robert Plant's *Shaken and Stirred*. It has some low synth-pedal pads that help me sort the low end, and the drums have a tight, natural sound. I also use a Bruce Hornsby song for the piano, guitar and vocal. If I can get that to sound right, then I know I have midrange really sorted out.

"Into The Fire" from Sarah McLachlan's *Solace* has a low end that is mind boggling. Some of the others that I use are Joe Cocker's "Unchain My Heart," and "King Of The Mountain" from Midnight Oil's *Diesel And Dust*. Last would be *Toy Matinee*, "Last Train Out" or really just about any of the songs from that disc. These are songs that I know inside and out and have instrumentation that helps me sort out the energy in the room.

In the end, though, I think the song or even the genre of music you use is irrelevant. The idea is to make whatever music you are doing translate the same from the PA to listener on a day-to-day basis.

M.L. PROCISE

Senior sound engineer at Showco Audio; M.L. has mixed acts such as ZZ Top, Michael Jackson, and Guns N' Roses.

Oftentimes, live engineers become very familiar with a certain piece of music, a passage that they can keenly identify the frequency response and sonic quality of and use as a reference when encountering different sound systems. They know how the music sounds through neutral listening platforms such as headphones or studio monitors and in controlled environments. If I subscribed to this methodology, I would take it another step further by knowing what the frequency

continued on page 110

Club Profile: The Catalyst

CLUB HOUSE: The cornerstone of the Catalyst sound system is the Ramsa WR-S52 console.

A LOOK AT A LARGE PLACE THAT MANAGES TO GET A SMALL SOUND

BY STEVE LA CERRA

If you're out on the road doing a club tour and make your way to the Santa Cruz area, there is a good chance that one of the stops will be the Catalyst on Pacific Avenue. With a capacity of 900 people, the Catalyst sounds smaller than it actually is. A high ceiling and a 15-foot deep, U-shaped balcony that runs along the side and rear walls of the rectangular 50- x 90-foot room give the Catalyst an open feeling with a lot of breathing room — even when it's packed.

Unlike many clubs or theaters that feature this type of balcony arrangement, you won't find the house mix position on the floor underneath the balcony or in the balcony itself opposite the stage. You will find the house console living towards the rear of the balcony section that runs along the right side of the room. This makes mixing the room a bit of a challenge, but certainly not an insurmountable task.

Visiting engineers who are working at the Catalyst for the first time without carrying production will find the house system well-equipped for mixing a vast majority of bands. House PA is piloted from behind a Ramsa WR-S52 console that accepts up to 48 microphone inputs from the stage (the snake is permanently installed behind the club walls and under the stage). Outboard gear at the house position includes a Yamaha SPX90, Roland SDE1000 delay, two dbx 160x compressors, two dbx 166A compressor/gates, and two Furman QN-44 noise gates (four channels each). A pair of Rane GE30 one-third octave graphic equalizers are also available, for use in tweaking the house EQ curve.

Unlike most clubs that have a bundle of insert cables laying behind the console for patching in the outboard gear, a TRS patchbay has been installed at the house position for easy interfacing of both the club's outboard processors, as well as any ancillary rack gear that you might have brought along with you. Catalyst house engineer Mike Sheehan is more than happy to help you get your toys patched into the system.

GENERAL CLUSTER

All speaker cabinets for the four-way house system are flown, but the arrangement at the Catalyst is a bit different from that of most clubs. In front of the center-stage area is a cabinet array consisting of two JBL folded-horn subwoofer enclosures (each contains two 15-inch JBL woofers) surrounding four three-way cabinets.

POSITION, POSITION, POSITION: The Catalyst FOH setup is beneath the right side of the balcony.

Each of the three-way enclosures holds two 15-inch JBL speakers, one Electro-Voice DH1A midrange driver, and two JBL compression horns. These cabinets are all front-loaded, trapezoidal enclosures, and are aimed at the floor area. In addition to this center cluster, there are two more clusters, each used as a "balcony fill." These fill clusters each consist of two trapezoidal cabinets (the same as used in the center cluster), but contain two 15-inch McCauley speakers, one Electro-Voice DH1A driver, and a single JBL tweeter. Sheehan notes that in light of the fact that the center cluster is aimed towards the floor, the balcony fills help to evenly distribute the mix to audience members sitting at tables in the left and right balcony areas. All speaker clusters are powered with Crown and QSC power amplifiers.

MONITORS BY THE BAY

Back in monitor land, a second Ramsa console (this one is a WR-840F) handles a total of eight mixes: six on the floor and two side-fill. Side-fill clusters are hung (helping to increase available floor space on the 28- x 18-foot stage) and are biamped with two 15-inch JBL speakers and a single 2-inch JBL 2445 driver. Sheehan has each of the monitors biamped using one side of a QSC MX 1500 on the low end and one side of a QSC MX 700 on the high end. Rane AC22 stereo two-way crossovers divide the spectrum at 1500 Hz.

Eight floor wedges are available (in case it's not already loud enough on stage); each wedge packs one 15-inch JBL and a 1-inch JBL 2425 driver attached to a 60 x 40 horn. There are also two biamped drum-fill enclosures each holding two 15-inch JBL

THE ROOM DOESN'T SOUND LIKE MUCH WHEN IT'S EMPTY. WHEN IT FILLS UP, THE BOTTOM END BECOMES A LOT TIGHTER. THE TOP END ALSO BECOMES MORE SMOOTH AND YOU CAN GET A MIX OUT OF THIS ROOM THAT IS (DARE I SAY) CD QUALITY.

woofers and a 60 x 40 horn on a JBL 2425 driver. For monitor EQ, the mix outputs of the Ramsa WR-840F are routed through Rane GE30's. Taking a cue from the house setup (pardon the pun), monitor land also has patchbay access: all group inserts, auxiliary outputs and matrix outputs are routed through the bays, easily accommodating extra processing gear.

The Catalyst has a pretty well-rounded offering of microphones including Shure Beta 57's and Beta 58's, and Shure SM56's, SM57's, SM58's, and SM91's (give those a spin in the kick drums), as well as AKG DT330's, Electro-Voice PL80 and 849's, and an assortment of direct boxes.

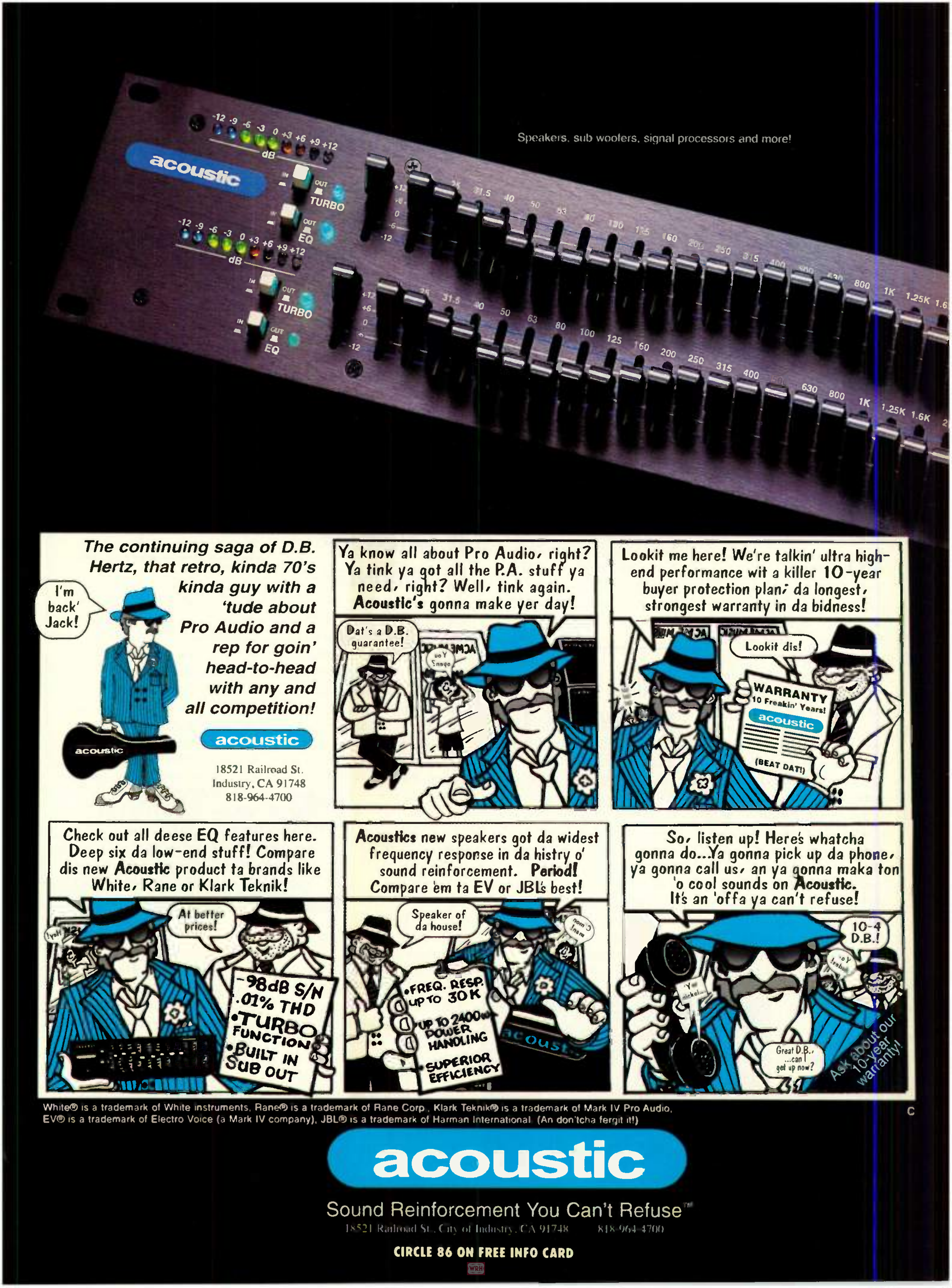
MIXING TIPS

Sheehan explains that when you are mixing this room you need to keep in mind that the left and right master faders of the Ramsa console are really used to feed the center cluster and the balcony fill — as opposed to left and right sides of the PA. Also, your mix position will be almost directly facing the right balcony fill, so there's a good chance that you will be running the center cluster fader a few dB hotter than the balcony fill fader (also, the center cluster has more area to cover down on the floor).

Keep in mind that less of the stage sound makes its way up to the mix position at the balcony area, so you need to check the balance between the vocals and the band — as well as between guitar solos and the band — downstairs on the floor. I found that if the solos sound too loud at the mix position, they are at a pretty good level down on the floor.

The room doesn't sound like much when it's empty, and, obviously, when it fills up the bottom end becomes a lot tighter. But it also seems that the top end becomes more smooth and you can get a mix out of this room that is (dare I say) CD quality.

Oh, and while you're there, try the Catalyst's in-house restaurant — you won't be disappointed. EQ



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Audio-Technica is now shipping new field-serviceable versions of its closed-back, dynamic ATH-M40 and ATH-D40 Precision Studiophones. Designated with the suffix "fs" (ATH-M40fs and ATH-D40fs), the new models allow elements, cables, and earpads to be replaced in the field if necessary. Both models utilize 40-millimeter drivers with neodymium magnets and copper-clad aluminum wire voice coils. Each pair is also equipped with an 11-foot cable terminated in a standard 1/4-inch connector. The ATH-M40fs features a flat, extended frequency response of 5 Hz to 28 kHz, a sensitivity of 100 dB, and an impedance of 60 ohms. The ATH-D40fs offers a frequency response of 20 Hz to 28 kHz, a sensitivity of 102 dB and an impedance of 66 ohms. Both models feature a maximum input power of 1600 mW at 1 kHz, and are offered at the manufacturer's professional net price of $175. For more information contact Audio-Technica U.S., Inc., 1221 Commerce Drive, Stow, OH 44224. Tel: 330-686-2600. Circle EQ free lit. #126.

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With more than 500 different instruments that can be changed by using a special algorithm for up to eight variations of each instrument, Clavia Digital Instruments' ddrum4 percussion controller features a number of improvements. Thanks to a new "ddrum Palette system," the user is able to change individual instruments with just one touch — without changing a complete "kit" setup. Users with Internet access will be able to expand their sound libraries free of charge by simply connecting the ddrum4 into a 'Net connected computer and downloading samples from Clavia's web site into the ddrum brain via MIDI. With ddrum's FlashRAM and SCSI option package, sounds can also be loaded and saved directly from CD-ROMs. For more details, contact ddrum's U.S. distributor, Armadillo Enterprises, at 923 McMullen Booth Road, Clearwater, FL 34619. Tel: 813-796-8868. Circle EQ free lit. #127.

ddrum4

Community CSX43-S2

A SENSE OF COMMUNITY

The two-way Community CSX43-S2 is a trapezoidal, bass-reflex enclosure housing a 15-inch ferrofluid-cooled woofer coupled to a 1-inch titanium diaphragm, high-frequency compression driver mounted on a 90- x 40-degree pattern control horn. The crossover between the two drivers is at 2 kHz. The CSX43-S2 has a sensitivity of 97 dB SPL at 1 W/1 m and it handles 200 W RMS and 500 W program power. Its frequency response is within ±5 dB from 45 Hz to 18 kHz and it has a nominal impedance of 8 ohms. The crossover incorporates Community's proprietary PowerSense DDP (Dynamic Drive Protection) circuit that monitors the power input to each driver to provide overcurrent and thermal protection. A stand socket is built into the carpeted enclosure bottom. For further information, contact Community Professional Loudspeakers, 333 East 5th Street, Chester, PA 19013-4511. Tel: 610-876-3400. Circle EQ free lit. #128.

TEKNIK ACHIEVEMENT

Klark Teknik's new DN 4000 is a dual-channel, user-programmable, 5-band parametric equalizer and delay line incorporating Klark Teknik's newly enhanced 20-bit converter design. Each channel has nine filters that consist of five fully parametric sections, high- and low-pass filters, and high- and low-frequency shelving filters, plus a delay line with a maximum delay time of 340 milliseconds. The LCD panel shows the actual frequency response of the equalization on both channels simultaneously. All parameters are accessed via dedicated selection switches under each display and adjusted by three rotary encoders for "Q," "Frequency," and "Level." A MIDI interface and remote interface port are provided

Klark Teknik DN4000

and up to 16 DN units can be controlled from one master. Price is $4600. For more details, contact Mark IV Professional Audio Group, 448 Post Road, Buchanan, MI 49107. Tel: 800-695-1010. Circle EQ free lit. #129.

MILES AHEAD

Housed in a compact two-rack-space chassis, the new Miles Technology MPR-450 power amplifier features six independent channels delivering total power in excess of 450 continuous watts. Each pair of channels is bridgeable for higher power with three, four, or five channels, while independent gain controls allow the level of each channel to be individually tailored. At 8 ohms, the MPR-450 delivers more than 60 watts of power per channel. At 4 ohms, it supplies more than 75 watts per channel. Bridged at 8 ohms, the MPR-450 provides more than 150 watts per channel. A heavy-duty R-Core power transformer helps to provide efficiency, while a proprietary circuit design, called PowerDirector, automatically supplies extra power to channels where it's needed. Price is $965. For further details, contact Miles Technology, 70 North St., Niles, MI 49120. Tel: 800-280-8572. Circle EQ free lit. #130.

Miles MPR-450

IN THE AIR

The 411 series UHF wireless microphone system from Azden features the all new 411UDR dual-conversion superheterodyne true-diversity receiver, the 411BT bodypack transmitter, and 411HT handheld microphone. The system is crystal-controlled, PLL-synthesized, and has 63 onboard user-selectable frequencies with a range of 794 MHz to approximately 806 MHz. The 411UDR is a true diversity receiver containing two separate receivers, each operating on the same frequency. It features a 1/4-inch and an XLR output jack with volume adjustment. Both transmitters are frequency-agile and transmit the 32.768 MHz tone necessary for the tone squelch circuitry in the receiver. The 411HT is powered with two AA batteries and will feature a charging station option that includes the NiCd battery. For more information, contact Azden Corp., 147 New Hyde Park Road, Franklin Square, NY 11010. Tel: 516-328-7500. Circle EQ free lit. #131.

Azden 411 Series

EQ LIVE

Sabine FBX-Solo Feedback Exterminator

SABINE HAS BEEN refining its popular FBX Feedback Exterminators™ for many years, and the latest versions include DSP-based filters and the choice of filter widths. Since one great advantage of DSP-based products is that the processing engine is physically small, Sabine has been able to really shrink the new FBX-Solo series of automatic feedback exterminators. These new processors are as small as a typical DI box but maintain much of the capability of their rack-mount predecessors.

The FBX principle is based on the nature of feedback — essentially an oscillation that typically creates a pure tone (at least until clipping occurs). Pure tones are relatively rare in music and speech program where harmonics form a major component of the sound. By analyzing the audio signal in search of pure tones that lack the harmonics of speech and most musical instruments, the FBX can detect feedback when it rises above the other audio signals. Once detected, the FBX determines the frequency of the ringing and then adjusts a filter to reduce that frequency. The reduction happens in steps so that only the necessary depth of filter is created, avoiding excessive equalization. A particular advantage of this method is that because the filters are very narrow (1/5 or 1/10 octave), they have less impact on the overall sound quality vs. the more common 1/3-octave equalizer. The filter notches remain until the unit is reset or (in the case of filters in the dynamic mode) a new feedback ring occurs and the filter is set to this new frequency and depth.

The FBX-Solo offers the opportunity to dedicate a Feedback Exterminator to a single microphone or instrument, leaving the rest of the mix unaffected by any feedback filtering. The SL-610 is specifically designed to be inserted into the line-level audio chain to provide feedback control exactly where you need it — a channel insert on the mixer, an auxiliary output feeding stage monitors, or directly on the output of a wireless microphone receiver. The SM-610 offers the potential to plug a microphone directly into the automatic filtering before it goes to the mixer. This is especially handy in situations where you don't have channel inserts or are working with a bare minimum of equipment. The SM-610 can also be used before a microphone is fed to the on-stage monitor and house mixers to provide both microphone preamplification and a line driver.

ROAD TEST

EQ LAB REPORT

MANUFACTURER: Sabine, Inc., 13301 Highway 441, Alachua, FL 32615-8544. Tel: 800-626-7394, 904-418-2000.

APPLICATIONS: Elimination of feedback in sound-reinforcement and monitoring systems.

SUMMARY: A miniature version of earlier FBX units that offers six automatic filters.

STRENGTHS: Very little setup required; choice of mic- or line-level versions; compact; inexpensive.

WEAKNESSES: No gain structure adjustment (line-level version) and less headroom than most EQs.

PRICE: $299.95, SL-610 line-level version; $349.95, SM-610 mic-level version; $379.95, SMG-610 guitar version (not reviewed).

EQ FREE LIT. #: 132

BY WADE MCGREGOR

Are there flies inside your speakers, or do you need a Furman Iso-Patch™?

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Furman IP-8 Isolated Patch Bay

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Setup of the units is straightforward. Insert the unit into the signal chain; increase the gain until feedback occurs (a filter is then automatically set); add gain until feedback reoccurs (another filter is set); and continue until all the filters are activated or sufficient gain-before-feedback is achieved. Once the fixed filters (default of four but user-selectable) have been set, a front-panel button marked "Lock Fixed" prevents these filters from changing frequency or depth. The remaining filters (total of six) will track any further feedback in a dynamic mode, ready for the inevitable changes that occur during a live event.

The FBX-Solo has fewer filters (six instead of nine) than the FBX-901 and FBX-1802 [see *EQ*, 12/94] and they also have less headroom (+11 dBV RMS) than the more expensive units. They do, though, share the same high-speed filter setting algorithms as the more sophisticated units and have a dynamic range of at least 92 dB.

Power is provided by a wall-wart style unit that connects to the rear of the unit with a solid Molex-type connector. The line-level SL-610 uses three-conductor 1/4-inch-phone jacks for the unbalanced I/O, while the microphone-level SM-610 uses a Neutrik Combo connector (XLR and 1/4-inch phone jack) for the balanced input and a 1/4-inch phone jack for the unbalanced output. Both units include a rear-panel switch to select 1/5-octave (speech) filters or 1/10-octave (music) filters. Up to six FBX-Solo units can be installed in an optional rack tray that occupies a single rack space and each unit measures 2.78" x 1.65" x 5.5" (6.95 x 4.13 x 13.75 cm), not including connectors or the front-panel switches and microphone gain knob (SM-610 only).

In spite of its small size, the FBX-Solo still offers users the ability to select the number of fixed filters (the balance being allocated dynamically) and the level of the internal noise gate (in five steps, including Off) by depressing buttons during power-up. The front panel includes Bypass, Reset, and Lock Fixed buttons and LED indication of level (in four steps including clip) and number of filters currently set, in addition to the status of the Bypass and Filter Lock buttons.

If you have the need for a feedback filter in your sound-reinforcement rig, but thought you couldn't afford an FBX, it's time to reconsider. The same caveats for FBX units apply: the system will briefly feedback while filters are brought into action and some instruments (such as flutes and organs) can accidentally activate the filters. However, it is hard to find a better way to deal with filtering for feedback when all other measures, such as loudspeaker and microphone placement, fail to achieve the required level of gain before feedback. EQ

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CIRCLE 39 ON FREE INFO CARD

IN REVIEW

DigiTech VTP-1 Preamp/EQ

EQ LAB REPORT

MANUFACTURER: DigiTech, 8760 S. Sandy Parkway, Sandy, UT 84070. Tel: 801-566-8800.

APPLICATION: Preamplify and "warm up" mic- and line-level signals for interfacing with both analog and digital gear.

SUMMARY: Studios based around budget digital gear (e.g., 8-track digital recorders and DATs) will find the VTP-1 a great way to go "digital direct," as well as replace the mic preamps in standard analog consoles.

STRENGTHS: AES/EBU and S/PDIF formats; 18-bit digital outs; clean-sounding analog tube preamp; musically useful EQ works well for general tone-shaping; includes both analog and clip metering; coolest looking box DigiTech has made in a long time.

WEAKNESSES: No bandwidth control for the two midrange EQs; 20k input impedance precludes using as a direct box with nonactive electric guitar/bass; supports only 44.1 and 48 kHz sample rates.

PRICE: $999.95 **EQ FREE LIT. #:** 133

BY CRAIG ANDERTON

THE DIGITECH VTP-1 immediately caught my attention with two features: lit analog VU meters straight out of the '50s, and 18-bit digital outputs that cater to the '90s. That nicely encapsulates this stereo preamp/EQ/A-D converter — the VTP-1 applies the best from a variety of technologies to that age-old question, "Now that we have a signal, how do we get it to the recorder?"

GETTING THERE FROM HERE

This 2U box has plenty of inputs and outputs. For analog, each channel has three ins: balanced XLR line, balanced XLR mic, and TRS (balanced or unbalanced) 1/4-inch phone. (If you prefer transformer inputs, DigiTech will provide you with technical info, including a parts list, at no charge; you can then have a qualified technician perform the modification.) Outputs are XLR and TRS 1/4-inch phone. Each channel also has a post-EQ, mono effects loop (+4 nominal) with TRS 1/4-inch jacks. In a pinch you can use the send as a post-preamp direct out, and the return as an input that bypasses the preamp (and disconnects the preamp from the signal chain).

For digital, there are both AES/EBU XLR and S/PDIF RCA connectors. A front-panel switch selects the format, as well as the sample rate (48 or 44.1 kHz); channel 1 is digital left and channel 2 is digital right. The digital outs allow going "digital direct" into anything from DATs to sound cards. Just remember that the VTP-1 wants to provide the master digital clock, so set the receiver for external digital sync. Also note that if you want to use a recorder's extended play (32 kHz) mode or go direct into sound cards at 22.050 or 11.025 kHz sampling rates for multimedia applications, these rates aren't supported.

UP FRONT (PANEL)

Both channels have the same controls. There are knobs for preamp gain (up to 66 dB mic, 26 dB line) and post-gain trim. The EQ has in/out and lo-cut (75 Hz) switches, as well as controls for low shelf (80 Hz, ±15 dB), hi shelf (12 kHz, ±15 dB), and two semiparametric mid sections, each with a frequency and cut/boost control (50 Hz–3.2 kHz and 500 Hz to 18 kHz, ±15 dB).

Switches for each channel include line/mic input select, phase normal/invert, 20 dB mic pad in/out, and 48 V phantom power on/off. The final control is the on-off switch.

For metering, the lit analog VU meter (which indicates average signal levels) gives the whole unit an exceptionally retro ambience. But as a concession to us modern types who like sounds with lots of percussive transients, there's a supplementary LED clip indicator (actually, it lights at 6 dB below clipping).

THE FINAL ANALYSIS

Like many contemporary processors, the VTP-1 mixes tube and solid-state technology — a 12AX7-based class A dual preamp and op amp-based EQ. The VTP-1 seems intended to combine a bit of warming with clean and very quiet amplification, rather than being optimized for the overdrive effects for which tubes are often used.

continued on page 110

DIGITECH VTP-1

continued from page 108

This is definitely more a preamp than a "processor." The EQ is excellent for general tone-shaping, but the lack of resonance (bandwidth) controls limits the ability to solve specific problems. Then again, if you want to use compression or a parametric, there's always the effects loop. One tip: to add some of that "air" that's so popular these days to "open up" a vocal, turn the upper midrange frequency control to 18 kHz, and add a *slight* boost.

Distortion characteristics are quite pleasing, even if you get a little carried away and start exercising the clip LED. Electronic sound sources such as drums and keyboards benefit from slight clipping in the VTP-1 by picking up a bit more "punch."

There's a lot of competition in tube mic preamps. While the VTP-1 can certainly hold its own in any shootout, digital interfacing is where it shines — its preamp tops the preamps used in the average console, and the 18-bit A/D converters are a step past the A/D converters used in budget digital recording devices. Take this box, two good mics, and a DAT to a piano concert, and you're set for high-quality live recording. Cutting vocals directly to digital seems to add a kind of presence that's hard to describe. (Incidentally, due to the 20 kohms input impedance, the VTP-1 is not really suitable as a direct box for guitar or bass unless they have active pickups; but considering that DigiTech also makes guitar processors, maybe their next generation will sport digital outs too.)

Making an affordable tube preamp with digital outs may seem like an obvious step, but DigiTech got there first, and has set the standard for others to follow. EQ

REFERENCE CDS

continued from page 94

response looks like via the input of a real-time analyzer of a particular track and then comparing the known response to that of the sound system. This thumbnail approach can expose the deficiencies of inadequate sound systems and afford the engineer the opportunity to attempt to "fix it" through system equalization. I don't subscribe to this.

What you are going to hear from a CD is not what you are going to hear from a live band. A CD through its system of processing, compression, and mastering is not representative of the transients that you encounter with the live band and avalanche of multiple transducers the sound system presents versus the control room listening speakers.

Although I love to listen to my favorite music over the sound system (in the past I have used *Faith Hope Love* by King's X for its wide frequency response), I think that you are much better off using a vocal mic, talking into it, and exciting the resonant frequencies that way, because you can use the same tools that you are using in the live situation: compressors, equalizers, and mics.

Pink noise is another way of exciting the room and it is probably a better tool to measure the reverb time in a room (especially one bandwidth at a time) than a CD. I use pink noise minimally, almost as more of a diagnostic to check the input and output of the system from left to right and bandwidth to bandwidth. By using a fairly slow integration time — say one or two seconds — you can see down to the dB if there's something wrong with the highs on the right side or the left or whatever. This really comes in handy when you are an engineer traveling from club to club, using a different system every day. It is the only thing you have as reference. EQ

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Horizon Music Upgrade ADAT I/O Cards

MANUFACTURER: Horizon Music, Inc., 230 N. Spring Street, Cape Girardeau, MO 63701. Tel: 800-821-3806.

APPLICATION: Analog input and output card upgrade kit for the original Alesis ADAT.

SUMMARY: These cards offer increased performance in terms of bandwidth and the ability to drive long cables.

STRENGTHS: High-quality, low-noise components; extremely easy to install.

WEAKNESSES: None of a technical nature.

PRICE: $349.95 **EQ FREE LIT. #:** 139

BY EDDIE CILETTI

HORIZON MUSIC, INC. is perhaps best known for its specialized line of audio cables, harnesses and interface devices (e.g., direct boxes). Recently, though, it has entered a different market by issuing the Horizon I/O card upgrade set for the "Original Formula" (OF) ADAT.

WHAT ARE THESE CARDS?

On the rear of the ADAT are two rows of 1/4-inch jacks, input at the top and output below. Each row requires a printed circuit board (PCB) on which are planted your garden variety TL084 quad op amps. The Horizon Music upgrade kit replaces both PCBs. In this high-speed garden are video op amps that do audio in their sleep. Their greatly extended frequency response is fertilized by low-noise, 1 percent resistors. While the ADAT chips are soldered, Horizon chose machined-pin sockets to provide the option of crop rotation [replacement].

The stock ADAT has unbalanced, 1/4-inch inputs and outputs that accept a nominal –10 dBv signal. (The multipin connector accepts balanced +4 dBu signals.) The Horizon cards have balanced inputs on both connectors; both outputs are high-level, +4 dBu nominal, and remain balanced and unbalanced at the multipin and 1/4-inch jacks, respectively.

While Horizon's features list places bandwidth and slew rate at the top (see table 1), the really important features are actually points 3, 5, and 7. (Horizon should de-emphasize the degree of importance of the bandwidth and slew rate specs in their promotional material. The key features are the balanced 1/4-inch inputs (with adjustable gain) and very low impedance outputs.) You may, for example, need the convenience of being able to connect a balanced, high-level signal to the standard 1/4-inch jacks. The Alesis input circuit design actually mixed the low-level (1/4-inch) and high-level (multipin) signals together. The Horizon circuitry routes the signal from the multipin through the gold-plated normals of the 1/4-inch connector, a more direct signal path that also minimizes noise.

If your machine must be located far from the console and/or patchbay, the low impedance of the Horizon output card will not sweat the long cable run, whereas the stock ADAT would prefer short,

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TABLE 1

1. DC (0 Hz) — 70 MHz input bandwidth
2. Up to 450 V/μS slew rate (true 3-dimensional sound)
3. Balanced input on both 1/4-inch and ELCO connectors (1)
4. Accepts mic- and line-level signals (up to 40 dB gain)
5. High-current output capability to drive long cable runs (2)
6. MAPS technology (patent pending)
7. Alesis ADAT dealer installation (optional) takes 10 minutes (3)

low-capacitance cables. Sexy, esoteric cable is the solution for gear that cannot be modified, but Horizon's approach is the real deal.

INSTALLATION

The installation is very user friendly and as fast as advertised. Be sure to observe the usual precautions — that is, unplug the unit, discharge yourself before touching the equipment, pay close attention to cable orientation, take notes if necessary, and be gentle.

Once the top cover is removed, the circuit cards can be easily removed by unscrewing the nuts around the jacks. Both are plastic, so do not use excessive force in either direction. Each card also has two ribbon connectors: one for power and the A/D/A interface; the other for the ELCO/EDAC multipin connector.

There are three "gotchas" if you don't pay attention.

1. Unlike the Alesis input card, Horizon's input card mounts the ribbon and 1/4-inch connectors on the underside of the printed circuit board (PCB). However...

2. The output card is mounted component side up.* (Don't give in to the temptation to flip the output cards.)

3. The Ribbon from the multipin connector to the Horizon input card must be flipped.

*The stock ADAT cards are both mounted component-side up.

I am not about to turn this review into a "which sounds better" piece. That kind of thing can make you blind; just ask Alanis Morissette. However, transient-rich percussion and stringed acoustic instruments will likely yield the most significant differences. I choose to emphasize the features that make the most practical sense: balanced inputs and low-impedance outputs.

Since there are filters on both sides of the A to D to A process, one could argue that the 70 MHz bandwidth, while impressive, might not always yield a significant increase in perceived performance. Distortion-rich instruments such as heavy metal guitar might actually tickle the filters to resonate in an uncomplimentary way. Since, even at the 48 kHz sample rate, useful bandwidth is limited to 22 kHz, there is also a limit to how much of the extended bandwidth will actually get through. Assuming some improvement, directional information from stereo miking techniques would be more accurately captured. Nuff said.

SURPRISE VIEW

When the ADAT is in Source/Input mode, there are two options: (1) the signal can be routed through the analog I/O cards only; (2) the signal can be routed through the digital audio converters (DACs). To toggle between analog and digital throughput, press Set Locate and All Input. The display will indicate each mode. Input mode permitted me to view both I/O cards on their own without digital artifacts. Most other digital tape decks — when not in Play — pass the signal from analog to digital and back again. It is for this reason that many users mix through their DAT machines (in Record/Pause) and that digital-to-analog converter sales are on the rise.

Another thing I learned about the ADAT is that there is a one nanoFarad capacitor across the 1/4-inch inputs. This cap loaded down my signal generator, which doesn't say much for either device, but I can tell you that the load presented by the Horizon's input card did almost nothing — a high impedance at all audio frequencies. The balanced inputs offer hum- and buzz-canceling properties.

As mentioned earlier, the low-output impedance circuitry alone make the Horizon upgrade worth investigating. From no load to a 600-ohm load, the signal at the ADAT's 1/4-inch output jack drops 5.25 dB, while the Horizon gives up only 1.25 dB. (The ADAT circuitry has a 510-ohm build-out resistor to protect the amplifier from short circuits.)

Note: Had the load also been a 510-ohm resistor, the drop would have been an even 6 dB. In terms of voltage, for example, 0 dBV is 1 volt RMS and a 6-dB drop would tip the scales at .5 volts RMS.

It seems a shame to "waste" all that bandwidth on a signal that must be digitized. These I/O cards, however, do stand on their own, not only as an ADAT upgrade, but also as an experimenter's kit. They also carry a 3-year warranty. Maybe I'll hot rod my keyboard mixer with 'em next! EQ

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CIRCLE 83 ON FREE INFO CARD

Bellari ADB3 Stereo Tube Direct Box

EQ LAB REPORT

MANUFACTURER: Bellari (division of Rolls), 5143 S. Main St., Salt Lake City, UT 84107. Tel: 801-263-9053.

APPLICATION: Interface guitars and other high-impedance output instruments with balanced, high-level, low-impedance XLR inputs, or "warm up" higher-level outputs.

SUMMARY: Does the job at a fair price.

STRENGTHS: Comparatively inexpensive; gain/pad/ground lift options; clean sound; good specs; steel chassis.

WEAKNESSES: Gain not continuously variable; with guitar and other low-level instruments, lacks sufficient punch for +4 inputs that want to be hit hard.

PRICE: $200 **EQ FREE LIT. #:** 135

BY CRAIG ANDERTON

IN CASE YOU'VE wondered why tube gear has to be so expensive, apparently the Bellari folks wondered too, and couldn't come up with a good reason. The result is the ADB3, a bare-bones tube direct box that costs $100 a channel. While it's not going to blow the socks off top-of-the-line models, it sounds better than you'd expect for the price.

INS/OUTS/SWITCHES

Inputs and outputs are basic: 1/4-inch unbalanced ins (extra credit for including paralleled jacks on the front and real panel) and rear-panel, transformer-fed XLR balanced outs (with pin 2 hot) capable of driving 600-ohm lines. Considering that direct boxes are often used with guitar and bass, stereo may seem redundant; however, this extends the usefulness to other applications (Chapman Stick players, take note). Besides, since the 7025 has two tubes in one bottle anyway, there's little reason not to include the extra parts and make the box twice as useful.

Each channel has a ground-lift switch, pad switch (0/-20 dB), and attenuation/gain switch (–20 dB/+20 dB). The input impedance is a righteous 4 MΩ so there will be no loading of guitar signals, and you can drive long cables (such as snakes) with impunity. The impedance drops to 100k with the pad switch in, but, of course, you would typically use the pad with lower-impedance sources (e.g., tape deck outputs). Unlike guitars, these are not affected by feeding lower input impedances.

With gain set to –20 dB, response is essentially flat from 20 Hz to 20 kHz. At +20 dB gain, the response falls off a tiny bit at the frequency response extremes. Noise is very low, even with the gain switched in. All in all, the specs are fine.

USING IT

Although the lack of a continuously variable level control reduces flexibility compared to something like a mic preamp, proper use of the pad and gain switches can handle many situations. For example, to use the ADB3 with a "hot" synth to warm up the sound a bit, simply switch the pad in and set the gain to –20. If the synth output is more anemic, then punch in the gain, switch out the pad, or both.

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CIRCLE 45 ON FREE INFO CARD

IN REVIEW

TL Audio EQ-2 Valve Preamp/ EQ

EQ LAB REPORT

MANUFACTURER: TL Audio, Letchworth, SG6 1AN, UK. Distributed in North America by Sascom Marketing Group, 34 Nelson Street, Oakville, Ontario, Canada L6L 3H6. Tel: 905-469-8080. WWW: www.sascom.com.

APPLICATION: Project and commercial studios.

SUMMARY: Dual-channel, stereo linkable, 4-band parametric, tube EQ with two sweepable high- and low-pass filters with mic, line, and instrument inputs.

STRENGTHS: Definitely tube; stacks of control; fast to use; fat; value for money.

WEAKNESSES: Twin overlapping band 4-band arrangement; curious LED metering.

PRICE: $2395

EQ FREE LIT. #: 136

BY ZENON SCHOEPE

HAVING EXPLORED nearly all the other applications of "affordable" tube processing, TL Audio has now attempted the holy grail of outboard —the parametric tube EQ.

This dual-channel, four-band parametric device can be linked for stereo, has two sweepable high- and low-pass filters, and employs six ECC83/12AX7 twin triodes divided as one tube stage per EQ filter and a two-stage output driver per channel. In typical TL Audio fashion, you also get two mic preamps with phantom power thrown in, front-panel instrument-level auxiliary inputs, plus XLR and unbalanced jack I/O, –10/+4 operation, and an insert on each channel.

The EQ-2 is a fairly rugged-looking 3U rackmount that you could probably take on the road without too much fear. Its controls are arranged in two identical rows to reflect the two separate channels. Each channel has pots-a-plenty, frequency, boost, and fully variable Q on each of the four bands, plus additional controls for input and output and the swept high- and low-pass filters.

When running stereo, control of the four parametric bands is handed to Channel A's pots, although input gains, output masters, and filters remain independent. Every band and each filter can be individually bypassed, and there's also a global bypass switch.

Input gain is variable from –20 to +30 dB for line and +10 dB to +60 dB for the mic inputs, and there is ±15 dB of level available at the end of the chain on the master output pot, which, regrettably, you don't always find on outboard EQs. The two filters are 12 dB/octave and sweep through 30 Hz to 1 kHz and 1 kHz to 25 kHz.

While each channel has four fully parametric bands, these are actually made up of two pairs of identical bands. Each offers ±15 dB of gain with fully variable Qs from 0.5 to 5. The LF and LM each cover 30 Hz to 3 kHz, while the HM and HF cover the span from 1 kHz to 20 kHz. This may seem like a curious arrangement to some, although users of older Amek consoles will recognize the principle and potential benefits. This arrangement, however, is not without some downside. It can be extremely useful to have two pairs of fully overlapping bands, which also partially overlap as pairs, to really concentrate the EQ (for example, in the voice band or on particularly difficult areas), but it does mean that you forfeit the more traditional and logical frequency spacing of a more usual 4-band equalizer.

Such a forfeit can translate into a bit of a culture shock operationally, and while you may choose to restrict the band ranges yourself, it never quite adds up to

the sort of instinctive and demarcated frequency areas that you encounter with a more traditional configuration. I know that some swear by this sort of arrangement, but I think the fact that it hasn't been adopted by more EQ manufacturers speaks for itself. It's the sort of thing you have to try and decide on personally.

IN USE

The tube effect is most apparent when whamming up just one band, and this produces a texture that is different from "tubeing" a signal completely across its spectrum. You can really misbehave and load-up full-frequency, arriving at quite a marvelous bit of processing that is continuously variable and adjustable. That is not to say, however, that the EQ-2 cannot be subtle: It's capable of a very classy, clean-with-a-hint-of-tube tone that characterizes most of the other TL Audio units.

For classic-type EQ'ing, broad band is undoubtedly best; although with the sharpness of bandwidth available and the overlapping bands, you can really pile the EQ into a section and stack settings for extreme phase shifts. Taking line- or, particularly, instrument-level signals through this works wonders for electric guitar and bass, and, of course, you can overdrive mildly as well.

This unit has TL Audio's combined peak/tube compression LED, which I still find hard to make sense of. The LED is a variable intensity type that starts to glow when a signal's inside the limits, but glows most heartily when the tubes are really working. Consequently, it tells what the tubes are doing rather than inform you about input level. I would prefer two separate LEDs to perform these two different functions.

The EQ-2 is best when used as a sweetening EQ across mixes, vocals, and individual sources. It's an excellent tracking EQ, very easy to get results with, and, because it has tubes, it's also a little more forgiving and interesting to use. However, it's not really decisive enough for ultra-fine corrective measures, which, except for fine-tune equalizing, is not such a bad thing anyway.

SUMMARY

The value for money aspect of this unit is bumped up by the inclusion of good quality mic preamps. These are decidedly smoother and fatter sounding than you're likely to encounter on a regular 8-bus board. The preamps won't make a U47 out of an SM57, but half-decent condensers can breathe a little easier and sound as good as they're ever going to. The fact that you can connect pretty much anything to the EQ-2 through the multitude of sockets is another great boon. Need I remind you that you're looking at two channels of tube EQ for a little more than a single channel of Super-EQ? EQ

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Actual Price

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ALESIS

IN REVIEW

Night Technologies PreQ³ EQ-Preamp

EQ LAB REPORT

MANUFACTURER: Night Technologies, International, 1680 West 820 North, Provo, UT 84601. Tel: 801-375-9288.

APPLICATION: Project and commercial studios, post, live sound.

SUMMARY: High-quality mic/line preamp with unique Air Band equalization.

STRENGTHS: Air Band EQ is remarkable; superb manipulation of mic signals.

WEAKNESSES: You have to use one to know what it's all about.

PRICE: $1595, 2-channel; $2595, 4-channel; $1295 for upgrade from 2- to 4-channel.

EQ FREE LIT. #: 137

BY ZENON SCHOEPE

ONE OF THE coolest thing about the Night Technologies PreQ³ is that it's modular. A single 1U can be filled with four preamps or partially loaded, as on the review sample, with just two channels, allowing additional channels to be added later straight into the precut and legended metal work.

The PreQ³ is more than just a mic preamp as it incorporates Night Technologies' Air Band shelf-type equalizer, which can also be applied to line-level inputs.

Connections are via balanced XLRs. The unit is pretty in blue and exceptionally well-made. Each channel has a cluster of small switches that select mic and line inputs, phantom power, phase reverse, a high-pass filter and a 20 dB pad. The pad and an additional Gain switch, which adds 20 dB to mic signals, makes for very versatile gain control when coupled to the input gain pot and a peak LED.

GET SOME AIR

The Air Band EQ section has two rotary controllers: one switches between 2.5 kHz, 5 kHz, 10 kHz, 20 kHz, and 40 kHz frequencies; the other adjusts boost at these turnover points. The maximum boost at the stated frequencies is 12.5 dB, but as these are effectively shelving-type filters, the effect and scope for adjustment is profound. The Air Band can be bypassed via a switch with an associated LED. If I have a criticism, it is that there is no power indicator. When running the box on line-level inputs, it's quite possible to have none of the LEDs illuminated and you can't tell if the box is actually on.

IN USE

A unique quality of the PreQ³ is that mic signals hit the Air Band equalizer before they hit the unit's gain circuitry. This means you can alter a mic's character very early on in the sound chain. That might not sound like a big deal, but it is significant, and it's also possible because the box is supremely silent.

Night Technologies claims this approach allows a cheap mic to be transformed in performance to approximate that of one costing very much more, but I've got to disagree. What this arrangement undoubtedly does is do wonders for wholesome top-end extension and definition on just about any mic you'd care to plug in. What it doesn't do is bring the fat and creamy character of a large capsule Teutonic classic to a standard condenser. That's not the end of the world, and the resultant sonic quality is still quite superb.

Adding Air Band to a mic in this manner is completely different from attempting to EQ it in the traditional way.

Perhaps it shouldn't be, but it is. Part of the difference has to do with the smooth and gentle nature of Air Band lift — five well-spaced frequencies and the variable boost combine to give endless permutations.

You can add broad presence that transforms the midrange on even a rudimentary condenser and can even give the complexion of a condenser to a decent dynamic simply because it gives you more of what is there already. Mics sound bigger and fuller. Higher frequencies, like 10 kHz, liven and open up the source. The 20 kHz and 40 kHz settings, while less obvious, add what I can only describe as "air."

The 40 kHz frequency would seem important only to bats, but because the band slope is so extended, at maximum gain it does impact further down the audio spectrum (albeit subtly and at the limits of my tired old ears). It's quite unlike anything I've heard before and, while I would stress that it's not blatant, if you leave it in on a program for any length of time, you miss it when you switch it out. One of the advantages of such high frequencies is that they are acting above the obvious noise band of sources or other gear.

SUMMARY

The PreQ^3's mic preamps are good, solid, and fast with a resounding bottom end. I reckon the real strength of this unit is in high-quality direct recording. The gain boost is particularly useful when trying to crank up on low-level signals such as room mics or distance sources.

The equalization offered with this device is truly transparent and serves as a very immediate and clever way of changing a mic's character. Plus, you don't feel bad about using it because you're only enhancing, and the resulting signal still lends itself well to any additional EQ'ing. Line inputs can also benefit from the processing, and I'm amazed at just how much you can change a sound without it sounding as if it's been EQ'ed.

The Night Technologies, Inc. PreQ^3 is a difficult unit to describe. If what I've said has caught your interest, then get to play with one and see what the fuss is about. EQ

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CIRCLE 224 ON FREE INFO CARD

Symetrix 620 A/ D Converter

MANUFACTURER: Symetrix, 14926 35th Avenue W, Lynnwood, WA 98037-2303. Tel: 206-787-3222.

APPLICATION: Twenty-bit analog-to-digital converter for studio use.

SUMMARY: Versatile A/D converter that has options making it good for straight audio or multimedia.

STRENGTHS: Good tonal sounds; smooth conversion; enhanced word size and noise shaping abilities.

WEAKNESSES: Eight-bit, 22.05 kHz isn't for everybody

PRICE: $995

EQ FREE LIT. #: 138

BY ROGER NICHOLS

SO, YOU'RE INTERESTED in 20-bit analog-to-digital converters, Eh? Well, you've come to the right place. Today we are going to talk about the Symetrix 620 20-bit A/D converter.

At first glance, you will notice that, for an A/D converter, the 620's front panel has more than its share of controls. Starting from the left of the front panel, you will find a pair of input level controls and a switch for selecting a fixed input level. This last feature is nice for times when you want to temporarily alter levels without disturbing your preset record levels. Next, we find a surprise: a switch labeled "Input Select" that switches between analog and digital inputs. It turns out that the 620 is more than just an analog-to-digital converter. The next selector is for sample rate. The choices are 48 kHz, 44.1 kHz, 32 kHz, and the never-before-seen 22.05 kHz, which we will discuss later. Output word size is selectable with the next control. Choices here are 20-bit (20), dithered 16-bit (D16), noise-shaped 16-bit (NS16), dithered 8-bit (D8), and noise-shaped 8-bit (NS8). Finally, we find an output Mute button and the left and right channel headroom meters.

The rear of the 620 is, of course, populated with the input and output connections. Analog inputs can accommodate either balanced 1/4-inch jacks or XLRs. The digital input consists of two connectors — an XLR for AES format and an RCA jack for S/PDIF. There is a push-button switch to select the desired input. The digital output also allows for S/PDIF or AES connections with a switch. In this case, both outputs are always hot. The switch selects the output impedance and digital data format that goes to both connectors. The only thing left on the rear panel is the AC connection via standard IEC electrical cord.

SO WHAT DO I DO WITH IT?

The 620 has three uses. First, as an external 16-bit converter for 16-bit recorders such as DAT machines, 8-track digital machines, and hard-disk recording and editing systems. Second, 20-bit mode allows high-resolution conversion for systems that allow for 20-bit data storage, or as a 20-bit feed digitally into a high-resolution DSP processor. Third, the digital input allows you to use the 620's DSP for word-size conversion and sample-rate conversion (44.1 kHz to 22.05 kHz) for multimedia audio uses.

Since the 620 is primarily an A/D converter, I thought that the first thing to do would be to listen to how it sounds. Initially, I listened to the 620 in 20-bit mode. I used my Apogee 20-bit D/A converters for the conversion back to analog. I have not yet mastered the art

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of listening directly to the digital audio bit stream. The tonal quality of the Symetrix 620 was superb. I listened to low-level audio and then goosed (very technical term) up the gain in the digital domain so I could hear the way in which the 620 quantizes low-level signals. The analog-to-digital conversion was very smooth. With state-of-the-art Delta/Sigma conversion and a Total Harmonic Distortion + Noise figure of –104 dB, you would expect nothing less.

In 16-bit mode, you can hear the increase in granulation of the low-level signals, but the 620 continued to perform well even when operating with crutches. It is interesting to change modes between 16 dither and 16 noise shape. It depends on the content of the music as to which one is the best to use. If you want the least distortion but can accept a higher noise floor, then dithering is the way to go. If the noise floor is the most important parameter, then noise shaping wins out.

Then there is 8-bit mode. Imagine how much a CD would store if all of the audio was 8 bits. And they would store even more if the sample rate was 22.05 kHz. The reason for the 8-bit mode is that it lets you hear what audio is going to sound like when you hear it on a CD-ROM. Finally you can hear the results when you mix, and then compensate for the lack of fidelity. In the "good old days," a producer would make a separate mix for the signal to compensate for the reduced quality over the radio. Now you can hear how bad your mix will sound, and you can turn up the solo, or the vocal, or whatever you need to turn up to get your message across.

As far as the 22.05 kHz sample rate goes, I have not been able to find a hard-disk system that will slow down far enough to allow me to directly input the 22.05 kHz signal. The Apogee converters would not track that low either, so I can't tell you what the 8-bit, 22.05 kHz sounds like. But, you know what? I don't care what it sounds like. If I ever mix to 8-bit, 22.05 kHz, please have me shot. [*Symetrix mentioned that the 22.05 kHz sample-rate function was put in based on requests from the multimedia sector. The rate "allows them to get a better than 8-bit sound by dithering into lower resolution recording programs — something like the Mac Sound Manager. The resulting audio winds up on CD-ROM games or sound bites." —Ed.*]

CONCLUSION

The Symetrix 620 A/D converter performs well and supplies useful features you won't be able to find elsewhere. In the "Bang For The Buck" ratings I would give the 620 a very high rating. If you don't have a 20-bit converter yet, then this would be a good place to start. If you already have a 20-bit converter, the added word size and noise shaping features of the 620 may make it worth the purchase anyway. EQ

AD INDEX

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

Desoldering Techniques

Getting over separation anxiety

BY MARK DE MARTINI AND EDDIE CILETTI

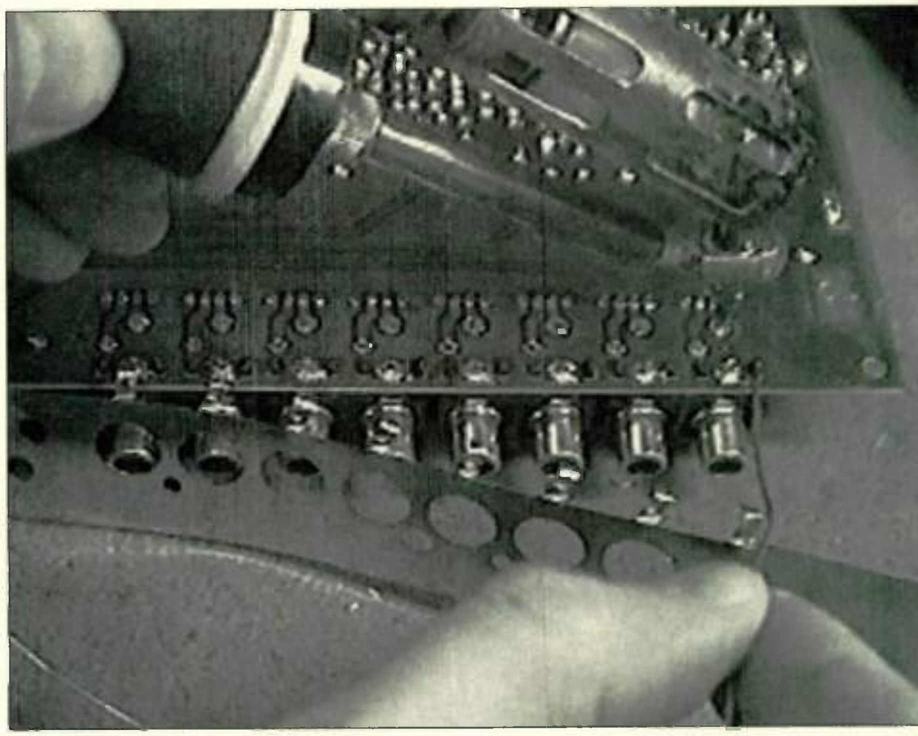


FIGURE 1: Using the Weller DS-700M to remove solder from the RCA connectors of the TASCAM DA-88 A-to-D card.

The March '96 *EQ* maintenance column recommended using a desoldering braid for removing op amps from console PC boards. This is a fine technique for single-sided boards, but for the double-sided type, the only sane way to desolder multilegged components is to use a vacuum-assisted desoldering tool. Prices for these tools range from the out-of-pocket affordable ($5–$30) to charge-card models ($300-$750 and beyond). (See table 1.) Radio Shack's hand-operated desoldering iron (catalog #64-2060) is fine for low-volume repair or modification work. For a "production-line" project (like changing all of the assignment switches in a console), a motorized vacuum tool such as the Weller DS-700M (see fig. 1) is recommended.

DESOLDERING TIPS

Keep the tip clean by wiping it on a damp cellulose sponge. Always apply solder immediately thereafter. Center the vacuum hole of the desoldering iron tip so that it fits over the end of the component lead. Press down firmly, but be sure to wait long enough to visibly melt the solder all the way through to the other side in order to get full evacuation. Inspect the desoldered joint to make sure that the solder has been cleared from all sides of the component lead. If not, it's best to start over by resoldering the joint, then try again, allowing a little more melt time.

FOOT LOOSE

Once all the holes look clear, you may still need to break the component leads loose from any residual traces of solder. Grab the end of the lead with a strong tweezers or fine needle-nose pliers and wiggle until you feel or hear a slight "snap" as the connection is broken. The lead must be free before attempting to remove the component, otherwise you may accidentally remove the "plated-through" section between the top and bottom layers.

If, for example, the component is a DIP IC, inspect any leads that are connected to the top-side pads. Place the tips of a fine-tip needle-nose pliers across a pair of pins on opposite sides

Source	Model	Style	Price
MCM/Chem-Wick Lite	21-320	braid: .1 inch x 5.5 feet	$3.59
MCM/Chem-Wick Lite	21-2050	braid: .1 inch x 50 feet	$19.50
MCM/Radio Shack	21-300/64-2086	desoldering bulb	$3.19/$2.79
Radio Shack	64-2120	desoldering tool (plastic)	$5.99
MCM	21-590	desoldering tool (metal)	$6.79
MCM/Radio Shack	21-3520/64-2060	desoldering iron with bulb	$6.95/$8.49
MCM	21-945	desoldering iron with tool	$27.25
Techni-Tool	Pace MP-1	desoldering iron/pump	$525.00
Techni-Tool	Weller DS-700M	desoldering iron/pump	$758.00

TABLE ONE: Desoldering tools at a glance. (MCM: 800-543-4330; Techni-Tool: 610-941-2400.)

of the IC. Squeeze the "shoulders" of the pins together to break them loose from any residual solder on the top pad. Repeat for each pair of pins.

Once the component has been removed, inspect the traces on both sides of the board to make sure no pads or traces have been lifted or damaged. Also inspect the leads of the removed component for remnants of top-side pads or through-hole plating that may have clung to the lead and been pulled out. It is much easier to repair damaged PC traces when you know where the damage is than to troubleshoot after new components have been installed.

HEAVY DUTY

On the subject of tools: If you are using the Radio Shack desoldering iron, shell out for the heavy-duty tip (catalog # 64-2062). It really does last longer. Tip life is determined by the quality and thickness of the plating combined with user abuse. Too much pressure will not melt the solder faster, but it will wear the protective coating and eventually erode the vacuum hole at the end. A badly eroded tip will not effectively clear plated-through holes and may cause damage to the PC board. If your iron's breathing becomes asthmatic, try poking out the tip end with a piece of solid wire. (Professional tools come with cleaning accessories.) A poorly maintained tool is useless.

Tweezers or "splinter forceps" (available from a medical supply store) are a great help in doing PC board work, not only because of their finely pointed, arrow-shaped tips, but also because an alignment pin prevents the tips from rolling. They are available in straight and curved versions and typically sell for six or seven dollars.

WARNING

If you have never done this type of desoldering before, *practice on junk before mutilating an expensive piece of equipment!* Check out your local electronic surplus/junk store for cheap, expendable PC boards on which to practice.

Mark De Martini is the chief tech at Sigma Sound Services, 212 North 12th Street, Philadelphia, PA 19107; Tel: 215-561-3660. Mr. Ciletti is a man who needs no introduction to EQ readers.

Build The High-Voltage EQ-Pre

How to build that high-quality mic preamp you've always wanted

BY JULES RYCKEBUSCH

Outboard microphone preamps are a hot ticket these days (see the June 1995 *EQ*), in large part because it's difficult for mixer manufacturers to include really good preamps on every channel and maintain a competitive price. Another reason is that even the best digital recorder needs an interface to the real world, for which a high-end mic preamp is very well-suited.

So, with all these top-notch outboard microphone preamps available, why build your own? First, it's always cool to build your own gear. Second, you can build the EQ-Pre, a state-of-the-art microphone preamp, for about $25 per channel and still have great features: phantom power, a phase switch, +30 dB of headroom, flat frequency response well beyond 200 kHz, and extremely quiet operation.

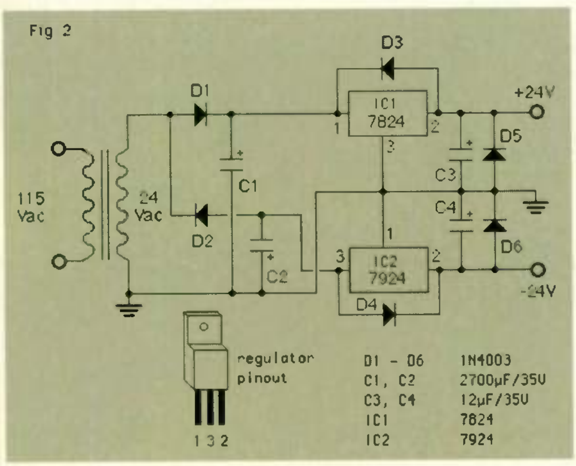


FIGURE 2

HOW IT WORKS

The heart of the EQ-Pre is the Burr-Brown INA103 instrumentation amplifier. Instrumentation amplifiers contain the functional equivalent of three operational amplifiers set up as a differential input, and the gain is set with one external resistor. Instrumentation amps have other characteristics that make them ideal for mic preamp applications, such as very high slew rate, wide bandwidth, and low noise+THD.

Another feature that sets the EQ-Pre apart from most commercial microphone preamps is the ±24 volt DC power supply. Increased supply voltages provide more headroom and better transient response, which result in greatly improved clarity and transparency.

The EQ-Pre has two separate gain stages, since minimizing the total amount of gain a single stage has to produce extends each stage's frequency response. The INA103 provides a fixed 30 dB of gain. A follow-on gain stage is adjustable from 0 to 20 dB. Although we could have made the INA103's gain adjustable by substituting a potentiometer for the gain-setting resistor, the pot would need to be a reverse audio taper type (which is difficult to find in small quantities) for the proper feel. Using a fixed metal film resistor mounted close to the INA103 achieves the quietest possible operation.

A Burr-Brown OPA-2604 (a state-of-the-art, FET-input dual op amp that can also run on 24 volts) provides the second gain stage, which is noninverting. A readily available 10k linear potentiometer sets the gain. The other half of the op amp provides a balanced output buffer.

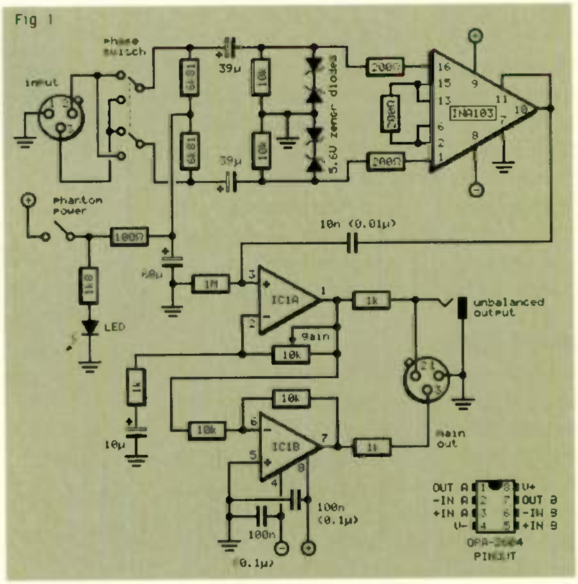


FIGURE 1

All resistors are 1 percent metal film and the capacitors are Panasonic HFQ series. Digi-Key (see sidebar) sells everything but the ICs, but these are available in single quantities from Burr-Brown distributors. The INA103 costs about $8 and the OPA-2604 costs about $3. See the parts list for further information.

CIRCUIT DETAILS

Fig. 1 shows the complete preamp circuit. The mic signal enters via a female XLR jack, then proceeds to a DPST phase switch that can reverse pins 2 and 3. Two 6.81k resistors couple in the phantom power (+24 volts, which works just fine; if you have +48 volts available, you can use that instead).

Four 5.6-volt zener diodes connected back-to-back across both input lines to ground protect the INA103 from any spikes produced from plugging/unplugging microphones while the preamp is energized. A pair of 200 resistors provide further protection for the INA103.

The 200 resistor across pins 13 and 6 determine the internal gain. Pins 2 and 15 are gain-sensing pins and need to connect as shown. Pin 7 is the ground reference pin and Pin 10 is the output pin. From pin 10, the signal couples to our second gain stage by a 0.01 µF (10

nF) capacitor. This capacitor also forms a high-pass filter with the 1M resistor, and helps eliminate any DC offset.

The second gain stage consists of one OPA-2406 stage and its associated feedback elements. The gain equals 1 plus the ratio of the 10k linear pot resistance divided by 1k. The 10μF capacitor in series with the 1k resistor also eliminates any DC offset and limits the preamp's low-frequency response to about 15 Hz. The 1k resistor in series with the 1/4-inch output jack protects the OPA-2604 output from shorts to ground. The second OPA-2604 op amp is a unity gain inverting buffer that provides the other polarity signal needed for a balanced output. Both the in-phase and out-of-phase signals go to the male XLR jack.

THE POWER SUPPLY

Fig. 2 is a bipolar power supply. The EQ-Pre uses Panasonic HFQ series capacitors, which feature very low leakage and a real low ESR (Equivalent Series Resistance). This makes them behave more like "ideal" capacitors. Both of the regulator ICs are 78XX/79XX series linear regulators made by JRC. Unlike most three-terminal regulators, they are sealed in a thermally conductive resin, so you can mount them without having to isolate their heat sinks.

CONSTRUCTION

For stereo, you can easily build two complete preamps on one Radio Shack experimenter's PC board (#276-168 — the kind where you solder the chips in and make a couple more connections to the IC pins).

The key to stellar preamp performance is proper grounding. Two independent continuous copper strips run under the IC chips; tie these both to ground near the ICs. The ground connection for all components that tie to ground (the 10k resistors and zener diodes) should be as close as possible to the INA103. Lay out the rest of the circuitry neatly using point-to-point wiring.

The power supply is also easy to build using point-to-point wiring. Decide where to mount the transformer, then secure the two main filter capacitors with silicone adhesive. Mount the heat sink close to the main capacitors, but make sure you leave enough space for air to circulate. Float the two 10 μF capacitors right off the regulators. Note the diodes across the regulators, which protect the regulators from damage due to reverse biasing.

Test the power supply before hooking it to the preamp. Check for proper polarity on the filter caps (they'll explode if reverse polarized for any length of time; I've seen it happen). If you read about 30 volts DC going into the regulators, life is good. The outputs should be within 5 percent or so of 24 volts. If all is well, connect the power supply to the preamp circuit board by twisting the +24 and –24 volt wires together and neatly running them to the board. Run the ground wire to the point that is closest to all the other ground connections. Use a separate +24 volt wire from the + regulator to the Phantom Power switch.

One of these power supplies can power about 12 preamps. Run separate power supply wires for each preamp board and don't build more than two preamps on one board or there could be trouble!

TESTING

Connect the EQ-Pre to a dynamic microphone (e.g., Shure SM58) and your console. Set the gain adjust control to minimum (about 30 dB of gain). At this gain you should be able to close-mic a loud vocalist and have plenty of headroom. Turn up the gain control, and the gain will increase smoothly up to a maximum of about 50 dB. Flip the Phase switch to verify it works (there may be a momentary loss of signal). If everything checks out, switch to a condenser microphone (e.g., AKG 451). Turn on the Phantom Power switch. Everything should sound clear and crisp. If the EQ-Pre works on a dynamic mic, but not a condenser, check your phantom power wiring. You should be able to measure +24 volts at pins 2 and 3 referenced to pin 1 (ground).

If everything checks out, congratulations! You now own a real high-end outboard microphone preamp.

Special Thanks to: Greg, Wally, and Darren at Greg Rike Productions in Orlando, FL.

Jules Ryckebusch is halfway through his four-year tour onboard the USS Newport News, a 688-class fast-attack submarine. Somehow, he still finds time to design stuff, cook for his lovely wife, and walk Roscoe, the little white dog.

PARTS LIST

Available from Digi-Key Electronics, 701 Brooks Ave. South, P.O. Box 677, Thief River Falls, MN 56701; tel. 800/344-4539.

All resistors 1/4W 1% metal film, values in ohms

100	100XBK-ND
200	200XBK-ND
1k	1.0KXBK-ND
6.81k	6.81KXBK-ND
10k	10.0KXBK-ND
1.8k	1.8KXBK-ND

Potentiometer, Clarostat 1/2-watt conductive plastic

10K pot	408N103-ND

Capacitors, All Panasonic HFQ series

10 μF, 50V	P5756-ND
39 μF, 63V	P5791-ND
68 μF, 63V	P5793-ND
2700 μF, 35V	P5753-ND

Voltage Regulators

+24V	NJM7824FA-ND
-24V	NJM7924FA-ND

Transformer

115 to 24VAC @ 1 amp MT3128-ND

Diodes:

Power supply	1N4003GITC-ND
Zeners	1N5232BCT-ND
LED of your choice	
Heat sink (two req'd)	HS112-ND

The INA103 and OPA-2604 are available from:

Insight Corp.
6925 Oakland Mills Rd., Ste. D
Columbia, MD 21045
800/677-7716

For Data sheets and specs call Burr-Brown: 800/548-6132

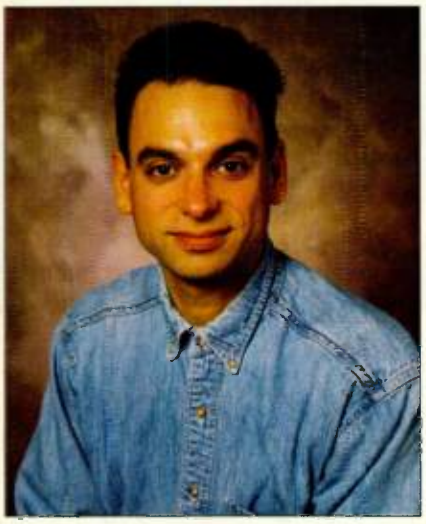
Miscellaneous

Hardware for mounting everything
Radio Shack PC board (see text)
Male chassis mount XLR connector
Female chassis mount XLR connector
1/4-inch phone jack

Patchbays: Real and Virtual

Clean up those cords — real or otherwise

BY EDDIE CILETTI



It's fairly easy to assemble a budget for the primary recording components — a mixer, a few multitracks, monitors, microphones, and multieffects. The design and cost of interconnection, however, is more elusive. Wiring materials are not expensive items compared to the process of wiring, a most labor-intensive task that, without "preproduction," could limit present flexibility as well as future growth.

A single, 96-point bantam patchbay starts out at an affordable $320 (list, unwired and without patch cords) and can end up costing $1500 to $3000, installed. A multipin or user-friendly solderless interface adds between $750 to $1250 to the patchbay price, before wiring. Prefab harnesses are cost effective, but length choices will be limited. Custom wiring will be entirely handmade and specific to your installation. Harness prices will range from an off-the-shelf $300 to a thought-provoking $2000. Cable length and quality of materials are some of the variables. As if this weren't painful enough, one patchbay is rarely adequate.

The traditional, hard-wired patchbay is appropriate for managing the number of analog sources and destinations typically found in a recording environment. Digital audio, being the new kid on the block, is not so steeped in tradition. Without some organized method of patching, that pile of digital gear you've amassed is probably tangled in a mess of spaghetti *no matter how neatly it started out.* If so, it's time to consider the interconnection options.

REAL PATCHBAYS

Traditional audio patchbays come in many shapes and sizes. My preference is for professional, bantam (.175 inch) connectors for their durability and packing density. All-metal connectors will outlast the plastic, pc-mounted "jackfields" that come standard with some consoles. The expense of bantam and professional 1/4-inch patch cords should not persuade you to take the "prosumer" route. In the long run, the investment in professional wiring will survive several reincarnations of your project studio.

Prosumer bays accommodate standard 1/4-inch plugs. With connectors on both front and rear panels, these bays are only a better deal because they facilitate installation — you can do it yourself using off-the-shelf interconnect and patch cables. No soldering or special punch-down tools are required. The per-patch-point cost is comparable with a pro bay, but they have no afterlife! Spring pressure in these jacks is not sufficient enough to self-clean the contacts known as "normals," the connection made from top to bottom rows when no patch cord is inserted.

Note: All patchbays should be vertically mounted to minimize particle contamination.

GIMME SPACE

Since the bandwidth of digital audio is roughly the same as analog video, a video patchbay is the appropriate way to route S/PDIF signals. There are, however, so many gotchas that the expense and inconvenience is hardly worth it. Balanced AES signals, for example, must be converted to unbalanced (S/PDIF) and back. This can be passively accomplished via transformers that are available from Canare for about $50 a pop (available from Markertek). Optical signals must be converted to S/PDIF, and this can only be accomplished with active electronics.

Video patchbays are more expensive than their analog counterpart. Each non-normalled jack costs about $10. Normalled and self-terminating jack-pairs are about $40. That means two rows of 24 jacks could cost either $480 (non-normalled) or $960 (normalled), plus the panel ($65) and patch cords (you don't wanna know!)

ROUTER BASICS

The answer for interconnecting digital gear is a router, or electronic patchbay. Routers have been directing MIDI traffic for some time and are most easily envisioned as a matrix table of rows and columns each representing inputs and outputs, respectively. Fig. 1 shows a five-by-five (5 x 5) input/output matrix. On the top row are outputs (sources), and in each respective column are letters, A and B, that represent two user presets to

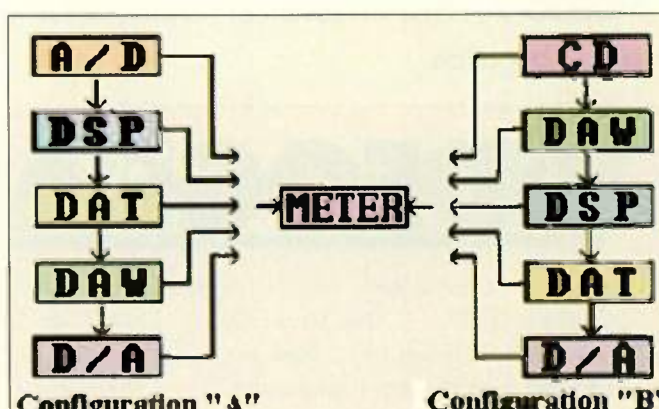


FIGURE 2: The same presets as in fig. 1 presented in block diagram form. Note that a router can send a single source to multiple destinations. In this case, any source can be routed to the meter without interrupting the preconfigured signals.

Inputs	Outputs	1 A/D	2 DAT	3 DAW	4 DSP	5 CD
1	D/A		B	A		
2	DAT				A & B	
3	DAW		A			B
4	DSP	A		B		
5	METER	A				

FIGURE 1: An input/output matrix. The letters "A" and "B" indicate two preset crosspoint configurations that can be more clearly seen in fig. 2.

the inputs (destinations). Note that output slot 5 is a source-only device (a CD player), while input slot 5 is a meter, a destination-only device.

Fig. 2 represents the same matrix only in block form. The number of inputs and outputs is typically symmetrical (8 x 8) just as a hardwired patchbay, but asymmetrical (8 x 1) routers, used as source selectors, are also an option.

The horizontal arrows in fig. 2 point to a meter to which any device can be routed (to confirm levels) without interrupting signal flow. A router can simultaneously feed one source to all destinations (as would be the case for making multiple DAT copies). When used in this way, the router becomes a distribution amplifier. On a hard-wired, passive patchbay, this is called a "mult." Since routers are active-technology based, power is available for the various interfaces, such as AES, S/PDIF, and Optical.

WISHFUL THINKING

By now you might be thinking that an analog router would be a cool breeze on a hot day. But don't think you can get out of cleaning patch cords that easily...Unfortunately, the sheer number of crosspoints required by a typical multitrack environment puts the analog router in a "world-class" orbit. Down here on earth, a more practical application is the Studio Technologies *Studio Comm*. It is a seven-input, stereo source selector that replicates a "facilities" panel typical of a world-class recording console. It has two control room outputs (for an alternate amplifier/speaker system), a studio/headphone output, a dub output, plus a microphone for talkback and slate.

VIDEO VIEW

In videoland, routers and patchbays coexist. The former makes life easier for the user, while the latter allows technicians to troubleshoot problems. (A bad router channel would make a tape machine seem "down" and a patch cord would be the easiest way to find out.) Because sound and picture go together, routers can be configured to move both audio and video. However, that's not how all those great camera angles make it to your TV screen. That bit o'wizardry requires a *switcher* or, as the Brits call it, a *vision mixer*. While all online video is referenced to a house clock (as mentioned in last month's column), switching from source to source must occur at the top of the picture — the vertical blanking interval.

Yeah, you might think this is an audio rag, but digital audio has much more in common with video. That's why products such as the Aardvark DAC have sync inputs. Word sync is another example. This point is relevant because, the Z-Systems Detangler makes a great "patchbay," but it can not be used as a glitchless digital source selector. Now that would be a fun product too!

SHORT CUTS

So much for my detour to the Mir Space Station. If your budget does not include a patchbay, you can get around it if:

- MIDI gear is the primary source
- The mono or stereo mix from samplers and synths (rather than individual outputs) is acceptable
- The mixer is amply outfitted with line inputs, tape returns, and busses
- Effects devices can be normalled to their respective sends and returns.

For these rigs, a floating signal processor, such as a stereo compressor/gate, can be connected to a pair of TRS "insert" cables so that it can be easily connected to either the mix bus insert or other "points" of interest. Have fun.

Eddie Ciletti feels that, "Life in the studio should be more than just a great place to grow mushrooms." His dark, damp e-mail addresses are: eddieaudio@aol.com and edaudio@interport.net. Coming soon, http://www.users.interport.net/~edaudio/.

PHIL RAMONE

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BACK TO NEW YORK

Once the recording had been finished, we used a Yamaha MDF-1 MIDI Data Filer to dump scene memories and dynamic automation moves from the 02R onto a floppy disk. Back at The Shire (my studio in New York), the MDF-1 was used to load this data into our 02R. We could have picked up the mix where we left off in Capitol, but for creative reasons we started a new mix with fresh ears. Had we done the session using Capitol's board, we would not have had the same flexibility to take the mix on disk and recall it exactly as we left it at the touch of a few buttons. Don't get me wrong, Capitol's board is a good one, and I have gotten some great-sounding music using it, but the recall process is considerably more complicated.

Since we wanted to do the remix at The Shire using the main channel faders, we used the 02R's "flip" function to swap the tape monitor and channel fader paths. Mixing on the 02R works like it does with most big-ticket digital consoles: you start with a scene memory (snapshot) and your automation proceeds with events (like channel on/off or dynamic moves) relative to that scene memory.

As you would expect, the automation runs via SMPTE. It so happens that the 02R's SMPTE input is an RCA connector, and the DA-88's SMPTE output (via the SY-88 sync card) is also on an RCA connector, so patching the two was easy. If we had been doing a session with our sequencer running virtual tracks, the 02R could also accept MIDI timecode and lock to that.

Knowing that the recording is going to wind up on a CD, you have to select a sample rate for the project with that in mind. It doesn't make any sense to do the normal trick of recording at 48 kHz and then mixing at 44.1 kHz. If you go through an analog console, you are coming out via analog outputs to, say, an Apogee converter and maybe laying the mix to DAT or PCM 9000 at 44.1 kHz. But in our case, once the audio is on tape, it is never going to leave the digital domain — it's just like on a hard-disk editing session. Your session sample rate determines what the output is going to be. We did the original multitrack session at 44.1 kHz so that we wouldn't have to do any sample-rate conversion to the mix for CD.

Because the 02R's signal path is all digital, it allows you to automate changes in EQ and panning, as well as channel on/off and fader moves. In fact, we used this ability to fix a pop on the lead vocal by briefly changing the EQ at a specific point in the song. By inserting different scene memory changes in your mix, you can also automate changes to other parameters (such as dynamics) as well. Your imagination is your only limit. The effects used in the final mixdown came from the 02R's two internal processors, so the recorded signals were never converted to analog for routing to and from an external processor. For those of you keeping score at home, the final mix at The Shire was put down to a TASCAM DA-20 Mk II DAT machine via the AES input.

Listening to the final mix, I am very happy with the result. While the musicians and Fran were focusing on the music and striving to get the most out of their performances, the board never once got in their way nor was there any downtime of any kind in either the recording or mixing of the track. The console retained the warmth of the instruments and the technology gave us the flexibility to walk out of the room with a floppy disk and some tapes and recall our work thousands of miles away.

I should point out that if I was called on to do a large orchestra date, another 02R or more would have been needed. But I think it's clear that the board does have its applications. And what this board says it will do, it does.

If you have any questions for Phil Ramone about the 02R, you can e-mail him at PRIoffice@aol.com.

GUERRILLA

continued from page 56

SCR. This little guy fails the diode test on my meter. This is not the case on the working supply. I jump into my Toyota for another trip to Tinkertronics for an NTE5011A zener diode. I pop it in, power up the ADAT, and, still, no front panel.

I'm about to go completely post office when the phone rings. It's Ed. I start relating each failure until he says, "Maybe you got the new diode in backwards?" Way to go Ed. Good punch. I switch the zener around, give it the juice, and I'm ready to roll tape. I call Paul at Pro Digital, "Cancel that UPS Red Label."

This troubleshooting trip was a team effort — a combination of perseverance mixed with the generosity of kind-hearted, intelligent people.

Bennet Spielvogel lives in Austin, TX, and owns Flashpoint Recording Studio. His e-mail address is ESideFlash@aol.com.

MI INSIDER

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heading of support, but just about all samplers these days have substantial, high-quality libraries available, and file compatibility makes even more options available.

FILE MANAGEMENT

With all those megabytes of sound, you can get lost without some way to find and load sounds. Roland offers a Quick Loading feature for marking samples (such as all those used for a particular project) in advance for instant loading later on. Ensoniq has macros for CD-ROM access as well as a bank option that pulls the sounds needed to create a bank of sounds (including from CD-ROMs), while Kurzweil's Advanced File Management System can search for any object on any SCSI drive, create macros for loading particular setups, and the like. The E4K has both a search utility that works just like the "find" command in a word processor and a "SoundSprint" function for quick auditioning and loading of up to 100 favorite presets.

There's also computer software such as Sample Search (Gallery Software) and Sonic Search (Sonic Science) to help manage sample libraries (archiving, cataloging, and search/retrieval).

HARD-DISK RECORDING

So far this is available with only a few samplers, including the ASR, Akai 3000XL/3200XL, and as an option on the Korg Trinity, but it probably won't be long before we see this showing up in more and more samplers. While onboard hard-disk recording can't compete with a spiffy computer-based system, it can be very handy when you want some vocals or guitars to go along with sequenced samples.

DSP

This includes standard (and always welcome) effects such as equalization, chorus, etc., however, many samplers go beyond the norm. The Roland S-760 (among others) has a time stretch algorithm for changing a sample's length without changing its pitch, as well as a sample-rate conversion routine.

Look for the option to resample using the DSP effects. One way to use this would be if you wanted a tough gated reverb sound on the snare and room reverb on the rest of the kit. Simply resample the snare through the gated reverb, then use the onboard reverb on the kit as a whole. In some cases DSP is an add-on; an extra $399 buys a reverb card for the Akai S3000XL and S3200XL.

SAMPLE THIS!

When you go to a store to sample a sampler, don't forget such factors as the keyboard feel (or, if a rack, editing function accessibility). And, of course, the sound of the unit is the ultimate arbiter — although these days samplers are more alike than dissimilar when it comes to sound quality. One quick check is to take a cymbal and transpose it way down (two octaves or so). The smoother and less grainy it is, the better. Also check the filter sweeps for smoothness and pay attention to controller options such as LFOs and envelopes, as these are crucial for shaping samples into something more animated.

In any event, whether you use samplers for dance mixes, orchestral emulation for sound tracks, or a compositional tool, the current crop has a lot to offer. Have fun!

Craig Anderton's latest book, a from-the-ground up revision of the classic book Home Recording for Musicians, is now available. For more info, check out his AOL site (keyword SSS).

EQ SHOPPER

EQ SHOPPER

CIRCLE 14 ON INFO CARD

CIRCLE 38 ON INFO CARD

CIRCLE 88 ON INFO CARD

CIRCLE 54 ON INFO CARD

EQ SHOPPER

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Wave Front Studio Monitors

"...I had exceptional directional imagery as well as wonderful transparency. Whether I monitored in the 80db to 85db range or took the speakers up to 110db, the propagation pattern was smooth but above all, at the end of a long day, my ears were not ringing!"

Tom Dowd, Producer
(Eric Clapton, Rod Stewart, Michael Bolton, Diana Ross, Ray Charles...)

What Have THEIR Monitors Done For YOUR Mixes?

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Cram That RAM

Now is the time to take advantage of low computer memory prices

BY MARTIN POLON

There has always been one element of computer systemology that conforms to the old saying about how "you can not have too much money or be too thin or have too much hair." Add to that, "You cannot have too much random access memory (RAM)." For most of the modern history of the personal computer, RAM was so expensive that the use of computers was limited by the amount of RAM available. Although this was less of a problem with early operating systems and application programs, today's Intel-based PCs running Windows 95 and Macintosh computers running System 7.5.3 require 8 MB just to run the operating systems properly.

Upgrades have been costly, finally settling into the groove of about $50 per megabyte in the beginning of 1994 so that a 4 MB upgrade would cost $200, 8 MB about $400, and so forth and so on. Prices would fluctuate, with an occasional deep fall as a new memory plant came online worldwide, but would rapidly stabilize. Except for a period when a fire in an Asian plant that made the epoxy cement used in memory chip manufacture caused a worldwide shortage and price hike, the pricing of memory chips has been a kind of "given."

The curiosity about RAM is that it has been the one thing everyone really wants and few can really afford for their computers. For those involved in recording and editing audio and video or multimedia or graphics, the demands of manipulation in RAM in real time are significantly more important than they are for the majority of computer users.

What has happened during the spring of 1996 has been the coming online and revving up of new memory plants and the planning or beginning of construction for memory facilities in Japan, Korea, Singapore, and Malaysia. The prices for chips have dropped by 75 percent in some cases. The norm for memory has gone from $50 per megabyte to as low as $12.50 per megabyte.

What has happened is that, in addition to the issue of ramping up of production of memory chips, the unrealistic expectations by chip makers of increased memory usage by those who have installed Windows 95 (W'95) has also spurred production. The demand for increased memory by W'95 *has* been substantial, but not at the levels that chip makers expected. At any rate, the price drop caused by excess supply vis-a-vis demand has started a virtual run on additional memory for both Macintosh and PC users.

Conventional advice at this time is very simple: Beg, borrow, or steal enough money to at least double the RAM in all of your computers, and, if possible and affordable, go to the max! One problem many studio users (and others) have with upgrading RAM is the necessity (in most cases) of virtually discarding the existing RAM in Macs and PCs.

Except for the most elaborate and expensive models, which have as many as eight RAM sockets or more, many computers have only one or two sockets. These are usually filled with the smallest values possible by the computer maker to control expense. Sometimes there is 1 to 4 MB of RAM soldered to the motherboard in addition, although that is less likely with Power PC- and Pentium-powered machines moving at 100 MHz or faster — that depend upon memory to enhance 32-bit/64-bit performance. Even multisocketed machines can have small chips installed that should be replaced by larger chips. Bottom line: Take whatever value your dealer offers for your old RAM and trade-up in class to more powerful RAM. You will not regret it!

What does more RAM do for you?

1. More RAM increases your operating speed in most cases. It does this in many machines via memory access interleaving. This can mean up to a 20 percent speed boost on some machines.

2. Speed gains are also made by having enough physical RAM to run everything you use on your computer. If a computer has to go back and forth to the hard drive while performing millions of computations, that slows down performance. Ditto the forced use of virtual memory schemes that the various operating systems employ where hard-disk space is used to substitute for inadequate amounts of installed physical RAM.

3. Adequate RAM also allows opening up various RAM caches used to speed up processor "scratch padding," enabler speed, operating system assist applications such as Speed Doubler (which needs as much as 2 MB), and CD-ROM precaching of repetitive data.

4. Application size can be increased to optimize efficient usage and to decrease the likelihood that application speed will fall victim to insufficient memory situations. If a RAM substitute such as RAM Doubler is used, it will not serve to compensate for physical RAM needed for actual audio and video manipulations. With adequate amounts of physical RAM, however, RAM programs do work much more effectively.

5. Audio and video applications will have enough room to run in RAM or in RAM disks and to manipulate (edit) large files of audio and video.

So how much RAM is enough and how long will the low prices hold? The minimum RAM that should be in any state-of-the-art Pentium or Power PC computer in a recording studio is 32 MB. More memory would be better, and there are many Macs running Digidesign or similar audio recording software that have 128 or 256 or 512 MB of RAM installed.

And remember this: Buy your RAM from reputable local or mail order suppliers — install it yourself only with careful and adequate static electricity protection and knowledge of your PC or Mac! EQ

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TASCAM

102 MKII / 103
Stereo Mixdown Cassette Decks

Best values for musicians, studio operators and production houses, the 102 MKII and the 103 consistently produce only the highest quality tape recorded output.

They Feature:

- 60dB signal-to-noise ratio combines with wide frequency response for high-fidelity sound reproduction using any type of cassette tape.
- Industry-standard Dolby B/C noise reduction and Dolby HX Pro sound technology extends high frequency performance up to 6dB and minimizes distortion.
- Advanced bias-sensing electronics automatically chooses optimal recording settings for the type of tape you load in—Normal, Metal or CrO2.
- Record/Mute autospacer automatically inserts 4 seconds of silence between songs or broadcast segments.
- Multi-function display clearly indicates transport mode, tape counter position, tape type and level indicator.
- Multi-counter with both tape counter and run-time modes.
- Independent L and R stereo level controls and master record level control.

Tascam 103 Advanced Features:

- 3-head system allows you to record on a tape and monitor it at the same time, without rewinding.
- MPX filter button eliminates pilot and sub carrier broadcast tones that can interfere with Dolby noise reduction.

202 MKIII
Dual Auto Reverse Cassette Deck

The 202 MKIII provides high-fidelity sound reproduction and a wide frequency response, as well as a host of features that help you dub, edit, record or playback onto/from one or two cassettes easily and efficiently.

- Normal speed and high-speed dubbing
- Autospacer automatically inserts 4 seconds of silence between songs or broadcast segments for pro quality tapes.
- Incorporates Dolby HX Pro sound technology to extend high frequency performance and minimize distortion on Normal, Metal and CrO2 tape.
- Allows you to quickly and easily create a professional-sounding composite tape from several sources. Functions like Intro Check, Computerized Program Search, Blank Scan and One Program quickly find the beginning of tracks you want.
- Twin two-head cassette decks in a durable rack-mount housing that can be used separately or in tandem during recording and playback for total flexibility.

—Play material on deck 1 while deck 2 records on one or both sides.
—Record simultaneously on both decks from an external master.
—Play back both sides of one or both decks in a continuous loop up to five times.
—Auto Reverse automatically reverses tape direction during play-back and record.
—Repeat rewinds tape and allows infinite looping during playback.
—Timer switch for unattended record/playback (timer required).

NEW! **302**
Double Auto Reverse Cassette Deck

All the features of the 202 MKIII, the new 302 adds even more recording and playback flexibility. Thats because the 302 is actually two fully independent cassette decks. Both decks have their own set of interface connectors, transport control keys and noise reducing functions.

- Auto-reverse capability on both decks
- Individual/simultaneous record capability–both decks
- Independent RCA unbalanced in/out for each deck
- Cascade and Control I/O let you link up to 10 additional machines for multiple dubbing or long playing record and playback applications.

CD-601
Professional CD Player

Frame-accurate cueing precision, extremely high-fidelity and a small form factor make the CD-601 ideal for post-production applications where sound effects and music are "flown-in" from compact discs. The CD-601 integrates with most post-production equipment including mixers, video editors and computer studio controllers.

- Balanced XLR and unbalanced RCA outputs
- Precision cueing control and Auto cue
- Linear motor-driven pick-ups eliminate dead air
- Optional RC-601 remote control adds additional features and conveniences
- Optional BU-2 RAM for instant start and seamless loops up to three minutes

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PMD-101/201/221/222/430
Portable Professional Cassette Recorders

The world standard for field recording, the PMD line is also the value leader. They all feature RCA line input/outputs, 1/4-inch headphone jack, built-in speaker, pause control, audible cue and review, tape counter, full auto shut-off and low battery indicator.

- All models except the PMD-430 have 1/2 speed playback/record capability. With 1/2 speed playback, musicians can slow down complicated passages for analysis. And when played back at 1/2 speed, the pitch is lowered by exactly one octave, so the notes are still musically correct—ideal for figuring out complicated solos or picking patterns.
- By recording at 1/2 speed, a three hour meeting can be recorded on a single tape. A built-in microphone and automatic level control make operation simple, and built-in speaker makes transcription convenient.
- 1/2 speed recording is equally ideal for churches, because 90 minutes can be recorded on a single side of tape—no interrupting your recording to flip the tape over. Line inputs make it easy to use and connect to your existing sound system.
- Three standard 'D' cell batteries provide up to 7-1/2 hours of operation and the optional RB430 rechargeable battery delivers up to 5-1/2 hours.

General	PMD-101	PMD-201	PMD-221	PMD-222	PMD-430
Stereo Mono	Mono	Mono	Mono	Mono	Stereo
Heads	2	2	3	3	3
Inputs/Outputs					
Mic Input	1/4-inch	Miniplug	Miniplug	Mini/XLR	1/4-inch
Condenser Mic	Built-In	Built-In	Built-In	Built-In	—
Remote Jack	—	Yes	Yes	Yes	—
Modular Tel. Jack	—	Yes	Yes	Yes	—
External Speaker Jack	—	Yes	Yes	Yes	—
Record Controls					
VU Meters	—	1	1	1	2 (Illuminated)
2-Speed Recording	Yes	Yes	Yes	Yes	—
Dolby B NR	—	—	—	—	Yes
dbx NR	—	—	—	—	Yes
Mic Attenuation	—	0,-10dB, -20dB	0,-10dB, -20dB	0,-10dB, -20dB	0,-15dB, -30dB
Ambient Noise Cont	—	Yes	Yes	Yes	—
MPX Filter	—	—	—	—	Yes
Manual Level Control	—	Yes	Yes	Yes	—
Limiter	—	Yes	Yes	Yes	Yes
ALC	Yes	Yes	Yes	Yes	—
Peak Indicator	—	—	Yes	Yes	—
Playback Controls					
Pitch Control	±20%	±20%	±20%	±20%	±6%
Bias Fine Adj	—	—	—	—	Yes
Tone Control	Yes	Yes	Yes	Yes	—
Half-Speed Playback	Yes	Yes	Yes	Yes	—
Memory Rewind	—	—	Yes	Yes	Yes

Telex

ACC2000/4000 Series Cassette Duplicators

Designed for high performance and high production, Telex's ACC Series (ACC2000/ACC4000) and (ACC2000 XL/ACC4000 XL) of expandable duplicators also offer easy maintenance and unsurpassed ease of use. The ACC2000 is a two-channel monaural duplicator, the ACC4000 is a four-channel stereo duplicator. Each produces 3 copies from a cassette master at 16 times normal speed and each can expand up to 27 copy positions (with additional copy modules). With the extra copy modules, you can duplicate up to 27 copies of a C-60 original in less than two minutes. And they copy both sides at once. The XL Series feature "Extended Life" cassette heads for increased performance and wear characteristics. They also offer improvements in wow and flutter, frequency response, signal-to-ratio and bias. Additionally the ACC4000 XL allows for either chrome or ferric cassette duplication. XL- models are available in stereo (ACC4000 XL) or mono (ACC2000 XL) versions.

Fingertip Operation

- Individual rotary audio level controls allow for an increase or decrease of audio levels as the master translates to the copies.
- Peak reading LED indicators allow quick and accurate monitoring of audio fluctuations.
- Side A or A/B select button let you set up for duplication of either 1 side or both sides of a cassette at once.
- Stop all tapes instantly at any point during the copy or rewind cycle.
- Short tape indicators alert you if a tape stops before the original does, identifying incomplete copies caused by jam or short.
- Automatic or manual selection of rewind and copy operation.

–Rewinds tapes to the beginning or end automatically (AUTO mode) or manually.
–In AUTO mode the copy button activates the entire rewind/copy/rewind sequence. In manual it starts copying immediately.

Easy Maintenance

- Slanted work surface and unique "heads-up" cassette platform allow less oxide build up on the heads and makes cassette loading and unloading much easier.
- Each cassette position has a three point tape guidance system that eliminates skew problems. Plus, when a tape is inserted, each cassette position is activated to prevent unnecessary wear and tear on the tape head mechanism.
- Audio and bias, along with head adjustments, are made easily from the top of the unit and a switch on the back engages the head and pinch roller for convenient cleaning.

ACC2000 Mono MasterModule:

- 1/2 track, two-channel monaural duplicator produces 3 copies from a cassette master at 30ips (16X normal speed).
- Expands up to 27 copy positions by adding ACC2000 copy modules (four positions each).
- Erase heads in the copy positions automatically erase existing audio as new material is being recorded.
- Track select, short tape indicators, auto/manual operation.
- Includes removable power cord and protective dust cover.

ACC2000 XL Mono Master Module:

- Same features as ACC2000, *plus*– Extended Life cassette heads.

ACC4000 Stereo Master Module:

- 1/4 track, four-channel stereo duplicator. Same features as ACC2000 Mono Master Module.

ACC4000 XL Stereo Master Module:

- All features as ACC4000, *plus*– Extended Life cassette heads. Can be configured for chrome or ferric cassette duplication.

ACC2000 Mono Copy Module:

- 1/2 track, two-channel monaural copy module.
- Each module has four copy positions with erase heads and controls for side select.
- LED displays indicate end-of-tape status for each pocket.
- Includes ribbon cables for connection to ACC2000 master and other copy modules.
- Includes removable power cord and protective dust cover.

ACC2000 XL Mono Copy Module:

- Same features as ACC2000 Copy Module, *plus*–Extended Life cassette heads. Connects to ACC2000 XL Master Module.

ACC4000 Stereo Copy Module:

- 1/4 track four-channel copy module. Has all the features of the ACC2000 Copy Module.

ACC4000 XL Stereo Copy Module:

- Same as the ACC4000Copy Module, *plus*–Extended Life heads. Configurable for chrome or ferric cassette duplication.

Copyette EH Series Duplicators

The popular Copyette series produces high quality, low cost cassettes in large quantities at nearly 16 times normal speed. This means you can reproduce both sides of a C-60 tape in less than two minutes. Available in two versions, the Copyettes are capable of duplicating either one cassette or three at a time. In addition each are available in both mono and stereo models.

They couldn't be easier to use. You simply insert the cassettes, press the START switch and they do the rest. They rewind all tapes to the beginning, copy, then rewind to the beginning again before stopping. The whole process can be stopped at any time by pressing the CYCLE button. Side Select feature allows you to set them up to copy one side of a tape or both sides at once.

Stereo Copyette 1•2•1

Weighing only 8 lbs. (3.6 kg), this unit has a durable, impact resistant housing and includes a removable power cord, carrying handle and protective cover. It also has an optical, non-reflective end-of-tape sensing system that provides gentle tape handling. A mono version is also available.

Stereo Copyette 1•2•3

This duplicator copies both sides of three cassettes at once, yet it's as small as the 1•2•1. It weighs only 12 pounds (5.4 kg) and includes a hard cover to protect the unit while not in use. It uses all DC Servo motors for the ultimate in reliability.

TASCAM

112 MKII
Stereo Cassette Deck

The classic "no frills" production workhorse, the 112 MKII is a 2-head, cost effective deck for musicians and production studios. Extremely rugged and reliable, the 112 MKII is ideal for production mastering and mixdown. It also features a parallel port for external control and an optional balanced connector kit means it is flexible enough to integrate into any production studio.

- Utilizes Dolby B or C noise reduction with Dolby HX Pro
- Automatically selects proper bias type, so you get optimal recording & playbank response with Normal, Metal or CrO2 tape.
- Gear independent input dials let you dial in stereo VU calibration with one dial. You can also adjust for channel specific calibration.
- Offers two Autolocator buttons and a MEMO IN control. These controls allow you to select two points on any tape for one button forward/reverse to wherever the action is. Additionally, RTZ (return to zero) quickly spools the tape back to 0000 on the tape counter.
- Rear-mounted RCA input/output jacks for easy connection to high quality sources.
- Optional LA-112 connector provides additional balanced or unbalanced XLR inputs and outputs. Installation is simple and requires no special tools.
- 25-pin D sub connector (parallel port) on the back, links the deck to the optional RC-134 remote control unit or for fader start from any mixer that use the same protocol.

112R MKII
Bi-Directional Stereo Cassette Deck

The 112R MKII is a sonically uncompromising, auto reversing and continuous play cassette deck. It offers the finest independent head auto-reverse design at this price level, plus it has extra dubbing and editing features that make it ideal for long program recording.

All the features of the 112 MKII *plus*—

- Three-head transport with separate high-performance record and playback heads. Manufactured from resilient Cobalt Amorphous materials, the independently-operating heads combine with precision FG servo direct-drive capstan motors to provide the highest standards of reproduction quality and performance.
- Frequency response is 25 Hz to kHz with less than 1% total harmonic distortion.
- Equipped with Hysteresis Tension Servo Control (HTSC) the 112R virtually eliminates wow and flutter. HTSC is an advanced servo control system that maintains consistent back tension on the tape all through the reel, combatting inconsistencies brought on by extreme temperatures and humidity.
- Super Acculign Rotating Head System allows recording or playback tape direction to be changed with one button. A single screw azimuth adjustment makes it easy to maintain the head alignment after many hours of continuous use.
- For unattended record/playback of material that is longer than one side of a tape, there are two features that spare you from constantly attending to the deck.

—Auto Reverse mode plays or records in both directions before stopping, switching sides on the fly.
—Continuous Reverse mode allows you to loop the tape during playback up to 5 times, or record in both directions, without pausing to flip the tape and re-engage the record mechanism. Both features are accessible from the front panel, with one-button selection.

122R MKIII
3-Head Stereo Cassette Deck

The standard for production and broadcast facilities, the 122 MKIII features smooth faultless tape handling mechanisms, a three head transport with high-performance Cobalt Amorphous record/playback heads and precision servo direct-drive capstan motors.

All the features of the 112R MK II (no reverse of course) *plus*—

- XLR balanced and unbalanced RCA inputs and outputs are selectable with the flip of a back-panel switch. There are 1/4-inch inputs on the front panel for simple and direct plug-in of line-level gear.
- MPX filter button eliminates polot and sub carrier broadcast tones that can interfere with Dolby noise reduction.
- Bias and level fine tuning for each channel. These tuners can be used in conjunction with the one-touch 400 Hz or 10 kHz oscillator adjustment signals to get proper VU calibration before or during each recording session.
- Record/mute autospacer automatically inserts 4 sec. of silence between songs or broadcast segments for pro quality tapes.

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Send cover letter, references and resume, along with samples of both field recordings and studio post-productions to: **WOSU Stations, Personnel Department, 2400 Olentangy River Road, Columbus, OH 43210.** Please indicate job number 631 and referral source in your cover letter. Applications accepted until position filled. The Ohio State University is an Equal Opportunity, Affirmative Action Employer. Women, minorities, Vietnam-era veterans, disabled veterans and individuals with disabilities are encouraged to apply.

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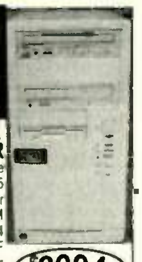

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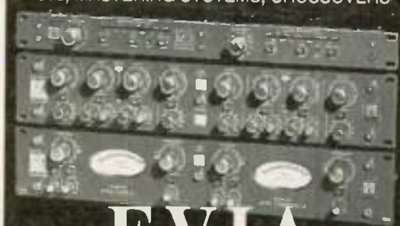

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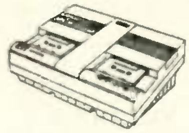

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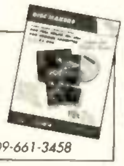

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ACROSS THE BOARD

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much as five hours of jitter and still come up with the right answer.

If a one or zero in was not the same as the one or zero out, then computers would not work. You would not be able to illegally copy software for your friends. You would not be able to copy from one hard disk to another without your bank balance changing randomly. ATM machines would dispense one amount of cash and debit your account by a totally different amount (well, maybe that one was a bad example).

If you are getting bad transfers, then it is faulty hardware or bad cables that are causing total destruction of the data stream. If an audio file on one medium compares bit-for-bit with an audio file on another medium, then it is the same file. They are the same, they will sound the same, there is no argument that will convince me otherwise. End of discussion.

It sounded like he was pissed. Did he ever explain the title to this column? Not really, but I'm not going to ask. You do it! EQ

SAMPLE CDS

continued from page 74

• Play idiomatically. Guitarists don't play 26 notes at a time, and drummers don't have four hands.

• Use a volume pedal to control overall dynamics rather than trying to do it all from the keyboard.

• Use pressure to add pitch bend to guitar patches, or vibrato to wind patches.

• Use appropriate signal processing. Send guitar patches through a guitar amp, or at least add some overdrive or compression.

• For vibrato, learn how to shake the pitch wheel manually instead of using the canned triangle wave vibrato. Guitarists use their fingers — so can you.

• Most samplers also have synthesizer-style processing options: filters, envelopes, LFOs, etc. Use these to shape your sound, and don't be afraid to layer synthesized and sampled sounds — the results can be striking (to convince yourself, layer an FM harp sound with a sampled one).

• Use samples recorded with different dynamics and correlate them to velocity, preferably by crossfading between the samples so there isn't an obvious dividing line. (Drummers take note: the Peavey SP has a special mode for triggering multiple drum hits at different velocities within a single sample, thus speeding response time and cutting down on polyphony demands.) Overlapping samples and crossfading in the overlap region can help smooth out the transition points between multisamples.

GETTING CREATIVE

In many parts of the world, particularly Europe, people who would normally have a difficult time playing music are using sample CDs and creating strikingly original forms of music, mostly centered around various mutant strains of dance music (techno, hip hop, ambient, jungle, etc.). In the process, the sampler goes from its original roots as an imitator of sounds to a creator of sounds, and even entire compositions.

Today's samplers are enjoying a resurgence that's overshadowing the synthesizer itself, in large part thanks to stabilizing RAM prices and the appropriation of mass market computer technology (CD-ROMs, SCSI ports, file translation programs, etc.). The tools are there, the sounds are there, and the music is there: now go get creative. EQ

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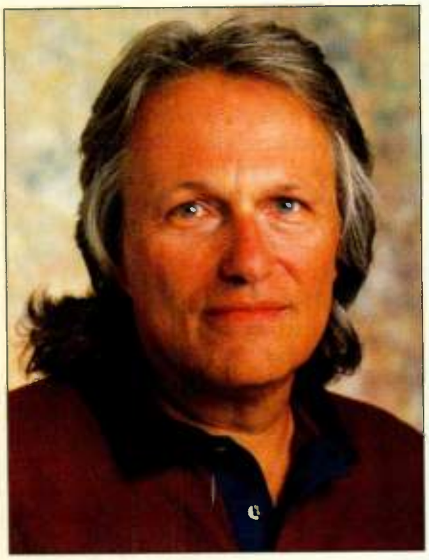
BY ROGER NICHOLS



What the hell does he mean by that? Does it have some bearing on digital audio timing and whether the AES stream is clocked from the leading edge or the trailing edge? Maybe we should just ask? "Hey, Roger, what does this mumbo jumbo mean?"

Well, I'm glad you asked. Mostly, it pertains to this column being late again. It seems as though every four weeks or so another month goes by. I have a good excuse this time, as I was out in California visiting friendly manufacturers in *Silicone* Valley. I also found out that it is not named after the area just above the female sternum.

Digidesign is hot on the release schedule for Pro Tools 4.0 — which is totally rewritten from the ground up and chock full of new features. One tour around the block with the new software and you won't know how you got along before. PCI versions of the Pro Tools hardware have been shipping for a few months now, with more PCI hardware on the way. Also new control surfaces are in the works that will make Pro Tools feel like a hardware console instead of a software program. I am working on the full exclusive review, so don't change that channel.

Sonic Solutions was also on the visiting list, although they are a little further North, in Novato, about 25 minutes past the Golden Gate bridge. Sonic has developed the first authoring tools for DVD. I saw a demonstration of the system and the first commercial software to be sent to the DVD pressing plant. DVD is going to be an impressive format. Your favorite movies in widescreen format with so many alternate language tracks that it will impress even the Hippo families. (If you get that one, e-mail me. I'd be impressed.) Multiple audio tracks of up to 96 kHz 24-bit to supply all of the surround sound you can stand is what mostly caught my attention.

Record companies saw lots of catalog reissues when CDs superseded the LP. Now they can re-release everything in multichannel surround sound. I am actually thinking about doing a few of them myself. With discrete surround channels like DTS or Dolby Digital, you can pan something just where you want it and it will stay there and sound good.

Sonic has also dropped their prices for an entry-level system, so I see the workstation wars really heating up in the very near future. There will also be a full Sonic Solutions review soon.

IT GIVES ME THE JITTERS

This jitter thing just won't go away. I still hear of people who insist that their digital audio sounds better when stored on one certain brand of hard disk. I hear tales of digital audio sounding better when transferred with SDIF-2 cables (three separate cables for left, right, and word clock with BNC connectors on each end) instead of AES. A major Japanese company who shall remain nameless (JVC) says that digital audio is better if transferred over aluminum wires. They say that aluminum works better than gold or silver. Maybe they hope you will ingest the aluminum, contract Alzheimer's disease, and forget that they made these claims.

Some people say that they can hear the difference between a CD-R that was cut at 1X and a CD-R that was cut at 4X. The glass masters at the CD plant are usually cut after transferring the audio from the CD-R at 2X. These same people say that the CDs pressed don't sound as good as the ones that are pressed from a 1X transfer to the glass cutter.

The theory is that the slight jitter in the spacing of the pits on the CD cause jitter in the final digital audio output. After looking at the schematics for a number of different CD players from Sony, Fisher, Denon, Kenwood, JVC, Marantz, and Luxman, I can only conclude that *this is not possible!* If your player's output jitters because of pit spacing problems then *it is broken — get it fixed or buy a new one!*

CD player timing is derived from an internal crystal that clocks the digital audio out of memory, not directly off of the disc. This crystal provides the reference for the servo that rotates the disc. The disc turns fast enough to decode the pits and place the data in memory at the same rate that the crystal is emptying the memory. If the memory is filling up, the disc slows down, and if the memory is emptying, the disc speeds up.

Every piece of digital audio gear works exactly the same way. With a TASCAM or ADAT, if the memory is getting too full, the rotating heads slow down. In all DAWs (Digital Audio Workstations), if the memory is getting empty, the computer gets more data off the disk. With a Sony 48-track or a Mitsubishi 32-track, the tape speeds up or slows down as necessary to feed the memory bin.

And finally: Jitter doesn't matter except for A/D and D/A conversions. If the receiver is properly decoding what the transmitter is sending, and a one or a zero going in gives you a one or a zero coming out, then it makes no difference whether the jitter is less than a nanosecond or more than a fortnight. As long as the data is buffered and reclocked with low jitter for the conversion process, it just doesn't matter.

When a bit is transmitted, the receiver attempts to discern the value during the middle portion of the bit time. There can be a lot of jitter and still give you the proper value of the bit. As an example, suppose our bit clock was from 6 AM to 6 PM. If you were told to go outside and look at the sky somewhere around 12:00 and determine whether it was day or night, you could have as

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