APRIL 1993

EQ

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EQ EXCLUSIVE

THE MAKING OF AEROSMITH'S GET A GRIP

BY JOE PERRY & STEVEN TYLER

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FEATURES

BAND IN A VAN

TECHNIQUES/WORKSHOPS

COLUMNS/DEPARTMENTS

On the cover: Aerosmith by Timothy White; Freight Train Jane by Ed Colver.

You're looking at the future of affordable digital multitrack.

Today it seems that everyone's jumping on the digital bandwagon. And for good reason. It sounds great, there's no generation loss, and it's state-of-the-art. But until now it's been very expensive—or even inferior.

So when we set out to design the future—we refused to accept anything but the best. And considering that our parent company, TEAC, is the largest manufacturer of professional audio recording and data storage equipment in the world, and was the first company to make multitrack recording affordable, we've got a lot at stake.

That's why TASCAM chose the newer 8mm tape format for digital multitrack recording. It's simply better than anything else. Why? The 8mm transport is the most compact and is designed to take the beatings of the start-stop-start-stop operations that characterize studio and post production environments. And we should know, because TEAC makes transports for both VHS and 8mm. We tried them both. In fact, tests show 8mm to be superior for digital audio multitrack recording. That's just the start. The 8mm format is superior in many ways. Like "Auto Track Finding" (ATF)—an innovative technology that ensures consistent, error-free operation by imbedding important control information during the helical scan. This maintains a perfect relationship between the tracking and program signals on your tape. What does that mean? Precise editing for punching in and out as well as the ability to exchange tapes between musicians and studios without synchronization concerns.

There's more. The Hi-8mm metal particle tape cassette is sturdier and protects the tape against dust and environmental hazards. The 8mm format takes advantage of technologically superior tape that characteristically has a higher coercivity and therefore higher retention than S-VHS tapes. That's why Hi-8 is a preferred format for backup of critical digital data by computer users worldwide. And that's why your recordings will last longer on Hi-8. Even more, with up to 100 minutes of recording time, Hi-8 offers longer recording length than any other format.

We could go on. But the point is that with over 20 years experience, TASCAM has quite an investment in multitrack recording. An investment that has paid off for musicians, recording studios and post production houses worldwide. We've put this experience to work in defining the future of affordable studio quality digital multitrack recording. And you can take advantage of it now.

The DA-88 digital multitrack system

TASCAM®
Take advantage of our experience.

For our free booklet, "Are you ready for Digital Recording?" write TASCAM, 7733 Telegraph Road, Montebello, CA 90640

CIRCLE 63 ON FREE INFO CARD

The Revolution Will Be Digitized

MORE THAN A FEW YEARS AGO I was involved in a revolution. As editor/publisher of *Modern Recording* magazine, I was part of a team that felt the heat and upheavel in music and technology, and jumped into the fray.

The revolution, of course, was a creative one. Electronics manufacturers, musicians, and recordists came together to overthrow the conventions of the period. Musicians wanted cost-effective multitrack machines and manufacturers were up to the challenge. *Bang!* Out came narrow format reel-to-reel machines. *Boom!* Look out, here came portable 4-track cassette units. Suddenly anyone with talent and desire could be part of the recording world. Recording studio voodoo was out and technical egalitarianism was in. And recording would never be the same. Long live the affordable analog revolution.

Part of *Modern Recording's* basic premise was to encourage and nuture technology to keep reaching even further. Equally important was to decipher the complex "secret" world of recording for our readers. It was an exciting time; breaking publishing rules, developing new guidelines (none existed for such magazines), and getting readers to come together. Today's leading magazines, *Mix* and *Pro Sound News* didn't even exist.

Today, most of those early recording and sound upstarts have not joined the Foreign Legion, but instead are still hard at work creating music and sound. Combined with new upstarts they form a powerful, solid core of the recording industry. But for some reason, until recently, much of the industry has taken them for granted. *EQ* magazine has not. Whether in $10,000 or $400,000 project studios, we know creative musicians, producers, live mixers, et al, are still searching for answers. Many of those answers are now upon us, as part of a new revolution — the affordable digital revolution.

Digital technology and more effective manufacturing techniques are combining to provide us with amazing products, including multitrack disk-and tape-based recorders, low-cost consoles, automation systems, and software galore. Never before has there been such a high degree of sonic integrity joined with such low cost and ease of operation. All of this enlightened technology falls squarely into the collective lap of the project recordist and live mixer. Enter *EQ* into the frame. Once again there is a publication breaking new ground for an audience who believes technology is a tool and wants to know how to use that tool most effectively. No problem, *EQ* is happy to set guidelines for the new revolution. We don't plan to forget our analog beginnings, but as digital rolls in, we'll be here as your personal A/D converter.

This magazine has little interest in simply reporting what has already been created. It would rather help you to create. If you've got questions, send them in; we'll get them answered. If it's new equipment you seek, look no further. *EQ* is here to help with applications and techniques — sharing information to assist you in doing projects faster, more efficiently, and better. Talent + Tools + *EQ* = Capturing Creativity.

Welcome to Revolution Central. Keep in touch.

Hector G. LaTorre
Executive Director

YOUR SHEP HAS COME IN

Thank you, thank you, thank you for the article "The Madonna Diaries" in the December 1992 issue, featuring mix-master Shep Pettibone.

Since he first burst on the scene in the mid-1980s, Pettibone, much like Madonna, has been one to anticipate almost inherently not only what people want to hear, but to be one step ahead in implementing it with his own style and lush production.

Anyone who's ever entrusted an LP track to Pettibone for remixing has never lived to regret it. When Bananarama's manager Hillary Shaw was looking for a new sound to score them a hit in 1990, I told her she had to let Shep Pettibone do it. Four months or so later, *Preacher Man* was out and climbing the charts.

Even if you don't like club music, you have to admire and respect him for his astounding talent and for his role in setting the standard for audio production.

Bill Pardue, Jr.
Fan Emporium
Marketing Assistant

MORE SHEP, LESS SEX

Regarding your article (or centerfold) on Shep Pettibone and whats-her-name (Dec. '92) and the letter from Steven H. (Feb '93).

I read the article and found it to be informative and, in some cases, amusing. Some of the Polaroid shots were funny too, but the blatant use of the pictures of — oh what *is* her name — were really uncalled for.

So what if other people want to think of her as powerful and lick her boots (I bet she would love that). She has no place in an engineering magazine.

I stopped reading *Mix* and, as far as I know, am a charter subscriber to your wonderful publication, but *please* keep to the subject.

Matthew Alan Warren
Psycho-Kitty Recording Services
Oakland, CA

CONFLICTING DATA

As much as I respect Craig Anderton, his "Computer Quagmire" advice was suspect.

Shep Pettibone and What's-Her-Name

"Invest in emulation and bridge software." Say what? Invest in emulators? How about pork bellies? Competition has forced software houses to cross platforms all by themselves. You can open and update your Lotus 1-2-3 or WordPerfect file from the office PC or your Mac without any emulation

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Imagine a rackmount mixer...

software. And then save it on a PC format. Between these platforms, the bridge comes standard with the major applications.

"You're probably going to need more than one computer platform." Buy two entirely different computer systems *and* the software to run on them? As my punky 4-year-old is fond of saying, "No way, José." Unless Craig is speaking about owning one of the upcoming Personal Digital Assistants, like AT&T's Personal Communi-

cator or Apple's Newton. Newton's scalable new Dylan Operating System looks interesting as hell.

"Apple may not be able to hold its own against the clone tidal wave." Worry about which cheap multitrack digital format will dominate, worry about your precious DAT masters playing back 10 years from now, worry about paying your kids' college tuition, but don't worry about Apple. Fiscal 1992 net sales are over $7 billion (and rising), with a net profit of

over half a billion dollars (and rising). Apple's profit margins are almost twice those of even the premium PC manufacturers like Compaq and Dell. Nobody makes as much money selling PC's as Apple. And hey, they've got a billion and a half dollars in the bank and no real debt. Just like me.

Commodore and Atari, despite their demonstrated capabilities, have missed the big boat. Sure, you can get the job done on them, but why be a rebel in computerdom instead of being where the action is?

Craig's comment that "something faster, better, and cheaper will always come along" hits the nail on the head. His advice to wait to "commit some big bucks to a third wave [not yet available] computer system" misses the point. Forget the future. It will always be more powerful and cheaper. If you want a computer, talk with friends who use them for the same reasons you will use yours, read up on the subject, and buy the most powerful one your budget will allow.

I'd also tell you to buy a better than average monitor, and recommend you try out a Mac.

David Correia
Celebration Sounds
Pawtucket, RI

[Craig Anderton responds: Yeah, I like the Mac too, but, although some file formats are compatible across platforms, running programs is an entirely different matter. Emulation is particularly relevant to Amiga and Atari users, since for a few hundred dollars they can run Mac and PC programs, not just exchange files.

Think of multiple platforms as "buying entirely different software and the computers necessary to run them." Many programs have not been ported to multiple platforms; multimedia developers in particular need to "speak" both PC and Mac (I know several who use both). I have Mac, Atari, and Amiga, and I find the combination extremely useful.

As the article stated, Apple is doing well — but there are 9 or 10 installed PCs for every Mac, and at the last NAMM show, the trend seemed to be develop for the PC first, then the Mac. Remember, VHS beat Beta not because it was better, but because it was cheap and plentiful.

Regarding waiting, I usually give the same advice as you, but I feel we're

in anomalous times — the next wave appears so imminent that one may not be able to amortize a big investment in a second wave machine before the third wave hits and you need to jump on that to stay competitive. The only point I'd add to what you said is decide what software you want to run, then pick the computer needed to run it.]

HARP ATTACK

Re: "Not To Harp, But...," Letters To EQ, December '92.

The January 1 edition of the *Los Angeles Times* reported that criminal charges have been filed by the city attorney's office against Chas Sanford for having a recording studio in his home. It appears that in addition to "the playing field to be leveled" and "friendly persuasion," possible prison time is now one of HARP's weapons against any competition.

Albert Wiegel
North Hollywood, CA

CITIZEN BROWN

Re: "I Had a Secret," December '92 issue, Letters to EQ.

Haven't we had enough gridlock with Congress? We need more in the music industry? "Creating a set of standard guidelines for booking and client relations? Documentation for sessions?"

Get a life, Teri Piro! *Sour grapes!*

If I can get superior results on a project at someone's home studio for a fraction of the cost of a full-blown commercial studio, then maybe it's time to start dismantling those huge white elephants that went up in the '70s and early-to-mid-'80s as a result of the recording boom. Major record companies were signing lots of acts back then and had big studio budgets. We "unsigned" musicians can't afford $150 rolls of tape and exhorbitant studio clock time.

Thanks to publications like *EQ*, we struggling artists and musicians can finally put together workshops (that's what the big record companies call indies), release, publish, and promote our own products independent of the majors' bureaucracy, red tape and general gridlock.

Peter Brown
Pompano Beach, FL

EQ TIPS

EDITED BY DAVE BRODY

A BETTER FOOTSWITCH

If you trip over footswitches, want a bigger target to stomp on, or need something more heavy duty than the average switch, here's an answer: the switching mats used by stores and supermarkets to do things like open a door automatically when someone steps on the mat.

You can get these types of switches from:

• Burglar alarm supply stores. This tends to be the priciest option.

• Surplus electronic stores. You might get lucky and find something at a ridiculously low price.

• Local distributors. Tapeswitch Corp. of America (100 Schmitt Blvd., Farmingdale, NY 11735; tel. 516-694-6312) makes just about any type of switch/mat configuration you could ever need. They don't sell direct, but can put you in touch with a local sales rep.

Mats generally use a normally open (or closed) SPST switching configuration, which is compatible with most modern, microprocessor-controlled gear (on power-up, many devices check whether a switch is normally closed or open and react accordingly). Guitar amps and older effects often require a latching action, or a more complex switch type (DPDT etc.). These are not suitable for use with switching mats unless you know how to build an appropriate interface.

Solder the right connector and cable to the wires coming out of the floormat, and you're ready to go. By the way, this type of switch works particularly well for punching in with tape recorders.   —*Charles R. Fischer*

RIDING THE WAVELENGTHS

When miking mid- to low-frequency instruments — particularly for multi-sampling — remembering a little physics can buy you a lot of "punch." Try locating your microphone one wavelength out from the sound source. This magic number is conjured up by dividing the speed of sound (about 1130 feet per second at 70° F) by the fundamental frequency of the instrument you're working with.

In other words, if you want to sample a horn playing an A=440, start with your mic about two and a half feet out from the bell of the axe (a tape measure really helps). Tweak to compensate for temperature and humidity (the speed of sound increases by 1.106 feet per second for each degree of temperature Fahrenheit). When you adjust your gain (or normalize your amplitude in software) you'll find that the "power" lost by single distance mic placement is restored. This happens because overtones are whole number multiples of the fundamental so, at the distance of the fundamental wavelength, all are in phase. The method will work as well for miking amplified instruments as it does for "unplugged" ones.

If you're not sampling notes but are instead recording musical passages, optimize your mic placement for the fundamental of the tonic of the key the instrument is playing in. Since other notes in the line will likely be "harmonically related" to the tonic, the instrument will sound strong throughout.

There are some practical limits to this business. As the frequency goes up the wavelength gets increasingly tiny until — at about 2 kHz (six inches) — you start to really hear the proximity effect of your microphone and it ain't worth doin' no more.

Similarly, at the other end of the frequency spectrum, the distances between mic and source become so huge that your gear and room are really put to the test. A bass trombone blasting out a low B-flat produces a fundamental wavelength of about 20 feet! Because sound level decreases by one half as you double the distance from the source, your mic pre-amp gain pot is going to get a real workout. And you're going to start hearing your room acoustics big time!

If nothing else, miking by wavelength will tell you some important things about the basic sound of your studio. And, coincidentally, you will discover one of the secrets behind some of the legendary British drum sounds. **EQ**

World Radio History

EQ&A

WHAT ME NORMAL?

Q *What are "normalled" and "half-normalled" patchbay connections? Can you include a diagram?*

Eugene Pereira
Amsterdam, Holland

A Jackfields are usually arranged in horizontal rows, each row having an equal number of jacks. Normalling refers to the signal flow path between pairs of jacks. It means that normally a signal will automatically flow between jacks without the necessity for a patch cord. Outputs (upper row) are normally wired to inputs (lower row). If no normalling is present, then every connection between jacks will require a manual patch by means of a patchcord.

One would normally like to have signal flow between microphone lines and microphone inputs, or between multitrack outputs and tape inputs, without having to use dozens of patchcords. These jacks are typically normalled together. When no plug is inserted in either of the paired jacks, signal continues to flow from one to another. When a plug is inserted in either jack the signal flow is interrupted, thus allowing for other sources to be inserted.

Half-normalling is commonly used in consoles because it provides ease of routine connection plus, flexibility when required. Insertion of a plug into the output jack does not cut signal flow, while insertion of a plug into the return or the input jack will cut off the original signal. This allows outputs to feed more than one destination, as well as allowing new sources to be inserted into outputs.

(See included images of half-normalling and signal flow, based on images created by Jon Gaines, Gaines Audio.)

Lewis Frisch
U.S. Press Officer and
Regional Sales Manager
Amek/TAC

TAKIN' IT TO THE MACS

Q *I have a fairly sophisticated Macintosh setup with Digidesign's SoundTools and ProTools, Opcode's Studio Vision and Mark of the Uni-corn's Performer. My MIDI interface is an Opcode Studio 5. My question is: How can I get Performer to work properly with Studio 5?*

Jeff Zinga
Springfield, VT

A Because you are using Studio Vision (and therefore have a Studio Setup), you are already halfway there. In order to get the Studio 5 working with non-OMS programs (such as Performer), you must first go to the "Compatibility Setup" option in your OMS Studio Setup program (under the Studio 5 menu bar). In this setup you have a choice of emulating a MIDI Time Piece and/or a Standard Interface.

Performer users will want to check "Emulate MIDI Time Piece." At this point you will have the option of separately routing the Ins and Outs of your MIDI gear to comply with the emulation mode. I suggest using the "Auto" button. This button automatically routes all of your MIDI devices, based on your Studio Setup routing (In and Out). It is a very quick and

WHAT ME NORMAL? Signal Flow

WHAT ME NORMAL? Half-normalling provides ease of routine connections.

Cheers subside.
Anticipated silence.
Fingers tense.
Lips to mike. All at stake.
But in control.
With the biggest name in rock'n roll.
45 years of tried and true.
Performance art.
For this one moment.
The moment of truth.
Fender Pro Audio.

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Fender

ELECTRONICS

efficient way to match your MTP settings to that of your Studio 5 (which should ultimately match, anyway). Also be sure to check that your clock speed is set to either 1 MHz or "Fast" mode. This setting should be the same as the clock speed in Performer. Make sure that in Performer, under the Basics Menu, you go to "MIDI Interface" and check "MIDI Time Piece" for each port on which a Studio 5 appears.

David Frangioni
Frangioni Enterprises

TWO FOR ONE

Q *Regarding the Lexicon LXP-15 review (EQ, December 1992). I use the LXP live, and a computer is used to send MIDI program changes to several MIDI devices. The LXP-15 seems to have a sluggish response to program changes, forcing me to edit sequences in order to have the correct program on time.*

Also, the LXP-15 sometimes picks up the program number of another device when sending program changes at the same time! (Every device in my system has its own MIDI channel.) Did your reviewer overlook these problems? Can you help me out?

Vincent Koek
Voorschoten, The Netherlands

A The first problem you describe (sluggish program response) is common to most digital processors — they must load a new program into the DSP. Until extremely high-speed microprocessors become inexpensive, this process will take at least a second or two in the affordable effects processors. You may have unrealistic expectations for the time required to make algorithm changes within a song.

On your second point: Of course I used the LXP-15 with other processors, sending each of them separate MIDI program changes. It's hardly something a comprehensive review could overlook! The fault you described did not occur in my unit. I used the 15 over several months with many different sources of program change messages. Perhaps you have a faulty unit, or perhaps another device is echoing program changes to the LXP-15's MIDI channel. Some devices will send a MIDI program message out whenever they change programs.

Wade McGregor
Contributing Editor
EQ

Steve Frankel from Lexicon also responds.

A Regarding the "sluggish" response time: We, as manufacturers, are fighting to make this as quick as possible. When you load a new algorithm (set of instructions) into a microprocessor (regardless of the manufacturer), you must "flush" the current algorithm, then load the new one. This takes time (approx. 150-800 ms). The LXP-15 may take a few milliseconds more due to the sheer complexity of the code that creates our reverb. This is why the 15 sounds so great. As far as the sending of multiple program changes, the LXP-15 does in fact respond to only one MIDI channel at a time, unless of course you set it to "Omni." Turn the PAGE knob to Sys(tem) to define the "MIDIchn" (1-16 or Omni).

For those of you you have our MRC MIDI Remote Controller, a computer MIDI sequencer, or even a keyboard with programming capability, and want to dump all your User Registers for storage, try this: Transmit the following string of Hex bytes to the MIDI IN of following Lexicon LXP processors (set to MIDI CH. 1):

LXP-1: F0 06 02 30 64 F7
LXP-5: F0 06 05 30 64 F7
LXP-15: F0 06 06 30 64 F7

Please let us know if there are any additional questions. International customers may contact one of our many distributors. We have distribution in every country in Western Europe and Scandanavia, several in Eastern Europe, Canada, Australia, the Far East, and the Middle East. US users (or foreign customers with big budgets) can also find us on CompuServe MIDI Forum B – Library 7 (Type GO MIDIBVEN).

In The Netherlands: TM Audio, Zomerweg 14, Ijsselstein Holland BV 3402. Tel: 31-3408-70717; Fax: 31-3408-70677.

Steve Frankel
Product Specialist
Lexicon, Inc.

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EQ PRODUCT VIEWS

DAT'S THE WAY

Panasonic's newest DAT machine is the SV-3200, a unit that, according to the company, is ideal for the project studio. The SV-3200 features 44.1 kHz/48 kHz sample rates (analog input), 1-bit A/D and D/A converters, a shuttle wheel with dual speed range and 400x-normal high-speed search. The 1-bit, 64 times oversampling A/D converters with antialiasing have reduced zero-cross distortion to assure clear audio signals at low and high levels. A high-resolution, 1-chip LSI 4-DAC system minimizes distortion and enhances linearity at low signal levels to help ensure quality playback. IEC 958 consumer format digital inputs and outputs (coaxial and optical for both) provide direct interfacing with CD players, digital audio workstations and other components. For complete details, contact Panasonic Professional Audio Systems, 6550 Katella Avenue, Cypress, CA 90630. Tel: 714-373-7278. Circle EQ free lit. #101.

LYNX IN THE CHAIN

TimeLine has introduced the Lynx-2 timecode module. Lynx-2 is completely compatible with all existing Lynx systems, including the Lynx Film Module, the Lynx Keyboard Control Unit (KCU), and the Lynx System Supervisor. Features include a wideband timecode reader, the capability to generate all timecode types and MTC, a 16-character dot-matrix timecode display, a built-in serial machine control interface, a built-in Gearbox processor that permits X-Frame and variable speed sync, a remote front panel option, a plug-in Film Processor option card, a plug-in VITC option card, and expanded front panel display status. For more information, contact TimeLine, Inc., 2401 Dogwood Way, Vista, CA 92083. Tel: 619-727-3300. Circle EQ free lit. #102.

SYNTH YOU ASKED...

Samson has introduced its new flagship model wireless mic — the UHF Synth Series. This multichannel system features an RF-level display in the front panel, making it easy to select the clearest frequency in crowded wireless environments. The visual display with multistage LEDs depicts the exact amount of RF interference on any given frequency. The Synth Series offers 74 available frequencies in both the receiver and transmitter — with 11 available for simultaneous use. New circuitry design is combined with dbx noise reduction for optimum performance. The Synth Series is available in single and dual receiver true diversity formats (both in one rack-mount space). Features on the belt packs and hand-held transmitters include group and channel selection, a mute switch, a sensitivity control, battery indicator, and an on/off switch. For more information, contact Samson Technologies Corp., P.O. Box 9068,Hicksville, NY 11802. Tel: 516-932-3810. Circle EQ free lit. #103.

DAT-URATION

Sony's new DTC-A7 DAT recorder has been designed with musicians and project recordists in mind. You can record analog signals at either 44.1 kHz or 48 kHz. At 44.1 kHz you can produce master tapes suitable for CD mastering. In long-play mode, recording time is doubled by using a 32 Hz sampling frequency. The DTC-A7 features 1-bit A/D conversion and advanced HDLC D/A converters. The unit also records absolute time for continuity across consecutive audio segments. The DTC-A7 retails for less than $1000 and fits in a standard 19-inch rack space. An infrared wireless remote is included. For the whole story, contact Sony Corporation of America, Sony Drive, Park Ridge, NJ 07656. Tel: 201-930-6432. Circle EQ free lit. #104.

TO DI FOR

Following up on the success of Tech 21's Sans Amp products, the company has just introduced the Sans Amp Bass DI for bass players looking for direct box convenience without sacrificing the warmth, presence, and punch of a properly miked bass amp. The manufacturer says

that the DI focuses on the most desirable industry-trademarked bass amp sound and provides flexibility within that particular sound spectrum. Trimmable internal controls adjust clarity, tonality and gain structure, which can be adjusted to range from crystal clear to heavy overdrive. Internal controls include Presence, Drive, Blend and Level. External features include Input, Parallel Output, Ground Connect switch and Active XLR switch. The 9V battery-powered unit can also be used as a transparent active DI. The Sans Amp Bass DI retails for $195. For the full story, contact Tech 21, Inc., 1600 Broadway, New York, NY 10019. Tel: 212-315-1116. Circle EQ free lit. #130.

GO APE

Bruel & Kjær has introduced a new series of Acoustic Pressure Equalization (APE) adapters for their 4003 and 4006 omnidirectional microphones. The new APE series consists of the APE 40, APE 30, and the APE CYL. Based on the original APE 50, the new adapters function as both spatial and spectral equalizers. The APE adapters are passive acoustic processors, employing diffraction technology to modify sound reception near the microphone diaphragm in order to change the mic's frequency and polar response. While the original APE 50 increases front-to-rear hemisphere sound reception for frequencies above 1 kHz, the APE 40 and APE 30 adapters increase directionality starting at 2 kHz. While the APE 50 provides increased frequency response at the 3.5 kHz range, the APE 40 delivers enhanced response at 4-5 kHz, and the APE 30 at 5.5-6 kHz. Rounding out the APE series is the APE CYL with enhanced frequency response at 500 Hz, 2 kHz, and 4 kHz. A complete set of B&K APE adapters in a small case is available as the APE 3/6 kit. Four stereo pairs of adapters in a single case are available as the APE 3/6-2 kit. For more information, contact Bruel & Kjær, c/o TGI North America, Inc., 300 Gage Ave., Unit 1, Kitchener, Ontario, Canada N2M 2C8. Tel: 519-745-1158. Circle EQ free lit. #131.

READ ALL ABOUT IT

Random Access Audio is a comprehensive new book that covers all aspects of computer-based editing and hard disk recording. Written by EQ contributor David Miles Huber, the book covers sampling and hard disk technology, practical editing techniques, digital synchronization, digital audio transmission, DSP, and MIDI. Information and relevant applications to multimedia production, music, video and the broadcast media are also thoroughly discussed. The book costs $26.95 and can be ordered by contacting Sams Publishing/Prentiss Hall, 11711 N. College Ave., Carmel, IN 46032. Tel: 800-573-2629. Circle EQ free lit. #132

SO NU?

Lexicon's NuVerb puts the company's digital signal processing technology into a card that installs in a Macintosh NuBus slot. The NuVerb card was designed to integrate with Digidesign's TDM digital audio bus system, allowing future direct connection with the Pro Tools or Sound Tools II systems. NuVerb can also work as a stand-alone processor with an AES digital I/O for connection with existing digital systems. NuVerb provides automation of parameters via software connected to a timecode source (i.e., MTC or DTL). In its single machine mode, NuVerb runs two proprietary DSP processors as a single unified engine. Additionally, the dual machine architecture allows the two processors to be split, allowing two other modes: Dual Mono and Cascade. For complete information contact, Lexicon, Inc., 100 Beaver Street, Waltham, MA 02154-8425. 617-736-0300. Circle EQ free lit. #133.

EQ PRODUCT VIEWS

PERFECT PITCH

Yamaha has introduced the SPX990 simultaneous multieffect processor. It's one of the first signal processors with 20-bit A/D and D/A conversion. The SPX990 features a card slot that accepts optional RAM memory cards, allowing transfer of programs and 100 user memory locations. The unit also offers all electronically balanced inputs and outputs. Improvements over the SPX900 include more precise pitch change through a new dynamic waveform analysis technique. For the full story, contact Yamaha Corp. of America, P.O. Box 660, Buena Park, CA 90622. Tel: 714-522-9011. Circle EQ free lit. #109.

KEEPING ON TRACK

Fostex has upgraded its X-28 multitracker with the introduction of the X-28H. The X-28H adds high-speed tape operation to the features of the X-28. The device is a 4-track cassette recorder that provides an 8-input, stereo-output mixer; 4-track simultaneous record capability; an autoplay function; and a backlit LCD in addition to dual speeds. By simply pressing rehearsal and stop buttons, you can alternate between normal tape speed and high speed. The X-28H is also available as part of a session kit that includes a recorder maintenance kit, blank tape, and microphone, cable and footswitch. To get on track, contact Fostex Corp. of America, 15431 Blackburn Ave., Norwalk, CA 90650. Tel: 310-921-1112. Circle EQ free lit. #110.

Suggested Retail Price $249.00 *

EQ. Clean and Precise. The right EQ to fix a track or shape a mix. Or to flatten the response of your studio monitors.

Introducing the Alesis M-EQ 230. The only EQ you should put between you and your music.

Featuring *two* 30 band channels for incredibly precise control. Each band is laser-trimmed to 1/3 octave ANSI/ISO centers, and features ±12dB of boost or cut so you can really dig in where you need to. Plus, to protect your speakers if power is interrupted, we've included Auto Power Muting.

And thanks to our exclusive Monolithic Surface Technology™ you get two channels instead of one in a one space 19" rack. For only $249.

MACKIE'S BACK IN TOWN

Mackie unveiled its new 8-Bus Series recording/PA consoles at NAMM. The initial production run in March is exclusively of the 24 x 8 configuration, with 32- and 16-input versions to become available shortly thereafter. Each 8 Bus Series console features in-line monitoring, 16 tape outputs (two per subgroup), internal 116 dB dynamic range, two completely assignable headphone monitoring systems, discrete low-noise, high-headroom mic preamps on all channels (including +48V phantom power), 100mm precision network faders, sealed rotary pots, BNC lamp connectors, and steel construction. An optional meter bridge is available. Retrofittable mute and level automation are currently in development. The channel strips contain in-place stereo solo, channel metering, 4-band EQ, MIX-B (monitor system), EQ SPLIT assigns HF/LF EQ to MIX-B, and six aux sends. And get this feature: the price is $3995. For more information, contact Mackie Designs, 12230 Woodinville Dr., Woodinville, WA 98072. Tel: 800-258-6883. Circle EQ free lit. #111.

GET ON BOARD

The new Antex Z-1 sound board has been designed for professional applications. It is the first DSP-based 16-bit stereo board to offer multiple compression formats, states the manufacturer. Real time 4:1 stereo compression is provided for the following formats: ADPCM, Microsoft ADPCM, and DVI ADPCM compressed formats. The Z-1 also offers 3.73:1 compression using CD-ROM XA/CDI. Files can be compressed at data sampling rates from 7.35 to 50 kHz — in software-selectable increments. The heart of the board is a 20 MIPS programmable Texas Instruments TMS320C2 digital signal processor. Texas Instruments currently lists an array of public domain algorithms for special digital effects such as delay, reverb, phase shifting, and more. For more details, contact Antex Electronics Corp., 16100 South Figueroa St., Gardena, CA 90248. Tel: 800-338-4231. Circle EQ free lit. #112.

EQ PRODUCT VIEWS

AMPLE SAMPLES

The new DSM 752 sampling mixer from Peavey was developed with design input from many professional DJs. The resultant sampling mixer provides higher "bit capability" and frequency response numbers. Most typical DJ mixers sample at less than 12-bit accuracy with a frequency response of between only 8 and 15 kHz. The new DSM752 delivers full 16-bit sampling at 44.1 kHz, resulting in samples of CD-quality, according to the manufacturer. Samples can be recorded and played back as one 8-second, two 4-second, or four 2-second segments. Every music channel on the DSM 752 features dual level-trimming controls. When playing a sample, the music level can be cut automatically or faded manually. The mixer also features a field-replaceable cross-fader, a total of six music inputs, level trim controls on all inputs, and more. To sample it for yourself, contact Peavey Electronics Corporation, 711 A Street, Meridian, MS 39302. Tel: 601-483-5363. Circle EQ free lit .#105.

LOTS OF SPIRIT

Folio is the newest addition to Soundcraft's Spirit line of consoles. It is also the most affordable, starting at under $500. The Spirit Folio is available with six mono and two stereo inputs in a 10x2 format, or with eight mono and two stereo inputs as a 12x12. The 12x12 is also available in a rack-mount configuration. The Folio is designed to be adaptable to a variety of applications including live performance, home recording, smaller venues, and clubs. High-quality mic inputs, balanced inputs, and a dynamic range of 108 dBu also make Folio ideally suited for direct-to-DAT location recording. Every input and master output on Folio has a 60mm linear fader for smooth, precise control of the signal. Additionally, each mono channel includes a mic input with Phantom power, 3-band EQ with a swept midsection, and a

high-pass filter as standard. Two full-feature stereo inputs can be used as effects returns or for stereo sources such as keyboards. For more information, contact Soundcraft USA/JBL Professional, 8500 Balboa Boulevard, Northridge, CA 91329. Tel: 818-893-8411. Circle EQ free lit. #106.

TRANSFORMATION INFORMATION

Toroid Corporation has introduced custom-designed transformers with "zero-noise" levels for amplifier and preamplifier applications. Toroidal transformers provide low profile and light weight as well as low audible noise. They also provide low magnetic stray field emission levels. Toroid's quiet transformers are available with or without encapsulation, with epoxy-potted center and threaded inserts, or potted inside a metal can with or without mounting tabs. Standard hardware is available up to a 5000VA power rating. Custom-designed transformers for chassis mount are offered up to 12,000VA single-phase, 60 Hz operation. For more information, contact Toroid Corporation, 608 Naylor Mill Road, Salisbury, MD 21801-9627. Tel: 410-860-0300. Circle EQ free lit. #107.

RAISE YOUR VOICES

The Alesis QuadraSynth is a new 64-voice, 76-key Master Keyboard that includes an ADAT multichannel digital interface and employs Alesis QS Composite Synthesis, which is a combination of additive and subtractive synthesis. It features 16MB of onboard sample ROM, with many new samples from Alesis's own sample library. A built-in QS Parallel Matrix Effects™ section, comprised of four completely independent signal processing busses, provides fully programmable multieffects. The manufacturer states that QuadraSynth is the first keyboard ever to offer true 64-voice polyphony, which allows for extremely complex sequencing without the problems of voice robbing. The QS Program Mode™ has 128 factory presets and 128 user programs, so you can program up to seven simultaneous effects that can be independently assigned to any of the four effects busses. There's more, so contact Alesis Corporation, 3630 Holdredge Avenue, Los Angeles, CA 90016. Tel: 213-467-8000. Circle EQ free lit. #108.

WHAT ABOUT BOB?

Bob Ludwig

"I figured if I was jumping
off a cliff anyway,
I may as well make it a
high cliff."

What's masterer Bob Ludwig been up to since he left New York City's Masterdisk late last year? Wonder no more: He's gone from the bright lights, big city and has moved up to Portland, Maine, where he has started his own mastering studio: Gateway Mastering Studio, Inc.

"Wherever I was working," explains Ludwig, "whether it was at Sterling Sound or Masterdisk, everybody thought I owned the place. I figured 'Why not?' My partner, Dan Crewe, moved up to Maine and my parents have lived up there for 25 years. It was a place I had always wanted to live and I figured if I was jumping off a cliff anyway, I might as well make it a high cliff."

Ludwig has had a temporary room open since January and has mastered both David Bowie's and Bruce Hornsby's latest efforts there. "It wasn't in the business plan to be running full tilt right away," says Ludwig. "But I've had a higher percentage of clients attending sessions here than in New York."

One thing that Ludwig could accomplish in Portland that would have been impossible in the tight confines of New York office buildings was the construction of Gateway's floating studio. "Though the floor of the studio is floating (separated from the room it is in)," remarks Ludwig, "the Duntech Sovereign speakers I have are on concrete pedestals that go down through the floor of the studio, through the floor of the room, all the way down to bedrock — now that's hard to do on the fifth floor of a Manhattan high rise.

"I set up the speakers this way because sound travels through hard surfaces quicker than through air, so normally, when you have the speaker set on the floor, some of the frequencies are transmitted to you faster through the floor than the ones coming through the air. This causes a minute, but measurable, amount of blurring. With the concrete pedestals not actually touching the floating floor, there is no transmission through solids. So the first sound you hear is pure sound from the air."

For the full scoop on this mastering facility, look for a future article by Bob Ludwig himself in these very pages.   —*Tony Savona*

SEE ME, HEAR ME, FEEL ME

David Gibson

"Everything you need will
be right in front of you,
floating directly
between the speakers
in the studio."

Medical professionals use it. The Pentagon's war division swears by it. Now, it seems as if the recording industry is getting in on the action. Virtual Reality, the buzzword of technophiles all across the world, is about to find its way into the lexicon of studio-speak. And if any one person should be credited as the visionary, it's David Gibson — mixer, educator, and, now, virtual reality pioneer.

Gibson began his foray into virtual reality by fostering it as an educational tool within his school, the California Recording Institute. By using dots on a Mac screen to represent parameters within each piece of studio equipment, Gibson was able to demonstrate the different structures of mixes that exist for varied styles of music. Soon the dots evolved into see-through spheres, and a detailed system for representing both instruments and effects was born. In its current condition, Gibson's "Virtual Mixer" uses boxes to represent reverberation, while oblong spheres signify an instrument that has been spread in stereo with fattening. The images can also be used as controllers for digital audio workstations, MIDI, and 100 percent automated mixers.

"Each image is designed to show how much space a sound takes up in the world of imaging," explains Gibson. "It demonstrates technical information about 'masking' and 'equalization,' so you can better utilize your work space and see where the music is."

Currently, enthusiasts use the Virtual Mixer by viewing images via a Macintosh monitor, but Gibson states that it's only a matter of months

before mixers and producers don 3-D glasses and interact with a visual display that is completely airborne. Says Gibson, "Everything you need will be right in front of you, floating directly between the speakers in the studio. And with the Data Glove, you'll be able to reach up, grab images and move them left and right for panning, closer and farther for volume, and up and down for pitch." Impressive, but does it do windows? In a way...

"With the Virtual Mixer, everything in the studio is in a single window," continues Gibson, "so you don't have to go from one window to another in order to access the mixer or set different sound effects." By keeping everything in one visual framework, the user is able to see when a large mix, such as an orchestra or a Pink Floyd anthem, is overcrowded with sounds. Images flash, enabling users to see the density of the mix at any given moment and make the appropriate settings to audio tools, such as compressor/limiters and noise gates.

So what began as an experiment in education has quickly evolved into a commercial breakthrough for David Gibson, whose Virtual Mixer is quickly developing into a feasible alternative to the tried-and-true contemporary mixing procedures. Even recording artists such as Peter Gabriel, Pete Townshend and Ozzy Osbourne have had their crack at the Virtual Mixer.

And what does this modern-day mixer envision for a recording age further on down the road? "Holograms," replies Gibson enthusiastically. "When you've got images floating around in front of you with no 3-D glasses on— that's when you'll really get goosebumps." —*Jon Varman*

GREG DROMAN with wife Marilyn Martin and his record label advance.

OTHER PEOPLE'S MONEY

Greg Droman

One day a limo that must've been a mile and a half long pulled up in front of my house and the president of Atlantic Records got out and came in to check out his ADAT studio.

Where do you get the money for your project studio? Producer/engineer Greg Droman managed to get by with a little help from his friends — and a record label.

Before dreams of label-funded studio gear go rumbling through your mind, remember that Droman is no stranger to the recording industry and certainly no stranger to Modern Records, the division of Atlantic Records that recently helped him in his endeavor.

"My wife, Marilyn Martin, did a song with Phil Collins for Atlantic Records a few years ago called 'Separate Lives,'" Droman explains. "Since then we've had a good relationship with the company and she is now recording a new album for a division of the company called Modern Records, the label on which Stevie Nicks records, among others."

It was because of this new album that Marilyn formed a writing partnership with Scott Plunkett, arranger

DAVID GIBSON mixes with a wave of his hand in his virtual project recording studio.

and keyboardist with Don Henley. The pair joined Droman in a tiny bedroom armed with a Fostex 8-track analog machine and created about eight demos that they played for the president of Atlantic Records, Doug Morris. He liked the demos so much that he introduced them to Paul Fishkin, president of Modern Records. "In fact," Droman adds, "they actually thought the demo sounded good enough to be released as the record."

"We explained to them that we needed a room where we could use real instruments," Droman continues, "because we wanted a more organic sound than you can get with drum machines and synth modules. They agreed and we were off and running."

Droman, who has worked with Fleetwood Mac, Jude Cole, and Hall & Oates, and who is also nominated for a Grammy this year for work done on Lindsey Buckingham's lat-

est solo album, already had a good start in his home studio. "Being an engineer," Droman states, "I had some equipment I would take with me on the jobs I did, like my API Lunchbox, so I wasn't starting from scratch."

Now his studio holds three Alesis ADAT's ("I'd like a fourth," Droman adds), a Sony PCM 2300 DAT machine, an API 560 graphic EQ and 512b mic pres, dbx 160 compressor/limiters, and a Summit TLA-100A tube limiter. For a console, Droman uses a Mackie 1604 and a Boss 16-channel strapped together. "I'm waiting for Mackie's 8-bus 32-channel board to become available," Droman remarks. "I've spoken with Greg Mackie and I know his products — they're the only ones I'd buy without hearing — based on how good the 1604 sounds."

The room itself, located in Droman's Sherman Oaks, California, home, is rather unique. "It's got so many angles, it's ridiculous," says Droman. "But we don't have any standing-wave problems." The studio's overdub room is sizable, about 18' x 7', and features a marble floor.

It's a good thing that Droman spent the money wisely: "One day a limo that must've been a mile and a half long pulled up in front of my house," Droman recounts, "and the president of Atlantic Records got out and came in to check out the place. Fortunately, he was thrilled with what we had done and loved what we had recorded so far."

Sounds like a dream come true, right? Droman has a few caveats: "I tried to keep the cost down because I don't think record companies are in the business of building recording studios. They just want a great record and they don't care how you get it. In fact, it's quite possible that they could be concerned that you're going to try to do it at home and it's not going to work out — that it may be substandard compared with what you could get at a big studio. It's safer for them for you to go to a larger studio, or so they believe. From our situation, we'd already done some records the traditional way and we both had track records with Atlantic. We could show that we knew what we were doing."
—*Tony Savona*

GUILTY.

AS THE BUILDERS OF D&R mixing consoles, others have accused us of being rather zealous. Maniacally dedicated. Over the top. Passionately committed to crafting the finest consoles at any price.

We plead guilty as charged.

After all, our shameless vendetta against smeared sonics leads us to the highly unreasonable length of phase-correlating every audio stage. We're the crazed console crafters who — in our unstoppable desire to eliminate RFI and other noise — starground every circuit on every console, with the aid of a custom-welded steel chassis.

And yes, we're the ones responsible for "high-def" EQs, floating subgroups, and other pioneering features which show so little regard for ordinary designs.

While overcoming the challenges of physics and the temptations of mediocrity, our unreasonable standards deliver what many consider to be the best consoles on the market. At the most reasonable prices.

You might think life could be lonely when you're guilty of a passionate pursuit for perfection. But along the way, we've met thousands of others who understand our intolerance for anything short of excellence. They are the thousands who have asked us to hand-craft them a D&R.

So if you've been accused of ridiculously high standards for your work, let us reassure you

Exhibit A: To find evidence of our remorseless commitment to sonic integrity, look no further than the new D&R Triton. Its transparency, flexibility, and unparalleled support put it in the same league as our flagship, the D&R Avalon.

that you're not alone. We're here whenever you need us. And we understand completely how you feel.

Exhibit B: When Digidesign was judging new consoles to use with their own 20- and 16-bit digital recording & editing systems, they knew the board would have to be good. Very good. And quiet. Very quiet. Their verdict? The D&R Orion.

D&R handcrafts consoles for recording, live sound, theatre, post-production and broadcast. Whether you own a world-class facility or a serious project studio, there's a D&R for your needs and budget.

CIRCLE 21 ON FREE INFO CARD

The PIG Man

STUDIO NAME: PIG Studio

LOCATION: The studio was built in the Beverly Hills living room where Charles Manson's family murdered actress Sharon Tate

MAIN MEN: Trent Reznor (Nine Inch Nails), Bill Kennedy (engineer), Sean Beavan (engineer), Chris Vrenna (studio assistant), and Maise (dog)

PROJECTS RECORDED: Nine Inch Nails

CREDITS: Trent Reznor As Nine Inch Nails: Pretty Hate Machine, *Broken* EP, *Fixed* maxi-single. As producer/remixer: Megadeth's "Symphony for Destruction" and "Sweating Bullets," Queen's "Stone Cold Crazy" and "Tie Your Mother Down," Machines Of Loving Grace's "Burn Like Brilliant Trash"

CONSOLE: 56-input Amek Mozart (all Amek MZ-15RN modules) with Supertrue Automation

KEYBOARDS: Prophet VS, Oberheim Obermoog, Xpander, Ob-8, Waldorf MicroWave, MiniMoog, Sequential Circuits Pro-One, ARP 2600 and Odyssey, Kurzweil K2000, Mellotron

SAMPLERS: Akai S1100 (2), Akai S1100 EX (2), E-mu Emax, Ensoniq Mirage

COMPUTERS & SOFTWARE: Macintosh Quadra 900 running Opcode's Studio-Vision, and Digidesign Pro Tools (8 channels) and TurboSynth, Alchemy, Unisyn

RECORDERS: Studer A800 MKIII 24-track (2)

DAT: Panasonic SV 3700

MICROPHONES: AKG-414, Shure Beta 58's and 57's, ElectroVoice RE20, Crown PZMs, Neumann U87's, Fostex stereo mic

OUTBOARD GEAR: Lexicon 300, PCM 42 (2), PCM 41, Eventide H3500 Ultraharmonizer and H3000SE Ultraharmonizer, Yamaha SPX1000, Lexicon LXP-1 and LXP-5, Zoom 9030, T.C. Electronics 2290 digital delay, Neve stereo compressor, Solid State Logic stereo compressor, Focusrite 131 compressor, Focusrite 115HD stereo EQ (2), Pultec EQs, Summit EQ

STUDIO NOTES: Says Trent Reznor, the musician/producer who *is* Nine Inch Nails: After much thought on the subject, I decided to invest the budgets for the next several Nine Inch Nails albums in an almost fully equipped project studio in which we (I, me, whoever) could write, record and mix. The design is fully modular so I can move things around as my mood dictates. Emotions change from record to record, so it makes sense to surround yourself with an environment that reflects your current state-of-mind. All of this is in theory, of course.

PRODUCTION NOTES: The secrets to getting interesting sonic distortion are held in old Neve EQs. I also like Focusrite EQs and Pultecs. The outboard effects that give me the most insane sounds are the Eventide H3500 and the Zoom 9030. The Zoom is absolutely excellent for strange and eerie vocals. For writing, composing, and digital editing, I have a modified Macintosh Quadra 900 with 8 channels of Pro Tools and 2 gigabytes of disk space. I run Opcode's Studio Vision, which has completely changed the way I write and record music (for the better, I think).

One of the nicest pieces of gear I have ever had the pleasure of dealing with is the Oberheim Xpander. It plays an integral part in making the music go to work. Another source of endlessly interesting sounds is Digidesign's TurboSynth, which is great, but sorely in need of an update. Some of the samplers and synths that really give Nine Inch Nails its punch are the OB-8, Mirage, Arp 2600 and an old, famous Mellotron. **EQ**

TRENT REZNOR WENT LOOKING FOR HIS
INDUSTRIAL MUSIC HEAVEN
HE FOUND HELTER SKELTER

THE PIG CREW: (From left to right) Maise, Chris Vrenna, Bill Kennedy, and Trent Reznor.

Jaquette Required

NAME: Bazzbo Productions
LOCATION: La Verne, CA
MAIN MEN: Tim Jaquette, chief engineer; Bob Somma, guitarist, arranger, producer
CREDITS: Tim Jaquette, touring and recording bass player, has been engineering for over 17 years, spent eight years as chief engineer at Studio Masters in West L.A., and developed a following that includes such artists as Paula Abdul, the Whispers, L.A.Reid, Babyface, Karen White, Pebbles, and Howard Hewitt. Recently he has developed an interest in video and film engineering, with such credits as *Young Riders* and the new Frank Marshall TV series *Johnny Bago*.

Bob Somma is an L.A. session guitarist with platinum record credentials. He has worked on many projects for Disney, Fox, MCA, and Virgin Records.
CONSOLE: D&R Orion 32 x 32 x 8 x 2
TAPE MACHINES: Sony APR-24 2-inch 24-track, Ampex ATR-800 1/4-inch 2-track, Panasonic SV-3700 DAT, Sony TC WR 690 dual cassette deck
MONITORS: Meyer HD-1's, Yamaha NS10M's, JBL 4406's
COMPUTERS: Macintosh SE-30 with 20 MB RAM, 40 MB hard drive; Sound Tools Sound Designer II with 65 MB hard drive
AMPLIFIERS: Yamaha P2700 (monitors), Carver power amp (headphones)
KEYBOARDS: Yamaha DX7, Kurzweil 1000, Yamaha KX88, Korg M1R, Roland D550
MICS: AKG Tubes, AKG 414's, AKG 452's, Neumann U87, Neumann KM 84's, Shure SM 57's, Sennheiser MD421's
VIDEO: Sony V9800 3/4-inch
OUTBOARD GEAR: Summit stereo tube limiters, UREI 1176 limiters, API mic preamps. Proteus 1, Akai ME20P patchbay
EQUIPMENT NOTES: Jaquette says: "We chose the Orion console after going out and doing some careful shopping around. It's very quiet. You can crank it as high as it goes and you don't hear residual electronic noise. A great feature is the EQ split capabilities, and it has a boost or cut of 16 dB on all four sweepable bands of EQ. I'm used to working on the Neve 8128 series and this console's functionality is very similar, yet it fit well within our budget, allowing us enough money for our other start-up costs.

"I really like the sound of the Summit stereo tube limiters. I also use 3M 996 tape, which I run at +6. They say 'Oh, +9!,' but that tends to sound too much like tape compression."
PRODUCTION NOTES: "We're planning on using our studio as a production studio only, therefore we don't intend to go to a wall-size rack. We're not going to be doing any real mixing here, just tracking. Our emphasis is on having the best equipment to get tracks down with. **EQ**

DUELING BAZZBOS: Tim Jaquette (right) and Bob Somma

Photo by Ed Colver

It's Back: The Second Annual

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We've Always Talked Performance.
Now Let's Talk Price.

World Radio History

We Came, We Saw...

...We checked out gear. Here's a look back at the NAMM that was.

BY CRAIG ANDERTON

This was the show where digital recording jumped into the mainstream, PCs outnumbered Macs, and a spirit of innovation returned with a vengeance. There's no way to cover everything, but we'll squeeze in as much as we can by looking at various trends (shown here in italics) and the products that exemplify them.

Following on the trail of the introduction of the Alesis ADAT and Tascam DA-88, digital recording — tape and disk — seems poised to take over and expand the project studio market, possibly as rapidly as the CD displaced the LP.

• ART announced the DR 8000 8-track digital recorder for $4000, which includes storage for 160 track-minutes and a tape drive for backup. A companion Windows 3.1 program edits tracks down to the waveform level.

• Akai's DR4d 4-track (expandable to 16) recorder requires an external hard disk, but is otherwise a self-contained SCSI peripheral. The DR4d includes four balanced analog I/Os, AES/EBU/SPDIF digital I/O, and a second SCSI bus for computer control — all for $1995.

• Session 8 ($3995), Digidesign's "digital studio in a box" offers four mic inputs, eight line ins and outs, four programmable inserts, 10x2 analog submixer, SPDIF digital I/O, audio interface card, and a 56001-based DSP card that supports eight tracks of CD-quality audio on a SCSI hard disk. The Windows 3.1 software (Mac is due later this year) includes a software patch bay, automated mixing, EQ, etc.; synchronizing with a suitable sequencer gives simultaneous MIDI and audio recording/playback. Session 8XL ($5995) is similar, but bundles a higher-end audio interface.

• OSC's Deck, formerly distributed by Digidesign, is now Deck 2.0 ($299) and available from OSC. It requires an Audiomedia or Sound Tools II card and allows for synchronous QuickTime playback. OSC also announced Metro ($225), a MIDI sequencer that works with Deck 2.0 to provide simultaneous recording/MIDI sequencing.

• The Spectral Synthesis Audio-Prism, a 12-track hard disk recorder/signal processing card, runs under Windows 3.1 (a Mac version is due later). The associated Prismatica software includes parametric EQ on every channel, and digital patching; all data and media are compatible with Spectral's AudioEngine. The projected price is under $4000.

• Yamaha's CBX-D5 ($2995) is a SCSI peripheral. Add a personal computer (Mac, Atari, or Windows 3.1), CBX-D5 compatible software, and hard drive for a 4-track recording system (expandable to 8 tracks). The CBX-D5 has four XLR ins and outs, digital I/O, four-band parametric EQs for each channel, and two SPX1000 digital signal processors.

As more studios generate large digital audio files, backup will become a major concern.

• Dynatek's RAID (Redundant Array of Inexpensive Disks) solves the hard-disk-crash-in-the-middle-of-recording problem by striping data across multiple drives and storing error-correction codes on drives that don't contain the associated data. It's an expensive — but effective — approach.

• Grey Matter Response introduced its Mezzo line of data backup products based on DAT and 8 mm tape, hard disks, and tape/hard disk combos.

Portastudio-style analog cassette decks will remain popular for budget recording, but PC-based "multimedia" cards will make budget digital recording affordable as well.

There were three major cassette-based multitrack introductions: the Alesis QuadraTrack (4 tracks with digital reverb, Dolby S, and 16 channels; $695) Marantz PMD720 (4 tracks, 10 channels, 2 speeds, dbx noise reduction; $679), and Tascam Porta 07 (4 tracks, compact size, $449). If you need lots of tracks with a multitrack cassette system, the Vestax MSX-2 ($180) synchronizes their syncable 4- and 6-track recorders to provide up to 32 tracks.

Several computer boards work with MPC-compatible sound cards to provide hard disk recording, editing, and synthesis for the PC; these could easily be the basis of small project and post studios.

• Roland's TAP-10 Audio Producer card ($600) delivers stereo 16-bit audio, a Sound Canvas-like General MIDI synthesizer, mixer, built-in effects (reverb/delay and chorus), and software to synchronize digital audio and MIDI file playback.

• Innovative Quality Software's SAW (Software Audio Workshop; $599) is a digital audio recording/editing program for the PC with playlist capabilities, real time digital mixing, 16 tracks, and detailed editing.

• The Cyber Audio Card from Alpha Systems Lab for the PC offers stereo digital audio recording, MIDI interface, 32-voice synthesizer, CD-ROM interface, mixer, and bundled software.

• DigiVox's Sound Impression software for the PC ($149.95) can record, play, edit, and process digital audio as well as play standard MIDI files and CDs.

How to turn your 8" to 15"

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CIRCLE 13 ON FREE INFO CARD

MI INSIDER

CRAIG'S ANNUAL NAMMINATIONS

And now, the time has come to bestow honor and recognition (sorry, no cash prizes) upon those companies and individuals who went beyond the call of duty in providing stuff for skeptical journalists to get excited about.

The **Take That, Digidesign! Award** goes to...Digidesign, who, by introducing the Session 8, has just become one of its biggest competitors.

Mackie won the **Best Show Freebie Award** with the Official Nerd Pack, containing a metal ruler, a comb, a nail file, a mechanical pencil, and a screwdriver labeled "Official cheap screwdriver which will probably break the first time you use it. But what the heck, it's free!"

E-mu garnered the **Pennies from Heaven Award** for being bought by Creative Labs (the Sound Blaster folks) for around $32 million.

The **Jimi Hendrix Memorial Summer of Love Award** goes to IX Development Technologies, whose Janus Machine lets you play backward guitar in real time.

The **Thank You, We Don't Need Three Different "Standards" Award** goes to The Great Apple Developer's Software Meeting. See, Apple came out with MIDI Manager but it didn't work very well, so Opcode did OMS but Opcode was licensing it, so MOTU introduced FreeMIDI, which is very cool and license-free — but do we really need another "standard"? So there was a big meeting to try and figure out a solution that wouldn't fragment the Mac market into little pieces. We'll keep you posted.

Roland received the **I Told You So Award** for realizing years ago that computer users would want better sounds for their computers — and bet on the PC while it was at it.

The **Swiss Army Knife Award** goes to the Toolbox from Third Coast Labs, a rack unit with two active direct boxes, rack light, two-stage power-on delay to avoid thumps, silent tuning, metronome, AC line voltage meter and filtering.

Jupiter Systems picked up the **These Are the Days of Miracle and Wonder Award** for Infinity, a looping program for the Mac that creates seamless loops for "unloopable" AIFF or Sound Designer samples. You have to hear it to believe it.

The **Soul of a New Machine Award** goes to DNA's Beat Blocks, which are $99 CDs with over 350 audio drum loops and a floppy disk for Mac, IBM, or Atari computers with Cubase-compatible "groove templates" for each loop. The first CD features the legendary Clyde Stubblefield, who helped define funk in his work with James Brown and others.

Epi-Graf accepted the highly coveted **Creativity Isn't Dead Yet Award** for the InfraMIDI Equator ($795). This box turns CD players, VCRs, LaserDisc players — anything you can control via infrared — into MIDI peripherals. The Equator records up to 120 codes from infrared remotes, which you assign to keys. Play the keys and the Equator sends the commands to the gear. Applications? Fly in a CD sound effect on top of a sequence, trigger VCRs along with sequences, use a MIDI keyboard as a master remote for all your gear, and so on.

The **Heard You Missed Me, Well I'm Back Award** goes to synth pioneer Tom Oberheim of Marion Systems. We got a sneak preview of his upcoming synthesizer project, and it looks like a winner — a 1U "mainframe" that accepts 8-voice analog synth plug-in modules (plug in 2 modules for 16 voices). But how do you explain the alternate mainframe he had with a whole bunch of edge connectors? Maybe "modular synthesis" is about to mean something different from what we're used to.

—*Craig Anderton*

Regarding Atari's Falcon030, Steinberg's Cubase works with the Yamaha CBX-D5 for recording and sequencing; D2D Systems showed its $300 stereo hard disk recording program (with nondestructive waveform editing, playlist, etc.) running with a Singular Solutions A/D64X stereo A/D converter for pro-quality digital recording; and Digital F/X's Digital Master EX (a 4-channel, 16-track hard disk recording/editing system) is now Falcon030/TT-compatible.

While the industry readies for the first 8mm machines from Tascam, the ADAT phenomenon has changed the industry; look for more companies to catch this wave.

• Digidesign is developing an interface between ADAT and Session 8/Pro Tools to transfer multichannel audio between both products in real time. This promises easy editing for ADAT users, and inexpensive storage for Digidesign buffs. And it sure looks as though you can use your ADAT as an 8-channel I/O for Digi products.

• Fostex's ADAT-compatible tape deck, the D-8, is oriented toward video/post work so it will cost a bit more than the Alesis ADAT because of extra video-oriented features (e.g., built-in sync).

• JLCooper's dataMaster ($749) slaves ADAT to MIDI sequencers or SMPTE (or ADAT can be the master), without giving up an ADAT track. A $149 Sony 9-pin option allows video editing systems to control ADAT.

• Steinberg/Jones sequencers will soon control ADAT via MIDI Machine Control commands, as will Opcode's StudioVision.

Signal processor manufacturers seem to be leaning away from multieffects and designing specialized processors that do less, but do it better.

• DigiTech's TSR-24 ($800) concentrates on reverb, but it is expandable with additional processing cards. Also new: the DHP-55 five-part harmony generator, which generates harmonies from chords (not just single notes). It sounds awesome with a guitar.

• The DSP-4000 Ultra-Harmonizer (approx. $5000) from Eventide lets you "software patch" various software modules to create a particular effect. Programs can crossfade from one to another, ending the dreaded "program change glitch."

• Bucking the less-is-more philos-

ophy, Korg introduced the A4 Guitar processor with up to 7 simultaneous effects and 60 programs. A bass version is also available.

• Roland's SDE-330 Dimensional Space Delay ($995) and SRV-330 Dimensional Space Reverb ($995) incorporate Roland's three dimensional sound localization technology to place sounds in a three dimensional field with conventional stereo playback systems.

• RSP Technologies' "Circle Sound" products (encoder, decoder, and controller/panner) are the latest stab at three dimensional surround sound. The main difference: Circle Sound works with non-encoded material as well as encoded (albeit less dramatically), and uses no delays or reverbs.

• The Zoom 9120 Advanced Sound Processor ($599) emphasizes reverb with its 16 different stereo effects, but also provides trigger-con-

trolled gating, delay time/tempo calculator, vocal eliminator, and other features.

In software land, PCs are proliferating — Windows 3.1 is everywhere. The Mac is still hanging in and the Falcon030 has given Atari a new lease on life, but the PC's day has arrived, particularly as a "front end" for digital recording and multimedia systems. Nonetheless, there were noteworthy developments on all platforms.

• Oktal's Multitude Pro (99 undo levels, loop playback/record, tape recorder control, tap tempo, etc.) is now running on the Falcon030, as well as on the TT030 and ST. Windows and Mac versions are due next.

• Dr. T introduced Omega II for the Falcon030. The Omega II features real-time editing even while locked to SMPTE. There's also multitrack graphic note editing and Fostex R8 recorder support.

• New from Opcode: Musicshop ($149.95), an entry-level notation printing/MIDI recording package for the Mac; a lower price for the Studio 5 multicable Mac interface ($995); an upgrade for Max to version 2.5 ($49); and Opcode MIDI System V1.2, which fixes the Apple Powerbook MIDI problems (except in the Powerbook 100).

• Musitek's Midiscan software reads printed sheet music and converts it into standard MIDI files. No kidding!

Keyboards don't stir up the excitement they once did, which has narrowed the field down to the serious players. Expect synth and sampler prices to remain in the $2000-$3000 "pro" range to offset the fact that fewer are being sold.

• Akai introduced five 32-voice samplers, starting at under $3000. They include the CD3000, the first sampler that comes with a built-in CD-ROM drive and five CDs to put in it.

• E-mu's Vintage Keys is the latest member of the Proteus family and features Mellotrons, electric pianos, Minimoogs, etc.

• Kurzweil's sampling option is now available for the K2000 and K2000R.

• Peavey's PCX-6 essentially combines their C8 keyboard, DPM-3 synthesis engine, SP sample playback engine,

A Fine Vintage

Everything old is new again...or that's what they say. And now that dance rhythms and house beats are once again taking over the nation's psyche (as well as its pop charts), musicians are looking for that kicking sampler that emulates the classic sounds of the '60s and '70s (with a little '80s thrown in). The secret, however, is a unit that sounds authentic without sounding too obvious, and if anyone could pull this off it's E-mu Systems, who, with its latest entry into the sampler market, breathes new life into vintage sounds.

Appropriately called the Vintage Keys Module, this is a single-rack-space unit that utilizes a pair of stereo outputs and four individual outs. It comes packed with 384 solid presets, including hefty bass sounds, great strings and brass, and other sounds that are reminiscent of the older Proteus modules. In essence, the Vintage Keys Module has all the classic analog keyboard instruments you would need to write and produce hit dance songs — as we discovered when we used the module recently to record Taylor Dayne.

Frankly, we've always liked E-mu's sounds, and an extensive library of usable patches was always available when we craved more variety in the

Putting E-mu's Vintage Keys Module to the test

BY SHEP PETTIBONE AND TONY SHIMKIN

sounds. They were a dream come true for those who did not have a huge selection of synths and keyboards cluttering up their studio space. The Vintage Keys wins points, however, where earlier Proteus models missed out. Where the earlier Proteus modules (XR 1, 2 and 3) had poor-quality front-face panels and a shallow depth (which made it hell to patch and unpatch the audio and MIDI cables from the rear when it was rack-mounted), the Vintage Keys has a cool, sturdy front panel and is deep enough for a rack mount.

If you have used any of the Proteus modules, then you'll find it extremely easy to start messing around with the Vintage Keys. Under the master menu, you can change your master tuning, find all of the MIDI functions, pitch bend, adjust modulation, find the controllers and footswitches, and so on. Under the edit menu, you can easily find your way around the parameters to customize these sounds.

The sounds are vintage indeed, though some are a little too clean for our taste. You have everything from an ARP 2600 to a Wurlitzer electric piano. Toss in a couple of Leslie speaker settings, a B3 organ, an Oberheim CP 70, and the Prophet and you've got all the vintage keys you might have wanted but couldn't find or afford. It also has Juno patches, which recall the 106, 60 or Super Jupiter, Fender Rhodes, MiniMoogs, MicroMoogs, MemoryMoogs, Taurus Pedal, M-12, Jaco Bass, and more.

The presets are all pretty good building blocks, but we would have liked to see (hear?) some basic patches of the vintage keys not layered with new synth sounds. In addition, the bass patches have subs and highs but lack the frequencies necessary to cut through a track. This is very easily fixed by combining or layering two sounds, such as a bass and an organ.

Don't worry about drums; the Vintage Keys Module has more than enough drum patches. These are interesting, although some are too effected. We would rather add effects than have to take them off.

The bottom line is that the Vintage Keys Module is an effective unit that is at home in both project and commercial studios. We recommend getting one before it becomes a vintage module itself.

Shep Pettibone wrote the cover story for our November/December issue. Tony Shimkin is his writing partner.

WHEN IT CAME TO BEHRINGER WE LET THE EXPERTS MAKE THE NOISE.

Leading experts in recording, film post-production and live sound are discovering the many advantages of the Behringer 2-channel and 8-channel DeNoisers. They know Behringer takes the noise out of the dirtiest signal path without altering the audio quality. Their reactions show why Behringer is now the most talked about name in professional audio circles.

"Simply lovely. Smiles all around. Room agreement was unanimous: We want this thing on all our tracks." **Mike Joseph—Editor REP, March 1992**

"I have used similar 'single-ended' devices on the mixes of 'Ghost' and 'Godfather III' and found the Behringer Mark III to be superior in every category—from ease of operation to final result.

"Consequently, I am—without hesitation—recommending to LucasArts/Skywalker Sound that they buy at least four channels of Behringer Mark III DeNoising for each mixing console here and in Los Angeles; a total of twelve mixing rooms." **Walter Murch—Film Editor and Music Mixer, LucasArts/Skywalker Sound**

"If the phrase *noise floor* is in your vocabulary and you would prefer that it was not, get a Behringer single ended noise reduction unit to the top of your *got to have one* list." **Robert Scovill—Sound Engineer/Mixer, Rush/Def Leppard**

The experts know why Behringer DeNoisers let them take the noise out and leave the audio quality in. Isn't it time you discovered all the good things Behringer can do for your audio?

THE PERFECT EAR.

BEHRINGER

CIRCLE 16 ON FREE INFO CARD

The Composer On Tour

The German-based Behringer line of equipment has been in existence since 1986 and recently became available here in the United States. I was introduced to the product line during my last tour with Rush, but began to utilize it more extensively working on the current Def Leppard tour. I recently was given the opportunity to put one of their premier pieces, the Composer ($479.99), to the test. Here are the results.

The preferred engineer of such heavy hitters as Rush and Def Leppard tests the mettle of Behringer's compressor

BY ROBERT SCOVILL

First off, I will say that I am a fan of compressors in all shapes and sizes and have tried as many as I could get my hands on from the likes of Fairchild, Pultec, Neve, Solid State Logic, Tube-Tech, UREI, Aphex, BSS, and dbx. You name it, I've tried it on a number of different sources and have particular favorites for given applications. In this hands-on report I will discuss some of the sources I tried the Composer on, and cover some of the proprietary circuits that Behringer has incorporated in the unit.

For those of you new to the uses of compression, I will say that Behringer manuals do a very comprehensive job of explaining the concepts and potential applications in a very down-to-earth method. The manual alone gets an A+ from me.

Because I was considering adding some of the Behringer line to the plethora of equipment that I currently have on tour, my first step was to sit down with our resident technical guru, Wally Bigbee, and open up the Composer and have a look inside to check out its structural integrity and component layout. We looked, too, for potential bugs that could result from traveling 500 miles each night in a semi, as well as from working in extreme temperature changes and smoke-filled environments. If you are going to strap this thing across your mix bus or the lead vocal at a live show, you had better be pretty confident that it is going to work on a daily basis, and that if it does fail, it can be repaired fairly easily and quickly.

EYES-ON TEST

At first glance, the Composer gives the appearance of a very sturdy piece of equipment. The rack-mount hardware is heavily reinforced and the finish is a scratch-resistant brushed aluminum. After removing the top panel, I took note of a well-supported circuit board with all of its nuts and bolts properly thread-locked to avoid their coming loose from the vibrations associated with travel. (This is a common occurrence in hardware that is not properly thread-locked and is found on some of the most expensive equipment on the road.) All the heat sinks are

BEHRINGER'S COMPOSER: Can it handle life on the road?

mechanically supported (when they are not, they are potential victims of vibration damage).

The back-panel XLRs are wired to the circuit board with jumpers. This is a big plus in comparison with PC-mounted connectors which, after a period of time on the road, invariably result in hairline fractures in the traces leading from them. The unit also uses readily available 3-terminal regulators in the event of power supply problems, and the unit displays many test points for easy troubleshooting. Also noticeable from the internal view is that the control knobs and switches are extremely well protected from front panel impacts — a nice feature that is normally not found.

One of the concerns in the technical construction of the unit is that the chassis is a "tub" — meaning that the removal of the printed circuit board is mandatory for repair and replacement of components. Also, unlike the XLRs mentioned above, the TRS connectors on the back of the unit are directly mounted to the PC board, which could result in some of the motion problems also mentioned above. Lastly, several of the capacitors could use a bit more mechanical support, and the use of Behringer proprietary integrated circuits could present a problemwhen it comes to repair.

CIRCUIT CITY

The Composer has taken three distinct tacks on the manipulation of the dynamic of the signal. The first is a newly developed IKA (Interactive Knee Adaption) within the compression circuitry. The circuit's purpose is to combine the "Hard Knee" compression concept with the "Soft Knee" feature to obtain compression that is program-adaptable, and thereby inaudible and very "musical," while at the same time allowing you to be creative in your dynamics processing. How is this achieved? Behringer claims that the ratio given on the front panel is not actually reached until the level exceeds the user-adjusted threshold by at least 10 dB, which accounts for the very smooth compression curve. This means that even with extreme ratios, the signal remains free of unwanted side effects such as smearing, pumping, and breathing.

Next is the IRC (Interactive Ratio Control), which is used within the expander section of the unit. Having an expander section in a compressor is a big asset because, when using a compressor, the noise floor is extremely amplified during pauses or sections that are well below threshold. This noise is even exaggerated when the compression ratio is inappropriate. The expander therefore works similarly to a low-ratio noise gate, fading out the source during pauses, etc. The expander in the Composer, however, is no normal noise gate. A normal gate or simple expander, even when used correctly, drastically cuts signals below the preset threshold. This is very noticeable when using it on vocals, where words can get chopped up in the process. Hence, Behringer is using a newly developed IRC whose ratio is dependent on the introduced program material and adjusted accordingly, automatically letting it be more tolerant to signals only slightly above the noise floor.

The last circuit is the IGC (Interactive Gain Control), which is the peak limiter section. This section has a wide range of applications, from speaker protection to eliminating peaks in a digital environment that can wreak auditory havoc (as we all know). This is a simple, one-knob threshold adjustment. The release time is about one second after falling below threshold adjustment. What could be more simple, right?

SMEAR TACTICS

When checking out compressors for given applications, the first thing I look for are the effects of the compression over a range of ratios and the threshold on the "image" fed to it. A lot of the compressors, when using extreme or semiextreme ratios and thresholds, give the source a bit of a smear. It's as if you took an inked rubber stamp and pressed it against the page, but twisted it a bit as you did so. You can tell what the image is, but the edges are a bit smeared. You can hear this if you compare the input with the compressed version of it on a pair of high-resolution headphones.

Image smear is something I have found in many of the early solid-state units that I have tried over the years. I must say, however, that many of today's units have minimized this

problem, and the Composer has done so exceptionally well, which I believe is very critical to good vocal compression.

HANDS ON

Okay, enough science. Let's get this thing hooked up and start twisting some knobs. Over a one-week period, I tried the Composer across a selection of inputs, ranging from individual instruments such as bass guitar and lead guitar to backing vocals and lead vocals, and had it inserted across a mix bus receiving a full program. I also used it when copying digital sources such as DAT and CD to an analog source such as cassette or portable multitrack. The inputs and outputs on the Composer are arranged logically on the back of the unit. It offers electronically servobalanced inputs and outputs on XLRs and TRS connectors. The unit can be used in a balanced or unbalanced configuration to facilitate a wide range of uses. Transformer-balanced outputs are available via the OT-1 option. There is also a switch on the rear of the unit for each channel to be selected between a -10 dBV±.316V operating level and a +4 dBu±1.22V operating level. Being a single switch prohibits operating the input and output at opposite levels, though. This switch also sets the in/out level meter to its respective reference.

The unit contains an external key send and return facility, which gives you the option of inserting an equalizer to do frequency-selective compression and de-essing, or, by just using the return, of driving the compressor from an effects bus or fader output. It is a nice feature to offer a send and a return, as opposed to just an external key input to facilitate both of these schools of thought of which I personally am a big fan.

The front panel controls are very logically laid out, with potentiometers being of the detent type. The Composer offers a single knob expander with a fast or slow release switch. The peak limiter section is also a single knob threshold. The compressor section is fairly standard for most compressors of this style available today: threshold, ratio, attack and release, and an output-level control. There are switches to choose between internal and external key sources and also a key listen

ROBERT SCOVILL puts the squeeze on the Composer.

switch that routes the key source to the output of the unit. An auto attack/release switch, an in/out, and a stereo coupling switch complete the front panel controls. Just a little application note here: Wouldn't it be nice to see a headphone output on compressors and gates that offer key source listen switches? Then you could actually listen to the key source while a unit is working, as opposed to having to replace the actual output with the key source — a bit of sore spot during a performance.

The "feel" of the unit is very solid, although the signal meters themselves are a bit suspect, being an 8-segment, widely spaced LED. Not my favorite style of meter, but probably a function of space economics as well as financial ones. I believe, however, that the meter could have been handled a little better. One small peeve that caught me by surprise was that to meter the input you have to bypass the unit. For example, when the unit is "out" of the circuit, you are metering the input level, and when the unit is "in" the circuit, you are metering the output. This could be a bit frustrating if you are level-checking during a performance because of the need to bypass.

The expander section seems to work well as long as the noise floor you are trying to negate is not too loud. If you are trying to negate a loud stage volume from a lead singer's mic when he is not singing, for example, you might want to gain a bit more control with one of the Behringer denoisers. I have used one with great success in this very application. I will say this, though: when I used the Composer's expander on backing vocals, it seemed to work very com-

prehensively. At any rate, this expander is a great asset — especially when using the unit with inner-ear monitoring systems — because of the ability to eliminate destructive ambience on open mics. I am going to be a bit greedy here, though, and state that I do miss the presence of an additional internal deesser circuit such as the one used in the BSS DPR-402. That unit does not offer the expander, though, so I suppose it all balances out. To have both expander and deesser in one unit would be heaven. (Is anyone listening?) The peak limiter section seems to work best when used sparingly while trying to eliminate peaks for analog/digital dubbing, for example, and as a component protector it appears to be quite comprehensive.

All in all, this is a very flexible unit and appeared very receptive to any type of source I used it on. It was very forgiving to ratio and low threshold settings. I think this unit could live as a cornerstone for a project recording studio, being used for tracking as well as for mix compression — especially when you consider its ratio of quality to affordability. It is a well-constructed, very functional, and audibly transparent unit that could be used in any number of recording or live-sound applications.

It got high marks from me. I'll take a couple.

Robert Scovill is the recipient of the 1992 Mix magazine Technical Excellence and Creativity (TEC) Award for Live Sound Engineer of the Year, and is currently striving for an endorsement deal from Tommy Armour.

"It's fat..."

Andora Studios, Hollywood

We're printing at 9 over 250 and it
   still doesn't sog up.
I know, the harder you hit it, the
   better it gets.
Listen to that high hat.
It's stiff.
Yeah, and it's not squashed.
No, it's not compressing at all.
That's pretty amazing.

What's amazing is the noise floor.
What noise floor? I don't hear
   any noise.
Right!
Hey, this sound's got everything
   I need.
It's got depth all right. You can hear
   everything—way back in there.
Clarity, punch, depth—that's it.

You heard it. Ampex 499.
I'd say it was audibly superior.
I'd say it just *sounds* better.

Ampex 499.
It just *sounds* better.

AMPEX

Short Takes On Hot Tapes

ANSWER THIS QUESTION: When you need to record…

1. You go buy a roll of tape and punch in.

2. You buy a full case of tape and realign your machine for that tape.

3. You won the lottery and now let your tech worry about all that.

No matter which type you are, if you are recording multitrack, the odds are that you're using that thin brown and black stuff. You are growingly intrigued by the new low-cost digital multitracks. But you've also read about the hot new analog tapes and wonder what a difference they could make in your existing studio.

You continue to record analog because:

1. MC Hammer made money with 1/2-inch and 1-inch analog.

2. Your machine is paid for and is making money.

3. You like the way analog sounds.

Any combination of answers is fine. Most projects are still tracked on analog machines until the final mix according to top West Coast mastering houses such as The Mastering Lab, Future Disc Systems, and Digiprep. Steve Hall of Future Disc reports that 75 percent of their incoming masters are now 1/2-inch 30 ips. Projects with adequate bud-

Ampex 499 & 3M 996 breath life into analog recording

BY WES DOOLEY

gets are often tracked on locked pairs of analog and digital machines. All machines have their own sound and producers use each to its own advantage.

So, what does this mean to you? Answer for yourself:

1. You are not out of the loop using analog.

2. Your choice of tape influences how you're going to sound.

3. You are out of the loop if you don't match your machine to its tape.

LONG LIVE ANALOG!

So what are your new tape choices and what's happening on the cutting edge of analog recording?

The most widely used analog recording tapes are still Ampex 456 and Scotch 226. They are available in 1/4-inch through 2-inch widths and most machines built in the past ten years are optimized for them. 3M 996 and Ampex 499 are a new generation of recording tapes that require more bias and different EQ for best results.

You can use these hot new tapes on a machine set up for 456 or 226, but you probably won't prefer the results. You will not get full use of the 3 dB additional headroom and the extra dB lower noise floor that these tapes can offer unless your machines are specifically aligned for them. Getting the best results out of your tape recorder is a bit like keeping a piano in good shape: you want to find a top technician and establish a relationship.

What difference do these new tapes make? See figs. 1-3 for the relative frequency response of Ampex 456, 499, and 3M 996 when tapes are interchanged in recording. For smoothest frequency response (and best headroom), your machine does need to be set-up for a specific tape. Any other type should be considered a special situation. See fig. 4 for a comparison of MOL levels for a properly set up machine versus a 996 to 456 transplant. (The Maximum Operating Level is defined as the point where 3 percent 3rd harmonic distortion is reached.)

Assuming that your machine has enough bias and EQ range (most newer model tape machines do, an 80-8 does not), what advantages can you expect to hear on your hot tape recordings?

1. They record

FIGURE 1: *Frequency Response — Ampex 456 used in place of Ampex 499 and 3M 996.* 0 level is 250 NW, speed is 15 ips. Alignment per factory instructions. The SOLID line was on a channel aligned for that tape. The DOTTED lines were on channels set up for Ampex 499 (HF loss, highest midrange sensitivity) and 3M 996 (most loss of highs).

FIGURE 2: *Frequency Response — 3M 996 used on machines set up for Ampex 456 and 499.* 0 level is 250 NW, speed is 15 ips. All alignments done by factory instructions. The SOLID line was on a channel aligned for that tape. The DOTTED lines were on channels set up for 456 (below the solid line at 1 kHz) and 499 (above the solid line at 1 kHz).

FIGURE 3: *Frequency Response — Ampex 499 used on machines set up for 3M 996 and Ampex 456.* 0 level is 250 NW, speed is 15 ips. All alignments done by factory instructions. The SOLID line was on a channel aligned for that tape. The DOTTED lines were on channels set up for 456 (a hump in the upper highs) and 996 (the lower dotted line).

FIGURE 4: *Difference in distortion for 3M 996 on a properly set up machine vs. recording on a machine set up for Ampex 456.* Freq. is 1 kHz, 0 dB reference is 250 NW, speed is 15 ips. The DOTTED line is with the stock 4 dB of overbias on a Tascam ATR60-16, the SOLID line is from 996 recorded on a channel set up for 456 with 3 dB of overbias for that tape.

FIGURE 5: *Noise vs. frequency for the Ampex 456, 499, and 3M 996.* Noise vs. frequency from 100 Hz to 20 kHz for three tape types at 15 ips. Top line is 456. Middle lines are 499 and 996. The bottom line is the residual noise of the machine with the tape stopped.

FIGURE 6: *Multifrequency third harmonic distortion at a fixed record level.* This run was at 250 NW on Ampex 499 tape. The SOLID line was done on a channel set up by the book. The DOTTED line was set up as recommended by several independent pro-audio technicians (well, one is more like a design engineer).

FIGURE 7: *Frequency response test for Ampex 499.* 0 level is 250 NW, speed is 15 ips. The SOLID line was on a channel aligned to Tascam and Ampex instructions. The DOTTED line was a channel set up for minimum third harmonic distortion.

about 3 dB hotter (3 dB higher Maximum Operating Level or MOL). Unlike digital, which hits a brick wall at maximum level, analog tapes have different MOLs at different frequencies. With analog tape, before you hit the MOL it starts compressing a bit. At the same time, the third harmonic distortion rapidly starts to rise (almost all of an analog tape's harmonic distortion is third harmonic). The dynamic blend of these two characteristics is part of what gives analog its distinctive sound. Since the new hot tapes have about 3 dB more MOL at all frequencies, you can move your record levels up 3 dB for a quieter remix. These hot tapes, when used with dbx or Dolby S or SR noise reduction, deliver a dynamic range that rivals digital for a lot less money. If you are not into using tape compression consciously, you can leave your record levels alone and enjoy 3 dB of extra safety margin while getting a more dynamic "live" sound off tape. (This is the approach that Doug Sax at The Mastering Lab recommends.)

2. They are quieter. Figure 5 compares the noise floor of 499 and 996 with 456.

3. The hot tapes have a better top end. This is noticeable in two ways. High frequency SOL (Saturation Output Level) is defined as the point where more record drive no longer increases the playback level. Both 499 and 996 have gained another dB or two SOL in the 10 to 15 kHz range at 15 ips. Also, on some 30 ips machines these high resolution tapes extend the high frequency response another 5 kHz.

4. The hot tapes have reduced print-through. Print-through is the pre- and post-echo that is heard on tapes that have been stored for a while without rewinding. For most music work this is not an overwhelming concern. On speech recording, however, it is critical. So much so that there are special tape formulations specially optimized for low print-through — such as 3M 966 and Ampex 478.

Both tape types are decent performers when it comes to modulation noise (background noise level that goes up and down with the signal level) and scrape flutter (like conventional wow and flutter but at a much higher frequency caused by stiction as the tape scrapes across the heads and guides). Tape output level consistency is good.

(For more information, check out Ampex and 3M's technical data sheets, or even better, look up Sam Wise's excellent article "Tape Talk" in the September 1992 issue of *Studio Sound*.)

What are the performance differences between the hot tapes? Like most competitive products they are more alike than different. Measurements show 499 tape has a higher MOL at lower frequencies, and 996 has the higher MOL at the high end. 996 has less print-through. Overall, 499 has a wider dynamic range. The most relevant question might be that of which tape sets up best on your recorder. The hot tapes are mechanically stiffer than 456 and 226. Some tape transports have tape-to-head contact problems with stiffer tapes. 3M 996 is stiffer than Ampex 499. Tascam reports that since 3M 996 tape came out earlier, they have enough experience with it to know that it works well. Han House at Fostex reports that Ampex 499 seems to be the best fit for their machines.

IT'S A SETUP
Which brings us to setup procedures. Three-head machines have the edge here. A three-head machine is much easier to align electronically, whether it be cassette or open reel. Since the record head is immediately followed by a reproduce head, you can see and hear the results of a tweak within fractions of a second without having to rewind the tape to check out what happened. When you are doing 8-, 16-, or 24-tracks, this is a major time-saver. Setting up such a recorder (such as the 15 and 30 ips 1-inch Tascam ATR60-16) for high performance becomes much faster. Most of these tests were done on an ATR 60-16 for that reason, and because of its excellent stability and performance. A Tascam MSR-16 two-head machine yielded comparable results, but the ATR 60-16 test results were used throughout this article.

Tape interchangeability and consistency are important because the most cost effective 8-, 16-, and 24-track machines are two-head machines from Fostex and Tascam. Since two-head machines are time-consuming to align, we tend to leave them alone for as long as possible. John Degele of Amperex Tape Recorder Service in Hollywood observes that people are often not as critical about a two-head machine's state of tune. You just can't hear the deterioration as easily. Thus we rely on the stability and consistency of the machine and of the tape. Doubly so, because two-head machines are factory set up on a statistical basis.

We will assume the factory wants to align two-head machines for 499 or 996 at 15 ips. The initial procedure is to individually set up a large number of machines while keeping good notes. After a playback alignment is done, a steady 10 kHz tone is recorded while each channel's bias is adjusted from minimum to maximum. The tape is rewound, and the point where the 10 kHz recording peaked in level and then decreased 4 dB below that level is found. The tape counter and the RTZ can make this easier. Once the track is correctly biased, record levels and high frequency levels can be adjusted. Then the process is repeated for the rest of the tracks.

The factory techs have done this to a large number of new machines and have established typical values for bias current, which they publish in the manual. New machines are generally stable, tape pretty consistent, and everything works adequately most of the time. But if you demand all the performance your machine is capable of and have relapped the head or have a worn one, you can't just plug in average values. Figures 6 and 7 illustrate the advantages you might gain by having your tracks custom tuned.

Go ahead and find out which tape sounds best on your machine. Build a relationship with a good tech because tape recorders are as complex and demanding as pianos or cars. They are another part of your sound and respond well to simple cleaning and regular maintenance. Now go get some tape and get aligned. Try it, you'll like it.

> *Wes Dooley is founder and engineering director of Audio Engineering Associates (AEA), which prides itself in its specialty sales and services for the professional audio community. Among their diverse list of clientele AEA boasts Stevie Wonder, Herbie Hancock, and the Beach Boys.*

Not The Same Old Song & Dance

AEROSMITH'S JOE PERRY AND STEVEN T ... **ABOUT GETTING AUDIO ANARCHY DOW**

TYLER TALK
W ON TAPE

AEROSMITH'S LEAD GUITARIST LETS THE MUSIC DO THE TALKING IN HIS PROJECT STUDIO

I've had something to record with at home ever since 1975. However, I didn't start building a serious project studio until about five years ago. And, although it's not a part of my Massachusetts home, it is close enough by that I can get to it whenever I want to.

One of my favorite things in the studio is a rack of Neve 1066 and 1073 EQs that I send everything through. I also have a bunch of effects, including Eventide's H3000SE and Roland's SDE-3000, that I go right to tape with. That way, if I want to record another mix I won't have to repatch everything. I also own a Lexicon LXP-5, which I used for a wild effect at the beginning of a song called "Fever" off of our new album, *Get A Grip*.

With *Get A Grip*, we were talking about going out of town to this place called Longview Studios. It's an old farmhouse with a bunch of 24-track rooms that we have used before for writing and recording. You go up there and play and sleep. It's all pretty cool, but I didn't like the prospect of spending the entire summer up there. I wanted to stay close to home.

Steven (Tyler) and I had already been writing at my project studio, the Boneyard, and I had just purchased a Macintosh LCII with Performer software, so I suggested that we keep working there. As it turned out, the band really enjoyed the close and personal experience.

IN THE BEGINNING

"Fever" was the first song that we developed for *Get A Grip*. Steven and I were just doing some demo stuff in the Boneyard about two years ago, and that's where that song and the new album began.

We write songs in various ways. Usually I'll have a few pieces of music that we start playing around with — with lots of guitars on it, as you may have guessed. And we'll sit there and start pulling stuff up and seeing what works. That's the way it's always been — for over 20 years.

Sometimes the demos are better than expected. For example, the last song on the second side of the new album, called "Boogie Man," is a little piece that we had done at a different rehearsal studio and that was taken right off the DAT machine. We always have a DAT machine running, just in case. This way, if we hear something we like, we can go back to it. "Boogie Man" had a guitar riff that we just hadn't planned on. Steve sat down and played the slap bass on the keyboards and I played guitar, and after about 45 minutes it gelled

into something really decent. What you hear on the album is the last two minutes of that jam right off the DAT. We played it once two years ago and that was that. If we had taken it one more step I would have written a melody for it, but it sounded so good that we decided to leave it alone.

We do a similar thing when we're recording with the whole band. Aerosmith always records live in the studio — everybody playing at once, with Steven doing a basic vocal. Then we sit back, take a listen to it, and then add to it and fix whatever's wrong. In fact, on some of the songs we recorded I did the lead first and then went back to record the rhythm track. This is how we get the best energy.

We used to do 30 takes, and then take the best performance of each track — the best first chorus, the best third chorus, the best lead, etc. Now we get it all in one whole take. These days, it's pretty rare that we actually cut up a song. We usually get it on the third or fourth take — one of the benefits of being in a band that spans more than two decades. The Boneyard helped too. Most of the demos we recorded there for *Get A Grip* were really templates. And in some cases, when we couldn't play out the original feeling that was captured on the demo, we would lift them right off the tape. If I got an effect one night that I couldn't reproduce, the demo made a great backup. (I have to get in the habit of writing stuff down — some nights I would bring up 24 tracks and I don't even keep track sheets!)

DRAWING THE LINE

Aerosmith contributes to the producing of its albums, and of course we benefit from the input of a talented producer. Every producer has a different style and every band has a different relationship with their producer. Basically, the way we work with Bruce Fairbairn, who did *Permanent Vacation*, *Pump*, and *Get A Grip*, is that we will have our songs at a point where we all like them. Then Bruce will come

THE INS & OUTS OF AEROSMITH'S PROJECT STUDIOS

By Gyro Gearloose (David Frangioni)

I began working with Aerosmith back in 1989, just as *Pump* was being released. I have worked on Joe Perry's, Steven Tyler's, and Tom Hamilton's project studios, as well as on projects for the entire band both in the studio and for tours. When Joe originally contacted me, we immediately assessed his requirements for a comfortable and productive working environment. What started out as a writing studio has now turned into a full-blown recording studio — complete with an accurate, acoustically treated control room, large recording area, and a powerful MIDI setup for composition.

Joe's 24-track recording studio is an analog/digital hybrid of vintage gear and new gear. The foundation is an Otari MX-80 24-track and an Allen & Heath 32x24 input Saber console. Effects include a Yamaha SPX-90, Eventide H3000SE, Lexicon LXP-1 and LXP-5 with MRC v 4.0, Roland SDE-3000, dbx compressors, and a couple of BBE's. He owns a Tascam DA-30 DAT machine and a Tascam 122 cassette recorder. Among other gear, there is a rack of Neve 1066 and 1073 mic pre/EQ modules and a rack of Tube-Tech components. These include a couple of compressors, mic pres, and associated dynamics modules. Joe loves them because they sound great, similar to Pultecs and LA-2A's, and because they are clean and reliable. During preproduction we put Steven through a Neve 1066 and a Tube-Tech compressor, and everyone agreed that it sounded great — upfront, big, and warm.

For MIDI, we're using a Mac LCII (10/80) with a Mark of the Unicorn MIDI Time Piece II interface handling the MIDI routing and SMPTE chores. Performer is Joe's sequencer of choice because it is extremely fast, powerful, and easy to use. Some of the sound modules include an Alesis D4, Korg O3r/w and O1/w, and E-mu Procussion.

As you might expect, Joe's guitar and amplifier collection is second to none. Usually, Joe will get the sound he wants with his own custom blend of guitars, amps, and effects. When he's got the sound, we just mic it and send it directly to tape. He has experimented with some speaker simulators, but none of them, as yet, meet his qualifications.

The studio is wired primarily with Mogami cable and Neutrik connectors and the patchbays are normalled to Joe's specifications. Monitoring is handled with several models from Tannoy and Yamaha along with Bryston power amplifiers.

Steven Tyler's studio is situated in a comfortable environment that doubles as a rehearsal space for the band. He and Joe live so close to one another that having two full-blown studios is redundant, so Steven's studio is very focused as a writing suite. His choice of keyboards include a Korg M1 and O1/w, as well as an Alesis HR-16 and D4 drum module. He has the usual assortment of reverbs, delays, and compressors along with a 4-track recorder and Yamaha monitors. The large, spacious room is a floating room, ideal for absorbing sound so the band can rehearse at full volume!

Gyro Gearloose, aka EQ contributor David Frangioni, is a Boston-based MIDI specialist/consultant whose clients include Chick Corea, Damn Yankees, Extreme, Bob Clearmountain, and Reggie Lucas (Madonna).

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in and listen for the rough spots. As part of the band, you get in the middle of the music, and you need someone with some distance from it. He comes in and listens with fresh ears and works with us to make it the best it can possibly be. If we had one part that we all thought was really smoking, and had something like two months away from it, maybe we would come back to it and see that it needed more. Unfortunately, we never have that luxury, so we get the next best thing: Bruce.

Personally, I think being a producer is such a harrowing experience that I'm glad I don't have to do it to make a living. "Producer" is like a catch-all title. It's a very important position, but I don't want to get any closer to producing Aerosmith than I am right now. Everybody in the band has strong feelings about it too. I don't think it's wise for us to get to the point where we don't have someone to act as a mediator.

I can relate to the problems that producers face. When I was with the Joe Perry Project, I was in charge of everything — and that taught me the value of working in a band like Aerosmith. I had a good time doing it, but I won't do that kind of solo work again. Maybe someday I'll put out a bunch of outtakes from the studio, but right now I just don't have the time. Besides, there's no need to put out anything that I can't release with Aerosmith.

DONE WITH ANALOG

Like *Pump* and all our other albums, *Get A Grip* was recorded analog. The reason why is simple: we prefer it. So far I haven't heard digital sound as good, at least when recording. I even hate the way a DAT sounds until it's gone to tape. With analog, there's a certain warmth that just can't be created with digital. And we tried. With *Pump*, we put everything through the AMS AudioFile. Listening to the record now, I think it sounds kind of harsh. This time around we didn't do any of that — we synced right to 1/2-inch tape.

All of your in-studio problems don't have to be solved digitally either. Just a few weeks ago we were getting ready to mix a song we had

Joe Perry's Project Studio Layout

AEROSMITH'S FRONT MAN TAKES US INSIDE HIS WRITER'S STUDIO AND SHOWS US THE TOYS IN HIS ATTIC

My writer's studio is a floating room over a barn that I use as a two-car garage. The room is about 40 square feet and is surrounded by a two-inch space that separates it from the rest of the barn, including the floor. The room is lifted off the floor by rubber stoppers at about 40 points. Why a floating room? The best way to get an entirely soundproofed situation is to put air between you and the rest of the world. There are things like water and lead that work too, but that's a little ridiculous when air will do the job.

The idea came from one of our past albums. We used a floating room in the recording of *Draw The Line*, which was recorded at an old nunnery called the Cenacle. There was no place to put the console, so we built a floating room and I liked the way it worked. Now I can crank it up to 120 dB before anybody'll hear me. If I want to have someone

over to write and play with, we have someplace to go and crank it out until three, four, or five in the morning and no one will complain. You all know how that shit goes — you pick it up and you don't stop for dinner.

I'm not looking to build a big studio. Joe's is close by enough that it's not really necessary for me to have one too. Besides, I'm really not into the tech-head thing yet — I have enough trouble remembering my own lyrics, and to start worrying about where the bus sends go is something I don't need right now. My short-term goal was to write for this album and have a place with a drum set, a couple of Korgs, and a toaster oven — a place where I wouldn't have any obstacles in my way. And that's what I did. I'm sure I'll get some small board when it gets real easy, but for now, if I get some idea that requires more than I've got, I'll work it up properly in Joe's studio.

My writer's studio is a fun place to work and I love it. I've been having this gentleman come in, named Marshall Ross, and he's carved this huge eagle for me there. It's got a 30-foot wingspan and a huge head that lights up. You can see it through the woods — it's the eighth wonder of the world.

We took a quantum leap from the old days to now. I used to spend a lot of time getting high and looking for outside things to make my insides right, but nowadays, instead of the infamous

Aerosmith lost weekends, I'll go find a friend to write with. I'd rather leave my studio with a cassette in my pocket and a couple of songs from my heart then have a lost weekend with someone whose name I can't even remember. Coincidentally, it's like the best of both worlds because you get sounds on tape in a project studio that you wouldn't normally get in a larger studio unless you were high.

WORK THIS WAY

For the new album, *Get A Grip*, we went to A&M Studios and put down an album's worth of songs. When we stepped back and looked at the big picture, we weren't sure if this was what we wanted to follow *Pump*. The whole band wished we could go back into Joe's studio (the Boneyard) for another month or two and keep writing because we were on a roll. The question was not whether the songs were any good but rather: Did we have the time to make them better? We decided to make the time. David Frangioni (who I like to call Gyro Gearloose because he could rewire a Walkman to record 4-track) had everything up and running at Joe's place, so we came back home and went to work.

The atmosphere in a project studio is absolutely better than the one in a large commercial studio. I want to have fun doing what I'm doing, or one day I'm just going to say "forget this." In the Boneyard, even if we get nothing onto tape, I know I'll still have a great time. And nothing is lost. Whenever I sit in a room with Joe and let the tape run, we get at least ten to fifteen solid seeds. For every song that was a hit over the years, I could get a tape from my vault downstairs and let you hear where it came from. We always have the DAT machine rolling, acting like a drip net catching the ideas.

I think Joe's room is perfect. It's small, about 25 x 9 feet, with a 12-foot ceiling. Joe's always looking for new stuff to put in it too. He was talking to Shelly Yakus at A&M about EQs and decided on Neve, and Gyro's given him
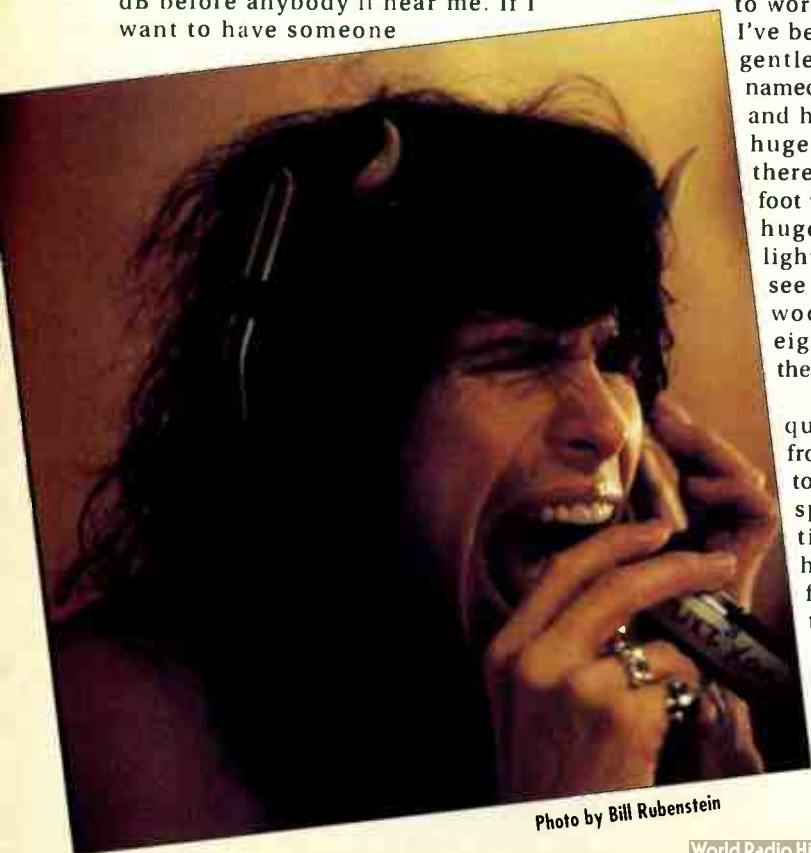
Photo by Bill Rubenstein

a lot of advice. I don't really have anything to do with picking out the equipment that goes into the Boneyard.

LET THE MUSIC DO THE TALKING

The secret to success for any band or songwriting team is communication. The relationship between Joe and me has always been about communication. Sometimes I have a hard time talking to him one-on-one, but I know I can always communicate with him musically. Music is such a strong language that when I lay down a drum beat, I'm saying something more than words can ever say. You can really let the music do the talking if you have an instrument in hand. I can answer him with anything he "says." That's the secret to teamwork between two people or five or whatever.

I think it's more difficult to work with a producer. Everybody likes to come in and rework your stuff to validate their punch cards and say that they were part of something. It's like a bunch of people telling you what the sugar cone beneath a big ball of vanilla ice cream tastes like and, oddly enough, it would taste different to you from what everybody else described. Everybody has their own interpretation of things and you've got to decide whether you take a hard-nosed stand and say, "No way man, this is me and this is the way it stays," or take chances. That's why we let people into our camp all the time. With each album, it gets harder and harder to listen to other people, but every album gets better and better, so a producer is definitely necessary.

Personally, I love producing. If I had the time, I would love to produce another band or my own album. Someday I'll put on the brakes and do it. When I do an Aerosmith album only a piece of me is shown. Being part of a band, you have to try everything. There are four other fingers in the pie and it's not the Steven show. There's a part of me that wants to do all the stuff I love. Someday I'll do it. I don't know exactly what it'll be; it depends on which side of the bed I wake up on that day.

GETTING VOCAL

When I record my vocal tracks, I don't

BACK IN THE SADDLE
By Bruce Fairbairn

There is never a dull minute with Aerosmith. In the studio they're always very willing and open to challenges. In fact, they almost demand to be challenged. Steven in particular is always working away at a song — throwing new ideas, parts, and arrangements at it.

Personally, I feel very comfortable throwing an idea out to the band. Sometimes we'll even laugh about it, but they'll always try it — no matter how crazy the idea sounds.

Usually, differing opinions or points of view are very difficult for everybody when you're recording. To Aerosmith's credit, however, this is not a major obstacle. With all the members of Aerosmith, once they've put their point out, and argued passionately for it, if it doesn't go their way they'll accept that and try to make the most out of the decision. This makes it a great atmosphere to work in, even during the explosive times.

This quality also makes working with the band exhausting — but extremely rewarding. Steven, especially, has so many ideas and moves so quickly through them that you have to be right there with him or you'll miss some gem that'll never be there again. It's very intense, but again, it's tremendously rewarding.

Every band is different in the studio. Aerosmith is definitely different from AC/DC, and AC/DC is different from the Scorpions, the group I'm working with now. If, as a producer, you plug whatever band you're working with into your system, you end up taking away their individual creativity and style. When I work with a band or an artist, we do things a certain way to a point, but then everything changes depending on who you are and what your style is.

Take Joe Perry, for example. Joe is one of the most creative and spontaneous guitar players I have ever worked with. There are plenty of guitar players who are good technically and who can execute well, but Joe plays on his feet. You can ask him to play a solo five times and each time it will be different but just as great. He's also always trying to open up new sounds. For example, on this album he must have used fifty vintage amplifiers. We dragged them into the studio, dusted them off, and fired them up. There were some great sounds that came out there — like the old Parc we split with an aged Epiphone Combo amp. It looks totally bizarre when you see it, but it sounds amazing when you get it right. I have to thank Joe for opening my eyes to the weird world of old guitar gear.

I didn't get the chance to work at Joe's project studio, but I feel that I know it well because I've heard so much music come out of it. Here's how we used it: First, I would come down to Boston and work with the band for a week or so, rehearsing. I would then identify some

areas that needed work and Steven and Joe would head back to Joe's studio to work on them and come up with new song ideas. Then they'd send it out to me in Canada and I'd come down and we'd rehearse again.

Their efforts at Joe's studio certainly paid off. In fact, some of the stuff from his studio ended up on the album in one form or another. The uniqueness of an idea combined with a sound that happened at a particular place in time was such that we couldn't hope to recreate it. There's great potential in his place. Now, if we could only get him to fill out track sheets...

In addition to Aerosmith's last three albums, Bruce Fairbairn has produced many of rock 'n roll's heavy hitters, including Bon Jovi, Poison, David Bowie, Blue Oyster Cult, and Gorky Park.

STUDIO DESIGN BY THE NUMBERS

THE BEST LAID PLANS

1 *A leading studio designer struts his stuff in fictitious rooms that exist only in the realm of EQ*

BY JOHN STORYK

Studio design is a field that involves a good deal of individual interpretation, so when *EQ* approached us and asked us to design rooms according to their specifications, we knew we had our work cut out for us to pick the best design from thousands that we could present for each room.

What you see here are the fruits of this labor of love. The specs and equipment were provided by the editors of *EQ*, but the comments were

provided by us. We hope these plans give you some idea of the thoughts and considerations that go into project studio design.

ROOM #1

Construction: 9' x 10', with a 9' ceiling; wallboard construction, with hardwood floor. Double windows at the end of one short side, double closets (sliding wood doors) along one side. Adjoins other bedrooms in pri-

vate home. Vocals and guitars need to be done in this room.

Console: Mackie 24 x 8 x 2

Recorders: Fostex GS16S; Sony PCM-2300 DAT; Carver PST-14 cassette deck

Monitor System: JBL 4208's with stands; Bryston 4BNPB amp

Rack(s): Symetrix 425 comp-limiter; dbx 172 SuperGate; Digitech VHM5 Vocalist; RSP Technologies Intelliverb; Peavey 20/20 effects

processor; Aphex C² Exciter; Furman PL-Plus Power Conditioner; Speck EQ16 equalizer; one 12-space rack with 48-point TT patchbay on wheels

Other: Console power supply; ART T28 Attack Module guitar amp; ART SGX 2000 guitar preamp; Alesis MMT-8 sequencer; Roland R8 drum machine; Korg 01/WFD keyboard; JLCooper PPS-100 SMPTE-to-MIDI converter; two mic stands; one three-tiered keyboard stand; two guitar stands.

Comments: We see room #1 as being pretty straightforward. It is a simple room in someone's house. The trick here is to take the closets out. If you can afford the equipment, you can afford to remove a few walls. The other important design feature is the additional glass at a splayed angle and a *matching* reflective plywood panel on the opposite wall. We would carpet this small room as well as apply some absorbing panels as shown. Depending on the exact rear door location you might add some splayed scatter-

ing treatment if there were room. This room will be relatively dead.

ROOM #2
Construction: 18' x 24', with wall-board construction on three walls plus one brick wall. 18' high (at peak) wooden beam cathedral ceiling. Hardwood floor. Adjoins family living area. Vocals can be done elsewhere.

Console: Amek Hendrix 40-input

Recorders: Tapeless room; Panasonic SV-3700 R-DAT

Monitor Systems: KRK 13000's with stands; Crown Power-Tech 1 power amp

Rack Gear: Drawmer DS-301 MIDI expander-gate; Orban 290Rx processor; Lexicon PCM-70; Lexicon LXP-5 with MRC; Behringer XR 2000 expander-gate; JBL M712 comp-limiter; DOD 231 seriesII 31-band graphic EQ; ART Multiverb Alpha effects processor; Roland SN-550 digital noise eliminator; two Radrack 12-space racks on wheels with 96-point patchbay

Other Equipment: Console power supply; Peavey Stereo Chorus 212 instrument amp; John Hardy M-1 mic preamp; Tech 21 SansAmp; Yamaha SY35; Kurzweil 2000; 4 mic stands; computer gear stands; two-tiered keyboard stand.

Comments: The secret to this room is in its size. Most people will take up every available inch in a room, but this room is quite large — even by professional studio standards. What we propose is to build a wall two feet from the back wall so the owner can get low frequency absorption or "venting." This way they won't have a runaway low-frequency reverb time. With a room like this, you can have powerful speakers, so you've got to soundproof. We recommend two additional layers of 5/8-inch gypsum board on resilient RC-1 channels.

ROOM #3
Construction: 15' x 20', cinder block construction with slab floor. Two

TOP TO BOTTOM: Rooms 1, 2, and 3.

windows each on two adjoining walls. Wooden beam ceiling 14' high. Garage-type structure, with one long wall adjoining a utility room. Vocals and guitar playing will be done here.

Console: Tascam 3700 32-input with JLCooper automation package

Recorders: Alesis ADAT; Tascam DA-60; Revox PR-99 2-track; Macintosh IIci; Mark Of The Unicorn Waveboard and MIDI Timepiece; Dynatek hard disk

Monitor System: Meyer HD-1 with stands; Yamaha NS10M (console-mounted); Hot House S400 power amp

Effects: Two Radrack 12 space racks with 96-point patchbay; Yamaha SPX 1000; SSL G383 mic preamp-EQ; Ensoniq DP/4; Aphex Compellor; Dolby Spectral Processor; Klark-Teknik DN410 Parametric EQ; Dynacord DRP-15 reverb unit; BSS DPR502 MIDI Gate; Rane Flex Series rack; Roland RN 550 effects processor

Other Equipment: Console power supply; Peavey SP keyboard and SX module; Ensoniq KS-32 keyboard MIDI Studio; two 12-space racks with 96-point patchbay; Fender Twin Reverb; one guitar stand; computer stands, platforms; 3-tiered keyboard stand; 6 mic stands.

Comments: This is a garage-type structure; so there really isn't any need for isolation because it is a separate building and I assumed it uses 8-inch block. All we did in the front was put some splays on to give it good symmetry and good first-order reflection scattering. Those splays pick up the glass lines because the weakest link of this room is the glass. To accommodate this, we added a second layer of glass on the outside. On the opposite side we did the same thing: we punched a hole exactly opposite the window, revealing the utility room that we are using as a vocal booth. Note that there is plenty of room for the gear and that the keyboards and gear are laid out on the nondoor side of the room away from the entrance.

John Storyk is one of the world's foremost studio designers, responsible for numerous professional studios, such as Electric Ladyland, Margarita Mix, and Howard Schwartz Recording, as well as for many project studios, including those owned by Whitney Houston, Ace Frehley, and Todd Rundgren. He heads up the Walters-Storyk Design Group, Inc., which is located in Highland, NY.

SUPERIOR DECORATING

② When shopping for studio furniture, shop ergonomically smart

QUIK LOK'S WORKSTATION
The WS/500 from Quik Lok Systems, is a multifunctional workstation that will organize keyboards, computer keypads, mouse pads, sound modules, power amps, CRTs, and small monitor speakers into a user-friendly assembly. The all-steel welded construction of the WS/500 is designed to be assembled by one person in about 15 minutes without tools.

The WS/500 is available as a complete prepackaged system for the MIDI musician with the full-blown keyboard/computer/sound module/multitrack recorder studio, or it can be purchased piece by piece. The base unit of the entire system, the WS/550 alone, makes an effective dig-

ital piano or clavinova stand with a wide range of height and width adjustability and a weight capability of 750 pounds.

For more information, contact Music Industries Corporation, 99 Tulip Avenue, Floral Park, NY 11001;

Tel: 800-431-6699 (in New York, Tel: 516-352-4110). Circle EQ free lit. #128.

RACKIN' IT UP
The RadRack series of studio furniture from Studio Plus combines ergonomic design with working efficiency. There are several RadRack models to choose from, and custom orders and special modifications are also available.

New to the line are two workstations and a Combo Rack that consolidates your equipment in one central location. The Combo Rack provides a total system for integrating small mixers or drum machines at desk level, with rack space below for amps and rack space at eye level for effects, patchbays, or other equipment that

requires easy access. The new SC-WK Combo Workstation combines the large Combo Rack with an angled table top and a mixer unit. The angled table top allows both racks to be directly at your fingertips without having to move your chair. The entire Combo Workstation provides 44 total rack spaces.

The two new workstations are named the SC6/10 and the SC10/12. The SC6/10 has six vertical top rack spaces and 10 vertical spaces below. The SC10/12 is taller and deeper, with 10 vertical spaces at eye level, larger flat desktop work area, and 12 vertical rack spaces below. This unit can be used as one unit to consolidate your equipment or can be attached to a tabletop to make it a workstation.

All RadRack products are handmade and come completely assembled with hard-wearing 12-gauge machined steel rack rails and heavy-duty bearing casters. They are finished in industrial-grade charcoal/granite zolatone.

For more information, contact Studio Plus, 1111 Rancho Conejo Blvd., #407, Newbury Park, CA 91320. Tel: 800-554-4990. Circle EQ free lit. #129.

RADICAL RACK: RadRack's Workstation

FIVE EASY SUGGESTIONS

3 How to know if your studio-design doctor knows his medicine, or if he's just an architectural quack.

BY FRANCIS DANIEL

How do you know if you are getting good advice from a studio designer? Here are some indicators to consider *before* you commit:

1 – Track record, of course. But even if you check out references (do it!) remember that a studio owner is going to have to be *very* unhappy to say outright that his place is a dog. And your expert won't give you the names of his failures, anyway. So go and scope out a couple of his or her rooms for yourself — and listen between the lines of what the owner or engineer says about it during your visit.

2 – The laws of physics still apply, no matter how many voodoo artists claiming to be studio designers tell you otherwise.

The voodoo crowd rarely show up dressed in feathers anymore, so how do you tell who they are? Since many of the basic issues that face studio designers also face professional acousticians and architects in many other contexts, you might ask your "expert(s)" what other kinds of acoustics and architecture they do, since incompetence doesn't last very long in the tough world of major-league building. This is the larger world that many professionals work in, and there is no training like it. Specialization with a broad background is a very good sign.

3 – You can also do a second objective check: Does your candidate belong to the appropriate *technical* organizations that deal with the scientific fundamentals? Membership in the Audio Engineering Society and the Acoustics Society of America are two very good indicators of someone who is keeping up with the basics, not just the buzz words. The pros will likely belong to both.

4 – OK, you have a candidate who has done some rooms *you* like and seems to be talking sense. Let's check out one of the very first tough questions he should ask you after the introductory chat is over: What's your construction budget? The answer to this question sets the framework for all his thinking in the ensuing exchanges. If money is too embarrassing to talk about, it is probably going to be hard to face other tough realities, like the following issue:

5 – Ask first not how your room sounds to you, but how it sounds to your neighbors! The word here is "isolation." If you need it and don't have it, you will find yourself in big-time legal trouble.

The time to think about this is *before* you commit, when you can still act on professional advice. We turn away calls all the time because the client has already bought a space that simply *cannot* be isolated.

If your candidate does not bring this matter up very early on, be wary: He may be afraid to tell you the bad news because it would risk losing the job.

To cite a recent real-life example, a drummer who had just bought a loft space with a wood joist floor and residential neighbors living below was in trouble. Period. In this case, an early telephone call was all it would have taken — now he's stuck with it. He wanted good news, and we had only very expensive bad news: We turned the job down.

So check out the expert before you commit to either the space or the expert.

Francis Daniel is a partner at ARcoustics, a studio design/build/products firm, and a senior associate at Shen Milsom & Wilke, an acoustical consulting firm, both in New York City.

DOING DALLAS

4

Creative interaction is the key to the design behind Dallas Austin's multi-project studio

BY PETER MAURER

DALLAS AUSTIN at the board.

My partners and I at studio bau:ton share one simple philosophy: treat every project as unique, with its own specific design, budget, and individuality. In a traditional commercial multi-room facility, the requirement is to design the space for client separation in order to protect the artist's privacy and creative environment. For a project facility such as Dallas Austin's in Atlanta, Georgia, the design requirements are quite different.

Dallas Austin is a collaborative individual and successful artist/producer with musical projects ranging from R&B to rap. He has his own company, Rowdy Records, and a rap group called The Highland Place Mobsters. It was at his request that we were assigned the task to design a facility that would accommodate producers, writers, and engineers who are all part of the same team. The lounges are shared and artistic ideas are created in an interactive environment.

A specific request was that Austin's facility, D.A.R.P., Inc. Studios, be designed to a "New York Street Scene" architectural theme. Throughout the design and construction process we made sure that this original idea was kept as an integral part of the project.

A personal studio is tailored more closely to the specific taste of its owner, while a commercial facility should have a somewhat broader appeal for a varying clientele. The professional project studio should have a personality, as it can be an aggressive marketing tool for producers, writers, and musicians to find and develop new talent. A project studio may include several writing rooms for MIDI production, which are available at low rates or sometimes made available for free to developing talent, to give them a comfortable opportunity to break into the music industry. After that talent has emerged, the tracking, overdubbing, and mixing can be done under the same roof.

When clients come to us with projects, we analyze their ideas and prepare a preliminary design scheme and cost estimate based on statistical data that we have accumulated and updated over the years. During this initial period, we are translating ideas into a possible real scenario. Most clients have great visions, but are inexperienced in gauging the expense involved. We attempt to educate our clients at every step of the process so that we stay within their particular budget requirements. Our greatest design challenge is to come up with alternative, economical solutions which still fulfill a client's initial dreams.
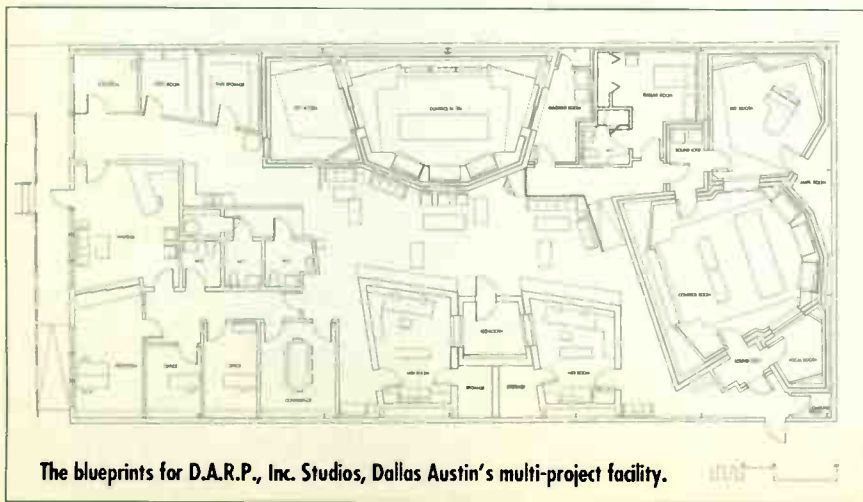
With the Dallas Austin project facility, we began with a building of 7380 square feet. Design began in March, 1992, and was carefully monitored by our project architect, studio bau:ton's Ross Brennan. Construction began in April and the first recording session took place in November. The facility is comprised of four studios — quite a step up from what most people think of as a "project" studio.

After completely demolishing and cleaning out the existing, free-standing building to an empty shell, the framing contractor was challenged with laying out a rather complex angular floor plan. It represents that back alley "New York Street Scene," with lots of corners and niches for hanging out in.

Floor-to-ceiling concrete block walls in the center seating area of the complex, which lies along the front wall of one of the control rooms and iso booth, and more block along the back wall of the large control room were chosen to take on the artistic finish as Dallas Austin desired. He hired two incredibly talented graffiti artists, Kenny Dred and JAZ who treated the concrete block walls in a spectacular and colorful way that perfectly finished off the street scene design goal. The high ceiling in the central lounge area is painted black and we added four skylights to bring in some natural light and to top off that urban feeling.

Studio A has a large control room with Austin's MIDI gear, a piano booth, a vocal booth, and a machine room. The console is a Solid State Logic 4000 E Series with G Series computer Total Recall and there are two Studer A827 24-track machines and two 2-tracks.

The blueprints for D.A.R.P., Inc. Studios, Dallas Austin's multi-project facility.

Studio B is a mix control room with a large overdub booth and a machine room. The console is a Solid State Logic 6000 E Series with G Series computer and Total Recall and the same complement of 24-track and 2-track machines. Both Studios A and B are equipped with almost identical, top-of-the-line outboard gear and effects processors.

Studios C and D are large, MIDI preproduction and composition rooms with a shared vocal overdub booth. Both of these studios have the expansion capability to be turned into low-cost mix rooms and are wired for 24-track recording.

The control rooms are finished in maple laminated plywood, some of it painted a hip black, but still showing the beautiful woodgrain against the dark blue, purple, and black fabric panels. The floors in the control room are carpeted with black Pirelli industrial rubber flooring behind the consoles. All the isolation booth floors are finished in raw maple with easily movable throw rugs.

And one last thing to remember: No matter how inventive the physical environment, both commercial and project studios rely on the people and the personalities that create the atmosphere.

Peter Maurer is partner with George Newburn and Peter Grueneisen in studio bau:ton (rhymes with Dow Jones). The L.A.-based architecture and acoustic firm has completed project studios for Peter Frampton in L.A., Walter Becker in Hawaii, and L.A. Reid and Babyface in Atlanta. studio bau:ton has also recently completed commercial facilities for the Record Plant, Post Logic, and 525 Post Production in L.A., and received an A.I.A. (American Institute of Architects) Honor Award for Studio X at Bad Animals, Seattle.

ACOUSTICAL MYTHS

5 A collection of time-honored studio design misconceptions — and the real truth behind them.

BY RUSS BERGER AND RICHARD SCHRAG

Acoustics can sometimes be a mysterious science. Logarithmic addition just doesn't come naturally to most of us, and the concepts of sound absorption vs. room modes and of reverberation vs. resonance aren't intuitive.

Little wonder, then, that applied acoustics — especially when the application is studio design — is full of myths, fallacies, and misconceptions. Sometimes it's a misunderstanding of the principles. Sometimes it's taking a grain of truth and using it incorrectly in a different situation. Sometimes it's solving one problem, but creating a bigger one in the process. Whatever the cause, a second look at traditional design concepts and construction techniques often reveals that what you thought was true about acoustics isn't always so.

Some of those misconceptions, though, have managed to become such standard practice that they give you ample opportunity to shoot yourself in the foot, acoustically, if you aren't aware of them.

MYTH 1: ABSORPTION FALLACIES

Absorption means reducing the sound, right? So putting some fuzzy material on my wall will keep my neighbors happy, right? Unfortunately, no. It's true that, when sound strikes a surface, some of the energy is absorbed and some is reflected from the surface. It is also true that some materials absorb more of the sound than others. In most cases, however, this may do a lot for the sound within the room, but doesn't help much when the problem is sound transmitted through the walls or ceiling of the room.

It's tempting to believe that soaking up all the sound will keep it from going somewhere else, and, in fact, increasing absorption does reduce the

Figure One

A   STC-56
B   STC-63
C   STC-48
D   STC-63

sound pressure levels from a given source. The rooms we live and work in generally have moderate absorption to begin with, though, so in a practical sense it is rarely possible to use "normal" finishes to make order-of-magnitude differences in overall room absorption. As a result, it is difficult to affect steady-state sound pressure levels in the space by more than a few dB with absorption alone. That doesn't mean that you can't make a room significantly quieter by changing the finishes. The harshness of a highly reverberant space doesn't stem from loudness as much as from factors like poor intelligibility and the direction and frequency content of the reflected sound.

Even in a completely absorptive (anechoic) environment, the sound pressure level at a wall surface still has a direct sound component that is dependent only on the sound energy that the source can produce and on the distance from it. No amount of absorption can reduce the level further.

Remember, too, that it is much more difficult to keep low-frequency sound from going through a wall than it is high-frequency sound. It is equally difficult to obtain very effective low frequency absorption over, say, a full octave or two. So the effect of absorp-
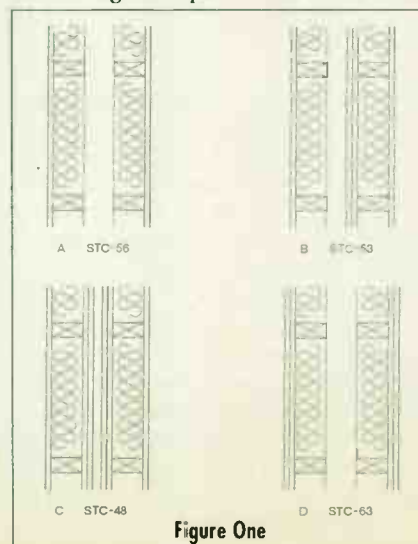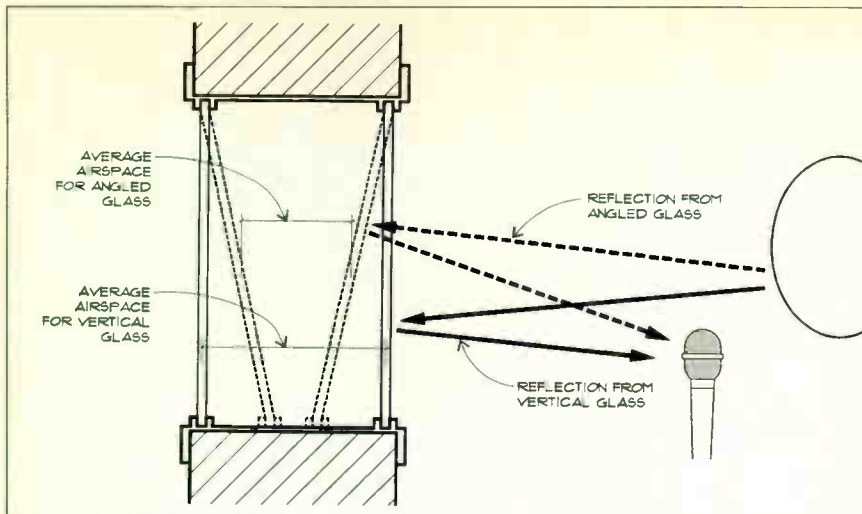
Figure Two

tion on sound isolation is at its least where you need it the most. Sound absorption can be one effective component of a larger noise control solution for problems involving mechanical equipment, because the sound power of the source is fixed. When dealing with voices or reproduced sound, however, an acoustically "dead" environment sometimes encourages you to speak louder or turn up the volume to compensate. This may offset any reduction in the overall room levels, or may actually make them worse.

The transmission loss through a partition is affected by the mass of the materials used, the thickness and assembly of the barrier, and control of flanking and structureborne paths. Absorption within the rooms in either side of the partition is a relatively minor issue. For sound isolation, there is no substitute for heavy, airtight construction, regardless of how you finish it.

MYTH 2: THE THREE-PANEL PARTITION
How many times have you seen articles on studio design in which "high-performance" partitions are detailed? Often these are touted as "triple walls" or described as a seemingly endless stack of different sheet goods with airspaces interspersed among them. ("We used the wallboard plus fiberboard plus wallboard, then a 1-inch gap plus wallboard plus rubber plus plywood, then a 2-inch gap plus...."). By serendipity, these walls may be sufficient for the needs of an individual studio, but they are not always a cost-effective use of materials or available space.

Take the example of a simple double stud partition. Starting with a single layer of gypsum board on the outside faces and cavity insulation

(fig. 1a), this wall has a Sound Transmission Class (STC) rating of STC-56. If we attempt to "improve" the wall by putting two additional layers of gypsum board on the inner face of one stud (fig. 1b), the STC rating actually *decreases* to STC-53. Following this "more is better" mindset, if we add two more layers of gypsum board to the inner face of the other stud (fig. 1c), the STC rating is even lower at STC-48. (Never mind the difficulty in actually building this version.)

So what are we doing wrong here? In a cavity wall, the transmission loss depends upon the mass (and stiffness) of the surfaces and the thickness (and absorption) of the airspace between them. In this example, putting gypsum board on the inner faces of the studs — creating a three- or four-panel wall — divides the airspace into smaller segments, and the low-frequency sound transmission loss (which in this case dominates the STC rating) is reduced.

If we merely add one layer of gypsum board to each outer face of the original wall (fig. 1d), an STC rating of STC-63 is achieved. This uses less material and less space than the four-panel wall, but gives significantly better performance. To optimize acoustical performance, *how* the materials are put together is often more important than *what* materials are selected.

MYTH 3: ANGLED GLASS
In traditional studio designs, interior windows — between a control room and booth, for example — very often have two panes of glass, with one or both tilted a few degrees from vertical. (Sometimes it's three panes — see Myth 2.) Several reasons are given for this design technique.

One reason is that taking the two panes out of parallel eliminates the resonances (standing waves) in the air cavity between them, which would otherwise limit the transmission loss at the resonant frequencies. In theory, this is a valid concern. In actual construction, however, there is always a practical limit to the overall thickness of the wall into which the window is built. Tilting the glass out at the top would put its center of gravity further out from the wall, and the structural support provided by the window frame and its attachment to the wall might come into question. Instead, the usual "solution" is to tilt the glass in at the bottom so that the two panes are very close together at the bottom of the window.

The result is an average airspace between the panes that is sometimes little more than half of what it could be if both panes were vertical (fig. 2). Since sound transmission loss through the assembly is highly dependent on the width of the airspace, the acoustical benefit of angling the glass is often negated by the reduced separation between the panes. Given an overall wall thickness, sound isolation
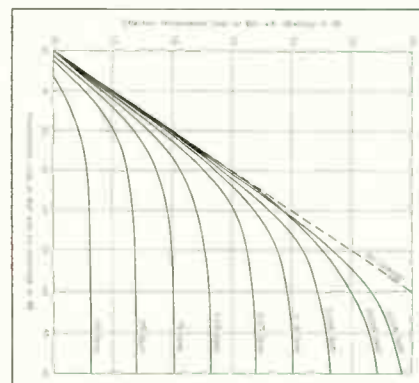

Figure Three

is maximized by maintaining the greatest overall distance between the panes.

A second reason for tilting the glass is to redirect reflections of sound from the window, which, owing to sightlines through it, is almost invariably at a height where reflections into microphones or listeners can occur. Unfortunately, the angle necessary to eliminate this problem is again usually more than the depth of the window frame can accommodate, and the detrimental reflection just occurs from a different point on the glass (fig. 2).

Actually, there are valid reasons to angle glass in double-pane windows, but they have nothing to do

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Reliable Music

November 17, 1992

Dale Kauffman
Crown Service

Dear Dale,

Reliable Music is a warranty service center for over 30 brands. As service manager, I spend quite a bit of time arguing with manufacturers about their warranty policies. So many of them want me to jump through all kind of hoops to service their equipment, and then pay me so little that I lose money on every repair. They want receipts, they want old parts, they won't accept NARDA forms, they provide me with miserable schematics (and no training), their parts departments constantly mess up my orders, and they expect me to put up with all this for virtually no money.

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And I'm giving up trying to tell them. I'll tell you instead what your fairness and support buys you; when a potential customer walks in my shop to ask my opinion on what type of amp he should buy, I answer without hesitation "Crown". If someone asks me how Crown amps are from a service standpoint, I tell them quite honestly they rarely break, and if they do, the customer will be taken care of. Quickly. At no charge. To their satisfaction. I'm confident about that, just as I'm confident a new Crown product will work well, that I'll be able to get parts and service manuals for it and that if it can't be repaired, you'll replace it.

What I'm trying to say is that my shop has your unqualified support, and customers know it, which is only one reason we sell so much Crown. If every company I dealt with understood this, my life would be so much simpler, but no one else shows signs of getting it so totally, completely right. My loss, your gain, I guess.
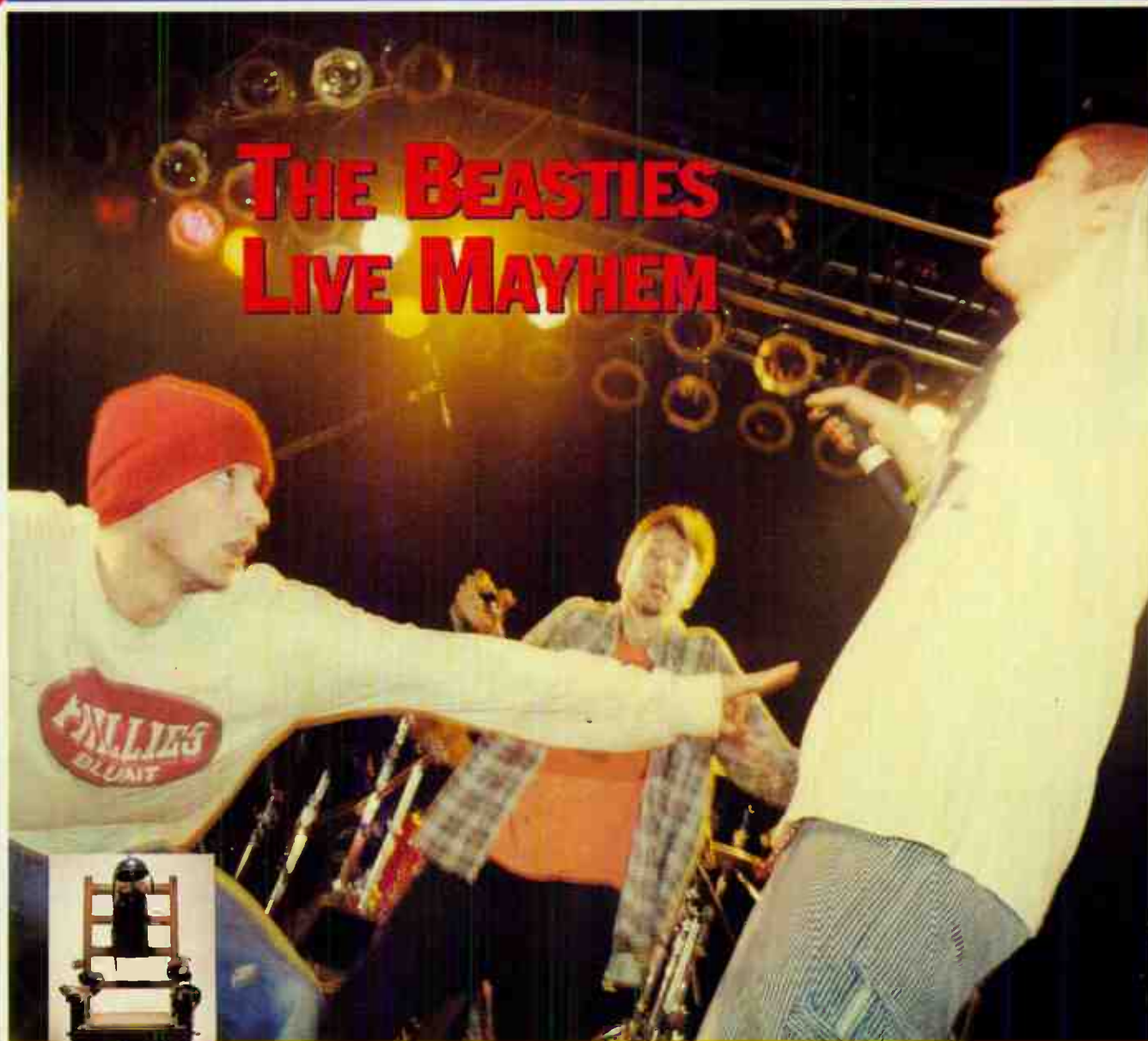
Thanks, guys.

Steve Stoeckel
Service Manager
Reliable Music

IN A BAND VAN

BAND

FLYING RIGHT

LIVE SOUND FOR THE GIGGING MUSICIAN

THE BEASTIES LIVE MAYHEM

INSIDE: *How To Avoid A Shocking Experience*
10 (OR SO) Tips On Using MIDI Live

BRINGING UP BEASTIE

■THE LIGHTS go down in New York's own Roseland Ballroom, a no-nonsense musical warehouse that has housed everything from the punk moshfests of the Ramones to the bluesy vibes of the Neville Brothers to the all-out funk jams of Prince. Now, as the outline of what appears to be a knocked-out elephant begins to glow a bright orange, a group that devotes equal time to each of the aforementioned musical genres takes center stage. The Beastie Boys, along with their mix-master, DJ Hurri-

cane, drive home a fierce set of songs that spans the musical spectrum of thrash, reggae, grass-roots funk, and the music they are best-known for — rap.

If the winds of change are retracting from the technology-driven rap shows of the late '80s and steam-rolling head-first into the realm of live, acoustic sound, then the Beastie Boys, comprised of King Ad Rock (Adam Horovitz), MCA (Adam Yauch), and Mike D ("the 'D' is for Diamond"). are the ones currently riding that wave to the extreme.

While rappers such as LL Cool J and Hammer have performed live acoustic renditions of their high-tech hits, bereft of the standard Akai MPC-60 backbeat, the Beasties have one-upped the ante by injecting a variety of musical styles into the mix. On any given moment, the souped-up "rap" act can be seen rhyming homeboy-style over sampled Zeppelin beats or Eagles rhythms when — without warning — a pummeling punk version of "Transit Cop" rears its head.

Is this a sudden rebirth? Or just another nostalgic, yet transient trip back to the days of old? Whatever the case may be, the Beastie Boys have made it clear that rappers and their ilk are determined to break out of the traditional role of "MC"

and earn their status as capable live musicians. Variety and vintage sounds are in — glitz and glad-handing technologies are out.

THE NEW FAT SOUND

"Some of the songs we played on tour had to do with our getting back to our roots," says resident Beastie Boy and lead punk vocalist, Mike D. "But a lot of the music was very new to us, derived from jams that evolved out of our project studio (G-Son, Atwater Village, California). We really didn't know what was going to work and what wasn't."

One of the show's many elements that did work — to maximum effect — was the sound system, which included Maryland Sound

MORE THAN JUST A MIC IN A SUITCASE, THESE RAPPERS ARE TOTING ALONG A BAND IN A VAN

BY JON VARMAN

Industries' MS12 speakers and an ATI Paragon console. After witnessing such rap acts as Public Enemy and Ice-T deliver incredible sets with supersonic distinction, the Beastie Boys decided that above all else, live sound was what mattered most. "There's just too many rap shows with bad PA systems," explains Mike D, "and a lot of today's rap audiences are burnt-out on what they're hearing because it just doesn't sound good."

In order to ensure top-notch production throughout the entire "Check Your Head" tour, the group employed the services of Mario Caldato Jr., a seasoned producer/mixer who immediately set out to organize a dynamic live sound event.

"I used Digitech's RSD 1900 throughout most of the tour," he says, "because it has a variety of great effects that can be manually altered. But for the hardcore stuff, I just laid low."

So, with all these different styles floating around, the one question that remains is: Where does the DJ fit in? After all is said and done, isn't he the engine that makes this boat float? "For the Beastie Boys, the DJ is just as important as the guitar, the bass, the drums, and even the congas," claims the group's mix-master and musical engine, DJ Hurricane. "Personally, all I need is a Technics turntable, my records, and a GLI mixer, and you've got yourself another instrument in the band." Not to mention a full-fledged jam-mate.

During the "Check Your Head" tour, DJ Hurricane's duties went far beyond the conventional realm of record-scratching and expanded into sections of the show that were formerly off-limits to DJs — namely, instrumentals. Here, he would skillfully blend in colorful cuts of artists such as Aretha Franklin and Stevie Wonder, while the Beastie Boys plucked and pounded away at their attributed axes. "My job is different from most DJs because I have to gibe with live musicians who are playing everything from jazz to funk. You have to know how to be on beat, because if you mess up, everyone's going to know."

Contributing to Hurricane's performance was a DJ coffin, suspended by springs and stocked with isolated turntables, that was designed to absorb a fair amount of the shock caused by rampant stage-stomping and other variables associated with a live show.

"If you're going out on the road, you've got to be precise every night," says Hurricane. "It's a lot different from DJing in a club, where you're standing in a DJ booth, doing whatever you want when you want it." To prepare for the tour's rap numbers such as "Slow and Low," "Shake Your Rump," and "Eggman" (their breakthrough hit, "Fight For Your Right [To Party]" was curiously omitted), Hurricane had 30 to 40 soundtracks made of each Beastie Boy album, completely devoid of the rappers' vocals. All that remained were the beats, rhythms, and samples, which enabled Hurricane to spin a soundtrack on one turntable while cutting in handpicked records (usually obtained in that town's record store) on the other.

"I have to have a lot of extra soundtrack records nearby so that if I hear a scratch on one of them during a show, I can just take it off, break it, and put another one on the turntable right away. If that record's damaged, I don't want to hear it ever again." For extra-bouncy stages, Hurricane also

had CD soundtracks of each Beastie Boy album ready, just in case the on-stage antics became too much for his shockproof coffin to handle.

Now that the Beastie Boys have laid down the groundwork for future concerts, they seem poised to break more boundaries. "We play what's in our hearts," says DJ Hurricane, who makes no apologies for the band's brazen style, "even if it's shocking to a lot of people. We'll jam on any type of song, no matter how strange or different it may be, and now our audience knows that."

But where does that leave the Beasties' one-and-only bona fide hit, "Fight For Your Right (To Party)"?

"Are you kidding?" says Hurricane, "we hate that song!"

THE BEAST MASTERS

When the Beasties unleashed their hip-hop hoe-down on the American public at large, it was a safe bet that many of the songs that thrashed and glided their way across the live stage would one day appear on wax...er plastic. After all, it's not every day that a rap group dons acoustic instruments to bang out punk tunes and Santana-style anthems as if the drum machine had suddenly gone out of style.

So when this triumvirate of rappers zoomed through the live circuit, DJ and band in tow, they brought along a Tascam MSR-16 recorder for the ride, counting on the system to record their every note, as well as the audience's every scream of "More, More!"

Most of the songs that were recorded for live purposes will eventually find a home on Beastie Boys videos, B-sides, film soundtracks, and, possibly, the band's follow-up to the *Check Your Head* record. While some of the act's rap material was taped on a DAT player, the live band material had to be mixed down and recorded on the MSR-16 in order to give the Beastie Boys' chief engineer, Mario Caldato, Jr., complete control. "The fidelity was good, the sounds were clean and it captured the excitement of a live song," says Caldato, who believes that some things you hear in a live show can never be obtained in the studio. "The liveness of a gig adds a bit of magic to an otherwise ordinary track."
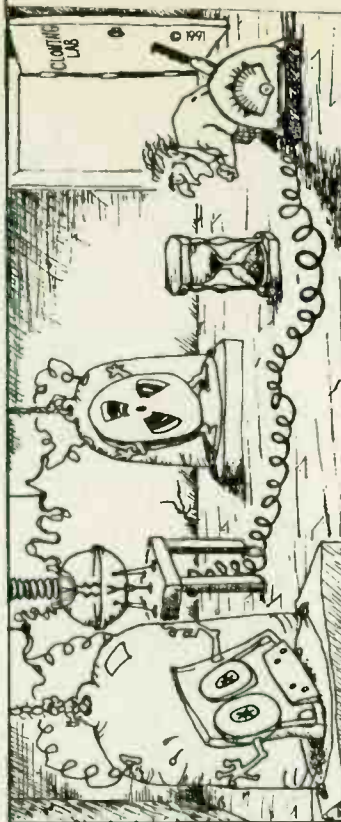
Everything, he claims, "went straight to tape—sink or swim." The audience had their own tracks (picked up with two room mics), while the drums came up in stereo, the keyboards and DJ went down on two tracks, and the bass went down on one. Additionally, four individual mic tracks were set up for the Beastie Boys and their main man, DJ Hurricane.

Tascam's 1/2-inch 16 track recorder features dbx Type-I noise reduction, auto punch-in/out functions and two speed settings. Other assets for rappers-on-the-run include dump and manual edit, spot erase, a sync lock switch, and bar-graph level meters with peak-hold indicators. It comes ready-made for external control via computer or the Tascam MIDIzER.
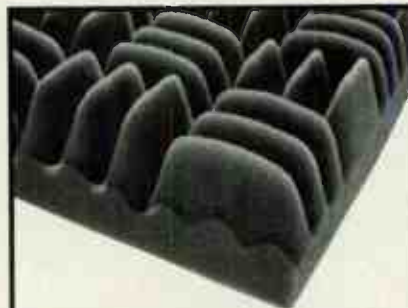
—Jon Varman

COME FLY WITH ME, PART 2

■ONCE THE loudspeaker enclosure is suited for flying, as was discussed last issue, the flying hardware system is the next consideration. For the most part, these systems can be categorized into three distinct groups: grid-truss (strap type) systems, stud-fitting systems, and modular mounted-truss systems.

A grid-truss, or strap type, flying hardware system makes use of an overall top truss from which the loudspeakers hang. Straps and cables hold the loudspeakers in place and adjust the tilt to the loudspeaker columns, while each loudspeaker is suspended by the one above it, and so on. Grid-truss flying hardware systems are easy to design and can appear to be cost-effective at first; however, this type of system can become very costly if used long-term. It is also difficult to control loudspeaker array direction and splay characteristics in a grid-truss system. For this reason, a grid truss system is very labor intensive and provides poor acoustic wavefront coherency.

Stud-fitting systems employ grabber clips and an assortment of small hardware components that rigidly lock the array together in various formats. Stud-fitting systems array the loudspeakers well and maintain a coherent wavefront. These systems, however, are very difficult to assemble, taking longer to assemble than other types of flying hardware systems.

The modular mounted-truss flying hardware systems use compact truss modules that are permanently affixed to the loudspeaker enclosure. The modular-mounted truss systems array the loudspeakers well and maintain a coherent acoustic wavefront. System designs are simple and assembly is intuitive. Such systems also assemble quickly and with a minimal number of crew members.

With a viable loudspeaker rigging system engaged, the next focus of attention is the use location. Safety is the primary concern. Loudspeaker array locations should be as far away from people as possible. Attachment to the venue structure should be appropriate, and be performed only by a qualified rigger. When assembling the loudspeaker array, be conscious of the surroundings at all times. Check and then double-check the array assembly to ensure that the rigging system is assembled properly, and that all attachment hardware is tight and shows no signs of wear or fatigue. Remember that the ultimate responsibility for any accidents that may occur falls on the owner of the loudspeaker flying system.

Other components of the rigging system include shackles, wire ropes, carabiners (oblong rings that hold a freely running rope), straps, round slings, clips, chain motors, beam clamps, and various other hardware. All of these must be designed and implemented into the system with the 5:1 design factor discussed in the previous issue. Also, all the components must be rated for overhead lifting. Aluminum carabiners and carabiners without locking gates have no place in a loudspeaker rigging hardware system. If forged eye bolts are implemented in the system they must be of the shoulder variety, and the angle of load must be thoroughly understood before the bolts are used. Shackles, quick-links, and clips must also be thoroughly understood, and the rules that apply to the direction of load must be observed. In many cases, the angle of the load will cause a decrease in excess of 50 percent in the safe working load of the component. Remember that the loudspeaker rigging hardware system is only as strong as the weakest link in the system.

The loudspeaker rigging hardware system can be the safest system in the

GETTING THE HARDWARE TO HELP YOU STRAIGHTEN UP AND FLY RIGHT
BY ANDREW MARTIN

IN A BAND VAN

world, but the venue must be able to support the load for the system to be of benefit. Specific load points should be available at any venue where loudspeaker arrays are to be flown. The load points should be engineered and load-rated with a design factor in place, and for a specific load angle. Many times, club venues will have forged eye bolts or suspension plates in the ceiling structure; these points must have a specified safe working load. Don't take the production manager's word for it, ask for proof; remember who is responsible if there is an accident. Larger venues may have a support structure constructed from steel, concrete, or wooden beams. Again, any suspension point should be load-rated and the 5:1 design factor must be applied. In any case, the work performed to attach to the venue structure should be done only by a qualified rigger.

Although flying loudspeaker enclosures in club venues can be a challenge, the benefits of a flying loudspeaker system are tremendous. The added intelligibility and increased gain distribution can turn a mediocre show into a brilliant success. Nonetheless, the process of flying loudspeaker enclosures must be approached with one primary concern at all times: SAFETY.

Andrew Martin is president of ATM Fly-Ware.

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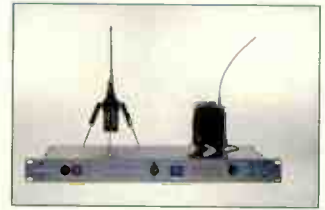

NEW GEAR FOR YOUR NEXT GIG

CUT THE CHORD

Shure Brothers has added the new Guitarist and Vocal Artist Systems to its arsenal

of wireless products. The company says the lines have been designed to offer quality audio and reliable RF performance to the first-time wireless buyer. The Vocal Artist System features Shure's L2/58 handheld transmitter, which matches the sound of its cabled progenitor, the famous SM58. The Guitarist System utilizes Shure's L11 body-pack transmitter and WA300 instrument cable which, the manufacturer states, provide clean, reliable signal transmission in a small, rugged

package. A third member of the line, the Presenter System, includes the L11 and Shure's 839W omnidirectional lavalier mic. All three systems operate on a single 9-volt battery that provides an average of 12 to 14 hours of playing time. To cut loose, contact Shure Brothers Incorporated, 222 Hartrey Avenue, Evanston, IL 60202-3696. Tel: 800-25-SHURE. Circle EQ free lit. #116.

EAR TAILORS

Future Sonics's Ear Monitors are custom-fitted in-ear stage monitors. They are high-fidelity stereo headphones that, when custom-fitted to the artist's ears,

transform the ear canal into an acoustical suspension speaker, states the manufacturer. Each pair is cosmetically fitted to be virtually invisible to the audience. Aside from their potential impairment of performance, the feedback and distortion caused by conventional monitor speakers and monitor amplification systems can be damaging to your hearing. Ear Monitors are designed to operate at safer output levels than conven-

---

tional monitor systems. Used properly, they help to protect your hearing and voice while enhancing the quality of the sound you and the audience hear. Since the system is custom-fitted, the purchase of a set of Ear Monitors involves an impression being taken of your ears by an audiologist or hearing center. List price for one set of Ear Monitors is $1500. For further information contact Future Sonics, Inc., 655 Danbury Ct., Newtown, PA 18940. Tel: 215-579-4414. Circle EQ free lit. #134.

MC YAMAHA

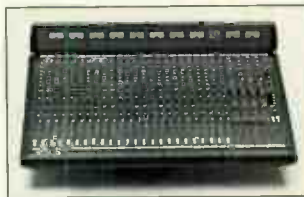Yamaha Professional Audio has introduced a new line of

four live production mixing consoles: the MC2403, MC1603, MC1203, and MC803. They offer 24, 16, 12, and 8 inputs, respectively. Channel gain has been boosted to 60 dB for enhanced signal-to-noise performance. The MC03 Series utilize analog meters for optimum visibility in outdoor applications, and a fourth output level meter has been added. Other new features include a left and right (L&R) mix function, mix (L&R) cueing, on/off switches on all channels and busses, sub outputs (L, R, and L&R), bus muting (L, R, L&R, and AUX 1-3), and the ability to jump the channel inserts for pre- or post-EQ operation. For complete information, contact Yamaha Corporation of America, Professional Audio Department, P.O. Box 6600, Buena Park, CA 90622. Tel: 714-522-9011. Circle EQ free lit. #115.

ESTEEMED SPEAKER

The CR Series and CRi Series loudspeakers from Celestion are portable and versatile. The line offers a variety of enclosures starting at $319 retail, designed to fit any general sound reinforcement requirement. The enclosures feature a unique curved front styling that enables arraying a series of cabinets in arches. All of the cabinets are firmly constructed, glued, and assembled with steel fasteners, and each has rigid internal bracing. A protective 18-gauge metal grille provides front protection, and enclosures are protected with molded stacking corners. Dual parallel 1/4-inch phone jacks and a high-frequency-level control are mounted into a recessed molded plate. All CR Series enclosures are constructed of plywood and covered in black carpet. CRi Series enclosures are constructed of multiply, void-free birch, and are finished in black-splattered acrylic. Each series includes a two-way 10-inch, two-way 12-inch, three-way 15-inch, and three-way 18-inch full range

system. Fifteen-inch and 18-inch subwoofer systems are also available, as are two-way 10-inch and two-way 12-inch floor monitor systems. For further infor-

mation, contact Celestion Industries, Inc., 80 Doug Brown Way, Holliston, MA 01746. Tel: 508-429-6706. Circle EQ free lit. #113.

TAKE YOUR SOLO

Following the success of the Solo Live and Solo MIDI mixing consoles, Soundtracs has premiered three new additions to the line: the Solo Monitor, Solo Live 8, and Solo Logic. The Solo Monitor is a 10-bus monitor console, designed for sound contracting and live sound applications. The input channels are equipped with 4-band EQ, signal-present

LEDs, four mute groups for scene muting, as well as a mic splitter. There is a one-band fully parametric EQ on all outputs, as well as individual metering. The Solo Live 8 features eight busses, a full-meter bridge, four mute groups, four stereo effects returns, and six discrete aux sends per channel. The Solo Logic expands on last year's Solo MIDI and features a full-meter bridge, full 12-bit VCA fader, and mute automation that includes update, trim and isolate modes. The Logic is also equipped with complete MIDI machine control. All three models are available in either 24- or 32-channel consoles and come with an outboard 19-inch rackmount power supply unit. For a Solo flight, contact Samson Technologies Corp., P.O. Box 9068, Hicksville, NY 11802. Tel: 516-932-3810. Circle EQ free lit. #114.

MAXIMUM MONITOR

The ITE-20 wearable stage monitors from Circuits Maximus Company, Inc., (CMCI) were designed by leading research audiologists specifically to perform as safe and effective stage monitors. The ITE-20 system consists of soft, durable ear seal flanges that precisely couple a very lightweight driver element to the ear canal. The flexible flanges softly hug the sides of the ear canal to form a triple seal without discomfort, says the manufacturer. Frequency response is smoothed by an acoustic damper from 80 to 15,500 Hz. The ITE-20's are available in either brown or beige to match skin tones. When they are in place they are comfortable and virtually invisible. Eartips are available in different sizes on special request to accommodate smaller- or larger-than-average ear canals. The eartips are easily removed and can be cleaned and reused or simply replaced with new ones. The ITE-20's can be used with CMCI's System 1200 control electronics, which provide a stagebox to accept a mic or line input, a 15-foot super flexible multicable, and a beltpack that gives the wearer complete volume and mix control of the monitors. A wireless version will be available in the spring. List price is $1200/pair. For more information, contact CMCI, 214 N. Park Drive, Arlington, VA 22203. 703-276-0125. Circle EQ free lit.#135.

PLAYING LIVE (AND STAYING THAT WAY)

■SEVERAL MONTHS ago singer-musician Jonathan Wild was booked to provide the entertainment in a brand-new banquet hall at one of northern California's wineries. After his equipment and speakers were positioned on stage, he looked around for a grounded outlet to power up his sound system. The "venue electrician," who happened to be the caterer, took off with the plug and, moments later, POW! Jonathan, who was kneeling in front of his equipment rack, got a faceful of smoke and a sick feeling inside. Needless to say, he was unable to perform without his equipment. The bright side was that he didn't get killed.

Why did this happen? It was a combination of unsafe wiring, complacency, and ignorance. I investigated this accident myself, and I discovered that the "helper" had forced a standard 120-volt plug into a 208-volt receptacle. I also found that three of the five extension cords used at the venue were missing the grounding prongs. For several seconds, the metal parts of the rack were energized at 120 volts above ground, and Jonathan's amplifiers, effects, wireless system and

mixer were seeing 208 volts. If he had put down a hand to steady himself, he would have gotten 120 volts across the chest — a normally fatal shock!

Reread the last paragraph and take note that I did not include unsafe equipment in the list of causes. That's because I designed Jonathan's rolling rack myself; every component is solidly grounded and the 14/3 power cable has a heavy-duty grounded plug. The equipment was safe, but the venue wiring was not. If all the grounding prongs were in place, a cir-

cuit breaker would have tripped immediately and prevented the equipment damage and shock hazard. Although a grounding prong is often the only protection against electrocution, I routinely find them missing on guitar amp power plugs (photo 1). An important thing to remember is that, although it is common to use a "ground lift adaptor" on the end of a grounded power cord (photo 2) to locate a noise problem, they should *never* be left attached to the plug for normal use. You could get killed as a result.

COMMON GROUND

Here's why everything should be grounded: Inside nearly every guitar amplifier, sound mixer, or outboard effects unit, there are one or more capacitors connected between the AC power circuit wires and the metal chassis. Although their purpose is to filter high-frequency noise and spikes, they will pass a small amount of current at 60 Hz, the power frequency. If the component is solidly grounded, this small leakage current is safely conducted back to its source by means of the "equipment grounding conductor" — the green wire that is connected to the

IF NOT HANDLED PROPERLY, YOUR GEAR CAN BE AS LETHAL AS AN ELECTRIC CHAIR

BY ERIC G. LEMMON

round grounding prong on each plug. If the grounding circuit is interrupted, either deliberately or by accident, the leakage between the 120-volt "hot" wire and the metal chassis will cause the latter to be at some voltage between zero and 120 volts, relative to ground potential.

Let's say you are holding your electric guitar neck in one hand, with your sweaty palm in contact with the metal strings. The strings are grounded to the pickup shield, and so on, and the cable is plugged into your guitar amp, which is now "hot" because of the missing grounding-prong. You reach out to grab a mic, which happens to be solidly grounded through its cable to the mixer, and POW!

You thought that clipping off the ground prong would fix your hum problem, but it almost fixed you instead, didn't it? Dust and moisture that get inside your equipment will significantly increase the leakage and the shock hazard, so keep the guts clean and bone dry. Never place drinks where they can spill into your gear.

POWER PLAYS
The best way to eliminate noise in a sound system is to have all equipment and instruments plugged into one outlet. Remember, we're talking about a small band now, not Metallica! As added insurance, when the outlet wiring is questionable, I will use my 1000-watt shielded isolation transformer (photo 3). An isolation transformer can literally be a lifesaver when a gig is outdoors, and the only available outlet looks like a self-help project. It may be prudent to use a portable
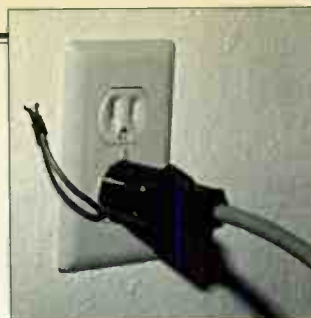

PHOTO 2: A ground-lift adaptor

generator in some cases.

Suppose you have had your equipment, power cables, and plugs checked out and repaired, and a qualified electrician has made you a power distribution box with, say, eight outlets on one end of a 75-foot-long type-12/3 SOW cable, and a heavy-duty grounded plug on the other. With this power cable, you can normally find a grounded outlet that hasn't been worn out by use and abuse. Try to find one close to the main panel.

Before you plug into a power outlet, check it first with a simple tester (photo 4). These testers cost about

PHOTO 1: Beware of missing grounding plugs.

PHOTO 3: An isolation transformer

PHOTO 4: A simple tester

these simple testers will give you assurance that the outlet is "hot" and the polarity correct, they have a serious deficiency: they *cannot* determine the quality of the ground connection. Testers that can check the ground cost around $200.

Perhaps the most important tester in my tool kit is a "Tic Tracer," a voltage detector made by TIF

five bucks and are available at most hardware and electronics stores. Although

Instruments (800-327-5060). When first turned on, it emits a slow peep-peep-peep, but when the detector "nose" is brought near any energized metal or wires, the peeping tempo rapidly increases to a steady tone. I use it just before sound check to make absolutely certain that all mics, mic stands, guitars, amp racks, stage
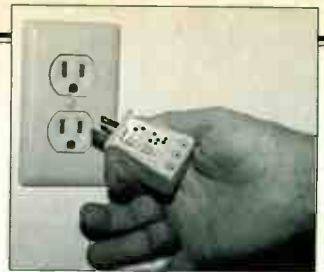
rails, dimmer packs, and lighting poles — everything that can be touched by the performers, the crew, or the public — are at ground potential (photo 5).

Almost every entertainer I know can recall at least one instance of getting a shock while handling a cabled microphone or

PHOTO 5: Testing the mics

instrument. They are lucky — they lived to tell about it. But, being smart about electrical safety is a heck of a better deal than being lucky — you'll live longer.

Eric G. Lemmon is the owner of Videotel Sound, an 8-track and MIDI recording studio near Lompoc, California, and is a consulting engineer who specializes in electrical systems for entertainment venue lighting and sound reinforcement. He is a member of the Institute of Electrical and Electronics Engineers, the Power Engineering Society, the International Association of Electrical Inspectors, and the U.S. Institute for Theatre Technology.

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FREIGHT TRAIN IN A VAN

BAND NAME: Freight Train Jane

MEMBERS: Tommy Bolan, guitar; Jamie St. James, vocals; Scott Werner, bass; Skip Tease, drums.

LATEST RELEASE: Currently preparing to shop a new 24-track demo engineered by Pat Regan at Fortress Recording Studio.

WHERE THEY'VE BEEN: The Roxy, The Whiskey, The Marquee, The Troubador, The Omni, Florentine Gardens, Spice, Goodies, FM Station, The Waters Club, Niterock Cafe, The Palace, Palomino, and various others.

WHERE THEY'RE GOING: Waiting on dates in San Francisco, Los Angeles, San Diego, Oklahoma, and Texas.

HOW THEY GET AROUND: U-Haul, Hertz, or Ryder

SOUND ENGINEER: Tom Dubielak

CONSOLE: ART Phantom 2408

CONSOLE RACK: ART DRX-2100; ART MDC-2001 (4); ART HD-31 (2); Lexicon PCM-60; Yamaha SPX-90; Yamaha REV-7; Klark Teknik gates.

SYNTHESIZERS: Casio PG-380 guitar synth and VZ-10M rackmount

EFFECTS: Guitar: DigiTech IPS-33B; Furman PL-8 (2); Kitty Hawk MIDI patchbay and mixers; ART SGX-2000, MDC-2001, DRX-2100, Multiverb Alpha, SGE-MACH II, DRX, I EQ, HD-15; Ashly SC-33 noise gate; BBE 822 sonic maximizer; Cry Baby Wah & Heil talk box; Rocktron Hush II C; Samson Concert TD wireless; ART X-15 MIDI Pedal. Bass: ART SGX-Nightbass; Rocktron Hush II B; Nady wireless; ART X-11 MIDI pedal.

LOUDSPEAKER SYSTEM: House or rental, depending on club.

AMPLIFIERS: Guitar: Engl; Kittyhawk. Bass: Yamaha; BGW; H&H.

OTHER GEAR: E.S.P. electric guitars; J. B. Player acoustic guitars; Seymour Duncan pickups; Kahler tremolos; D'Addario strings; Spectraflex cables; CB Labs pocket rocket (headphone amp); Gripmaster Riff Grip; Black Mountain instruments (dulcimers); Raxxess Metalsmiths; L.P. Music (percussion, etc., etc...); Dunlop picks.

BEST LIVE EXPERIENCE: Bolan remarks: Playing the 1993 Winter NAMM Concert Event at The Marquee Club, where Freight Train Jane gave away $5000 in free gear sponsored by my endorsers. It was great to see the looks on people's faces as they won.

WORST LIVE EXPERIENCE: Continues Bolan: Without a doubt, the night we played this L.A. club. First, the truck broke down and all our gear was unloaded into a new truck. In the process, both of my pedalboards got run over *twice* by the new truck as the gear was being reloaded. They still worked even though parts were smashed and beat up. Next, in front of a huge crowd, we blew the house power three times (I have the video to prove it). To top it off, our publicist, Joan, went flying down a flight of stairs backstage. Luckily, she looked a lot better than my pedalboards did when the "gig from hell" was finally over.

TOUR TECH TIPS: Bolan states: The Phantom is a great board because it's quiet and has tons of headroom for live use. It also has eight monitor/aux sends so we can do a complete monitor mix without an extra board. It's durable enough to handle heavy touring use. Also, I place my MDC-2001 last in the chain before my power amps. This allows me to crank my amps up and get the tubes cooking. Because of the 2001's design, however, I can back the volume off on the 2001 and get a great live-stage volume with all the sustain and crunch I need — without blasting my soundman out of the water. ☿

FRONT (L TO R) PLEASE INCLUDE: Jamie St. James, Tommy Bolan, Scott Werner, and Skip Tease. Back, left to right Ludwig and Roger Southall. Hall outside Fortress Rehearsal/Recording Studio, North Hollywood, CA.

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LSD TRIPS IN A VAN

BAND NAME: Life, Sex & Death (LSD)
MEMBERS: Stanley, vocals; Bill E. Gar, multi-string basses and backing vocals; Alex Kane, The Plank and backing vocals; Brian Michael Horak, drums.
LATEST RELEASE: The Silent Majority (Reprise Records/Warner Bros.)
WHERE THEY'VE BEEN: United States and Canada
WHERE THEY'RE GOING: They are opening for Poison on their U.S. tour.
HOW THEY GET AROUND: 15-foot Econoline tour truck. Sometimes they rent a tour bus.
SOUND ENGINEER: Andy "Pinball" Morrison from Buffalo, New York.
CONSOLE: 40-channel Soundcraft 500

RACK: Klark-Teknik EQ's; dbx 160 XT; Brooks Siren crossover; Yamaha SPX 900; dbx 463; Samson VHF wireless system; Eventide Harmonizer 3000S; Roland SDE 3000; Yamaha Rev 5; Lexicon PCM 42.
SYNTHESIZERS: Roland 300; E-mu Proteus I and II.
LOUDSPEAKER SYSTEM: Electro-Voice MT Manifold series; Ampeg 8x10's; JBL; Pignose amplification; Celestion.
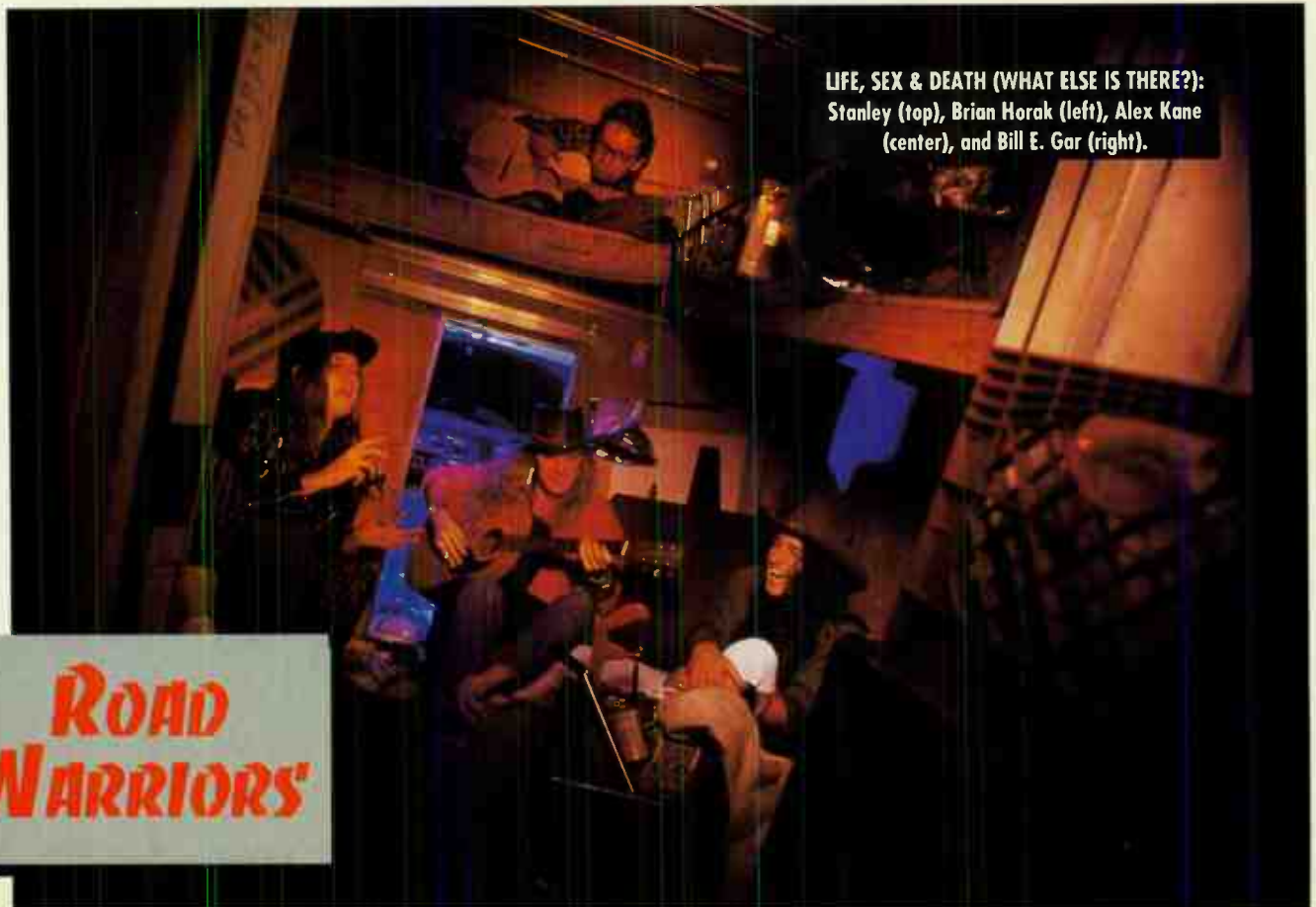**AMPLIFIERS:** Marshalls; Smith; Crest Audio; Ampeg SVT II; Crown; BGW; Peavey.
OTHER GEAR: Shure Beta 58, Balalaika; Hohner Golden Melody harmonica; Gibson semi-hollow body; Ludwig Vista-lite Clear (4-piece).
BEST LIVE EXPERIENCES: *Bill E. Gar* — Livestock Festival in Tampa, FL; Hammer-jack's in Baltimore, MD; Cain's Ballroom in Tulsa, OK; City Limits in Dallas, TX. *Stanley* — The next show, the next small town, the next big city. *Alex Kane* — Any night when all my gear actually worked correctly.
WORST LIVE EXPERIENCES: *Bill E. Gar* — The night in Texas when the sound system blew up and couldn't be repaired. The cables under the console looked so much like spaghetti, an Italian grandma would be proud. *Stanley* — The gig we were banned from playing. *Alex Kane* — Getting back into the "tour van," where you can imagine the stink of it all. *Brian Michael Horak* — The time in Melbourne, Florida, when I got the chicken pox and couldn't play.

TOUR TIPS: *Bill E. Gar* — Andy Morrison says, "Trust your ears first, then check out the VU meters." *Stanley* — Try not to break things. *Alex Kane* — Since we're a very physical act and tend to be destructive (a very positive and purifying — though expensive — method of venting primal rage), I know how much abuse my stuff can usually take. I won't let anyone but my tech (Johnny Bullet) play my gear as I feel it interferes with the somewhat metaphysical bond between player and gear. If your stuff goes down on stage, roll with it and keep a sense of humor. *Brian Michael Horak* — At the end of our set, I go into a drum roll — I literally make the drums roll across the stage.

LIFE, SEX & DEATH (WHAT ELSE IS THERE?): Stanley (top), Brian Horak (left), Alex Kane (center), and Bill E. Gar (right).

ROAD WARRIORS

Photo by Blake Bogdanovich

PLUGGING INTO MIDI, LIVE

■MAKING BETTER use of those MIDI ports on the back of your processing gear? Taking control of keyboard submixes on stage? Resetting your digital cross-over between acts? Here are a few ideas for plugging applications into those MIDI ports that may save you time when the curtain's about to rise.

1. Ever wish for a new mixer that had muting groups? MIDI gates can be inserted into channels of your current mixer, allowing them to be assigned to muting groups using a MIDI remote control; at least until your budget can stretch to a completely new mixer. LA Audio's MIDI Gate can be inserted into 16-input channels (per 1U device) to add MIDI-controlled muting and it also works as a conventional gate or provides MIDI triggers from the audio signal to drive samplers for background vocals or drum sounds.

2. MIDI-controlled EQs are affordable enough that in those really difficult situations, a MIDI EQ could be inserted into the channel and the presets for each extreme recalled with a single button. If you need dynamic EQ changes that follow the performer, assign the filters to faders on a MIDI remote control and set the fader travel to go between normal and those extreme EQ posi-

tions. This can keep your eyes on the performance without finding you've overshot the EQ setting.

3. If you need more effects units than you have effects returns, then it might be time to consider a MIDI-controlled audio patching matrix. These units, such as Anatek's SMP-16, can take

Rane's RPS-4 and the Alesis DataDisk SQ (top)

all those inputs and either parallel them in various combinations or swap them between effects returns as they are required. Once memorized, these combinations can be controlled from MIDI program change commands. Then you just have to press a button between songs instead of diving into the connector panel.

4. MIDI audio patching can also be used to reroute the mixer outputs. Audio switching matrices are common in theater applications where loudspeakers may be placed all over the stage,

each used for only a single sound effect. Onstage monitors sometimes get into nearly the same situation; where an extreme downstage left wedge is only needed for the solo in the closing number and another is simply for the percussionist in one break. Rather than use up mix busses, assign the sends to a switching matrix that changes the signal routing for these special numbers, returning the mix busses to other monitor feeds during the rest of the show. The audio patching matrix can also be used to mute sends until they are required.

5. Once you really max out the MIDI capability of your rig, it will be time to add a MIDI Sys-Ex storage unit to store those carefully built presets onto a floppy disk. If the very worst should happen (i.e., one of your main effects units dies) a substitute of the same model can be borrowed and your presets downloaded from a floppy disk. Alesis makes the DataDisk SQ, a unit that fits nicely into 1 U

of any effects rack to give you one less thing to worry about. If you don't think you have that many presets — yet — then at least consider making a written list of the parameter values in all your carefully-crafted presets. The SQ's built in MIDI sequencer can also be handy for generating anything from a test sequence of program changes to controller macros that change effects parameters on cue during the show.

6. MIDI sound modules can be pressed into service as tone and noise generators. This can come in handy when a video crew shows up unannounced and asks for a feed from your mixer. The second thing they will ask for is a tone so they can set levels. Most synths can produce a steady tone once you've turned off all the modulation. This tone will make everyone's meters agree on where 0 VU is.

7. Label your rack of effects, it can really save your sanity when things get strange. Attaching small labels to each effect unit between the rack bolts that display the MIDI channel and a default MIDI program number will allow you to quickly recover from MIDI mismanagement or the warm-up act's overzealous mix maniac.

8. If you need lengthy cables to send MIDI signals across the stage or to the house mix position, you will need an adaptor such as those made by Lone Wolf, JLCooper and Anatek. These units will lift the 50-foot-

TEN (OR SO) TIPS FOR MAXIMIZING YOUR MIDI ON STAGE
BY WADE MCGREGOR

cable-length barrier constraining conventional MIDI cables.

9. MIDI remote controls, such as the Lexicon MRC, Fostex Mixtab, JLCooper FaderMaster, and others, can be used to reduce all the front panel controls on your effects units to the few that actually need to be changed during the show. They can also give you control over many parameters that are inaccessible from the front panel of some units. Devices that lack front panel control for some or all of their dynamic controls do, however, feature a learn mode that will assign an incoming MIDI control message to the selected parameter. Some of these devices include the Lexicon LXP-1 and LXP-5 and Niche ACM MIDI attenuator.

10. Don't feel you have to be limited to faders for MIDI control. If you have simple MIDI requirements, Rane's RPS 4 remote program switcher converts a switch closure to a MIDI program change command. Four switches can call up four different MIDI programs in your effects rack and be mounted wherever they are convenient — such as an empty bit of a mixer strip.

And one for good measure... Footpedals, footswitches, and pushbuttons are a control tool for synths and samplers, and these can also be used at the house mixer for muting, level changes, and program changes. The most obvious applications of the footswitch is stepping between programs in the effects units, but it can also mute the reverb, cut the echo regeneration, or sustain that last chord while the band leaves the stage. If you already have your hands full mixing, perhaps you have overlooked this part of the choreography.

recently finished recording, "You Gotta Love It." Steve and I thought we had overdubbed all the parts. We hadn't played it back and we had put it away because it originally wasn't intended for this album. So, the day before we were supposed to be done mixing, we put the song up and there was nothing on it. None of the keyboard parts, none of the strings, none of the background vocals. Nothing. We figured it would take days to re-create it.

Enter our hero: engineer Brendan O'Brien. He came in and played us the demo so we could hear what it sounded like. He suggested we get a couple of keyboards and start screwing around, but we called all over and couldn't get hold of any. We were starting to pull our hair out! So we ended up flying the keyboards off the demo. Steve figured he could place the vocals in about two hours, but Brendan suggested flying them, too, and it sounded good to us.

But we weren't on easy street yet. We placed the demo side-by-side with the new tape and the tracks were slightly out of time. The SMPTE wouldn't lock up. Brendan flew it in and every four or six seconds he had to stop the tape, fly it in, and slap one reel with his hand because it was moving too fast. I hadn't seen that done in 15 years. It worked great and took him 45 minutes tops.

ON THE ROAD AGAIN

With the new album just released, we'll be going out on tour to support it. Right now I'm not sure exactly how we're going to do it — everytime we go out we do something different.

One thing I know will be different: Whenever I have any effects like reverb or echo, I would rather have it go into the board rather than through pedals. I prefer to have it go straight to my amp and then have whatever effect I want put on at the monitors. This time around I know I'll have a system like that set up.

What I'll probably do is set up a console backstage just for my guitar effects. That way, the signal can go right to that console, where we can add the effect and adjust the EQ. I figure the chain will go: guitar, amplifier, microphone, my backstage console, and finally, the house mixer.

PERRY-SPECTIVE

The reason that Aerosmith has stayed so strong for so long is that everybody's got an equal say. Everybody's got their strengths (Brad, for example, goes really wild with the videos). If anybody is given too much free rein, then they'll get carried away with it. I think back over the years to some of the decisions that have been made on

> You get sounds on tape in a project studio that you wouldn't normally get in a larger studio unless you were high.

stuff I really loved, and it causes a lot of sleepless nights. But I know it makes a better album, and therefore a better band.

Well, I gotta go now. With this album just finished and a new record deal, I'm going to be busy for a long time.

Permanent vacation my ass. **EQ**

like to go in and just throw them down. I like to do them at my own pace, which is pretty quick anyway. I prefer to have six or seven tracks and keep singing the same song with different voices and in different ways.

I like after-hours vocals. It's what I did on *Get A Grip* and it's what I did on *Pump.* I go in there after the band leaves — so it's just me and an engineer. That's when I can have the most fun. No one's listening and there's no pressure. After hours is when I can get closest to a song and its real meaning. The emotions that come out of me then always are in sync with the song.

When we head out on the road, as we'll be doing shortly to support the new album, I fall into this little place I call my road voice. With it, I can't hit the high notes I can hit now, but I hit another kind — a raspy, bluesier high. I used to have a tiny switch on my mic that I could hit with my ring finger and it would switch me between straight signal and DDL, but now I don't do any of that. The thing I'm looking forward to on this tour is trying out some in-ear monitors.

SWEET AGGRESSION

We had a problem with *Pump* in that it was too bright, which is one of the problems when transferring analog material to digital. Digital hurts. Digital burns. Rock and roll has a certain warmth that gets lost in digital recording.

Sometimes in our analog mixes, we're coming off the board on tape and we're mixing my screams and Joe's lead guitar, which are both high register, and we've got them where we want them — really loud. You transfer that to digital and it bites your ear off. So you roll some of it off in mastering and it loses something.

In the old days we went through so much outboard gear, and some of it sounded real good. When I listen to something like *Toys in the Attic,* I can hear things that I never heard before. In that sense, digital is a wonderful medium, but I haven't been able to master it myself, nor do I know anybody who can go from analog to digital and make it sound really good — especially for us, because we like to record and master really hot. Case in point: *Get A Grip* is 4 dB up from *Pump.* This is our most aggressive sounding album to date. It's gonna make ears bleed.

CATCH A WAVE

Now that the album's out, I'm going to go surfing — or at least it feels that way. I've got my surfboard, which is the album, I've got a wave, which is the tour, and now I've got to ride this mother into the shore. If you're going to bother standing up on the board, you might as well take it as far as it'll go.

It's rough sometimes, but I still love what I do. There are times when my ears burn from exasperation and I get so frustrated, but it all pays off when I listen to the album at the end.

Somewhere along the line I get this dream that I can make something work and I go at it with a vengeance. And I've always been the sort who, if you tell me I can't, will. It's that rebellious nature that I've had since I was a kid. It's what got me into so much trouble. Now I'm here to do it again with the new album: to go where no band has gone before. **EQ**

World Radio History

Further Hums and Buzzes

Your studio system really isn't much more complicated than your stereo system. Audio problems can be solved easily provided you keep in mind that system complexity is due only to the sheer number of devices connected. So if you followed all of the "rules" and still have problems, don't get frustrated; be patient and read on.

HOW TO TROUBLESHOOT

The most common audio problems are hum and buzz (H/B) and we'll tell you how to track down and resolve some of these simple problems. To begin with, check the list of tools you'll need on the next page.

You might think we're getting ready to install a system based on this "grocery list." Well, if you have nagging problems, it may be necessary to disassemble your system. Before doing so, make sure that everything is labeled clearly and permanently. Wire replacement is critical, so you will want to make sure that everything is secured.

Reusable cable ties will hold wire harnesses in place until we're ready to permanently tie things down. To label wires or harnesses, use the wire markers — not bits of tape that dry up and fall off. The markers' bright yellow color is easy to spot. Buy all you can because they don't stay on the rack long. The flush cutters are used to cut the excess off the wire ties and labels. Cut as close as possible to avoid sharp edges that cause

the next worst thing to a paper cut. The fuses will save your speaker system from destructive blasts. The value chosen will provide an adequate safety net for most nearfield monitors. Wire nuts can be used to join the fuse holder to the speaker cable in case soldering isn't your bag.
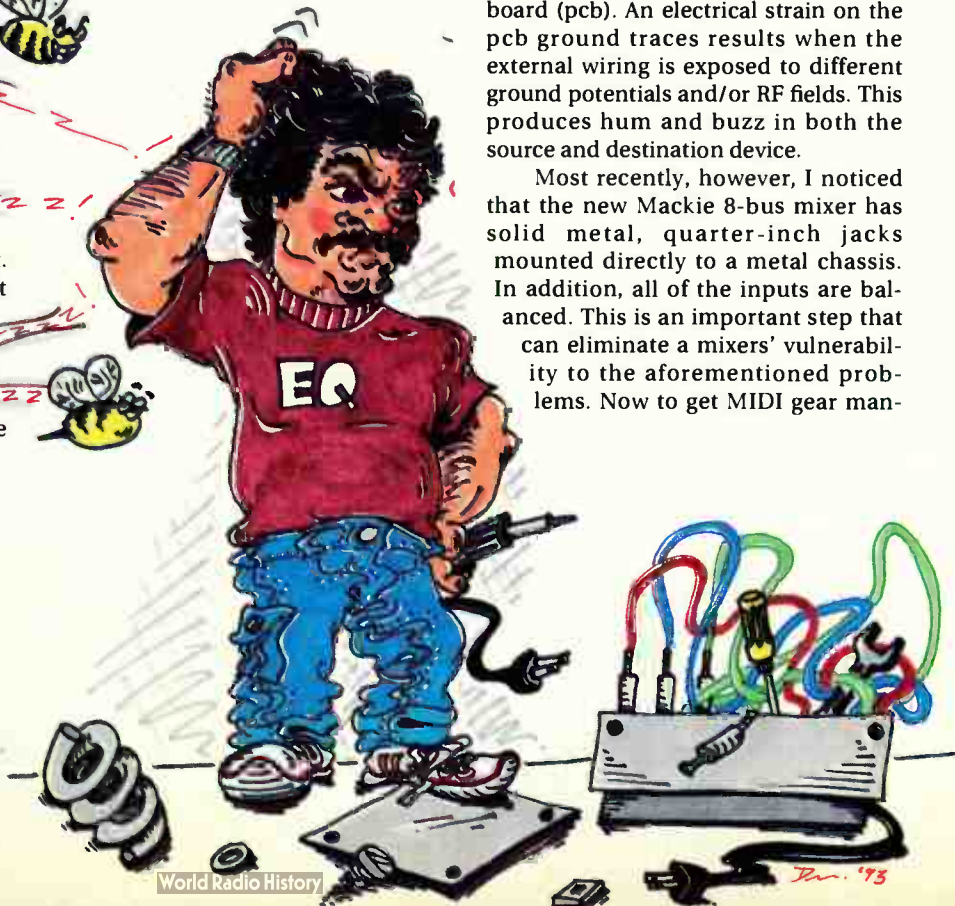
GROUND ADAPTORS

Ground adaptors are designed to create a three-pin, grounded AC outlet when only two points are provided. Used as intended, the wire or metal tab is connected to the screw that secures the cover plate. In audio applications these adaptors are more commonly used to interrupt (lift) the ground connection provided by the round prong on the AC power plug. For this application, I recommend that the wire or tab be removed so that it doesn't make an intermittent connection. Now it is a Ground-Lift Adaptor (GLA).

Usually the GLA acts like a Band-Aid, providing only temporary relief of ground-related problems. But hums and buzzes literally come and go with the weather. For the purpose of this article, GLA will be used to determine the "weak link" in the chain, but will hopefully not be the final solution.

Warning: Readers should understand that lifting the ground on individual pieces of equipment via a ground adaptor is against electrical safety code regulations. (Yes, even though everybody does it.) Doing so is meant as a temporary measure employed only for the specific purpose of tracking down hums and buzzes. Systematically connect each piece of equipment until the offending piece is found, then consult the manufacturer.

Note: Most equipment is designed to be built by cost effective, mass production methods. For example, just about every manufacturer of MIDI-oriented gear uses plastic-insulated, quarter-inch jacks for inputs and outputs to facilitate final assembly. These jacks are mounted directly to a printed circuit board (pcb). An electrical strain on the pcb ground traces results when the external wiring is exposed to different ground potentials and/or RF fields. This produces hum and buzz in both the source and destination device.

Most recently, however, I noticed that the new Mackie 8-bus mixer has solid metal, quarter-inch jacks mounted directly to a metal chassis. In addition, all of the inputs are balanced. This is an important step that can eliminate a mixers' vulnerability to the aforementioned problems. Now to get MIDI gear man-

There's more than one way to plug a noise problem

BY EDDIE CILETTI

Illustration by Donald Guth Jr.

MT-16Xpandable

MATRIX functions:
16-FOLD MERGING:

MIDI routing with no compromise for professional use including a single edit facility for all MIDI channels through MULTI CONVERTING

16-FOLD PROCESSING

All processing functions can be combined without limits at any time and are available for any input/output

ROUTING:

All of the sixteen MT-16X inputs can be merged without limits and they can be distributed to any outputs. The programs can be recalled either via MIDI, remote or the foot switch. Every input has its own seperate MIDI on/off function with an integrated ALL NOTES OFF generator.

PROCESSING:

SPLIT: 8 SPLIT zones per input.
TRANSPOSE: Up to +/- 64 halftones.
FILTER: Individual filtering of MIDI data for every input, output and MIDI channel.
VELOCITY: switch/limit/offset.
DATA CONVERTER: Controller Converting.
MANIFOLD: Multiplies the MIDI transmit channels.
SEND FUNCTIONS: PROGRAM CHANGE, VOLUME, MIDI CLOCK SEND DATA, TR-Transparent MIDI reset.

OPTIONAL EXPANSIONS:

On request, the MT-16X can be optionally upgrated to a combination of a MIDI Matrix/Processor and a Sequencer Player. It uses the universal MIDI File Standard for its internal format. This assures full compatibility to top software sequencers. Its built in 3,5" floppy drive (720 KB/1,44 MB) is PC compatible and allows loading of songs from this computer system.

PLAYBACK:

Hundreds of songs can be loaded into, if expanded to full 16 MB. Once loaded, any of them can be started instantly by the footswitch, MIDI program change or from the included remote control. Any of the 64 tracks can be assigned to any of the MT-16X outputs simultaneously.

RECORDING:

Recording of MIDI data is possible with high resolution simultaneously on all inputs. With UNIVERSAL DUMP any system exclusive data of different MIDI instruments can be safely stored and archived.

FURTHER OPTIONAL EXPANSIONS:

– FORNET, Fibre Optical Realtime Network:
  (MIDI, Network to expand up to 64x64 IN/OUTPUTS)
– SMPTE
– SCSI HARD DISC
– MEMORY CARD

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CORRIGAN MARKETING

World Radio History

ufacturers to follow Mackie's lead...

THE EASY WAY OUT

A simple list of "Do's and Don'ts" to refresh your memory:

1. Do keep your audio wiring away from power cables and transformers (wall blobs, etc).

2. Do keep power amps and supplies away from mixers and tape machines.

3. Do try to keep wiring neat and don't be afraid to take it apart and start again.

4. Don't use a GLA on a power strip. Use them on individual pieces, if at all.

5. Don't use a GLA on a device with a two-prong AC plug. (Unless reversing the plug makes a difference — which is scary — using a GLA is just plain silly!).

In almost all cases of tracking down the source of hum and buzz, you will be asked to do the following:

1. Try a GLA on a device that has a three-prong AC plug.

2. Remove the device from the rack.

3. Disconnect the device from the existing wiring and try new wiring that makes a direct path, carefully avoiding all wall blobs and wiring.

A picture is worth a thousand words; so, too, is one good example. Everyone is familiar with a guitar's sensitivity to picking up H/B. The pickup is designed to convert string vibrations into electricity through an electromagnetic field. Current is generated in the many turns of wire around a metal pole, much like a spool of thread on a bobbin. The unfortunate side of this is that it also picks up radiated power fields from other sources. These fields are generated by power transformers, which are also made up of coils of wire wrapped around a metal pole piece.

Balanced audio wiring is like a humbucking pickup. Both attempt to reject or cancel radiated fields.

ZOOMING IN

The mixer and the power amp have tremendous amounts of gain that we can use to "zoom in" on system noise. No matter how careful one is, the potential for speaker-damaging transients is great. If fuses have not been installed, do it now. Periodically reset the system gain to "Normal" to regain perspective.

SIMPLE, LABOR-INTENSIVE SOLUTIONS

The two common sources of hum and buzz are radiated hum fields and ground loops. Transformers (from power supplies and wall blobs) introduce hum into nearby audio wiring. This symptom may not be apparent until all channels are opened. Ground loops occur when two devices are not at the same electrical potential. Before the audio wiring is connected, this potential difference can be measured in AC volts. Once the audio wiring is connected between the two devices, current begins to flow, degrading the effectiveness of the audio shield.

The "generic" mixer referred to has an assignable stereo mix bus with a control room output. Here's how I usually start an "investigation":

1. All input faders down.
2. All effects sends and returns off.
3. Control room knob up full.
4. Stereo mix fader(s) up full.
5. Power amp up full.

If there is H/B that "follows" the stereo fader, deassign the channel faders from the stereo bus. These switches are usually labeled "L/R" or "Mix." If the hum gradually goes away, you need to determine whether the noise is being generated from within the mixer, or if the wiring is acting like an antenna. Let's pursue the latter...

Reassign the channels to the mix bus. Now it will be necessary to disconnect everything except the monitor feed to the power amp. (You did label everything, right?) Listen while disconnecting all inputs, inserts, direct outs, busses, tape inputs and outputs, FX sends, and returns. The H/B will gradually decrease if the wiring is picking up radiated hum or if there is a ground loop. Reorienting the wiring should help.

If there is still residual H/B, does it continue to follow the mix fader? If so, the cause could be the close proximity of a power amp, power supply, or video/computer monitor. Move them away from the mixer or turn them off.

What's left may be any of the following: a ground loop between the mixer and power amplifier, a power supply problem, or a grounding problem within the mixer.

For the next test, turn the control room level completely down and the amp gain to max. It may be necessary to put your ear right up to the speaker. Since the only audio cables remaining make the connection from the mixer to the power amp, try rearranging or replacing them. To check for ground loops, try a GLA on the power amp or mixer (if either has a three-prong plug). Also, if this applies to your situation, try removing the mixer, power supply, and/or power amp from the rack.

If that doesn't do it, and you're still reading, try turning off the mixer. The power supply will take up to a few seconds to shut off. If the hum dies immediately, it is likely a power supply problem. If the mixer has an external power supply, you need to have it inspected. Consult your dealer or authorized service station for a swap or repair. If the supply is internal, consult the manufacturer, a dealer, or service center in your area.

One final note. At least one mixer that I am fond of has an internal grounding problem that manifests itself as hum when all channels are assigned to the mix bus. The solution to the problem is to add individual ground wires to each module. This not only reduces the hum by about 15 dB (on a 32-input mixer), but all who implement the modification claim an improvement in the low-frequency area. **EQ**

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L MIC   R MIC
(ANY POLAR PATTERN)

SPACED PAIR

MUSICAL
ENSEMBLE

ANGLE

L MIC   R MIC

SPACING
(DIRECTIONAL MICS)

NEAR-COINCIDENT PAIR

MUSICAL
ENSEMBLE

L MIC   R MIC
(OMNI MICS)

BAFFLED PAIR

Mention stereo miking and most people in popular music start thinking string quartets or symphony orchestras. And yet stereo miking can enhance any music style — whether you're recording acoustic instruments or heavy electric guitars. True-stereo methods use two or three mics (rather than several close-up mics) to capture the overall sound of a sonic event.

For example, if you record a group of background singers with two

Get realistic ambience and natural sounds with stereo miking

BY BRUCE BARTLETT WITH JENNY BARTLETT

mics, during playback you'll hear sonic images of the singers in various spots between your monitor speakers. These image locations — left to right, front to back — match where the singers were during the recording session. The result is a greater sense of space and realism. A true-stereo recording is like a group photo; panned mono tracks are like a paste-up of individual photos.

WHY MIC IN STEREO?

Panning mono tracks is convenient but often sounds artificial. The size of each instrument is reduced to a point, and each instrument might sound isolated in its own acoustic space.

You can improve the realism by miking some of the instruments or singers in stereo. That is, pick up certain parts of the ensemble with a stereo pair of mics, then overdub several of these stereo pickups.

Stereo miking works well on these sound sources:

*drum kit (overhead)

*piano (out front and in line with the lid, or over the strings)

*soloist or singer/guitarist in a live room

*backing vocals

*horn and string sections, or an entire orchestra

*vibraphone, xylophone, percussion

*Leslie organ speaker

Stereo miking has many benefits:

*You sense there's a musical group playing together in a common ambient space.

*The miking, being more distant, captures the timbres more naturally.

*The size of each instrument is reproduced. Big sound sources, such as a piano, sound much more real in stereo.

*Front-to-back depth is reproduced.

*The recording includes time cues for localization. That's if you use noncoincident mic techniques (covered later).

*The recording includes the sound of natural room acoustics.

Stereo miking has some drawbacks as well. You must adjust the balance between performers by moving them toward and away from the mics during the session. This is more time-consuming and expensive than moving faders for individual tracks after the session. Also, the musicians in each stereo pickup are not acoustically isolated from each other. So if someone makes a mistake, you must rerecord the whole ensemble rather than just the musician who messed up.

CHOOSING A TECHNIQUE

The accompanying sidebars and fig. 1 describe several stereo mic techniques. The method you choose depends upon the stereo effects and features you want, and not so much upon the type of instrument you're recording. Consider the following:

Sharp imaging: coincident, near-coincident, or baffled pair.

A warm sense of ambience: spaced, baffled, or near-coincident.

Mono compatibility: coincident or SASS.

Convenience: stereo mic.

STEREO MIKING ACRONYMS

A-B: Spaced pair.

XY: Crossed coincident unidirectional mics.

MS: Mid-Side; a forward-aiming mic matrixed with a side-aiming bidirectional mic to get left and right signals.

SASS: Stereo Ambient Sampling System, two ear-spaced PZM mics with a foam baffle between them.

OSS: Optimal Stereo System, two ear-spaced omni mics with a hard padded baffle between them.

ORTF: Office de Radiodiffusion Television Francaise (French Broadcast Organization).

Lowest cost: two mics on a stereo bar.

Deep lows: spaced or baffled omnidirectional condenser mics.

Remote control of the stereo spread: MS or Soundfield stereo mic.

Inconspicuous: spaced pair, or spaced boundary mics on the floor.

RECORDING PROCEDURES

Follow these procedures for stereo overdubs:

1. Pan the left mic hard left, the right mic hard right, and record their signals on two tracks.

2. Adjust the acoustics around the instruments. If necessary, add some padding or hard reflective surfaces.

3. Place the musicians around the stereo mic pair where you want them to appear in the final mix. For example, you might overdub strings spread between center and far right and horns spread between center and far left. Place the mic pair far enough away to pick up all parts of the ensemble equally.

4. Experiment with different miking heights (to vary the tonal balance) and miking distance (to vary the amount of ambience).

5. If some instruments or vocalists are too quiet, move them closer to the stereo mic pair, and vice versa, until the balance is right.

6. If the monitored stereo spread is too narrow, angle or space the mics farther apart. Do the opposite if the separation is excessive.

Repeat these steps for other groups or instruments you want to record in stereo. When you're done, sit back and enjoy a more believable illusion than you're used to hearing. **EQ**

STEREO MIC ARRANGEMENTS

To record stereo, you can use either a single stereo mic, or two mics in a stereo array. Here are four ways to set up two mics (Figure 1):

*Coincident pair: two directional mics angled apart, with their diaphragms aligned vertically and grilles almost touching. Example — two cardioids angled 135 degrees apart.

*Spaced pair: two mics of any polar pattern, spaced a few feet apart horizontally and aiming straight ahead. Example — two omnis spaced 2 to 10 feet apart.

*Near-coincident pair: two directional mics angled apart and spaced a few inches horizontally. Example — two cardioids angled 110 degrees apart and spaced 7 inches (the ORTF system).

*Baffled pair: two omnidirectional mics, ear-spaced, with a hard or soft baffle between them. Example — OSS or SASS.
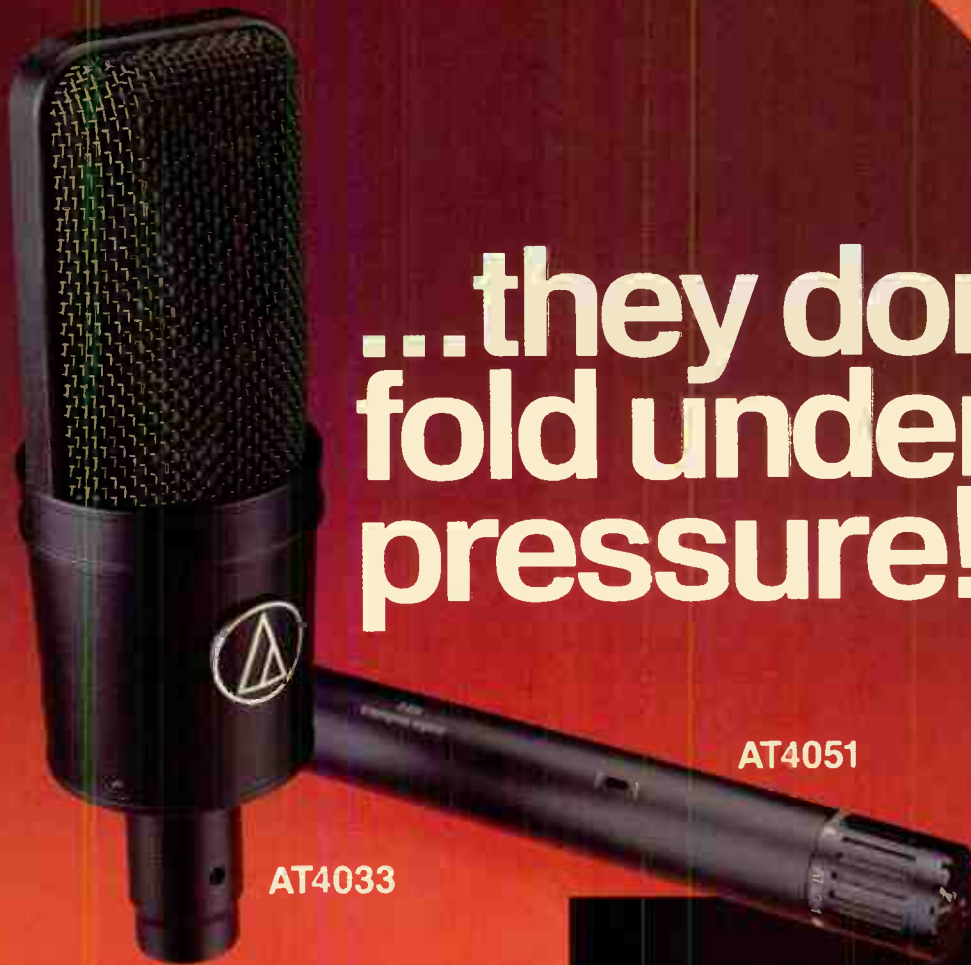
You can mount both mics to a single mic stand by using a stereo microphone adapter (stereo bar, stereo rail). The adapter allows the mic pair to be moved as a unit without disturbing their arrangement. Most adapters let you adjust the microphone spacing and angling.

A stereo microphone combines two mic capsules in a single housing for convenient stereo recording. Simply place the mic a few feet in front of the source, and you'll get a stereo recording with little fuss.

Most stereo mics are made with coincident mic capsules. Since there is no horizontal spacing between the capsules, there is no delay or phase shift between their signals. Thus the microphone is mono-compatible: The frequency response is the same in mono or stereo. That's because there are no phase cancellations if the two channels are combined. This is important if your recording will be heard on mono TV speakers.

Stereo microphones are available in many configurations, such as XY, MS, Blumlein, ORTF, OSS, Soundfield, and SASS (see second sidebar). MS stereo microphones and the Soundfield microphone let you remote-control the stereo spread, and vary the stereo spread after recording.

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CIRCLE 24 ON FREE INFO CARD

Studio Software Report

PROJECT: MULTIMEDIA

The electronic global village is here now.
Will your project studio find a niche there?

by Jeff Burger

Multimedia is taking the world by storm, and bringing with it a tidal wave of new opportunities for skilled talent. As today's desktop computers become increasingly adept at delivering digital audio and video, it's clear that the current winds of change merely represent an early warning: multimedia technology is developing into a full-fledged tsunami representing an entirely new market for musicians and studios alike.

What exactly is multimedia? The definition that works best across the board is that multimedia is the integration of graphics, text, video, sound, and microprocessor-based interactivity in various combinations.

If that definition seems a bit like defining printed media as sheets of paper upon which ink is applied in recognizable patterns, it is. But just as there are countless print variations—books, magazines, newspapers, fiction versus non-fiction—multimedia is best defined according to its myriad uses. The following look at the various markets for multimedia and the audio needs it requires is designed to help you chart a clear course to new outlets for your audio goals, skills and equipment.

Business Presentations

The largest established market for multimedia is the business presentation—typically a sort of slide show on silicon steroids. The primary uses of the business presentation are high-tech sales presentations at the conference room level, and supporting a speaker using a projection system to reach a larger audience. The proliferation of more powerful laptop computers is also arming more and more salespeople and managers with presentations that can literally be pulled out of a brief case for one-on-ones with prospects and clients.

This type of business presentation follows the practice of desktop publishing. It's not only done largely in-house, but it uses software like Aldus Persuasion or Microsoft PowerPoint. These applications let the user create (and/or convert from existing documents) a series of still images, and sequence them together so at delivery, simply clicking the mouse will step through the presentation.

The combination of do-it-yourself production with variable pacing has typically caused this type of presentation to be mute, the person delivering the presentation, of course. This is beginning to change however, due to several factors: keeping up with the competition, improved presentation software with audio

Editorial

By Tim Tully

"What's new," I asked my pal, Grumpmeier. He was on heavy deadline and had to have a bunch of music done in half the time it should really take.

"S.O.S.," he said, indicating this was not a novel situation, and continued staring at his computer monitor.

"Thank god for computers, eh?" I said brightly, looking to break the palpable tension in his project studio. "With all your software, you should be able to knock this stuff out in time for lunch."

He turned from his monitor just long enough to give me a look usually reserved for the terminally feeble-minded.

"The stuff's all too complicated," he finally growled, turning back to the screen. "I can't figure out whether to pull down the Transfrabulate menu, go into the Overtly Quantize dialog, set the Gridlock Masking function to one value less than my shortest note and tighten up every third beat in measures with velocities above 105," Grumpmeier explained.

"Or, I could just redo the take," he mulled absently.

"B-but doesn't all this gear make you more productive," I stammered, "and help you get more work done faster?" Confusion was turning me redundant.

"These guys and their 'Features Wars' are driving me nuts." Grumpmeier shot back impatiently. I resisted observing that Grumpmeier's condition made this seem less a drive than a short stroll.

"Look at these," he waved a pair of thick tomes at me. "The new manuals for the upgrade to my sequencer says there's seven different ways to start playing a sequence from bar 25: I key in a time, or click on a ruler, or pull a menu, or hit Command-Option-Tab-Click...." He was frantically turning from Reference to Tutorial to Quick Start. "I can't figure out if one's any better or worse, or if I'm missing something by not using another, or...or...."

He took a long breath. "I'm supposed to be dreaming up melodies here," he said, wistfully, "and I can't even get past loading my patch maps. I spend more time looking through the manual for some function at the bottom left of a dialog box on some sub-sub-menu than I do composing.

"And look," his voice rose again. "I got a IIfx here, and it takes this sequencer five seconds after I stop playing the music to reset the screen—and I haven't even changed anything!"

I thought a diagnostic approach might calm him. "Do you think generating all those colors, and notation, and flying windows and things slows it down?"

I couldn't tell if Grumpmeier thought my question unworthy of a response, or had just sunk deeper into his learning curve.

"How about using their entry-level program?" I offered helpfully.

"Can't sync to SMPTE," he murmured, almost inaudibly. "It lost a few features, but it's no easier to figure out."

I left Grumpmeier's studio with no more thoughts of lunch: lean and fast; powerful and professional.

Hmmm. ∎

World Radio History

and video support, and increasing skill levels to match. Despite this, the market for sound in these instant presentations is largely consumed by music libraries.

Where music is needed is the kind of business presentations that require more theatrical qualities: those in trade show booths and at large corporate events. To pull these off, a company will typically hire outside producers. While the less creative of these producers slide by on library music, the more ambitious use original music to realize the dynamics and synergy that will move an audience in the style of Hollywood. The count is about 50/50 in terms of live spokesperson vs. pre-recorded narration.

Interactive Kiosks

Interactive kiosks represent another multimedia market that's gaining momentum. These are essentially stand-alone boxes with which the public can interact to get information and even products. While the user is typically only aware of a monitor and simple user-interface—a touchscreen or a handful of buttons—the kiosk is hiding a fairly normal computer (sometimes in conjunction with a laserdisc player). These days, kiosks are automating everything from the information booths at the Smithsonian, to movie previews at your local video store, to the vending of state lottery tickets. While there's nothing that says a kiosk has to have sound, the heat is on to make these electronic drones as compelling an experience as possible—and that means audio, at least in the form of spoken narration.

Computer-based Training

For years the computer has been automating the corporate chores of orienting new employees and training existing workers to do new tasks. The combination of the visualization benefits of video with the computer's abilities both to present information selectively, and to test comprehension, offers many benefits—especially in larger organizations. Again, there is no mandate for sound in these productions, but narration (perhaps even multilingual) and music make the training process much more effective and enjoyable.

Education

Traditional education is embracing multimedia for the same reasons as is corporate training: a more compelling experience yields greater comprehension and retention of information. This need is even more pronounced when the goal is to educate a generation that's been weaned on arcade games and music videos.

The possibilities for multimedia in education are abundant and apparent. Sadly though, while education is the market in which multimedia could have the greatest benefit for the human race, it's simultaneously the one with the fewest dollars to spend on it. Several states, including Florida and Texas, have nonetheless mandated that certain portions of funds, traditionally allocated for textbooks, be spent instead on tools such as laserdiscs, computers, and educational programming to run on them. Ambitious teachers are even taking it upon themselves to create HyperCard-

style stacks that allow students to build their own audio-visual equivalent of things like book reports. Another upside is that this supports the idea that computers don't replace teachers, but serve as tools for more effective education.

Home Entertainment

The area in which multimedia has both the greatest market potential, and the least success to date, is home entertainment. The model in which multimedia will eventually succeed in consumer electronics has several distinct differences from other markets—and especially from business presentations.

First, while the business world embraces computers, the intimidation factor of a computer dictates that the home version of multimedia be delivered on the home TV/stereo by a familiar-looking interactive player. This player is based on CD-ROM technology and can also do double-duty: playing standard audio CDs and showing Kodak PhotoCDs (Kodak's new standard for viewing 35mm slides on the living room TV). The technology is also evolving in such a way that CDs will likely soon replace the VCR as a movie rental vehicle.

Second, where the presentation market is based on do-it-yourself, the home entertainment market revolves around the consumption of pre-programmed material. Indeed, part of the challenge is convincing Joe and Jane Public that they need more in the way of interactivity than their current VCR/TV remote provides. One impetus in that direction is the evolution of the huge Nintendo/Sega game market toward an audio-visual quality that can compete with Hollywood. Parents concerned with the ailing educational system and the "latchkey kids" syndrome will not only invest in better technology for their children, but will begin exploring more mature "edu-tainment" titles.

Clearly, interactivity in the home media mainstream will gain ground only insofar as producers begin delivering titles that utilize the medium's potential. Interactive fiction, for example, is a genre with which we have little experience and few collected works. Such productions not only bring new creative issues to the core script, but technical issues as well. Soundtrack music, for example, must now accommodate a non-linear story line that's guided by user choices. User interface design must also consider the way the audio domain works with visuals, and contributes to the overall intuitiveness of the presentation. Creating soundtracks for interactive home entertainment products is perhaps the most challenging of all the incarnations of multimedia, and it also has the potential for the greatest reward.

The Project Studio Meets Multimedia

Your entry into multimedia will most likely be a gradual one, rather than some instantaneous flick of a switch. Asking yourself the following questions can help you set your goals.

First, will you remain a specialist in the audio portion of multimedia or strive to become a well-rounded multimedia producer? While many producers come from a musical background, there's a lot more to it from the standpoints of creative, technical, and business chops (not to mention equipment costs) than hanging out a shingle. The alternative is to present yourself to producers as an audio specialist who can add value to a team effort. As multimedia productions become more ambitious, the teamwork model of the movie studio is becoming more and more prevalent.

Second, will you create specialized soundtracks for individual work-for-hire projects or produce generic library music for the use of others? Composing for specific projects can be both rewarding and a pain when it comes to pleasing clients on deadline. Clip media of all kinds has shown itself to be increasingly valuable in the do-it-yourself presentation market where people often have little creativity or production skill. Music libraries also have value to even the most experienced producers, as evidenced by the thousands of commercials and industrial videos that rely on the licensing of off-the-shelf music. This path, however, must be accompanied by an associated mass market plan (possibly via a distributor that specializes in clip media).

Third, will you invest the time and money embracing the numerous and often-complex computer-based delivery media, or will you just provide basic sequences and master tapes that others convert and integrate into the final production? The former is an easier path, but the latter will make your skills and services more marketable.

Finally, does a particular incarnation of multimedia tug at your heartstrings more than others? Soundtracks for business presentations fall into the same category as those for industrial video and commercials: to some it's a good living, to others it's hell. Similarly, there seem to be people who can deal with computer games and people who can't—and very few in between. On a larger scale, the challenges that interactivity brings to some forms of new media are embraced by some and frustrating to others.

The bottom line is that multimedia is only going to get stronger. In this decade the line between computer, television, and telephone will continue to blur until we have all-purpose information/entertainment appliances. As the global village becomes accessible in every living room and every office, the need to integrate high-quality audio content into the media stream will make today's audio market pale by comparison. The real question then becomes one of when—rather than if—you will embrace multimedia as a vehicle for your work. ■

Jeff Burger is a former founding editor of EQ, a contributing editor to NewMedia magazine, and owner of Creative Technologies—a multimedia production and consulting firm based in Northern California. His new book, The Desktop Multimedia Bible *(Addison-Wesley) is a definitive work on the subject.*

World Radio History

The Next Generation

The project studio is moving into your computer—expansion cards can record, process and mix your sound like never before.

(text rotated, right margin) ILLUSTRATIONS BY LINDA BLECK

Yamaha CBX-D5 Hard Disk Recorder (no price)

Yamaha Corporation of America announced it will ship the CBX-D5 Digital Recording Processor during the second quarter of 1993. The unit is designed to work in conjunction with a PC, Mac or Atari computer.

The CBX-D5 is a recording system with simultaneous two-track recording and four-track CD-quality playback. During digital audio recording, processing, and playback, the CBX-D5's onboard co-processor takes on much of the "load" from the host computer's CPU. The end result is that the CBX-D5 can be used with less costly computers such as the Apple Macintosh SE-30 and Atari TT, as well as any Windows 3.1-compatible computer. When used with a Mac or PC, the unit can also double as a MIDI interface.

The CBX-D5 stores recorded audio on a hard drive which is connected directly to the unit. Data is transferred between the hard drive, host computer, and CBX-D5 via a standard SCSI connection. As a standard SCSI device, the CBX-D5 can be connected to multiple hard drives (up to seven SCSI devices, including the host computer). The CBX-D5 simply requires a hard drive with an access time of at least 30 milliseconds.

The unit is a professional-quality, 16-bit system, supporting sampling rates of 48, 44.1, and 32 and 22.05 kHz. 18-bit D/A conversion is used with 8x oversampling digital filters. The CBX-D5 also provides on-board digital signal processing (DSP) and digital equalization (DEQ) with 82 different reverb and modulation effects, in addition to a separate four-band digital parametric equalizer for each audio channel. All these features can be controlled in real time.

The CBX-D5 was developed with software support from Mark of the Unicorn ("Digital Performer" for Macintosh computers) and Steinberg Jones ("Cubase Audio" for the Atari computer and for Windows 3.1-compatible computers) and hardware support from Dynatek Automation Systems (hard drives and storage devices for the CBX-D5).

Yamaha Corporation of America
714.522.9011

Digital Master EX (no price)

Digital F/X has announced Digital Master EX, a 16-track, 4-channel direct-to-disk audio recorder and editor. The system's features include non-destructive editing, variable speed playback and record, SMPTE synchronization with chase-lock, four independent digital and analog inputs and outputs, 4-channel event playlist editing, true graphic waveform editing, and DAT backup and restore, all working under a custom operating system. The Digital Master EX uses standard SCSI hard disk drives and is compatible with Atari Computer Corporation's ST, STe, TT and Falcon 030 computer systems.

The system works with a variety of SMPTE sync types, multiple digital I/O formats and sample rates, giving it the ability to be completely integrated into existing multitrack analog or digital audio systems. The complete Digital Master EX system consists of three 19-inch, single-space rack modules. Custom circuitry is employed in the electronically-balanced analog modules to provide the sonic quality of digital with the warmth of analog. The digital module has a 2-channel optical input and four discrete coaxial digital inputs and outputs.

Digital F/X
800.274.4339

Audiomedia LC (no price)

Digidesign has announced Audiomedia LC, a stereo, direct-to-disk, CD-quality recording and playback system for Macintosh LC-class computers. The system has extensive editing features and powerful digital processing functions, and includes Digidesign's industry-standard audio editing software, Sound Designer II.

The system consists of two components: the Audiomedia LC digital signal processing card, designed especially to fit the LC-PDS expansion slot, and the Sound Designer II software. The Audiomedia LC card contains a Motorola 56001 DSP chip running at 33 MHz for all audio processing functions. For audio input and output, the card includes stereo mini jack analog connectors. Audiomedia LC differs from the Audiomedia II card in that it does not support Digital I/0 or 48 kHz sample rates, due to limited real estate on the board. Features of the Sound Designer II software include non-destructive "playlist" editing, SMPTE trigger, real-time digital EQs, pitch shift with time correction, and more.

Digidesign
415.688.0600

SampleCell II ($1,995)

Digidesign has introduced SampleCell II, a stereo 16-bit, 32-voice, 32-MB sample playback card for the Macintosh II and Quadra families of computers. A basic SampleCell II system features dynamic digital filtering and eight polyphonic analog outputs. The package also includes a special version of Digidesign's Sound Designer II software for looping and other sample-editing functions; two CD-ROM sound library discs, including over 650 megabytes of ready-to-load instrument samples and sound effects; and a catalog of more than twenty other CD-ROM libraries available for SampleCell II from third party sound design companies.

The SampleCell Editor software, in addition to loading and configuring samples in the card, provides a direct link to Digidesign's Sound Designer II sample editing software, as well as Jupiter Systems Infinity sample looping software.

Each SampleCell II card can be fitted with up to 32 megabytes of standard, Mac II 4-megabyte RAM. (The $1995

price includes no on-board RAM.) The unit will work in conjunction with Digidesign's popular Sound Tools II, Audiomedia II and Pro Tools systems. The original version of SampleCell was designed to work with Apple's MIDI Manager, but since Opcode's OMS and most professional sequencers offer specific MIDI support for the unit, SampleCell II does not require MIDI Manager.

When Digidesign's TDM system is released later this year, SampleCell II owners will be able to add optional digital effects such as reverb (see Lexicon NuVerb, below), delay, EQ, internal mixing and integration with MIDI and digital audio tracks, and digital outputs in the form of software and hardware modules (including a forthcoming A/D - D/A converter/patchbay from Apogee Electronics).

Digidesign
415.688.0600

Lexicon NuVerb ($1,795, $495 TDM option)

Lexicon has announced a Macintosh NuBus card designed to integrate with Digidesign's Pro Tools or Sound Tools II systems via the upcoming TDM digital audio bus system. NuVerb can also be configured with AES/EBU digital I/O for connection to existing systems.

In addition to the cost savings of a processor that requires neither case nor power supply, the NuVerb takes advantage of the host Mac's screen for editing all effects parameters in the form of a complete effects-routing block diagram.

The unit features such effects algorithms as random hall, random ambiance, stereo adjust/delay, rich plate, split chamber, dual delay and compression. These algorithms are said to include variations on Lexicon's PCM-70. 224XL and 480L processors.

The NuVerb is a true stereo device containing two proprietary digital signal processors. The unit's most computation-

ally intensive effects use both DSPs at once, while other effects allow the two processors to be split in either a dual mono or cascaded configurations for additional flexibility. The NuVerb can also automate effects changes via SMPTE timecode. Setup, effects changes and parameter glides can all be automated in real time.

Lexicon, Inc.
617.736.0300

MicroEditor 2.1 (Contract $200, upgrade $50) (MicroSync $1,500)

Micro Technology's upgrade for its MircoSound Digital Audio Workstations for the PC can record within a mix, to allow overdubbing within a multitrack project. Automated effect sends allow individual segments to be selected, played through any outboard effects unit, and returned to the mix. The wet and dry sounds will both link to the segment, allowing the user to alternate between the two at any time. MicroEditor sound files can be converted to 8- or 16-bit multimedia .WAV files, and the 16-bit files can be used within mix projects. MicroEditor 16-bit sound files will retain the default to insure compatibility with previous MicroEditor files.

MicroSync ($1,500) is an optional synchronization board that allows MicroSound WorkStations to vari-speed resolve to external SMPTE timecode and Video blackburst. With MicroSync, MicroSound is capable of full slave operation, chasing during record and playback. It regains Sync-Lock within one second after high-speed shuttling, accurate within 1 ms over two hours at any sampling rate. Vari-speed capability allows chasing film in order to locate hit points accurately. All SMPTE frame rates are supported. MicroSync installs in any ISA/EISA-bus computer running MicroSound, includes a five-foot cable to the I/O module and interfaces to

SMPTE and video using standard RCA connectors. All existing MicroSound users may add MicroSync as an upgrade.

Micro Technology Unlimited
919.870.0344

FWB SCSI Jackhammer ($999)

FWB announced the shipment of SCSI JackHammer for the Macintosh. SCSI JackHammer is a RISC-based accelerator card that supports Fast and Wide SCSI-2 hard disk drives which can transfer data up to 20MB per second.

The SCSI JackHammer is priced at $799 when purchased with one of FWB's Hammer drives and at $999 when bought separately. It is compatible with all Macs with NuBus slots, including the new Quadras, and supports all brands of SCSI hard drives. The board ships with a special version of FWB's Hard Disk ToolKit SCSI utility software.

As a NuBus bus-master, the SCSI JackHammer is able to control activity on the SCSI bus in the background without intervention from the Mac's CPU, thereby off-loading the CPU's role in managing SCSI transactions and keeping it free to perform other processing tasks.

FWB, Inc.
415.474.8055

MIDI Express Multiport Interfaces (no prices)

Mark of the Unicorn, Inc. has introduced the MIDI Express and MIDI Express PC interfaces for Macintosh and IBM PC computers respectively. The MIDI Express and Express PC are true, multicable interfaces. Using up to 96 MIDI channels, users can connect keyboards, drum machines, synth modules and more—all simultaneously, without the need for an additional MIDI switcher.

The MIDI Express and Express PC are actually three devices integrated into a single economical package: a multicable MIDI interface, a MIDI merger/patchbay with extensive merging and filtering capa-

bilities, and a SMPTE tape synchronizer with adjustable freewheeling. The combination of these interrelated functions into a single device makes the MIDI Express both easier to use and more economical than separate units.

The 1U MIDI Express is based on Mark of the Unicorn's MIDI Time Piece II. It features a front panel panic button, click-to-MIDI converter and a footswitch input.

The Macintosh version of the MIDI Express is a four-in, six-out MIDI matrix with 64 incoming and 96 outgoing MIDI channels. The unit features a 16-scene memory with front panel Bank and Scene select buttons. The preset and user-definable memories allow use of the MIDI Express when the computer is disconnected. A Stripe SMPTE button on the front will start the unit's SMPTE generator, and will also work when the computer is turned off or disconnected.

The Mac MIDI Express has an internal power supply, and is compatible with virtually all MIDI software available for the Macintosh including Performer, Digital Performer, Unisyn and Mosiac.

The MIDI Express PC is a six-in, six-out MIDI matrix with 96 available MIDI channels. The unit includes an ISA card that fits any 8- or 16-bit slot, and is powered directly from the computer, eliminating the need for a separate power supply. A Windows Multimedia Extension driver allows the use of the MIDI Express PC with a wide variety of Windows-based MIDI software.

Mark of the Unicorn
617.576.2760

Coda Finale 3.0 ($749)

Coda Music Technology announced a major enhancement to Finale, its music notation program for the Mac. Version 3.0 is currently in beta, and will ship upon quality assurance approval. The

Version Updates

LATEST SOFTWARE AND FIRMWARE VERSIONS

Minimize time-consuming system crashes! Banish annoying bugs! Take advantage of all the R&D that developers put into your equipment!

You can't get the most out of your studio gear without the latest software. SSR's Version Update list tries to include all the application and operating system ROM software you might find in a studio, the most current versions of each one, and why and how to get ahold of it. Not every entry is as complete as we'd like, but we attempt to give you the most salient information available.

Manufacturers can have their products listed in the Version Update List by sending information to: Studio Software Report, PO Box 8607, Emeryville, CA 94662-8607. Preferred media (in descending order) for submissions to the list are: 1. Text files on Macintosh or PC 3.5" floppies, 2. Fax (510.450.0301), 3. Print on paper. You can also send E-mail on PAN to SSR. Include the name of the product, whether the software is a computer application or internal to a piece of hardware, the latest version, release date of that version, cost to owners of the previous version, system requirements, the new version's important fixes and features and the name and issue of any recent magazine reviews you want people to read.

PRODUCT	V.	R.D.	$	REQ.	FIXES/FEATURES	REVIEWS
ALCHEMY Passport Designs 415.726.0280	2.6	9/92	$99	Mac Plus^ big hard drive	Supports Pro Tools and RasterOps Media Time, new fft functions	KYBD 6/88 12/89
AUDIOMEDIA Digidesign 415.688.0600	x.x	11/92	n/c	Mac II	Program discontinued. Upgrade to Sound Designer II	
BEYOND 2.0 Dr. T's Music Software 617.455.1454	2.1			Mac II		KYBD 1/92
CADENZA FOR WINDOWS Big Noise Software 904.730.0754	1.1	5/92	$10 or n/c <60 days from purchase		Small fixes	EM 6/92; HSR 8/92
CADENZA FOR DOS Big Noise Software 904.730.0754	2.5g	4/92	$10 or n/c <60 days from purchase		Small fixes	EM 5/90
CAKEWALK PRO/WINDOWS Twelve Tone Systems 800.234.1171	1.03	1/93	n/c	PC/clone, Windows 3.0+, MIDI interface	Fix VGA display, edit controller fill Other	
CAKEWALK FOR DOS Twelve Tone Systems 800.234.1171	4.01	1/91				KYBD 2/90
CAKEWALK PRO FOR DOS Twelve Tone Systems 800.234.1171	4.0	1/91				
CAKEWALK ALIVE FOR DOS Twelve Tone Systems 800.234.1171	4.01	1/91				
CUBASE ST Steinberg-Jones 818.993.4091	3.01	1/93	$10	Atari TT & ST/ Mega (1 Mbyte/ Monochrome only)	64 track sequence	KYBD 10/89
CUBASE AUDIO/MAC Steinberg-Jones 818.993.4091	1.1	1/93	$200 new	Mac II SE30, 2M RAM Large Hard Disk	Score Printing bug fixes	
CUBASE MACINTOSH Steinberg-Jones	2.5	1/93	$60	Mac Plus, 2M RAM	384 ppqn; 8 mixer maps; MIDI mixer; 29.97 SMPTE sync	

V. = Version #. A version number preceded by a "<" indicates the cost to owners of that version or earlier; version numbers preceded by a ">" indicates the cost to owners of that version or later. >prev means "from the previous version." Upgrades from earlier versions may cost more. **R.D.** = Release date. **$** = Cost of update to registered owners. **REQ.** = Min. hardware and software required. **FIXES/FEATURES** = What's cool about this version. **REVIEWS** = Recent magazine reviews. Abbreviations: n/c= no charge; Kybd= Keyboard; EM= Electronic Musician; CMJ= Computer Music Journal; HSR=Home & Studio Recording.

PRODUCT	V.	R.D.	$	REQ.	FIXES/FEATURES	REVIEWS
CUBASE WINDOWS Steinberg-Jones	1.0.1	1/93	$60	PC/Clone; Windows 3.1	GS mapping	
CUBEAT Steinberg-Jones	2.0	4/92	$27.50	Atari ST/Mega	Scaled down Cubase sequencer	
CUE Opcode Systems	3.01	12/90	>3.0 n/c, <3.0 $50	Mac Plus		
DECK Digidesign 415.688.0600	1.04		n/c	Mac II, IIx, IIcx, IIci, IIsi or IIfx	Works with all Digidesign hardware	KYBD 11/90
DIGISYSTEM INIT Digidesign	2.3	12/92	n/c	Mac IIs and Quadras*		
DIGITAL PERFORMER Mark of the Unicorn 617.576.2760 Fax: 617.576.3609	1.2	1/93	n/c	Mac II, 5M RAM Digital Waveboard or Digidesign hard disk system	Complete new Manual; all performer 4.02 features: Humanize, Scale tempo; custom consoles; 4chnls on Digital Waveboard; 2 chnls of Audiomedia II	
DIGITAL WAVEBOARD Mark of the Unicorn 617.576.2760	1.0	4/92	$1495 new	Mac II, 5M RAM, hard disk, A/D D/A converter	Hard Disk recording	
DM-80 (ROM) Roland Corp US 213.685.5141	1.10	4/92		DM-80 8- and 4-track hard disk recorder	DM-80 operating system firmware	
ENCORE/MAC/WINDOWS Passport Designs 415.726.0280	2.6	1/93	$99 >prev	Mac Plus/PC Windows 3.1	Proprietary True Type Font, "Anastasia"	KYBD 6/90
EPS 16+ (ROM) Ensoniq 215.647.3930	1.3	7/92		EPS 16+	Intelligent backup & restore, SCSI improvements	
EZ VISION Opcode Systems 415.369.8131	1.01			OMS	Entry-level MIDI sequencer	
GALAXY Opcode Systems 415.369.8131	1.2.2	1/93	$15 or n/c	Mac Plus; OMS		
GALAXY PLUS EDITORS Opcode Systems	1.2.2	1/93	$99	Mac Plus; OMS	Set#2 Editors: K2000, E-mu Procussion, Alesis D4, Roland U-220	KYBD 4/92
JAZZ IMPROVISATION SOFTWARE MiBAC Music Software 507.645.5851	1.5.8	5/92		Mac, multitimbral MIDI Synth	Improved transmit routines, drum note sustain, improved printing, 12 styles, tempo and countoff control, humanize rhythm	MacWorld, 11/90, EM, 9/90, KYBD 10/90, Downbeat, 10/90
K2000 ROM Kurzweil 310.926.3200	2.0	7/92	$150	K2000	Editing features, MIDI	KYBD 3/92
KCS Dr. T's Music Software 617.455.1454	3.5	10/91		Amiga		KYBD 10/86, 2/87
KCS LEVEL II Dr. T's Music Software		3/89		Mac		
KCS OMEGA Dr. T's Music Software	4.0			Atari		
MACPROTEUS FRONT PANEL Digidesign	1.0f2	12/92		Mac II, IIx, IIcx, IIsi, IIci or IIfx		
MASTER TRACKS PRO 5 Passport Designs 415.726.0280	5.1	12/92	>prev: $99	Mac Plus	Auto mixer, SMPTE insert, enhanced Step Editor, transpose map, velocity editor	
MASTER TRACKS PRO FOR WINDOWS Passport Designs	4.6	1/92	>prev: $99	IBM AT, PS2/clone, MPC, Windows 3.0, DOS 3.1Microsoft	Issues commands to Start, Stop MCIplayer to play WAV. Audio in Windows	
MASTERLIST Digidesign	2.3	12/92				

PRODUCT	V.	R.D.	$	REQ.	FIXES/FEATURES	REVIEWS
MASTERSCORE II Steinberg-Jones 818.993.4091	2.0	1/93	$89	Atari ST/Mega	Score Printing, bugs	
MAX Opcode Systems 415.369.8131	2.5	1/93		MAC II	Flexible MIDI Programming	KYBD, 4/91
MIDISCOPE Kurzweil Music Systems 213.926.3200	1.5		n/c	Mac	MIDI data analysis	
MIDIMIXR 7s CNSLE MAC/PC Mark of the Unicorn 617.576.2760	1.1.1/ 1.02	1/93 1/93	n/c n/c	Mac PC/clone	Control MIDI Mixer 7s ditto	
MIDI TIME PIECE DA Mark of the Unicorn 617.576.2760	1.2	1/93	n/c	Mac	Set up MIDI Time Piece	
MIDI TIME PIECE ROM Mark of the Unicorn 617.576.2760	1.0d	1/93	n/c	Mac		
MIMIX Steinberg-Jones	1.06	4/92	$5,995 new	Atari ST/Mega	Mixing automation	
MOD FACTORY Creacent Engineering 201.746.9417	1.0	3/92	$250 new	Eventide H3000 Ultra-harmonizer	New H3000 functions: gain-ducked delay and reverb, envelope-controlled filter, audio-rate LFOs	
MOSAIC Mark of the Unicorn 617.576.2760 Fax: 617.576.3609	1.1	1/93	$195 for Composer owners	Mac II 2.5M RAM (Sys6) or 3M RAM (Sys 7) hard drive, ATM	Prints notation on ImageWriter,PostScript or QuickDraw printers	KYBD 2/93
MUSIC TIME MAC Passport Designs 415.726.0280	1.0	9/92		IBM AT, PS2/clone, MPC, Windows 3.0, DOS 3.1Mcrsft mouse	Auto mixer, SMPTE insert, enhanced Step Editor, transpose map, velocity editor	
MUSICATOR GS/ WINDOWS THINKWARE 415.255.2091	1.0	12/92	n/c	Mac II	various feature enhancements	
OFFICE MANAGER White Crow Inc. 800.424.0310	2.5	12/92	n/c	Mac II	various feature enhancements	
OBJECT MOVER Kurzweil Music Systems 213.926.3200	2.0			Mac	K1000/1200 series librarian	
OMS Opcode Systems	1.2	1/93	n/c	Mac IIs, Quadras	Fixes problems with Powerbook modem port	EM 3/92
PERFORMER Mark of the Unicorn 617.576.2760 Fax: 617.576.3609	4.1	2/93	n/c	Mac Plus 4M RAM hard drive	partial solo; solo button in all edit wdws popup menus for track switching	KYBD 8/90
PRO TOOLS SETUP Digidesign 415.688.0600	1.2	4/92	n/c	Mac IIs, Quadra		
PRODECK Digidesign	1.15	12/92	n/c	Mac IIs, Quadra, Pro Tools Card	16 channel support, post spotting, 1.0 bug fixes	
PROEDIT Digidesign	1.15	12/92	n/c	Mac IIs, Quadra, Pro Tools Card	16 channel support, post spotting, 1.0 bug fixes	
Q-SHEET A/V Digidesign	2.01			Mac Plus, SE1, SE/30, IIs		
QUICKTOPIX Optical Media International 408.376.3511	1.0	3/92	$8,500 (w/ hardware) new	Philips CDD-521 CD Recorder, Mac, PC or UNIX	CD premastering software produces CD of any format on desktop: Red Book, CD-ROM, CD-ROM XA, CD-I. Photo CD and CD-Write Once.	

V. = Version #. A version number preceded by a "<" indicates the cost to owners of that version or earlier; version numbers preceded by a ">" indicates the cost to owners of that version or later. >prev means "from the previous version." Upgrades from earlier versions may cost more. **R.D.** = Release date. **$** = Cost of update to registered owners. **REQ.** = Min. hardware and software required. **FIXES/FEATURES** = What's cool about this version. **REVIEWS** = Recent magazine reviews. Abbreviations: n/c= no charge; Kybd= Keyboard; EM= Electronic Musician; CMJ= Computer Music Journal; HSR=Home & Studio Recording.

PRODUCT	V.	R.D.	$	REQ.	FIXES/FEATURES	REVIEWS
S-770, S750 (ROM) RolandCorpUS 213.685.5141	2.13	2/92		Roland S-770 and S-750 samplers		KYBD 10.90
S1100 (ROM) Akai 817.336.5114	2.0	5/92	$650	S1100 sampler	Combines RAM & hard disk (incl magneto-optical) sampling, varispeed play, pan, fade, more.	
SAMPLECELL EDITOR Digidesign 415.688.0600	1.30	1/92	n/c	Mac IIs, Quadra, SampleCell Nubus Card	Direct drivers for Studio Vision, Digital Performer	
SOUND ACCESS Digidesign 415.688.0600	1.10		n/c			
SOUND DESIGNER II Digidesign	2.5	12/92	>2.xx: n/c <1.xx: $195	Mac IIs, Quadra	New time compression/expansion, pitch shifting	
SOUND DESIGNER IIPT Digidesign	2.5	12/92	$995 for Pro Tools	Mac IIs; Quadra	New time compression/expansion, pitch shifting	
SOUND DESIGNER IISC Digidesign	2.10	1/92	n/c	Mac IIs; Quadra	Stereo sample editing software for SampleCell	
SOUND EDIT PRO MacroMedia 415.442.0200	1.0	1/92	$349 new	Mac Plus, Sys. 6.0.7	Edit 16-bit audio, non-destructive editing, DSP effects	
SOUND STAGE Turtle Beach Systems 717.843.6916	2.0	1/93	n/c	PC w/ Windows 3.1	feature fixes, larger buffers	
STUDIO 3 DA Opcode Systems 415.369.8131	1.01	9/89	n/c	Studio 3 Interface		
STUDIO 5/OMS SETUP Opcode Systems	1.1.3	1/92	n/c	Studio 5 Interface	More OMS device names	
STUDIO VISION Opcode Systems 415.369.8131	1.43	9/91	>1.3 n/c; <1.3 $29	Mac SE or II with Digidesign card; OMS	New Control Bar with new control buttons, new manual, lock markers to bar, beat or SMPTE, editing during playback, reclock free-time sequences.	KYBD 1/91; EM 2/91
TIGER CUB Dr. T's Music Software 617.455.1454	1.1	10/91		Amiga		
TIGER Dr. T's Music Software	1.21	2/91		Atari		KYBD 2/90
TIMEBANDIT Steinberg-Jones 818.993.4091 Fax: 818.701.7452	1.0	4/92	$495. new	Mac II, LC, SE/30 and Classic I	Time Correction, Compression, Expansion, Pitch Shift and Harmonisation effects for Sound Designer II files	
TIMECODE READER (ROM) EnterTec Inc. 804.353.7133	1.1		$179	PC/Clone DOS, Windows 3.x	Captures SMPTE to Clipboard	
TRACKCHART Opcode Systems	1.03	4/92	n/c	Mac Plus	Sys 7 compat., 32 bit clean	
TURBOSYNTH Digidesign 415.688.0600	2.00			Mac Plus, SE, SE/30, II, IIx, IIcx or IIci	Sound creation and editing system for samplers	KYBD 10/88
VISION Opcode Systems 415.369.8131	1.43	9/91	>1.3 n/c; <1.3 $29	Mac Plus	New Control Bar with new control buttons, new manual, lock markers to bar, beat or SMPTE, editing during playback, reclock free-time sequences.	KYBD 7/89
WAVE FOR WINDOWS Turtle Beach Systems	1/93	2.0	$39	PC/clone Windows 3.1	DSP reverb, delay, pitch shift	
X-OR Dr. T's Music Software	2.1A			Mac, PC clone		KYBD 1/91
X-OR Dr. T's Music Software	1.12			Atari		

Macintosh MIDI Madness

By Peter Freeman

Mac Models Multiply—MIDI Mavens May be Mired.

Recent months have seen a sudden proliferation of new Macintosh models: LC IIs and IIIs, Performas, PowerBooks and more abound. Amidst the Apple marketing frenzy, clouds of confusion and question marks have loomed over Mac/MIDI musicians and studio owners shopping for a new computer. This comes not only from the wide array of features offered by (or in some cases, absent from) the new Macs, but also from questions about the MIDI-readiness of these machines.

To alleviate as much of this confusion as possible, we've compiled the following information regarding the MIDI and digital audio compatibility of the newest Macs. Although the information is as up-to-date as possible, certain new Mac models are so new that the major music software companies have not, at press time, completed final software/hardware compatibility testing on them. Nevertheless, we were able to gather final information on the majority of the new models.

LC II and LC III

■ Opcode and Passport report that the LCII and LCIII both appear to work just fine with their MIDI software. Opcode hasn't tested Studio Vision—its MIDI/digital audio sequencer—on these machines yet since this would require the special versions of Digidesign's Audiomedia II card for the LC series (see Next Generation). Digidesign reports that the card works OK on the LC and LCII, but cautions that these machines are a bit slow for the task of handling simultaneous MIDI and digital audio. Digidesign has not yet completed testing

the LCIII with their products.

Color Classic

■ The newest addition to the Classic family has been found to work quite well with MIDI. This makes it especially attractive as an entry-level or secondary music computer.

IIvi/IIvx/Performa 600

■ Serial port troubles plagued the early IIvx's, resulting in problems with time code and dense clumps of MIDI data. Apple has now partially fixed this problem, so the new vx's are OK for most MIDI work. However, to ensure complete MIDI compatibility with the IIvx and Performa 600, you need version 1.01 of Apple's software fix called "System Enabler 001," released on November 25, 1992. All the new Macs mentioned here ship with some version of the System Enabler because System 7.1 needs it to recognize the new Macs properly. But for those who may not have the latest version, Performa 600 owners can call 800 SOS-APPL, and IIvx owners have to go to an Apple dealer for their fix.

Centris 610/650

■ The Centris models have tested reliably with MIDI applications. However, since the Centris 610 doesn't support full-size NuBus cards, Digidesign cautions that its digital audio hardware won't work with that machine. The Centris 650 has no such limitation, and is therefore compatible with Digi products. Opcode (whose Studio 4 and Studio 5 MIDI interfaces can be configured to communicate with the Mac at speeds many times that of MIDI) notes

that both Centris models will communicate reliably with Opcode MIDI hardware at speeds up to 4x MIDI.

Quadra 800

■ After initial testing, the Quadra 800 appears to be compatible with MIDI and digital audio applications. As with the Centris models, up to 4x MIDI communication speeds are possible with the Studio 4 and 5. The Quadra 900 and 950 only worked reliably at up to 1.8x MIDI, so this is obviously an improvement.

PowerBooks

■ From the beginning, doing MIDI on a Macintosh PowerBook has not been a winning proposition. There seems to be a bottleneck at PowerBook serial ports, and the word is that doing large System Exclusive data dumps will definitely be problematic, but "light" sequencing may be OK. I've tried it, and it seems to work acceptably.

The original bottleneck still plagues the otherwise-sexy machines to some extent. Fortunately, Opcode's latest release of OMS offers a partial solution. OMS version 1.2 completely fixes MIDI problems, with the modem port only, on all PowerBooks except the 100. It doesn't help with the 100 because that model only has a printer port.

Two points are clear: first, the printer ports on all models of PowerBooks continue to be unreliable for MIDI work, and second, only MIDI software that specifically supports OMS can take advantage of the improved performance offered by OMS 1.2. It's no good simply installing OMS onto your PowerBook and then running non-OMS applications like Steinberg's Cubase or Mark of the Unicorn's Performer and expecting them to work smoothly.

Duo Dockers 210 & 230

■ Among the most exciting new models in the Mac family are the PowerBook Duo 210 and 230, whose "docking" capabilities make them viable both as portable and desktop computers. Happily, Digidesign has reported that the Duo Dock has tested favorably with their Pro Tools system and MIDI software, and that the modem port on the Mini Dock works fine for MIDI.

However, Opcode again recommends OMS 1.2 for completely reliable operation on the Duo modem ports. No final information was available about the MIDI-worthiness of the Duo's printer ports at press time, but it appears that they may both be OK for moderate densities of MIDI data. ■

NEXT GENERATION

retail price will remain at $749; an academic edition sell for $250. Upgrades will be available for $99. The enhancements in Finale 3 are centered on making the package easier to learn and use. Improved flexibility, more notation options and new functions were also added.

To ease the learning and use of Finale 3, the program's user interface adds a new document display window, reorganized menus and tools, added on-screen rulers, and simplified controls. Two forms of on-line help were added. First, extensive Balloon Help was added for System 7 users. Second, a Message Bar instructs users on the use of each Finale tool.

Finale's new Playback Controls (which replace the old Playback tool) give users straightforward, easy-to-operate tape deck-style controls to manage Finale's MIDI playback. The Instrument List makes it easier for users to assign MIDI instrument sounds.

Two other enhancements help users apply "articulations" such as staccato and accent marks. Apply Articulations makes it easy to attach articulation marks to many notes at once. Smart Articulations "snap" to the right spot on the notehead and move automatically when the note's stem direction changes.

Software Directions

Where have all the sample editors gone? Is there MIDI coherence coming on the Mac? Does even the Shadow know?

Whatever happened to sample-editing programs? **Digidesign**, the original Mac sample-editing gods, may drop sampler support from future versions of their programs, to concentrate on hard-disk audio. They never did embrace SMDI, the high-speed sample exchange protocol that would have given their software the speed needed for the '90s. Passport's **Alchemy**, optimized for samples and which *does* works with SMDI, never learned to play from disk. On the IBM side, things are not even as good.

Who's to blame depends on who's talking. Some think software developers don't want to deal with sampler manufacturers and write drivers for every sampler out there. Others think sampler makers write obscure, difficult-to-support protocols, making software writers' jobs miserable. Some think the market for sample editors just isn't very big, because users aren't willing to design their own samples anymore, and just grab things off CD-ROMs.

The answer is probably a bit of all of these, but whatever the reason, it's a shame to see this valuable genre of musical tool disappear. Without the tools to plumb its creative possibilities, the best sampler in the world is just a synthesizer with lots of memory.

One positive sign: Digidesign's **Turbosynth**, the amazing "electronic studio on a screen" software, is slated for a big upgrade this April. The new version will support all the Digidesign cards, read and write stereo AIFF, Sound Designer and Sound Designer II files, and add an "all pass filter" module. This is a "diffuser," used to develop reverb patches. It won't be a Lexicon,

but should be a handy tool. The bad news is that Turbosynth will still play from RAM only, and won't support any samplers except for Digidesign's **SampleCell I** and **II**.

■ Another hope on the horizon is Jupiter Systems' **Infinity**, a new DSP program for the Mac (compatible with all Digidesign cards) that uses algorithms with polysyllabic names to do nothing but create essentially seamless loops in any sample. The program, currently in beta, claims a "looping tool...for every type of sound," that preserve samples' "liveness." The different tools let you adjust loops during playback; create seamless loops in orchestra, chorus and other non-periodic waveforms; loop sounds with clear harmonic series, like winds, brass and bells; crossfade with an intelligent scanner that automatically finds a loop point and more. If it can do in the field what it did at the NAMM demo, Infinity could be a winner. **Jupiter Systems: 713.961.5215**.

■ Digidesign is also developing a set of **DSP plug-ins** for use with Sound Designer II, Pro Tools, Studio Vision and other third-party applications. The plug-ins give these applications functions like noise reduction and other effects. The Digis are also writing a new **sound output driver** for **QuickTime** that lets any Digidesign sound card record and play audio for any QuickTime application (Adobe's **Premiere**, Passport's **Producer** etc.). QuickTime movies currently can only play 8-bit Mac audio—not bad, but it ain't love. Digidesign is waiting for Apple to release its new Sound Manager and QuickTime 1.6, projected for this June.

■ Some startling developments in Mac compatibility took place at NAMM. Mark of the Unicorn announced **Free MIDI**, a Mac system addition designed to go head-to-head with the Opcode MIDI System (**OMS**), by providing multiport MIDI inputs and outputs for sound cards, and—like **MIDI Manager**—provide system-level links among simultaneously-running music programs.

The one important difference was that Opcode licenses and supports OMS for a fee, but MOTU was willing to give their software to any comers for no charge at all.

At the same time, **Apple** distributed a chatty memo to its developers, apologizing for dropping the ball on its own music system software (**MIDI Manager** and **Sound Manager**) and for the poor MIDI performance of some of its newer Macs (see: *Macintosh MIDI Madness*, this issue). Better times are ahead, the memo declared.

In light of this, Apple and several developers met to plot a common course of action. Nothing official has been announced yet, but it looks like a single standard for multiport MIDI and inter-application communication will emerge—supported by everybody. Now if we can just get everyone to agree on how many video frames there are in a second...

■ In the new uses for technology department, the **Voyager** Company has published a CD-ROM version of **"A Hard Day's Night."** It's got the whole film, in glorious black and white, as a Macintosh QuickTime file, and while it plays, you get to watch the script roll by, and even access various critical essays and trivia about the film. CD-ROM cognoscenti will wonder why they didn't take the next logical step: put MIDI files on the disk so the songs could play out as General MIDI sequences or let the user print sheet music in any key. Think about the possibilities. And then consider what we could do with a CD-ROM of *"This is Spinal Tap,"* or *"The*

Commitments," or "*The Rocky Horror Picture Show.*"

■ Multimedia authoring is turning out to be the neighborhood bully of Macintosh software. No matter how powerful your Mac, a QuickTime file or two will bring it to its knees. Even those '030-based Macs we thought were so powerful less than a year ago are getting bruised by these multimedia maulers. So, Apple's new mid-priced 68040-based computers are going to be welcome to those of us skulking around in the streets and alleys of multimedia development.

I got my hands on one of Apple's newest and baddest Macs—a **Centris 650**—about a week after they were released. The word, most assuredly, is: Fast. In fact, this 68040 machine, running at 25 MHz, beats the now-discontinued Quadra 700 on almost every benchmark. It's quite a novelty to run Passport's Encore, for instance, and see the on-screen manuscript effortlessly keep up with playback. Or to "feel" the additional power while you edit samples in **Alchemy**. Or hear and watch digital samples, animations and slide transitions unfold in Passport's **Producer** without dropping a single frame of the **Quick-Time** movie.

The 650 has three NuBus slots, and a Processor Direct slot. It can hold up to 132Mb of RAM and up to 1Mb of video RAM (for 16-bit color on 16-inch monitors). The smaller and slower **Centris 610**, based on a 68040/20MHz processor without the floating point unit, has only one expansion slot but may be a likely choice if you don't need the RAM or slot expansion prowess of the 650.

In Mac country, these systems are "cost-effective." In the real world, they "ain't cheap." Base-level configurations of the 610 start at under $2000 and starter 650's begin under $3000. Heavy users will want the upscale 650 with 8Mb RAM, a 230Mb hard drive, and Ethernet on the motherboard for $3559 retail. Mmm. Speed hurts.

As if Apple hasn't introduced enough buyer inertia into its product releases, making new Macs appear (and disappear) every few months, the February 22 issue of *MacWeek* spilled the beans on

two new Macs scheduled for release this summer with special features designed to attract multimedia mavens.

These new 68040 Macs will reportedly boast an onboard digital signal processor (DSP) designed to supercharge QuickTime capture and playback, and add speech recognition and speech synthesis to the Mac's arsenal. For those of you who've been keeping up with computer-based digital recording, it takes only a small leap of imagination to guess that some enterprising company might turn that DSP to the task of direct-to-hard-disk audio recording (assuming Apple hasn't hamstrung access to the chip).

For gadget freaks, it's also reported that Apple will release an inexpensive black and white video camera that attaches to the top of your monitor and automates the process of capturing QuickTime video in 256-shade grayscale. This should be a boon to industrial productions. Now, corporate executives who actually had to traipse around the office to humiliate employees will be able to get in everybody's face without ever leaving the leather-upholstered sanctity of their corner offices. Isn't technology wonderful?

Prices weren't available yet for these super-multimedia Macs.

■ **Digidesign** and **Alesis** agreed to develop an interface between the **ADAT** and Digidesign's **Session 8** and **Pro Tools** direct-to-disk products. The interface will be sold by Digidesign and allow direct, real-time, synchronized transfer of multi-channel audio between both recorders. It will give users the benefits of both tape- and disk-based recording, combining ADAT's familiar interface and robust, low-cost storage with Digidesign's extensive editing and processing. ADAT owners will be able to move their tracks directly to a Session 8 or Pro Tools system, edit and EQ them, and send them back to the ADAT. The new product will let Digidesign users store and back up multi-channel recordings onto inexpensive S-VHS tape in real time, and integrate into the ADAT Worldwide Network.

Release tentatively scheduled for later this year. No suggested retail price or product name has yet been set. ■

Shape Designer, for creating music graphics with user-definable playback behavior, is improved, and additional MIDI recording options are available.

Finale 3 will operate under System 6 or 7. System 7 support includes Balloon Help, TrueType versions of music fonts Petrucci and Seville, complete icon families, support for virtual memory and all basic Apple events, and 32-bit compatibility.

Current owners can purchase upgrades directly from Coda or from Coda dealers. MusicProse Owners who upgrade to Finale 3 get a $249 rebate, and a $50 rebate is available on purchases of Finale 2.6.3. Users who bought Finale 2.6.3 after January 1, 1993 will get a free upgrade to Finale 3 when it's available.

Coda Music Technology
800.843.2066

Notator Logic (no price)

Ensoniq is shipping the substantially re-written Notator Logic for the Mac. Designed by Emagic, Logic integrates sequencing and notation, and assists in controlling MIDI setups.

Logic offers unlimited tracks of any length, unlimited sequences and polyphony per track and 960 ppqn resolution. Logic can edit in real time, using reversible, non-destructive parameters, and lets the user see and edit sequences using the event list, matrix or hyper edit or notation.

The program allows editing in real time, using reversable, non-destructive parameters. Logic recognizes triplets automatically, has unlimited, styles, unrestricted text input anywhere on page, WYHI-WYG (What You Hear Is What You Get) performance, and much more.

Notator Logic will be available in April 1993, and will support the Atari 1040 series, the TT, and the Falcon 030.

Ensoniq Corp.
215.647.3930

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Cubase

Taming the Wild Narrator

Making strong words for a strong medium

BY CRAIG ANDERTON

There's more to multimedia than music — narration, for instance. Since a chain is only as good as its weakest link, dazzling graphics and CD-quality audio don't count for much if the narration is weak.

Fortunately, the new breed of hard-disk recording devices with built-in DSP functions are ideal for enhancing narration. Here are seven techniques that can turn narration tracks from "ho-hum" to "oh, wow!"

CONTROLLING P-POPPING

Close miking and bass boosting give an intimate, FM DJ ambience, but also create the ideal conditions for p-popping. Here are three ways to control it:

• Gain changes. Speakers say "P" sounds consistently, making them easy to identify among the other sounds. Drag the mouse across the audio region containing the P, and lower the gain by about 50 percent (more if necessary). This reduces the pop but leaves the timbral quality intact.

• EQ changes. Pops are primarily low-frequency energy. Defining the pop region and cutting all frequencies below about 40 Hz will remove boominess from the pop. Sometimes doing this to the entire track will eliminate all p-pops in one stroke without noticeably changing the narration's overall "warmth."

• Splicing. To remove the pop entirely — if it, for example, mars an otherwise sexy, breathy part — define the same part of the "P" sound as shown under "Gain changes," and cut it so that the rest of the file "closes up" around the cut.

If you're using Sound Designer, make sure Smoothing is on when you perform any of these operations.

PHRASE-BY-PHRASE NORMALIZATION

Compression or limiting can give a voice "punch," but the trade-off can be a pinched, flat quality that obscures the speaker's inflections and tends to sound boring over time.

A better option is to pick individual phrases (they're easy to see because there will be spaces between phrases) and normalize each one to full gain. This will maintain a high average level, but leave the dynamics intact. (Note: If the narration runs too long, normalize before you close up spaces between phrases to shorten the total time.) If one word sticks out way above the other words in a phrase, lower its gain to bring it into line before normalizing the entire phrase.

FROM SOPRANO TO TENOR

If you hoped for James Earl Jones, but Bart Simpson showed up instead, try pitch-shifting the track downward by about 10 to 20 cents. (Activate any time-correction option so that words don't become longer, just lower-pitched.) This can give a deeper, more authoritative timbre.

REMOVING SIBILANCE

This is a cousin of p-popping, where too much treble (or use of "exciter" boxes) causes "S" sounds to dominate the track. Like "P" sounds, these are easy to see: the waveform has a dense, spiky appearance. Define the region that includes the "S," and lower gain by about 50%.

THE CLINTON MEMORIAL INHALE-KILLER

When Bill said he "didn't inhale," he may have been talking about doing narration. Some breath inhales before

SPEAKING OF NARRATION: Ending P-popping with splicing (below) and reducing sibilance with gain change (right).

words are only natural (rumor has it that one of Barbra Streisand's engineers once removed the inhales on a vocal track, only to be told to put them back in because it made the track sound unnatural), but sometimes inhales can be distracting enough to overshadow the words that follow.

You can do the gain change trick

Parametric EQ

Filter Type:

Sample Rate: 44100 Hz

Cutoff Freq. 41 + Hz

"P" Sound

Ending P-popping with EQ (above).

mentioned earlier to tame "P" and "S" sounds, but I've found that defining the inhale region and applying a fade in works even better in most cases.

REMOVING NOISE

It's tempting to remove background noise between words or phrases since it's so easy to just cut it out, but this can sound choppy, like a noise gate. It's better to fade out to silence, and fade up from silence. An alternate approach is to cut your narration tracks completely dry and after denoising them, add a tiny bit of reverb. This will cover up the transitions between signals, noise, and silence.

INFLECTION AFTER THE FACT

Suppose you have the greatest narration track of all time, but the ad agency later decides to emphasize the product name by lengthening it the last time it appears. Using a time-scale function can do the job. Alchemy's time-scale algorithm is particularly good; Sound Designer has to time-scale an entire sound file, so the

Gain Change

Change -10.02 db, or -68.45 %

"P" Sound

Killing P-popping with gain changes.

best option is to copy just the word to be scaled to a separate file, time-scale it, and paste the time-scaled version back into the original file.

Working on narration is yet another way to impress a client, save a session, create a better product, and of course rack up another hour of billable studio time! Although nothing substitutes for getting a good narration track in the first place, for turning okay narration into great narration, there's nothing like hard-disk editing. **EQ**

MI INSIDER

and SX II stereo A/D converters. The latest DPM-3 upgrade creates the DPM-4, which supports 32-voice operation.

• Roland's DJ-70 Sampling Workstation ($3495) for live DJs and remixers features 16-bit sampling, autolooping and truncation for instant "groove" samples, keyboard, 8-track sequencer, 24-voice polyphony, up to 4 MB of RAM, and a disk drive.

YES, THERE'S MORE...

Some products didn't fit into any trends, but who cares? They're worth noting.

• Vestax and Marantz showed DCC cassette decks. Can tape — even digital tape — survive against the optically-based MiniDisc?

• Generalmusic Sound Engineer, a MIDI-controlled analog mixer (under $10,000), includes specialized software for the Atari (Windows and Mac versions are due later) with screens for panning, levels, input gain, graphical EQ control, aux sends control, etc. All of these parameters can be controlled via MIDI or SMPTE.

• Alesis announced the Monitor One nearfield studio reference monitor, with a 6.5" low-frequency driver and 1" soft-dome tweeter ($395/pair). A rubber coating provides a nonskid surface and reduces acoustic reflections.

• TC Electronics's TC8201 ($4995) is an AES/EBU interface analyzer and test generator with Windows 3.1 PC-

> Digital recording — tape and disk — seems poised to take over and expand the project studio market, possibly as rapidly as the CD displaced the LP.

compatible software; it tests for adherence to AES/EBU and EIAJ CP340 digital audio transmission standards.

• MIDI Solutions Inc. has a line of MIDI accessories — MIDI thru, HR-16 to D4 sysex converter, merger, MIDI-controlled relay, mapper, velocity curve converter, and several more — that range from $29 to $99.

• The PM-216 Automated Audio Patchbay Mixer from CM AUTOmation ($799.95) lets you patch audio devices into the 16 ins and outs, then change patch routings under MIDI commands.

• If you play to a click track but hate headphones, here are two options: the Q-Logic MIDI Metro, a 1U rack-mount unit whose lights change in time with the beat; and Visual Conductor, whose LEDs imitate the gesture of a conductor's baton.

• MIDIman's Video Syncman ($399.95) combines a MIDI sync box and screen burner. It allows the internally generated SMPTE to sync to house sync, incoming video, MIDI Time Code(!), or internal clock.

• Tech 21's Bass DI ($195) features tube emulation, as well as several controls to customize bass sounds in the studio.

• Many drummers consider Kat drum pads to be among the best. Their new dk10 has 10 pads and easy programming for under $500.

• Wireless MIDI is back: The Aquila MR1 MIDI receiver ($910) and MR2 Diversity model ($1330) provide wireless MIDI transmission over approximately 250 and 400 feet, respectively.   **EQ**

A10k Race That Everybody Wins

Acquiring a production mixer for under ten big ones used to be a challenge; any console that was considered professional came in on the wrong side of this outer limit. But times have changed. Integrated circuits have improved continuously, offering higher levels of performance at ever-lower prices. Manufacturing techniques have been rationalized through skilled producers like Peavey and Mackie entering the market, and venerable manufacturers like Soundcraft being acquired by market-savvy giants like JBL — all within the same time period.

The result is a veritable bonanza for anyone seeking a production console. More audio firepower is being provided at the lowest prices ever, with surprisingly little sacrifice when it comes to specifications, feel, and features. Luxuries like automation are now offered as a built-in feature included in the $10,000. It's a great time to be alive, at least from an audio bang-for-the-buck perspective.

Of course you're probably reading this magazine for more specifics, so it's down to the business of distinguishing between the various offerings. A few questions will immediately divvy up the numerous consoles into discernible categories.

The first question is: How much space do you have available? Are you after a compact configuration, or do you have enough room to put a full-size console into service? Most under-ten-grand consoles tend to fall into the diminutive category, but there are a couple — most notably Peavey's AMR Production 800 and, to a lesser extent, Tascam's M3500 and M3700 — that might be oversized for some personal production facilities. Big consoles are nice. They give you plenty of room to work, and it's awfully easy to sort out which control performs each function and grab for the right knob without looking.

A second consideration is automation. Until recently the only form of automation available in this price range was MIDI muting, which simply turns a channel, subgroup, or effect-send on or off using MIDI note on/off or program change commands. This simple-sounding feature offers surprisingly large benefits. It enables tracks that are not in use at a given moment to be cut off, which greatly reduces background noise. Any noise that might remain when the signal is present is masked by the higher-level material, so the ear doesn't hear it. The result is surprising; if you take the trouble to mute all 'dead air' on every track, you'll achieve near-digital results even when recording on a 'home' multitrack. MIDI muting is available on consoles of very modest cost, like Tascam's M2500 models.

If MIDI muting is enough for you, there are differences in the systems. The Allen & Heath GS3 is the only console of this group that internally stores real-time mute changes (the rest depend either on "snapshot" recall of presets or the use of an external sequencer or other MIDI source for real-time changes).

But automation has gone beyond mere mutes: It's now possible to acquire a console that stays within your budget yet boasts true fader automation. There are different levels to fader automation as well. The most advanced systems offer "flying" or moving faders that keep pace with every change that's been recorded. This is a feature you definitely *won't* find for under $10,000. But there are systems that keep track of all fader changes and that allow you to update

No, it's not a downsized marathon, but these under $10,000 consoles can put anybody out front in the studio technology race

BY J.D. SHARP

THE LOGICAL CHOICE: Soundtracs Solo Logic console

WORKSPACE: The M3700 console from Tascam

them by using two lights per input that show when you are above, below, or right at the stored level (the null point). To make changes, you simply bring the fader to the null point, press Update, and begin making changes.

If you decide to go the automation route, there is the further consideration of fader resolution. Automation takes the continuous action of a slide fader and converts it into a finite number of steps. MIDI automation systems, such as the Niche add-on or the Fostex DMC-100, are bound to the MIDI-imposed limit of 128 steps, or seven bits. If you assume that you're trying to achieve an overall dynamic range of about 100 dB, this would average out to less than one dB per step. In fact, your ears require that there be more steps at certain levels in order for those steps to remain smooth and undetectable. There's also the complaint that too few steps cause 'zippering' or 'staircasing' of the audio if the fader is moved too quickly. Although suppliers have figured out ways to minimize this effect and smooth it out, there's no question that more steps are highly desirable.

The next level up is 8-bit automation, offered by Soundcraft in its new Spirit Auto series. The units use MIDI and special software by either JLCooper (for the Mac) or Steinberg (for Atari) to "steal" the eighth bit from MIDI (this requires a dedicated MIDI out to the console). The number of steps for total fader travel increases to an acceptable 256, yet the ability to send and receive data over MIDI is

retained. Tascam's M3700 console also features 8-bit resolution when working to its internal memory, but only 7 bits over MIDI. This is rectified by adding JLCooper's optional Pro 3700 circuit board and optional software to the M3700, which offers features similar to those of the Spirit Auto running Cooper software on the Mac. These features include off-line mix editing, which many think is the whole point of getting automation in the first place. Without it you are limited to the updating of a single mix, using the console's built-in memory; if you aren't crazy about the way you mixed the second verse, but you like how the first verse turned out, you just rerecord the second-verse changes. With off-line mixing, you can copy the mix from the first verse and paste it onto the second verse. You can also combine sections from different passes, such as putting the first mix of the intro together with the third mix of the verse and the fourth mix of the chorus, and so on.

We wouldn't bring this up, except that it's relevant to the next step in fader resolution, namely the 12-bit, 4096-step detail offered by both Allen and Heath and Soundtracs in their GS3V and Solo Logic products, respectively. There's little question that both consoles provide absolutely professional response from their faders, but there is a problem: since MIDI can only handle 8 bits at best, it's useless for anything in the 12-bit world other than bulk dump and load for backing up mixes. Both these products as yet

provide no means to merge and cut/paste mixes together. They're working on it, so you need to decide whether to go for more fader resolution today in the hope that tomorrow you'll have the software you'd like, or to acquire a system with less resolution that has all the bells and whistles in the here and now. After all, the single-pass systems do allow you to update a mix endlessly, and you can punch in and out whenever desired, so it is a very workable way to end up with a perfect balance between tracks.

One final consideration: Tascam's M3700 console is the only one that provides an internal method of storage (a built-in 3.5-inch floppy drive). The others depend on MIDI Sys-Ex dump/load for archiving and for recall of a mix. If you're not computer-oriented, the M3700 makes the most sense.

The rest of the console selection process revolves around features. A number of recent designs combine in-line construction with eight subgroups. An in-line console offers one or two inputs plus an output for each input "strip." There are some serious advantages. If you take a look at both plain and automated versions of the Soundcraft Spirit, Allen & Heath GS3, Studiomaster P7, and Soundtracs Solo consoles you'll find a Tape Out for each input, so a 24-in console also has 24 outs.

All of the above in-line boards have another benefit: each input is dual, with both a main and monitor (tape) input per channel. Although only the main input can be assigned into the subgroups, both ins can make it into the Mix (L-R) Bus, effectively doubling the number of inputs for mixdown. Once again this can be done without patchbays, saving a bundle.

As you come down in price, some of this convenience is sacrificed, although all functions can be duplicated with a patchbay and a tad of ingenuity. This is where Greg Mackie's new 8-Bus consoles fall (the 16•8, 24•8 and 32•8). These feature 8 outs with double connectors, so it's easy to hook up a 16-track machine. There are also direct outs, although the convenient front panel button that directly assigns an input straight onto a tape track is omitted (that's where the patchbay comes in). Mackie has raised the stakes in the EQ arena by making one of the midbands fully parametric. The extensive list of features and quality

engineering touches make these consoles well worth looking into if you can live without immediate automation and MIDI muting, and are willing to put up with slightly less convenience.

A bit further down the scale some of the nifty features start to roll off. Tascam's M2500 mixers, available in 16- and 24-input configurations, are a good example. MIDI muting is provided along with 100 snapshot memories. There are 2 inputs per strip plus additional effect returns, so there are plenty of inputs for virtual tracks. EQ splitting is gone, but it is possible to reverse inputs so that you can at least apply EQ where it's most needed. Specifications, while still excellent, are a notch below consoles in the higher price categories. Some controls don't feel quite as smooth. But with a bit of skill, it's still possible to achieve digital-level signal-to-noise performance out of most mixers in this category, and it remains very simple to hook up and access as many as 24 tracks of recording.

Finally, there's the sound of each console. Each console has its own circuitry, with variations in noise, distortion, and phase performance. EQ circuits in particular offer different action and timbral results, and microphone preamps are the source of endless discussion among recording aficionados. These differences lend a certain character to the sound, no matter how advanced the engineering. Unfortunately, there are very few opportunities to directly compare consoles in a controlled environment. Fortunately, the level of circuit design in all of these products is quite good, so there's no way to get burned; it's more a matter of taste, opinion, and preference. Perhaps your best assurance of happiness is to purchase any console with the understanding that you can return it if it fails to live up to expectations. Chances are, though, that if you do the research, you'll be keeping it. **EQ**

For more information on these products, use the free lit. numbers below:	
Company & Product	**Lit. #**
Allen & Heath GS3 and GS3V	117
Mackie 8-Bus Series	118
Soundcraft Spirit Auto Series	119
Soundtracs Solo Logic	120
Studiomaster P7	121
Tascam M2500	122
Tascam M3700	123

Understanding DCC: Byte By Byte

Photo by Peter Monroe

Reducing data without reducing overall sound quality

BY LEN FELDMAN

It's been said that the decade of the '90s will be the decade of data compression. Actually, the term "data compression" is a bit of a misnomer. A more appropriate term would be data reduction, or data bitrate reduction. Well, whatever you want to call it, what's it all about? Let's take it from the top.

When you play a compact disc, the laser pickup and all the associated digital circuitry has to handle roughly 1.4 million bits per second. A CD can, in fact, hold about 680 MB of data!

When Philips began working on its Digital Compact Cassette (DCC) project, one of their objectives was to make the new DCC machines able to play back older, analog cassettes. That meant using a tape transport (and tape) that moved along at the snail's pace of only 1-7/8 inches per second. There was no way that you could cram 1.4 megabits of data onto 1-7/8 inches worth of cassette tape, so something had to give.

THE HEARING THRESHOLD

Humans can hear sounds over a very wide range of amplitudes, but there are some sounds that, while measurable and recordable, are below the so-called threshold of human hearing. In recording a CD, all sounds are converted to digital data — even sounds that are too soft to hear. In designing the DCC format, Philips engineers reasoned that if humans can't hear a sound, then there's no need to record it. That would reduce the required data rate considerably. Figure 1 illustrates this principle. The two vertical arrows represent sounds at different levels. The midfrequency sound is loud enough to be heard — it rises above the "threshold" of hearing. On the other hand, the low-frequency sound is below the threshold of hearing. (Generally, bass sounds and treble sounds need to be louder than midfrequency sounds in order for us to hear them.) That second, low-frequency sound need not be recorded since we can't hear it anyway.

THE MASKING EFFECT

There's another phenomenon of human hearing that enabled Philips to "throw away" even more data in designing the DCC system. That phenomenon is known as the masking effect. Here's how it works: If two sounds are produced at the same time with one softer than the other and close in frequency, you will hear the louder of the two sounds but not the softer one. Figure 2 helps to explain this effect. The louder sound, represented by the taller arrow, actually raises or modifies the threshold curve. A sound that would normally have been heard if it were the only sound around (represented by the shorter arrow in fig. 2) is now below the threshold of hearing because of the louder sound that's being produced at the same time.

So, if there were some way to continuously analyze sounds that you want to record, and if you could discard sounds that are both below the threshold of hearing and are being masked by other, louder sounds, you could reduce the amount of data that

had to be recorded. The overall system that performs this task for the new Digital Compact Cassette format is called Precision Adaptive Sub-band Coding, or PASC for short. By properly using these two psychoacoustic principles, Philips engineers found that they could "throw away" fully three-quarters of the data that would have had to be recorded in the case of a compact disc or an R-DAT recording.

Deciding what data to throw away is what makes the PASC system a bit complicated. PASC divides the audio frequency spectrum into 32 bands. The system then checks the signal levels in each of these narrow bands to determine which bands contain audible data and which don't. By rejecting signals that are not audible, PASC can assign its quantization "bits" to the remaining, required signals. Basically, PASC assigns or allocates digital "bits" to each signal in the order of signal strength above the audibility threshold.

DCC DIFFERENCES AND SIMILARITIES

DCC cassettes, whether blank (recordable) or commercially recorded, have the same length and width as the more familiar analog cassette tapes. The DCC cassette shell is narrower than an analog cassette, however, since the top of the shell is flat and is without the "step" found on analog cassette shells. A DCC cassette has hub holes that are accessible only from the underside, so unlike the case with analog cassettes, you can't flip a cassette over to "Side B" when "Side A" has been completely recorded or played. Furthermore, a sliding metal shutter covers the hub holes and tape, protecting the tape against dust, dirt, fingerprints, and tape slack. The sliding metal shutter operates pretty much like the slider on the 3-1/2 inch floppy disk used in computers.

TRACKS GALORE!

DCC tape is divided into upper and lower sectors, corresponding to the familiar Side A and Side B layouts of an analog tape. Even after discarding three-quarters of the digital data that would normally be recorded onto a

IT FIGURES: Figure 1 (left) and Figure 2 (right).

CD, the amount of digital data required for DCC recording is still pretty large, so no fewer than eight tracks of digital audio data are laid down, each only 185 microns wide. Don't think of these eight tracks as being separately addressable; the digital data is distributed among the eight tracks on a continuous and simultaneous basis. A ninth auxiliary track containing control information, as well as other nonaudio data, is also applied to each sector. During playback, the DCC tape head reads only a 70-micron portion of each track. That makes transport accuracy less critical, reducing the possibility of mistracking even when playing back DCC tapes made on another DCC recorder.

To handle the narrow-track pitch of 195 microns (185 microns for each actual track and 10 microns of spacing between adjacent tracks), DCC machines use precision record/play heads fabricated using thin-film technology. This technology is similar to the lithographic processes used in the manufacture of large-scale integrated circuits, or LSIs. Unlike the case with analog cassettes, no erase head is incorporated in DCC machines; in recording on a previously recorded tape, new digital data simply overwrites the old data.

In addition to the nine tracks associated with recording and playback of each side of a DCC tape, a wider left-channel and right-channel pair of tracks is provided at the lower portion of the thin-film head for playback of conventional analog cassette tapes. It is this configuration that makes DCC machines "backwards

Now that many recording engineers have heard DCC tapes and compared them with the same program material on CDs, that skepticism is fast disappearing.

compatible," or able to play analog cassette tapes. The DCC standard does not specifically exclude the possibility of some manufacturers offering a DCC recorder/player that will also be able to record analog cassettes. Currently, however, all the DCC decks that I've seen so far offer digital record and play capability, but only play capability for analog cassettes.

ERROR-CORRECTION CAPABILITY

Like all digital audio recording formats such as CD and DAT, the DCC standards call for powerful error-correction capability. Fully 50 percent of the data recorded on a DCC tape is devoted to redundant information that can be called upon when error correction is required. The error correction is so powerful, in fact, that an entire track of the eight tracks used in the DCC system can be missing without affecting playback accuracy.

There was a lot of skepticism among pro audio people when the principle of DCC was first announced, but now that many recording engineers have heard DCC tapes and compared them with the same program material on CDs, that skepticism is fast disappearing. As I said at the outset, data reduction is the way things are going in audio and, for that matter, in video as well. Whatever HDTV system we finally choose for the U.S., you can be sure that data reduction will be applied to the audio channels of that system.

Next time, we'll take a look at the other new digital format: the MiniDisc, developed by Sony Corporation. **EQ**

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ACOUSTIC MYTHS

continued from page 64

with improving sound transmission loss through the window. One is to alleviate flutter echo between the window and other acoustically hard surfaces on a parallel wall. Another is to reduce the multiple visual reflections that can occur between parallel glass surfaces. Good room geometry and finishes can fix the first problem, and good lighting can fix the second.

The acoustical characteristics of the glass, the mounting details, and the interior perimeter absorption have a much greater effect on the sound isolation of the window than does the angle of the glass.

MYTH 4: "TRANSPARENT" MATERIALS

The sound absorbing properties of standard building materials are often given as a Noise Reduction Coefficient (NRC) rating. Unfortunately, this standard measurement takes into account only speech frequencies and ignores the extremes of the frequency spectrum. More important, it measures the absorption of a material or assembly in a test chamber with random incidence of sound on a relatively small sample.

In practice, absorptive materials are often placed on walls where the sound is almost always at "grazing" incidence, or nearly parallel to the surface. When you drop a rock into the water it sinks, but when you throw it parallel to the water, it will sometimes skip along the surface. Sound behaves in much the same way: many materials that appear "transparent" based on NRC ratings or porosity are actually highly reflective to sound at grazing incidence.

One example is perforated metal, which is frequently incorporated into prefabricated modular acoustical enclosures to provide an "absorbent" interior surface. If a modular room is shaped to provide a reflection-free zone (RFZ) for a specific listening area, or if loudspeakers are mounted near the perforated metal surfaces, sound will strike the surface at grazing incidence and the absorptive properties will be rendered much less effective than intended.

Whenever acoustical test data is used to select a material or product, make sure that not only the numbers but also the test itself are appropriate for the specific application. If the test

Figure Four

conditions don't match the intended use, field performance may be quite different from what you'd expect.

MYTH 5: THE FIELD-FABRICATED DOOR

Doors are always the weak link in the sound isolation of an acoustically critical room. Anything that's operable cannot be built as solid and airtight as a fixed component, and real-life products don't seal completely or stay in perfect alignment.

To make matters worse, some manufacturers promote "acoustical doors" with ratings based on tests in which a nonoperable door panel is fixed into an opening. Seeing this, many people (including some studio designers) have made valiant but futile attempts to improve a door's sound isolation performance by making the door panel better. Years ago, it was not uncommon to see two solid core wood doors bolted together with a layer of "machine rubber" sandwiched between.

What is usually overlooked, however, is that the door itself is rarely the limiting factor. The acoustical leaks are almost always worse at the seals around the perimeter of the door. Even

Figure Five

the best field-applied door seals can quickly go out of adjustment and won't consistently maintain optimum contact and closure between the door and its frame.

Figure 3 shows the effect of such leaks on sound transmission loss. If we consider a 3'-0" x 7'-0" door with a gap around its perimeter of only 1/64-inch, the gap represents only one percent of the total area. This is enough, however, to effectively reduce an STC-36 door to an STC-29 rating. More importantly, if the door is beefed up to stop an additional 10 dB of sound, the composite transmission loss increases by only 1 dB. In other words, improving the door panel rarely affects the overall performance, since the perimeter seals can't keep up.

Sound-rated doors — in which the door, frame, and seals are manufactured as an integral unit — are the only reliable means of getting acoustical performance that is significantly better than a relatively simple door panel and field-applied seals. Alternatively, using multiple doors in a vestibule arrangement or keeping the door opening separated from the noise sources will help obtain appropriate sound isolation.

MYTH 6: MOSTLY RIGHT IS GOOD ENOUGH

Failures in studio construction happen more frequently from lack of attention to detail than from an error in the overall design. One typical example is that of building a drywall partition. It's carefully erected with isolated stud framing, filled with acoustical insulation, and finished with multiple layers of drywall carried from the floor slab all the way up to the metal deck above.

Later the electrician uses his claw hammer to run some conduit through the wall, and the plumber puts in a sprinkler pipe or two. You note that there are some gaps around these penetrations, and that the drywall doesn't fit into the corrugations at the deck, so you issue instructions that all gaps are to be stuffed with insulation. It seems as though that would be harmless enough, but you've probably just wasted half of the effort and materials that went into the wall.

The insulation provides sound absorption, but it isn't a barrier to sound transmission through and around the wall. Even though a 3/4-inch gap along the top of a 15-foot length of wall represents only one square foot of opening, stuffing it with insulation instead of sealing the gap

EQ MARCH **109**

World Radio History

can limit the wall's overall performance by more than 10 dB. Figure 4 shows two field tests of a drywall partition. The only difference between them is the manner in which the head of the wall was sealed to the deck above — initially the gap had been stuffed with insulation, but later a barrier designed to conform to the gap was installed and sealed airtight into place. This single modification improved the sound isolation from STC-31 to STC-44.

What's important in facility design and construction is balance. There's no point in putting a great door in an inferior wall, or vice versa. And the best, most expensive partition is only as good as its leakiest electrical box. As the sound isolation requirements of a room increase, the effect of an acoustical weak link becomes more and more devastating. Each of the components must meet the required performance, or all will fail collectively.

MYTH 7: YOU CAN'T HEAR HEAT

From the standpoint of audio fidelity, it is desirable to minimize the length of the cables that connect a loudspeaker to its amplifier. What better place, then, for the amplifier but directly beneath the speaker? Unfortunately, if you fall into this trap, saving a few feet of speaker wire may cost you dearly in acoustical problems.

Temperature gradients and air movement between a speaker and listener can drastically affect the sound field. In particular, they are likely to cause perceived shifts in the acoustical stereo image, much like heat rising from hot pavement can distort an optical image. Putting amplifiers directly beneath the monitor speakers allows them to to vent heat directly in front of the speakers and the thermal turbulence creates audible distortion. Similarly, the heat generated by some mixing consoles (coupled with poor ventilation design) ironically renders them unusable where accurate monitoring is required.

This same phenomena is often observed where air diffusers for the heating, ventilating, and air conditioning (HVAC) systems have been located incorrectly in a room. Figure 5 shows the amplitude difference in a time-delay spectrometry measurement (TDS) caused solely by cycling the air conditioning systems in a room where the supply diffusers dumped air directly between the speaker and the listener.

In any critical monitoring environment, even seemingly "nonacoustical" heat sources and air flow must be carefully controlled to maintain a sonically neutral sound field.

BEWARE THE ACOUSTICAL MYTH
Individually, the examples in this article may help you avoid specific pitfalls in studio design and construction. Collectively, they serve to illustrate the dangers of blindly believing what you are told or what you see at a world-famous studio. The "it's always done this way" approach may not be based on sound acoustical principles, let alone be the best means to achieve results.

Any time an acoustical myth can be identified and replaced with a little common sense or objective proof, acoustics as a science will become a little less mysterious, and we'll all have one less acoustical "truth" that only appears to be true.

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World Radio History

The Taxman Always Rings Twice

Dealing with the ides-of-April blues may be a little more difficult this year for the project studio owner

BY MARTIN POLON

The middle of April signals only one thing to most audio practitioners: "the taxman cometh." Recent revisions to the enforcement of the tax codes have been made by the Internal Revenue Service (IRS), based on a landmark January 1993 tax decision of the U.S. Supreme Court. This increases the possibility of scrutiny of the operations of a home business or home studio, while decreasing the options for deducting the expenses relating to the space used by the home enterprise.

The U.S. Internal Revenue Code, despite almost continuous tampering by Congress and other special interest groups through the years, still provides for complete recognition — for the purposes of deduction — of all legitimate business expenses incurred in running a recording studio, as long as the studio itself is a free-standing enterprise! So the operators of studios that are separate entities at a different location from the principal or any other residence of the studio's owner can deduct all costs related to the nonstudio physical plant. These costs include, but are not limited to, land, physical structures, heating, cooling, telephone, AC power, security systems

and fencing, cleaning and janitorial services, and building maintenance. That is, of course, so long as the studio shows some profitability over what the IRS considers a reasonable time frame.

The proverbial "fly in the ointment" appears, however, when the studio is in some way affixed to, or part of, the owner's home or lot. Then any expense that is "shared" in some way with the residence can be called into question. There have been a number of "tests" required under the Tax Reform Act of 1976 for home businesses such as studios, to validate the deduction for the space utilized. These tests require that the studio space be the taxpayer's principal place of business, or be used by both clients and customers in dealing with the taxpayer, or be a separate studio structure not attached to the dwelling unit. All of this takes on real importance when you consider that the personal and project studio revolution owes much of its growth to the tax advantages designed for the home office "electronic cottage."

The spin that the Supreme Court decision this year throws on the taxpayer is that now the old tests still apply, but the home studio space must be the place where the taxpayer's most important business is conducted and it must be the place where the owner/operator spends a substantial amount of his or her working hours.

The IRS did agree not to apply the new interpretation of the rules retroactively before tax year 1992. The new guidelines will be utilized to aid in making examinations (a term preferred by the IRS to audit) of those filings suspected of improper use of the home business deduction.

Three issues directly relate to the use of the so-called "home office" deduction for studios at this time. First, if any other sources of income or expenditures of time off premises exist for the studio's owner/operator, the home studio's significant financial and operational viability must be proven in writing. This will be the most difficult issue to overcome for many home "hobby" studio operators.

And that is exactly the category the IRS is targeting. Second, the loss of the home studio space deduction complicates the deductibility of home studio equipment acquisitions and other related operating expenses. The IRS could argue that these deductions are much harder to justify if the taxpayer cannot justify the priority of the studio space itself. Third, filing for the "home office" deduction creates a category of double jeopardy for the personal/project studios operating illegally in residential neighborhoods. Of course, these guidelines are general and each case will be evaluated by the IRS as to the specific circumstances. Contact a certified tax expert for further assistance.

The bottom line is that the justification of the home studio space as a tax deduction will require that the income from the studio be the most important element in the taxpayer's income flow. Unfortunately, most home facilities have evolved as personal musical creative environments. Unless the owner of the studio is independently wealthy — and few are — the other sources of income that support both the individual and his or her studio and music will take precedence and could cause the disallowance of the home studio space deduction by the IRS. Even more ironical is the fact that the very elements that legitimize the home enterprise deduction on the Federal level — written proof of outside income and client activity in the residential studio — place the facility and its owner/operator at risk for local and/or state zoning violations.

Clearly a case of "the government giveth and the government taketh away!

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Generalmusic S3 Music Processor

Today's pro keyboards represent exceptional value for the money, yet they also have a certain similarity that sometimes makes it difficult to get excited about new products. Most synths follow a "workstation" model that consists of a sample + synthesis sound engine, sequencer, and effects. The S3 is no exception, so what's to differentiate it from all the other keyboards competing for your disposable income?

As it turns out, the S3 is a very different kind of keyboard. All synths make tradeoffs, but the S3 is one of the few that trades off studio tweaking for live performance. Let's see exactly what that means.

BASICS

The first thing you notice is the packaging: 76-note keyboard, 8 sliders for real-time control, big backlit display, and a black matte finish that is virtually nonreflective under stage lighting. A waltz through the 350 ROM sounds (programs) is impressive, thanks to 208 samples in 6 MB of ROM — vibrant acoustic instruments, resonant analog sounds, good drums, lay-

ers, splits, and a responsive keyboard with polyphonic pressure and release velocity.

The user interface also includes over 90 buttons that are logically grouped and unambiguously labeled; overall, menu/cursor navigation is pretty straightforward.

The S3 is "multitasking" in the sense that you can load sequences while playing, but also in that the keyboard remembers edit screen settings. You can jump from, say, editing a sound to recording a sequence, and when you return to editing, the screen will be where you left it. The S3 won't remember anything unless you save it to the 1.64 MB floppy disk drive, however, since all memory (even the 2 MB of user sample RAM and 2 MB of program storage/sequence RAM) is volatile. The memory does not use standard SIMMs, and cannot be expanded.

OPERATING SYSTEM

The lowest level of the S3 food chain is the Waveform. Putting two of the same Waveforms together creates a Sound, which can be edited. A Perfor-

mance combines multiple Sounds; ten Performances make up a Bank (which can also store an associated Song), and there are a total of ten Banks. The sum of all this is a Setup.

The S3 takes a streamlined approach to sound and MIDI channel management by using a "track" metaphor. A Performance has 16 tracks, which can represent up to 16 layers of a sound, 16 channels of sounds acting like an expander module, 16 channels of master keyboard MIDI outs, 16 sequencer tracks for internal recording, or 16 sequencer tracks for external MIDI playback. Although this requires some compromises when layering sounds on playback from a sequencer, it is a simple and logical approach.

In keeping with the live performance theme, the S3 stores lots of sounds in an onboard "sound library" that shares 2 MB of memory with the sequencer. This memory can hold a maximum of about 250,000 sequenced events, or around 2,000 sounds (besides the 350 in ROM). It hardly seems likely you'll use more than 1,000 sounds, which still leaves room for about 125,000 events. (For comparison, a typical bass/drums/chords sequence for live use typically requires around 5,000-10,000 events.)

Concerned about how to find the proverbial sound needle in the library haystack? Don't be. The S3 has a "find" and "find again" function that searches on text strings, and lets you audition any patch. All keyboard manufacturers should implement the library/search concept—it's great. (Speaking of computer-like functions, there's also an "undo" button, and a clipboard that can hold 6 edits for subsequent pasting, as with envelopes, filter settings, etc.)

The 192 ppq sequencer does the usual: record, quantize, play back, and so on. It can record tempo changes and load SMF files from Atari or MS-DOS format disks, however, and has a "microscope mode" for

editing individual events. All these are welcome features. The biggest limitation for sequencing is the 16-voice polyphony, although I've found it adequate for most applications (besides, Generalmusic claims that an imminent software revision, available free to current owners, will boost the polyphony to 32 voices).

The S3 does a thorough implementation of General MIDI. The 350 ROM patches conform to the General MIDI spec, with bank-select messages allowing selection of alternate patches (e.g., several different types of pianos).

EDIT PARAMETERS
The S3's voice architecture consists of two detunable oscillators, amplitude envelope, pitch envelope, filter envelope, LFO, pan, and dual filters (each of which has five modes). The multi-segment envelopes and associated graphic displays, versatile filter structure, and excellent resonance characteristics are all strong points.

Unfortunately, the oscillators must be the same waveform. You can't mix, for example, piano and acoustic bass at the voice level; you have to create separate piano and acoustic bass Sounds, and balance them while in a Performance. If you want to edit one Sound while listening to the other in context, you can't—you must jump between Edit and Performance modes.

INS AND OUTS
There are left and right unbalanced 1/4" audio outs that pass through dual effects processors (one reverb, one DDL) and four other auxiliary outputs, which are presumably dry because auxiliary outputs generally drive outboard signal processors. (Incidentally, the overall output level seems fairly low compared with other instruments. Generalmusic said, however, that latest production models

have 6 dB more gain than the initial production run; current owners can get a no-charge board swap.) Two sets of MIDI in/out/thru connectors let you drive a total of 32 channels and merge incoming MIDI data (or process it on the way to the output). The optional Sample Translator disk loads into memory and lets you get samples in and out in several ways— via MIDI over SDS, or directly from Atari disks (Avalon or Sound Designer format) or SampleVision MS-DOS disks. Sample Translator also lets you transfer samples to other SDS-compatible instruments, edit start and end loop points, zoom in or out on the waveform (very handy for catching "dead time" at the beginning of a sample), adjust level, assign/deassign the sample to different keyboard ranges, and retune the sample by adjusting the sample rate. There's no SCSI port, so expect to take some time with big SDS sample transfers.

USING IT
Setting splits is easy (only two button presses, and one key press to indicate the split point); selecting or muting tracks simply involves clicking or double-clicking, respectively, on the associated track button. You can solo the highlighted track (even a muted one) at any time by pressing the Solo button. Simple.

Doing quick tweaks of multi-setups is a snap, thanks to the faders. You can edit volume, attack, release, filter 1, filter 2, and pan from the front panel in real time by pressing the associated parameter button and varying the desired slider. Each slider corresponds to one of the sounds in the multi; selecting the parameter to be tweaked, and moving the slider, changes the associated sound. This approach lets you do a task such as edit all volume or filter settings in a multi just by moving faders. (Additionally, another mode lets you use

the faders to adjust different parameters in a single sound.)

One problem: the pitch wheel response has a very wide dead band in the center, so you can't wiggle it to get vibrato, and the effect of moving the pitch bend wheel and not having the pitch change as soon as you expect it to is disconcerting. I'm sure a future software update could fix this if enough people complained.

THE BIG WRAP-UP
There's much more we could talk about — this is a machine with a lot happening under the hood — but we have only so much space. So it's time to put the S3 in perspective.

Of course, no synth can be everything for everybody. Despite the S3's gorgeous sound quality, hard-core tweakers may be wary of the emphasis on the single-waveform voice architecture. But as a live Performance keyboard, the S3 shines—from the comprehensive master keyboard features to the easy-to-read display and quick parameter selection/tweaking process. The lack of battery backup means that you'll have to reload everything from disk if someone kicks out the power cord, but that's life. Your consolation is that if the disk is in the drive, the S3 will load it automatically when power returns.

At $3495 ($2995 for the S2, which has a 61-note keyboard), the S3 is relatively expensive, and there are many other fine keyboards in the same price range. Yet the S3 brings something new to the party along with sound quality, and some of its features can't be found anywhere else.

Choosing a keyboard is a highly subjective matter, and only you know whether the sounds and interface suit your view of the world. But one thing's for sure: the S3 is a keyboard that merits serious consideration if you're in the market for a pro-level synthesizer.
—Craig Anderton

Peavey PC 1600 MIDI Control

MANUFACTURER: Peavey Electronics Corp., 711 A Street, Meridian, MS 39301. Tel: 604-483-5365

APPLICATION: For mixing MIDI sequences or live performances from a hardware control surface.

STRENGTHS: Ready-made setup presets let you easily perform a series of otherwise complex MIDI mixing and controller functions. Cool real-time Sys-Ex editing feature for a number of instruments, effects devices, and MIDI mixers.

WEAKNESS: Although hardware MIDI controllers don't have all the whiz-bang features that larger automated consoles have, the Peavey PC-1600 does what it's supposed to do and it does it rather well.

LIST PRICE: $349   **EQ FREE LIT. #:** 125

Since my personal roots in the music business stem from studio recording, I've always had a keen interest in music consoles and mixing systems. Because I'm also a devout techno-puppy, this interest has spilled over into the newly emerging field of MIDI-based mixers and controller systems.

For the most part, I've been rather underwhelmed with MIDI mixer designs. This is due, in part, to the fact that systems have only recently begun to offer a reasonable degree of dynamic control over level and other limited functions,

while most project studio consoles boast about offering only the simplest of features.

Fortunately, there's a simple and straightforward hardware device that can ease the process of mixing when MIDI instruments, effects devices, or MIDI-controlled digital audio levels are involved. This tool is known as a MIDI controller. In its most basic form, a MIDI controller is a physical control surface that's made up of a number of data sliders and function switches. These sliders can be assigned to control any number of MIDI controller parameters between the values of 0 and 127 (such as volume, pan, pitch bend,

after touch — you name it). Also, a set of associated function switches can be programmed to perform such MIDI functions as channel mute, solo, program change, note on/off, and so on.

Devices like these are often useful tools for electronic musicians in the project studio, allowing a recorded sequence to be mixed in real-time from a single control surface, or on stage, allowing MIDI settings to be mixed or recalled during a performance. Additionally, they can be a real boon to those who mix digital audio and MIDI via MIDI controller messages, as is sometimes the case with such digital audio workstations as Pro Tools.

Of course, these devices don't always have all the answers to your mixing prayers. Hardware MIDI controllers have their own quirks. To me, the biggest of these is that even though you're dealing with a hands-on hardware surface whose controls can be assigned to perform a wide range of functions, when you move beyond those basic functions you're often placed in the position of having to mentally juggle the intuitive world of physical controls and the not-so-intuitive world of keeping tabs on MIDI controller numbers and Sys-Ex codes.

Of the MIDI controllers that have come onto the market over the last few years, the one

that has most caught my attention is the PC 1600 from Peavey Electronics. One of the main reasons for this is that instead of offering only 8 data faders and switches (as have been available on past units), this box offers immediate control over each MIDI channel by giving us 16 long-throw (60 mm) data sliders and associated function switches.

CHECKING IT OUT

One of the 1600's simplest and coolest features is its ability to automatically transmit the proper control or switch function by your simply moving a slider or pressing a switch. Gone is the need to manually make a controller active in order to change a parameter. Just make your move (or any series of combined moves) and the proper commands will be sent — and fast!

Each of the 1600's data sliders and switches can be individually assigned to a MIDI controller or command from the control surface's right-hand side through the use of an alpha-style data wheel, four-direction cursor buttons, six edit command buttons, and a 40-character/2 line LCD display screen.

So far it sounds pretty simple, huh? Well in its basic operation it is. When you turn the unit on, the device will default to an initial "00" setup preset: "Volume with mute." When using this preset, each of the 16 data faders will correspondingly vary the volume of each MIDI channel, while each function switch acts as a channel mute button. This sets the selected channel's MIDI volume level to "0," thereby effectively muting it.

Preset "01" is similar to the first with the exception that the channel switches will act as solo buttons. In this mode, once a switch is pressed, all other MIDI channel volumes will be set to "0," which has the effect of muting all of the nonsoloed channels.

Preset "02," known as "Volume in groups," is programmed so that faders 1-12 will control their associated channel volumes. Fader 13, however, acts as a subgroup master for channels 1-4, while fader 14 is the master

for 5-8, fader 15 for 9-12, and 16 for 13-15. The switches are simply used to mute their associated fader and sub-group.

Preset "03" falls into the "extremely useful but could be better" category. As one might expect with a preset known as "8 Vols & 8 Pans," sliders 1-8 are used to control their respective volumes, while sliders 9-16 are used as pan controllers for channels 1-8. This is great, but there's no preset for controlling MIDI channels 9-16. I know that the 1600 is user-programmable, but it would have been a nice touch to throw that in at the factory.

Preset "04" is fairly novel. When it is selected, the PC 1600's switches will act as note-on keys for a 2-octave C-Major scale, while the accompanying sliders are used to control the volume of each note. This seemingly useless feature allows the device to be used as a control surface for triggering sounds from a sampler. (Got any other ideas?)

When we get to Presets "05," "06," and "07," the applications begin to change. Respectively, these presets will allow you to run through program changes, vary aftertouch, and pitch bend notes by simply moving the appropriate channel's data fader.

Once we get to Preset "08" and beyond, we begin to get to one of the PC 1600's most powerful features. I'm referring to its ability to act as a real-time, Sys-Ex editor for creating your own custom sound or effects patches. All you need to do is move the corresponding sliders or buttons.

Up to 50 presets can be stored into the 1600's internal memory. As shipped, 41 of these presets are templates that can be used to edit instrument or device patches in real time by moving the data sliders. A list of these instruments includes: Peavey's V3, V2, Autograph EQ, and DPM SX; Cubase sequencer remote control functions; Lexicon's LXP-1 and LXP-5; Korg's M1, M3R and 01W; JLCooper's MegaMix; Roland's MKS-80; E-mu's Proteus 1, 2 and 3; and Yamaha's SY 99, DX-7, and TG-55.

If these presets don't match up to

World Radio History

World Radio History

Bryston 7B PRO Power Amplifier

EQ LAB REPORT

MANUFACTURER: Bryston, Ltd., 57 Westmore Drive, Rexdale, ON, Canada M9V 3Y6. Tel: 416-746-1800; Fax: 416-746-0308.

APPLICATIONS: Power amplifier for critical listening applications.

SUMMARY: A high-power, single-channel amplifier capable of driving low impedances and multiple loudspeakers.

STRENGTHS: Robust construction, excellent sound quality, relatively easy to service, and comes with a comprehensive warranty

WEAKNESSES: Input level adjustment is not intended for regular use, requires 3 U of rack space.

PRICE: $2195   **EQ FREE LIT. #:** 126

Bryston has long been associated with high-end recording studios and audiophile home stereo systems, where its amplifiers have been in service for decades. Bryston's 20-year warranty even applies to secondhand units. I have had a good deal of experience with Bryston amplifiers in both recording and sound-reinforcement systems, and have had only a small percentage of breakdowns, all of which were dealt with promptly by the factory. The previous generation of Bryston 3B and 4B amplifiers gave me reliable service even in demanding sound-reinforcement applications. The new versions of these amps, with the added title of NPB for the professional version and NRB for the regular (Hi-Fi) version, seem to be easier to service and have fewer wiring harnesses to contend with. This should allow these amps to enjoy even longer service than their predecessors.

I was sent the Bryston 4B NPB for a quick listen and the Bryston 7B NPB for a full evaluation. These amplifiers are similar in design and layout; the 4B is a 2-channel unit providing 250 watts into 8 ohms and the 7B a single-channel unit supplying over 500 watts into 8 ohms. The 7B is specifically designed to drive very difficult loudspeaker loads that can range down to impedances of less than 1 ohm. Both amplifiers are three rack-units high, just over 15 inches deep, and weigh about 40 pounds.

The NPB version of these amps uses a Neutrik 3-conductor, 1/4-inch phone jack in parallel with an XLR-type connector for the actively balanced input. A tiny front-mounted trim control recessed into the front panel of the amp adjusts the input level. This control is a little difficult to set, as it is accessible only by a small-blade screwdriver. I would have preferred a knob mounted on the rear of the amp, away from untrained fingers but easier to adjust. In most applications, however, this is only adjusted once, when the amp is installed, to set the final system gain structure.

The front panel of the amp also has a power on/off button and a single LED that displays the presence of power (green), peak output level (yellow), and fault conditions (red) in the input signal, such as RF or DC, or output signal, such as clipping or shorted loudspeaker cables. This provides a simple, yet comprehensive system for indicating the amplifier's status without resorting to level meters or an array of status LEDs.

The output configuration of the 7B is well thought out, providing a choice between maximum voltage swing or output current. Although a single-channel unit, the 7B has two pairs of five-way binding posts and a switch to select between two modes: a series mode for loads above four ohms or parallel mode for loads down to a

one-ohm impedance. The 7B was designed to drive difficult loudspeaker loads that draw a great deal of current, such as the popular UREI studio monitors and the multiple drivers common in many subwoofers. Both output modes will provide the same gain as other Bryston amplifiers, which, in an emergency, allows users to swap out any Bryston amp with any other Bryston amp and achieve the same gain, albeit with differing maximum power capabilities.

The balanced inputs of the 7B NPB can have the signal polarity reversed or ground-lifted with the flick of a rear-panel switch. This is very convenient when tracing noise faults or interdriver polarity problems in a loudspeaker system. The 7B I had for review goes against the AES polarity convention for the XLR-type connector, although the polarity switch can immediately compensate for this or for other polarity inversions — such as those between different loudspeaker manufacturers. Bryston is changing its input polarity convention and will now be producing its amps with XLR pin 2 positive relative to pin 3.

The output from the 7B is impressive. Each unit is shipped with a data sheet of its individual test results following final burn-in at the factory. Distortion figures are well below 0.01 percent (typically below 0.004 percent) at all audio frequencies, and output power is 570 watts into 8 ohms or 800 watts into 4 ohms. The noise floor is over 105 dB below the rated output. Basically, the amplifier does not get in the way of the audio signal and merely provides approximately 30 dB of gain to drive even difficult loudspeaker loads. There is, however, protection against DC and RF signals reaching the loudspeakers, which, if it happened, could easily result in a catastrophe. Through the audio band (20 Hz to 20 kHz), I measured less than 1/4 dB of variation in the frequency response. One 7B I was sent did have a problem

operating in parallel mode, but all I had to do was ship it back; there is never any charge for repairing a faulty Bryston, even if you're not a reviewer!

These new Bryston amps produce less heat when idling than their predecessors, mostly owing to the use of toroidal power transformers. The inclusion of a new circuit that limits the in-rush current when the units are switched on prevents tripped circuit breakers when the amp rack is powered up and eliminates big noises if the whole sound system is accidentally powered up at once. Bryston has continued to produce convection-cooled amplifiers and, with the change to toroidal power transformers, these amplifiers do not produce any acoustic noise in any mode of operation, even at full power!

Improvements have been made over previous models for servicing as well. The new design does away with the DC fuses in the older Bryston amps and reserves the rear panel for I/O connectors, switches, and the AC line fuse. The AC power cord is a 14-gauge cable terminating in a detachable IEC connector. This provides sufficient current capacity for this amp's dynamic requirement and also means that permanent installations can disconnect all the cabling when removing an amplifier for service. The output modules can be replaced by removing nine Robertson-type chassis screws. Bryston will exchange or repair these modules, saving the cost of shipping the whole amplifier for the sake of one faulty output module. In the case of the 7B, however, both modules are necessary in order for Bryston to match the gain.

In situations where high-efficiency loudspeakers are used, amplifiers spend most of their time producing milliwatts — not megawatts — and it is often this dynamic performance that sets the brute-force amplifier apart from a critical monitoring

amplifier. A clarity is apparent in these Bryston amps at low volume, a clarity, that can be masked by the more-raw-output-power designs, perhaps due to Bryston's combination of Class A and A/B operating modes. Music can be full of low-level subtleties, and if those details are important to you, then you should consider more than just the maximum output level when comparing amplifiers.
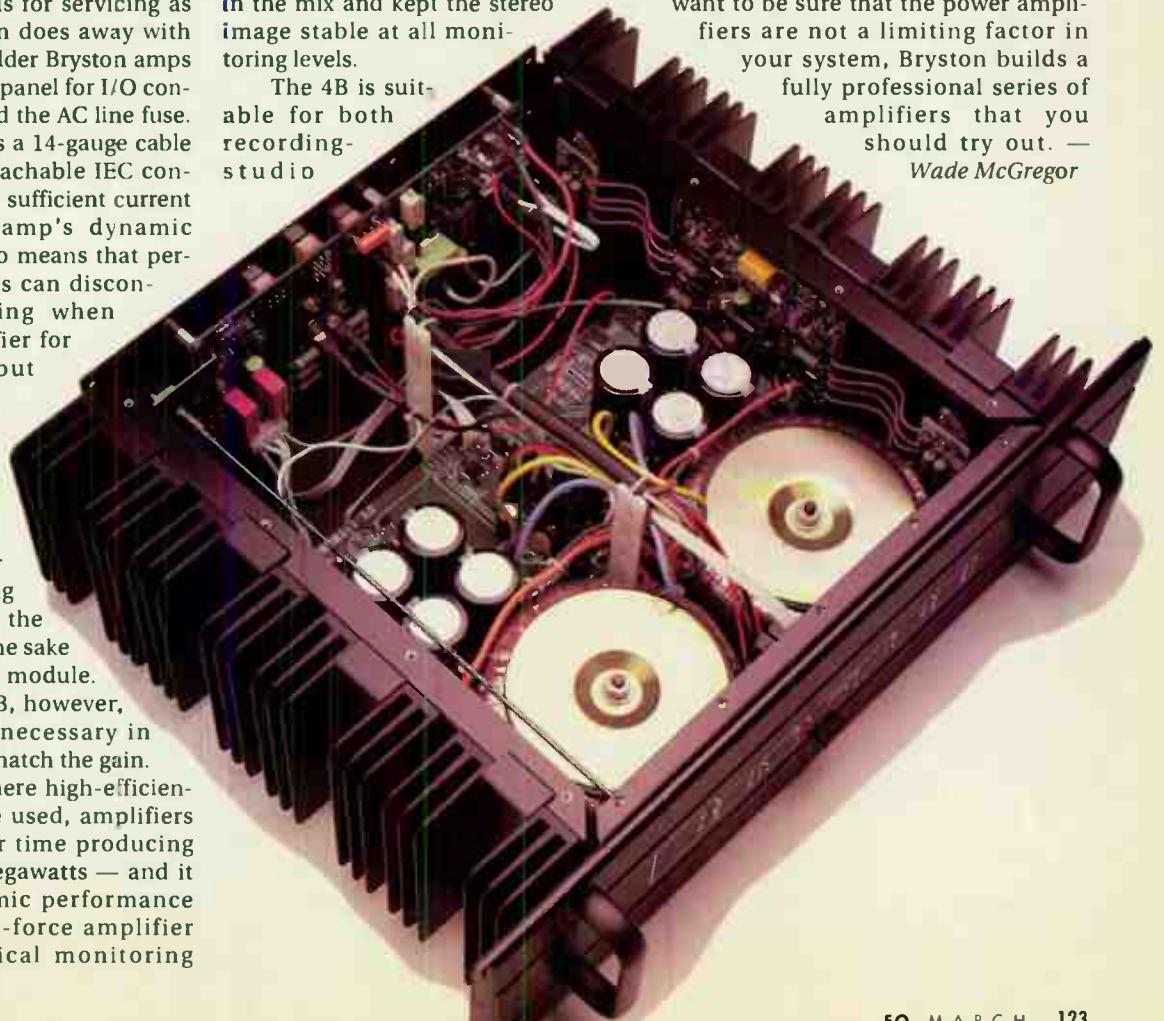
The 7B is able to drive difficult loads without sounding strained. This quality was even noticeable on mediocre power-hungry stage monitors. I used the 7B in one application where very small monitor wedges were being driven in parallel; the loudspeakers had never sounded so good and there was even enough gain to please the guitarist. The same amps, brought into the studio, unmasked details in the mix and kept the stereo image stable at all monitoring levels.

The 4B is suitable for both recording-studio

monitoring and sound-reinforcement systems where the loudspeakers have impedances greater than 4 ohms. Where loudspeakers provide difficult loads or are run in parallel, the 7B will provide the current these loads require. In a typical three-way active-crossover loudspeaker system, a Bryston 3B (120 watts per channel into 8 ohms) would be used to drive the high-frequency loudspeakers, a 4B to drive the low-frequency loudspeakers, and a 7B to drive the subwoofers.

Bryston continues to refine its product, making improvements with each new generation. And, although it has not entered into the race for the world's smallest, lightest, and most powerful amp, they are producing an excellent product for a large number of amplifier applications. If you have optimized your loudspeakers and want to be sure that the power amplifiers are not a limiting factor in your system, Bryston builds a fully professional series of amplifiers that you should try out. — *Wade McGregor*

REVIEW SHORT

Yamaha TG100 Tone Generator

"One of the coolest aspects of this box is the fact that it can be directly connected to a computer via a built-in MIDI interface."

Since its inception, I've never been a big fan of the newly developed General MIDI specification which is used to standardize sound patches on Multimedia sound modules and cards. The whole concept behind the General MIDI convention is that whenever a music sequence is played it will sound the same or reasonably similar when any sound generator that conforms to this sound patch specification is used. For example, a grand piano patch (program change #1) will always be heard as a grand piano and won't surprise you by being a Martian-ray-gun effect from hell.

The reason that I've never been much of a fan is that most General MIDI sound generators have sounded too cheap and twinky for my taste — that is, until I met up with the TG100 tone generator from Yamaha ($449). The sounds that come from this small, two-pound box are fairly impressive.

On the tech-side, the TG100 is a 16-voice (28-simultaneous note) sample playback unit that, according to the spec sheet, offers 192 preset voic-

es. In reality, the unit only offers the usual 128 voices in the General MIDI default mode. The other 64 voices can only be accessed when piggybacking the TG100 onto Yamaha's Disk Orchestra and Clavinova series of instruments.

Besides the sound quality, one of the coolest features of this box is that it can be directly connected to a computer via a built-in MIDI interface. This unique feature makes the TG100 a good module for lightly equipped studios, as it eliminates the need for a dedicated interface by connecting to an available serial port on any Mac, IBM-compatible, or PS/2 computer. If you're willing to temporarily give up the mouse's serial port, it could even act as an interface for your laptop computer. As one would expect, the TG100 also has standard MIDI ports, so that you can expand your system by plugging it directly into the existing MIDI chain.

Another option, that might be of use (though I'm having a hard time coming up with one) is a front panel Audio-In jack and volume control that lets you mix audio signals in with the box's internally generated sounds. Often, such an input would compromise on sound quality, but an A-B comparison showed that no corners were cut here. I couldn't tell the difference between the TG100's output and the direct output of a CD player.

The device's sound quality is tight and punchy at the bottom end and present at the top end, however, it lacks a little in the midrange department. Not too much, but its slight midrange shyness gives it a bit of a rough, in-your-face edge. The ten sep-

arate drum setups (of which only eight are accessible to General MIDI) are good enough to stand on their own, but would also be a useful addition to any percussionist's palette. The eight global reverb settings, including pairs of hall, room, plate, and delay effects, were likewise more than decent. All in all, the sounds and effects are in line with most high-quality synth or sample playback modules.

It's interesting to note that when comparing the TG100, Roland MT-32, and a Proteus General MIDI patch bank, I discovered something that should have been obvious to me — that even the cheesiest multimedia sequences could be greatly beefed up by simply layering several General MIDI devices together. The general effect of the combined sound patches and percussion produced a richer and more interesting texture. Designers of high-end multimedia presentations could really benefit from this simple and effective trick of the trade.

There's just one downer. Unfortunately, the designers didn't opt to backlight the device's LCD display. This simple design flaw often literally leaves you in the dark. All in all, though, the TG100 is a quality sound generator that would be a great entry-level MIDI sound device, as well as a useful addition to any existing MIDI system. Thumbs up!

— *David Miles Huber*

Contact the Yamaha Corp of America at PO Box 6600, Buena Park, CA 90622-6600; tel: 714-522-9011. EQ free lit. #127.

PEAVEY PC1600

any or all of your instruments, the controller will let you edit your own Sys-Ex data command strings by entering the appropriate Hex codes and assigning each fader and button to a control parameter. Sounds good to me; my programming skills, however, are practically nil and Hex is pretty much Greek to me. I have heard that Peavey is currently compiling a library of additional instrument edit templates and will eventually make these available to users. For more info, I recommend consulting your local dealer and having him contact Peavey on your behalf.

Other features of the PC1600 include the ability to save up to 100 faders and switch settings as snapshot "scenes" (handy for stage work). Extensive MIDI filtering and controller data merging are also built into the box, while two jacks on the rear panel make it possible for a footpedal and/or footswitch to be assigned to a controller or to turn a switch function on or off. Incidentally, any and all of the 1600's internal programming can be dumped to and loaded from a data storage device via Sys-Ex.

MY 2 CENTS

In a nutshell, the PC 1600 does its job flawlessly and is easy to use, although I feel that the manual could do a better job of explaining the deeper program aspects in easy-to-understand language. Controller and Sys-Ex data is transmitted really fast (often, when editing on certain instruments, you might have to wait while the receiving device tries to catch up with the 1600's edit moves).

Since the unit has only one MIDI IN and an OUT, it makes sense that its mixing and edit capabilities could best benefit from a system that has a MIDI patchbay or an interface that has patching capabilities. If you're like me, the constant need for plugging and replugging MIDI cables could get old really fast.

Simply said, if you're into MIDI mixing and would like a hands-on control surface with bonus patch and effects editing capabilities thrown in, the 1600 is a real bargain. Now all that Peavey needs to do is add more instrument edit templates to its list and then distribute these on disk to its dealers.
—*David Miles Huber*

Ad Index

PAGE	BRAND	INFO	PHONE #	PAGE	BRAND	INFO	PHONE #
103	Acoustical Solutions	01	800-782-5742	14	Genelec	37	508-650-9444
41	AKG	02	415-351-3500	94	General Music	39	708-766-8230
24,25,17,2	Alesis	3,4,5	213-467-8000	21, 71	Illbruck	40	800-662-0032
3,13	Amek	06,07	818-508-9788	74	Instit. of Audio Research	76	212-677-7580
47	Ampex	08	415-367-3809	91	IMC/Akai	41	817-336-5114
37	Aphex	13	818-767-2929	103	JRF Magnetic Sciences	42	201-579-5773
73	ART	15	716-436-2720	40	KABA	43	800-231-8273
10	Ashly Audio	09	716-544-5191	84	Lexicon	44	617-736-0300
12	Audio Control Industrial	10	206-775-8461	27, 89	Mackie Designs	46,45	206-487-4333
104	Audio Institute of America	11	415-931-4160	92	Miditemp	74	615-355-8756
103	Audia Technica	12	216-686-2600	83	Musician's Friend	47	800-776-5173
75	AudioTechniques	78	212-586-5989	65	Panasonic/Ramsa	52	714-373-7278
117	Bananas At Large	14	415-457-7600	7	Peavey	48	601-483-5365
43	Behringer	16	516-932-3810	104	Polyline	50	708-390-7758
12	Caig Laboratories	17	619-451-1799	128	Rane	51	206-355-6000
110	Center For the Media Arts	18	212-807-6670	49	Roland	53	213-685-5141
20	Conservatory Recording	19	602-496-6508	68	RSP Technologies	49	313-853-3055
66	Crown	20	219-294-8000	39	Russ Berger Design	54	214-661-5222
31	D&R	21	409-588-3411	9, 11	Samson/Soundtracs	56	516-932-3810
117	DAT Audio Gallery	22	213-829-3428	87	SAS Industries	57	804-582-6139
35	dbx	23	415-351-3500	97	Sennheiser	58	203-434-9190
131	DIC Digital	25	201-224-9344	132	Soundcraft	59	818-893-8411
98	Digital Audio Labs	24	612-559-6104	19	Sunn/Fender	60	714-990-0909
16, 20	Disc Makers	26, 27	215-232-4140	96	Sweetwater Sound	61	219-432-8176
51	Discount Distributors	28	516-563-8326	76	Symetrix	62	206-282-2555
115	DOD Electronics	29	801-268-8400	5, 100, 101	Tascam/TEAC	63, 64	213-716-0303
8	Dolby	30	415-558-0200	39	Tech 21	65	212-315-1116
30	Drawmer	32	508-650-9444	71	The Cloning Laboratory	66	616-929-1761
105	Dynatek Automation	31	416-636-3000	74	The DAT Store	67	310-828-6487
15	E-mu Systems	77	408-438-1921	87	The Recording Workshop	68	614-663-2510
81	Ensoniq	33	215-647-3930	116	Vestax	69	707-427-1920
70	Europadisk	34	212-226-4401	45	Walters-Storyk	70	914-255-2255
107	Full Compass	35	800-356-5844	38	Whisper Room	71	410-997-1440
79	Furman Sound	36	415-927-1225	38	World Media Group	72	317-353-1113
124	Gemini Sound	38	718-851-6000	80	Yorkville Sound	73	716-297-2920

ACROSS THE BOARD

many MB you will need to store any amount of stereo audio at any sample rate, what frequency is G#4 or any other note, figure out how many percent down do you VSO your tape machine to get a half-tone down, and tell you how many milliseconds there are between two marks 3/4-inch apart on a piece of tape running 15 ips. The only thing I couldn't get it to do is tell me how fast 30 ips is in furlongs per fortnight. Studio Pal is a must have for any studio.

DUBBING DOWN

Every so often, (once a day) I have to make a DAT-to-DAT copy or some other form of digital audio transfer that requires a connection between a professional and consumer machine. Most professional machines will ignore SCMS or other consumer flags in the digital audio bit stream, so that part of the transfer is OK. The part of the interface that most people miss is that of proper level and impedance matching between the devices. Most of us don't have a problem with using an impedance matching transformer with a microphone or a level/impedance matching box to connect consumer audio gear to professional gear. When dealing with analog audio, you can hear the mismatches easily.

Digital interfacing is a different story. The frequencies are above 5 megahertz. Your ears may still be good, but I can't quite hear that high any more. (It must be from listening to too much video sync.) Simply connecting a cable with an XLR on one end and an RCA on the other end between the two machines may have let you squeeze by in the past, but that is not the right way to do it. I found some impedance matching transformers made by Canare (the cable people) that are designed to match the 110-ohm balanced 5-volt levels of AES professional, to the 75-ohm unbalanced 1-volt levels of SP/DIF (IEC958) consumer.
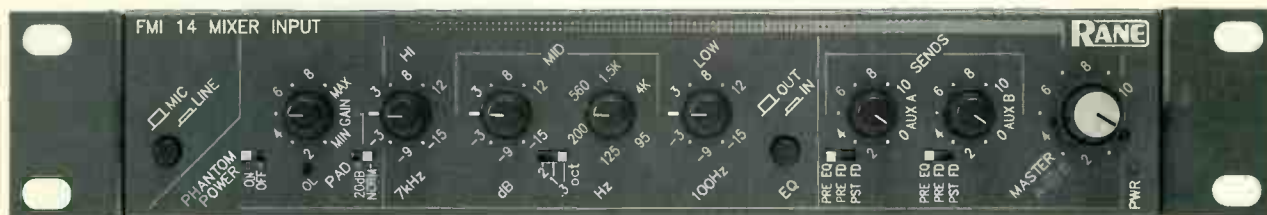
Once in a while, when listening back to digital transfers, I would hear a little "zip" or "click" or a faint hissing sound that would pan from one side to the other. These noises would not be on the original tape. (Sometimes they are on the original tape and I wish that they would go away in the transfer.) These are caused by the loss of proper synchronization between the play and record decks, which is directly attributable to the level and impedance mismatch in the interface between the two pieces of equipment. After I inserted one of these transformers in the digital signal path, all of the problems disappeared.

Remember: These transformers don't change the format of the digital audio between professional and consumer, they just clean up the levels. If you need to change formats, then you need the Digital Domain FCN-1 format converter, but that's yet another story.

I hope you don't mind if I cut this a little short, I have about 164 new products that I want to review including a CD-ROM recorder. Yup, record your own CD-ROMs for fun and profit. By this time next week I'll be slaving over another hot recording console, working my fingers to the bone — *in Maui!* I'll be working with Walter Becker on his solo album. We have to get it done by summer, in time for the 1993 Steely Dan tour. On our days off we will be searching for a suitable cliff off which to sacrifice one of Walter's old 3M digital machines. You have to keep those Digital Gods appeased, now, don't you! **EQ**

And the Check Is In The Mail...
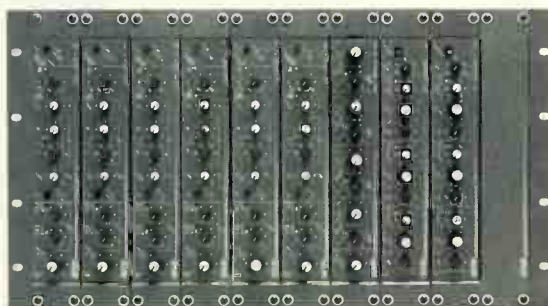
Donald Fagen's album hasn't hit the streets yet. It's on its way...promise.

BY ROGER NICHOLS

A couple of columns ago I stated that Donald Fagen's album would be done by the time that issue crossed your desk. I lied. It is now the end of February and we are just finishing the mix of the final tune. It wasn't the mixing itself that took so long, but all of the little things leading up to it. You know, the recording part of the project. It seems that there is no way around it. All of the elements must be recorded before you can mix. I knew there was a catch.

We mixed Donald's album at River Sound in New York. This is not a plug,

but I think that everyone should record there so they can pay for all of the cool toys that we got to use during the mix. The console is a Neve 8078 that came from the old Motown West studios in Hollywood. It is a nice warm sounding console. The drawback with Donald's project was that the board only has 40 inputs and we had 48 plus tracks chock full of instruments waiting to be mixed. The 8078 has a monitor section that can be routed to the main mix bus for additional inputs, but the gain structure is not the same and the half-size faders are not automated. Since the main part of the console is fitted with Flying Faders, I thought it would be neat (sorry, I meant to say "bitchin'") to have a little outboard rack of automated faders that could be run by the main console automation and control the audio feeds to the monitor section of the console.

River Sound's tech wizard, Phil Burnet, got an additional 18 Flying Faders from Neve and went to work on the idea. Six of the faders fit into an open space in the console, but the other 12 were mounted into a little side car that stuck out at a right angle to the console. It took a couple of days to match the gain and sound of the monitor section to that of the main part of the console, but with a little help from the SIM machine, we came out from under the console with 60 Flying Fader inputs. A stack of Focusrite, Meyer, GML, Pultec, Avalon, and Radio Shack (just kidding) EQs finished off the monitor section inputs. So, next time you try to mix something at home and wish you had just one more input for the zither-wingle, remember that the same thing happens in the big studios.

This marks the sixth album that I have mixed to CD-R (recordable CD). There must be a lot of CD-R machines out in the market place because blank CD prices are dropping like a stone. About 18 months ago, blank CDs were about $80 each by the case. Last week the going price for a box of ten blanks was $25 each.

The reason I like this format so much is because it can't be erased. No longer do I have to wish that I hadn't

gone over that earlier version of a mix. All I have to do now is remember to record it onto the CD.

Besides printing the mixes to the CD-R, I also had my Apogees feeding a Fostex D-20-B timecode DAT machine and an Akai DD-1000 optical disk recorder. These other formats provided back-ups for the master (the CD-R). Along with the mix, I printed timecode from the 48-track onto the Fostex and Akai. I did this for a couple of reasons:

First, the timecode reference on the optical disk would be the same for every version of the mix. If at some later time, for example, Donald wished the vocal was louder in the second verse, we could easily drop in that section from the "vocal up" version of the mix. I would only have to find the edit points on one version of the mix. Since the timecode reference is the same every time, we print a mix substituting any other version in the same spot.

Second, timecode on the DAT would allow us to lock up the mix to the original 48-track. If Donald decided that it was a mistake leaving off the *digidoo* solo, we could add it to the mix without remixing the whole song.

REVIEW BYTES

I am finishing up a review of Digidesign's DINR noise reduction software, and so far I am very impressed. I used it on Donald's album to eliminate a buzz on the Fender Rhodes he played on a song called "Counter Moon." The buzz was from a ground loop that we just couldn't get rid of. I transferred the entire Rhodes track (eight minutes of stereo) into SoundTools, let DINR learn the noise characteristics during a quiet spot in the track, processed the file, and transferred it back to the 48-track. All of this was done in the digital domain and the whole process took less than thirty minutes.

For you Mac users out there, I found the ultimate desktop accessory for use in the studio. It is a software calculator called Studio Pal (Harmonic Systems, Inc., P.O. Box 488, Fairfax, CA 94978-0488. Tel: 415-485-5242). It will do timecode calculations, tell you how

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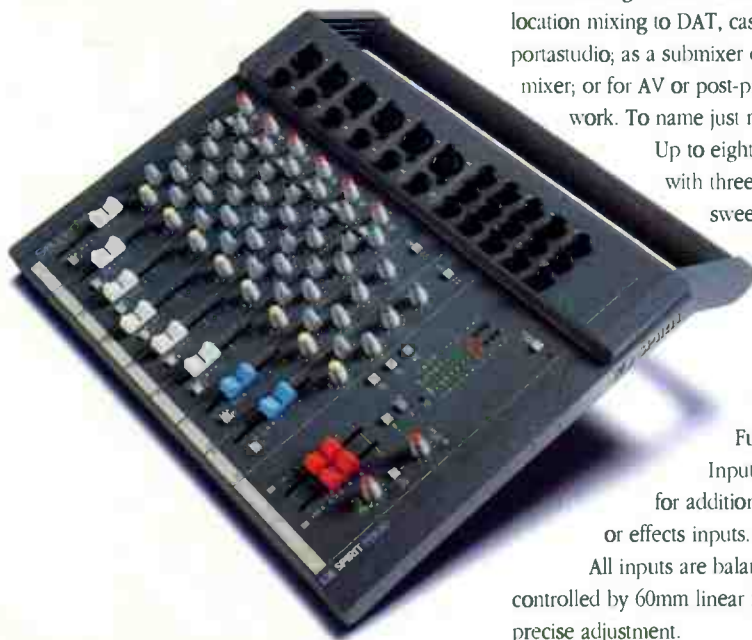Inputs are perfect
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