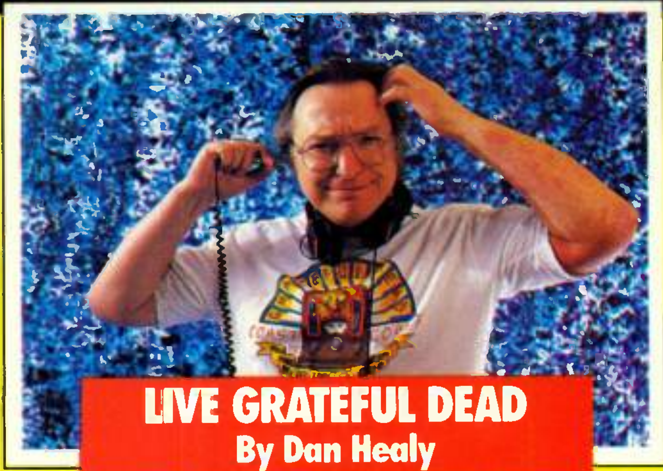
PROJECT RECORDING & SOUND TECHNIQUES

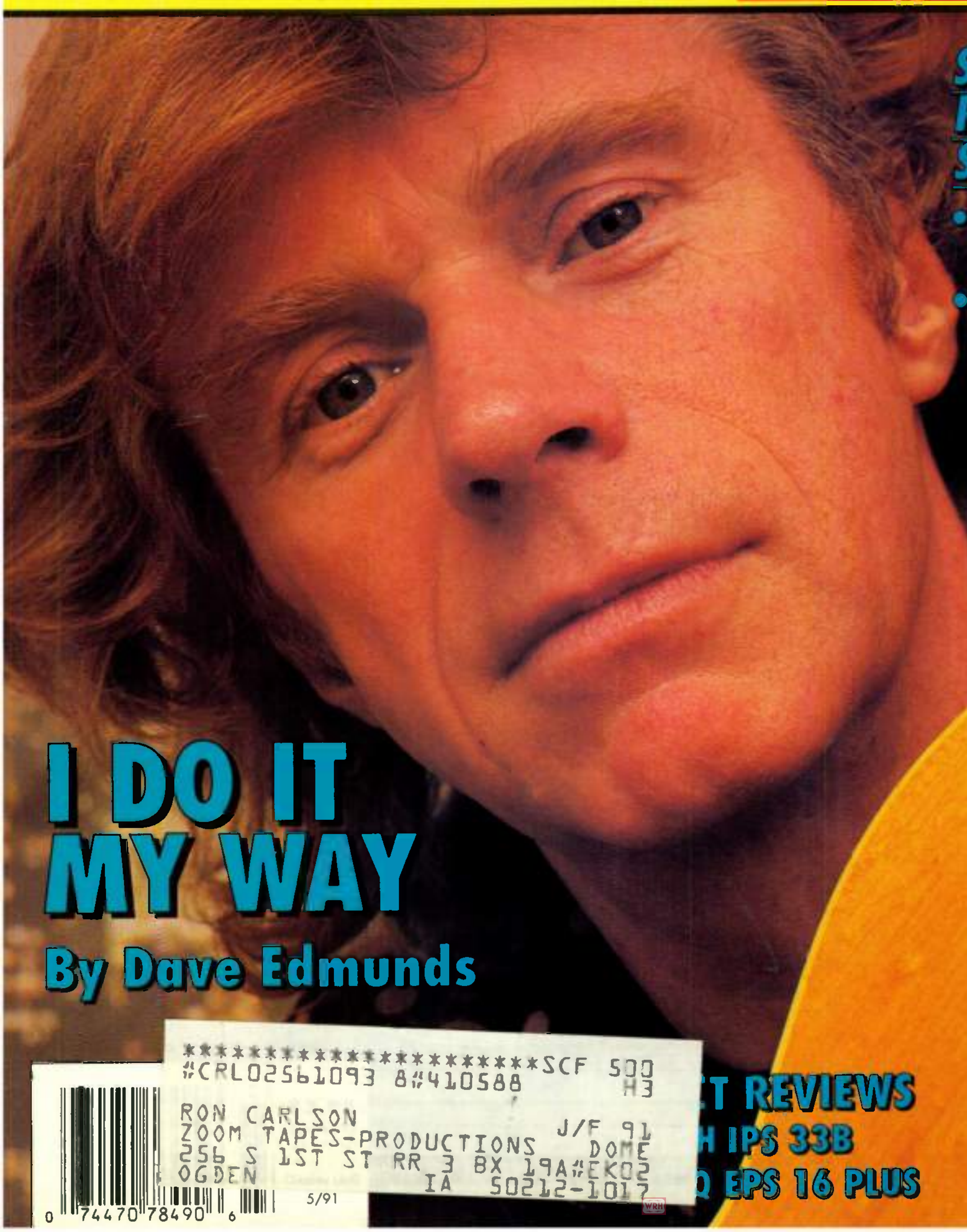
EQ

APRIL 1991 · PSN PUBLICATIONS



LIVE GRATEFUL DEAD

By Dan Healy



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ROLAND R-880 REVIEWED

By Roger Nichols

BUILD YOUR OWN EQ

By Craig Anderton

I DO IT MY WAY

By Dave Edmunds

T REVIEWS

H IPS 33B

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PROJECT RECORDING
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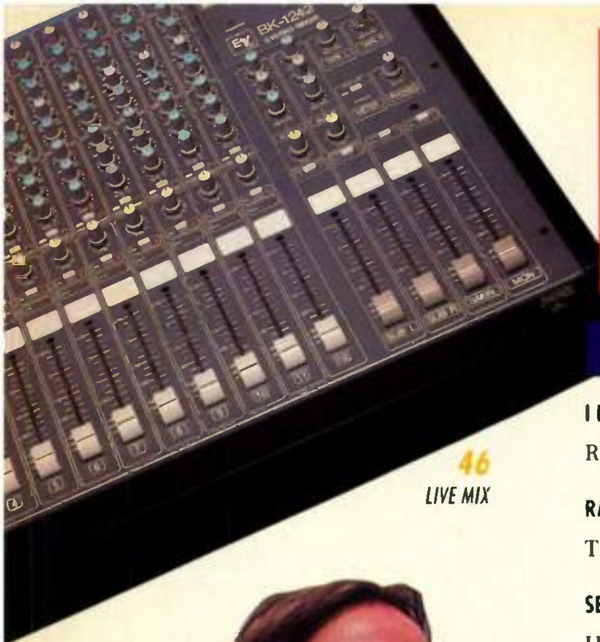
VOLUME 2, ISSUE 1
APRIL 1991



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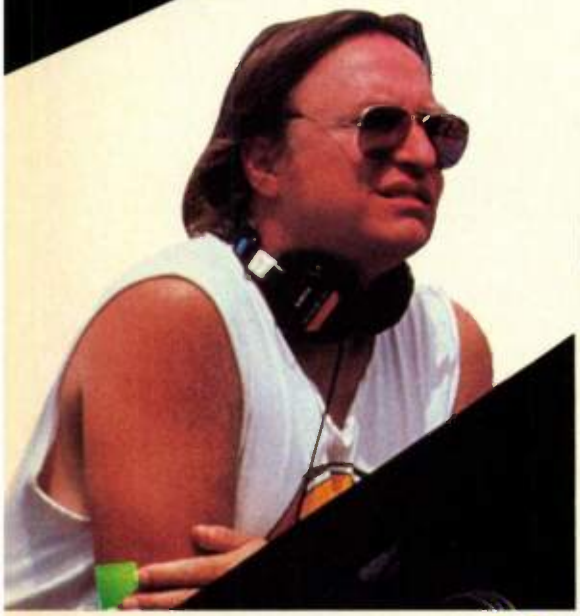

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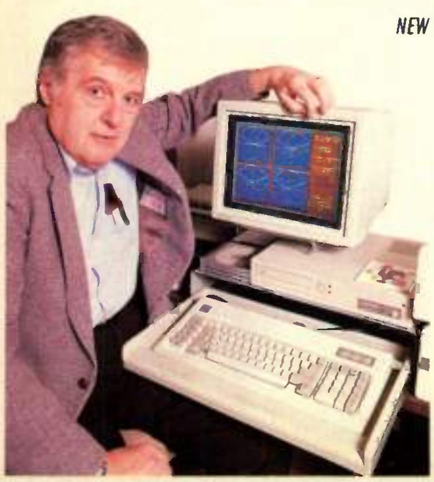

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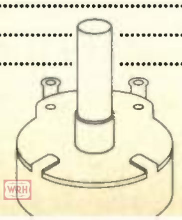

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THE RE27N/D

Neodymium-Aligned Technology and Variable-D® Design are Combined to Create the New RE27N/D

Twenty years ago, Electro-Voice introduced the legendary RE20, which soon became an industry standard in broadcast production and recording. Today, the Variable-D design concept pioneered by the RE20 is still world class.

The new RE27N/D not only utilizes this time-proven design, but takes it one step further with the addition of EV's N/DYM® technology. Electro-Voice was the first audio manufacturer to harness the power of this rare-earth super magnet. N/DYM actually delivers four times the power of conventional magnets. The RE27N/D also offers three switchable filters, one high frequency and two low frequency. Due to the increased sensitivity provided by N/DYM, the switchable filters enable the selection of either a flat high-end response or a shelving emphasis above 4 kHz for enhanced vocal presence, and the option of two low-frequency rolloffs.

The net result: a microphone that is designed in the tradition of the RE20, but exhibits higher output and even wider frequency response, providing a high-performance version of the Variable-D design concept.

N/DYM and Variable-D are the ingredients of a perfect blend — the new RE27N/D.

For additional information, contact Ivan Schwartz of Electro-Voice at 616/695-6831.

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CIRCLE 15 ON FREE INFO CARD

Our new professional DAT re
you'll have to look at DAT in a

<u>Time Code.</u> It's about time someone perfected time code for DAT. And Sony's new PCM-7000 DAT recorders have done just that. They make time code based editing easier than you ever thought possible. They allow you to read and generate SMPTE, EBU or film rates which can be prestriped, poststriped or recorded simultaneously with your audio. They even let you translate from one time code rate to another during playback. All of which means if you're not using a Sony professional time code DAT recorder, you may want to take a closer look at what you're missing.

<u>Speed and Size.</u> When speed counts the PCM-7000 recorders are the tools you want. They have a shuttle speed 175 times play speed, which lets you locate cues or lock to other equipment faster than with open-reel systems. They also come with helpful menus and self-diagnostics for fast set-up and easier maintenance. But speed isn't the only issue. Size is also important. Unlike reel-to-reel recorders, you can fit any of our new DAT recorders into just 5¼" of rack space. In addition, each DAT tape can fit two hours of stereo digital audio into a package smaller than a standard audio cassette, saving you plenty of storage space. And since DAT tape costs about one-third of analog open-reel tape, most facilities could save enough in the first year to pay for the recorder.

<u>Instant Start.</u> If you need "On-the-Air" or "On-the-Fly" cueing, you can equip our new DAT recorders with Instant Start. It's simple to use. Just mark and trim the starting point, then press PLAY. You'll get instantaneous audio output with absolutely no start-up "wow." To make it even more convenient, you can initiate Instant Start with a fader-start or GPI.

<u>Chase Synchronization.</u> With the internal Chase Synchronization option of the PCM-7050 and 7030, you can press a single button to lock to any time code based equipment—whether it's a VTR, ATR or a second Sony PCM-7000. You can also enter or capture an offset instantly and maintain synchronization with the time code data or from an external reference.

<u>The System.</u> The PCM-7050, 7030 and 7010 time code DAT recorders are all designed and constructed for professional use. They each offer clear advantages in sound quality and operating economy over conventional analog recorders. They're smaller and faster than analog recorders. And when you add the RM-D7300 edit controller to the PCM-7050 and 7030, they create a powerful electronic editing system. No other DAT manufacturer offers you this kind of system approach because no other manufacturer looks at DAT quite the same way as Sony. For more information, call the Sony Professional Audio Group at 1-800-635-SONY.

<u>Professional Inputs and Outputs.</u> It goes without saying that you can connect these DAT recorders with a variety of I/O's—like balanced XLR analog I/O's or optional digital I/O's, including AES/EBU. This enables you to transfer audio to digital VTR's and just about any other professional audio equipment you desire.

corders have so many features, whole new way.

External Control. Our new DAT recorders offer you several external control options—an RS-232C port for computers, parallel connectors for external synchronizers and a 9-pin serial port on the PCM-7050 and 7030 for compatible video editors. So you can control our DAT recorders from just about any source you choose.

Off-Tape Monitoring. A sophisticated four head design lets you monitor off-tape as you record. So you'll always be confident of the quality of the recorded signal.

Electronic Editing. Plug the PCM-7050 and 7030 into the RM-D7300 edit controller and you'll be able to capture edit points in RAM memory, trim with the jog wheel and preview edit points before committing them to tape. You'll also be able to set the crossfade time from 0 to 999 ms and digitally adjust the audio level for the smoothest transition possible. It's definitely the best way to edit DAT.

Audio Quality. We've minimized phase distortion in our new DAT recorders by giving them 18-bit D-to-A converters with 8 times oversampling—and A-to-D converters with 64 times oversampling. And because they're digital, frequency response is extremely flat from 20Hz-20kHz, dynamic range exceeds 90dB and "wow and flutter" levels are so low they can't be measured. Which sounds pretty good here, but sounds even better on DAT.

SONY.

BUSINESS AND PROFESSIONAL GROUP

CIRCLE 49 ON FREE INFO CARD

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LETTERS TO EQ

GO FOR IT

It is 3:00 a.m. here in Mt. Pleasant, Michigan. I have just finished reading your December 1990 edition front to back. I love it! I especially love the article by Nile Rodgers.

In the last few years I have thought about expanding my musicianship by becoming a producer also. Now I am certain. I am so moved by this article and all the things I've learned in just this one edition, that I've made a big decision. I'm moving to Detroit. I'm transferring from my school here (Central Michigan University) to Wayne State.

The music industry needs me and I need music. EQ needs people like me and if it wants to become my pipeline to the industry, I'll need EQ. So put your feet up Hector, because you've found a home also. Thank you.

Joey Lee Cunningham,
Mt. Pleasant, MI

A-FOR-V

"The Five Day Demo" User Techniques article (February 1991) was very interesting to a video user like myself who doesn't post audio like the author, Paul Lehrman. Paul, 3/4-inch U-Matic video has standard audio on channel 2. That's why audio dub is channel 1 only. Next time you'll want a VCR with audio insert capability. Knowing the capability you need will make it easier to decide what gear to rent.

John McHugh, TV33,
St. Louis Park, MN

COLLECTOR'S ITEM

It was with great interest that I read Martin Porter's article "Legacy: Voice of a Generation" in the February 1991 issue of EQ magazine. The article was superb — it was both fascinating and informative.

I have owned a Shure 55-S for about ten years and recently required a second, identical microphone. My problem has been locating sources for information concerning the Shure 55-S series mics. I believe that the 55-S microphones I have were produced during the late 1950s or during the 1960s. I am quite interested in locating the original mics, perhaps from the 1937 - 1945 period. More important, however, is obtaining any and all literature relating to this classic mic. I am seeking articles, original specification sheets or booklets. In short, I have become a 55-S freak and need a fix!

I have read EQ with great interest since the first issue.

Great job. Keep up the good work!

Charles Granata,
Clifton, NJ

Charles: You can contact Shure at 222 Hartrey Ave., Evanston, IL 60602; (708) 886-2200. We find that the best way to locate old articles is working in the reference library: search under microphones, audio and other related topics. Good luck.

SQUELCH DEFENSE

Richard Grula's article on wireless microphone tips in the February issue covers some important points, but he rather unfairly discards the usefulness of the squelch control by equating it to a leftover CB ham radio knob.

The squelch control on a wireless microphone or guitar system serves a very important purpose. By properly setting the squelch, you have set the wireless system to perform at its optimum under the existing circumstances. Since most wireless systems operate under frequencies that are used by other RF devices, the squelch serves to set the sensitivity of the receiver to pick up its intended transmitter and not the other RF signals floating around out there. While this might "decrease the operational range" of the system, it does so to keep the microphone or the guitar system from experiencing a nasty "hit" from a strong source of interference that might not even be on the same frequency. With the squelch wide open, even a minor dropout of the microphone or guitar signal will come roaring through with "ear-shattering VHF white noise." Any other loss of signal, whether it is a weak battery or someone turning the transmit-

WRITE TO US

Write to: Letters to the Editor, EQ, 939 Port Washington Blvd., Port Washington, NY 11050. EQ welcomes letters but reserves the right to edit them for clarity and space.

EVEN WALTER BECKER IS TALKING ABOUT SOUNDTRACS.

Soundtracs IL Series

"You have to be careful about what you spend for a console in a home studio. You want as clean a signal path and as versatile a board as you can get. The Soundtracs IL 4832 is logically laid out, easy to get around on, has great sounding EQs and prints a very clean signal to tape.

We use a 32-track digital recorder — the IL 4832 made the most sense. It provides a 32 buss design in an extremely affordable package. It looks great in the room, too."

As a founding member of Steely Dan, Walter Becker is known for his uncompromising point of view. So choosing a console for his personal studio in Maui was a carefully considered decision.

Soundtracs IL 4832 features an inline design that produces a pure, transparent sound. Its 32 Busses allow total flexibility for maximum ease of use in a variety of recording situations.

The IL 4832 comes standard with patchbay and delivers up to 104 inputs with EQ and Aux on mixdown. The board is also available in a 36 mainframe format.

Sonic purity, versatility, maximum inputs and operational flexibility. These are the reasons why even Walter Becker has so many good things to say about recording with the IL 4832.

SOUNDTRACS PLC

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CIRCLE 29 ON FREE INFO CARD

ter off, will result in the same scenario. The correct way to set the squelch is to turn the receiver on, turn its transmitter off, and raise the squelch control until the receiver just goes into "quieting"; that is, the RF noise goes away. Setting it to a lower position will only allow some of the interference to get through; setting it to a higher position won't buy you any more protection and will decrease the useful range of the system. It is a good idea to leave the receiver idling at the setting for a few minutes to see if there is some intermittent signal out there that occasionally pops up. The best time to do this, from a technical standpoint, is when all other electronics on stage and in the club are fired up. This includes amps, lighting systems, other wireless units (do them one at a time), special effects and the like. Setting the squelch when nothing else is on will only give you a false sense of security.

By the way, Grula's second tip — keep the receiver as close to the transmitter as possible — is a great idea. But remember that digital audio processing devices are transmitters all by themselves. Wireless receivers have to be separated from them by at least two or three feet.

Keep up the good work. The world needs the information.

John F. Phelan,
Director, Technical Markets,
Shure Brothers

READER CHALLENGE

I recently purchased a used 16-track tape recorder from an electronics whiz who picked it up at a government auction. The manufacturer is SANGAMO, model SABRE III. It was built, I'm assuming, for military intelligence gathering, as each track is hooked up to individual FM radio receivers.

The problem is that it's not simul-sync capable; when you record, you record on all 16 tracks. Also, if you record on one track and then try to overdub on another track, the information on the original track will be recorded over and lost. However, it appears that the manufacturer did allow for upgrading. On the front panel of each input module there is a hole punched for a toggle switch and print that reads — REREC ON/OFF — but no switch, just a plastic plug in the hole.

I have no idea who or where SANGAMO is. Can anybody help me?

Dennis Bonnet,
San Antonio, TX,

Readers: Forward any clues to our editorial offices.

BACK TO BASICS

As a recording student at Towson State, I would like to thank you for your informative and educational publication. I'm writing to ask whether you plan to run articles covering fundamentals of how audio equipment works.

Debbie Nachman, Baltimore, MD

Debbie: Check out John Woram's new column, starting this issue.

EQ&A

Q Could you describe the difference between M-S (mid-side) and X-Y stereo miking techniques? Also, is there an audible difference between using two coincident microphones and just one stereo microphone?

Walter Daniels, Langhorne, PA

A Both the X-Y intensity stereo miking technique and the M-S scheme were invented by A.D. Blumlein, whose comprehensive Patent #394325 (1933) laid the foundation for two channel stereo as we know it. Blumlein also introduced the A-B technique, which refers to a left/right spaced pair of microphones.

The X-Y technique refers to a coincident pair of directional microphones, aligned vertically one diaphragm above the other, and traditionally at an acceptance angle of 90 degrees. The M-S technique is another coincident arrangement, with a microphone (of any directional characteristic) pointed forward and a figure 8 microphone pointed at 90 degrees. M-S has a number of advantages over the X-Y technique, but it does require a special matrixing sum-and-difference network to recreate the normal left and right signals. M-S allows the apparent image width to be adjusted electronically, accompanied by a subjective change in the reverberation and hence the feeling of distance. You could call it kind of an audio "zoom lens," used either at the recording site at microphone level, or after tracks are on tape at line level (depending on the capability of your matrix).

Special stereo microphones give more precise coincidence, and allow the manufacturer to more accurately match the frequency response of the capsules. Great care is taken with uniform off-axis and directional characteristics, which is vitally important since so much of the sound is off-axis (picked up edgewise).

A compact disc has been prepared that showcases the audible differences between various miking techniques on the same source material; for more information contact Performance Recordings (No. PR-6-CD) through Harmonia Mundi USA, (213) 559-0802.

David Ogden,
Product Manager
AKG Acoustics

Q I run a small Macintosh SE-based MIDI studio with a lot of rackmount synthesizer and signal processing gear. I've been using Passport's Master Tracks Pro along with the Opcode Studio 3 interface, but now I'm interested in upgrading to Mark of the Unicorn's Performance 3.5/MIDI Time Piece computer interface combination because of the eight independent MIDI channels that the MIDI Time Piece provides. However, I have a few questions: First, do you need a Mac II computer to take advantage of the MTP's "fast" 8-output mode? Second, since clients bring in sequences created on various sequencers (Vision, Beyond, Cubase, Master Tracks Pro, etc.) can these other programs drive the eight individual outputs, or will I be limited to using just two outputs? Are there other companies supporting the MIDI Time Piece? Or is it a situation where different programs will not work with different interfaces?"

Andy Colon, Plantation, FL

A The MIDI Time Piece (MTP) enhances your current studio setup in several ways: Along with the eight independent MIDI outs, you also get a sophisticated MIDI merger/patchbay with data muting and rechannelizing MIDI processing effects, a SMPTE-to-MIDI converter, SMPTE generator, and a Fast data transmission mode.

The MTP Fast data mode allows the Macintosh and the MTP to communicate as fast as the Macintosh will send and receive data. On a Mac Plus or SE this rate is approximately 150 percent faster than the standard one megahertz speed of other interfaces. On a Mac II (or higher) this rate can be 200 percent to 400 percent faster than one megahertz.

Currently, Dr. T's Beyond sequencer supports the independent MIDI out cablization scheme of the MTP; Opcode's Vision and Studio Vision, Steinberg-Jones' Cubase, and Passport's Master Tracks Pro 4 all plan to implement MTP support in upcoming revisions.

Access to all eight MTP MIDI inputs and outputs is available and completely configurable even when using a sequencer that does not fully support the MTP. However, using such a sequencer only allows access to 16 MIDI channels per serial port, totalling 32 MIDI channels if both the modem and printer port are connected.

Susan Patalano/
Daniel Ross
Mark of the Unicorn

Q I have heard that the SCMS (Serial Copy Management System) on the Tascam DA-30 DAT machine is disabled on the digital inputs. Is this true? Also, is SCMS only activated when recording digitally from CDs, or from any digital sources? I'd sure like to be able to make digital copies of my master tapes that are dumped to DAT via Digidesign's Sound Tools.

Albert Eigenreich, Erie, PA

A The Tascam DA-30 has two types of digital I/O ports available. The RCA connectors are for the S/PDIF or consumer interface and they pass on SCMS information. This was originally intended as the port that allowed a CD player's digital output to be connected to a DAT recorder. As such, it was the primary source of the arguments that resulted in the adoption of SCMS. The DA-30 also has a pair of XLR connectors which are for the AES/EBU format digital interface. These connectors do not pass through any SCMS data and allow any number of DA-30s to be connected for duplication purposes.

As for using Sound Tools, if you have their DAT I/O, you are working

This is where your questions get answered. Send your query with your name and address to: EQ Editorial Offices, 939 Port Washington Blvd., Port Washington, NY 11050 with S/PDIF. However, the output of Sound Tools, unlike a pre-recorded CD, doesn't contain copy inhibit (SCMS) data. Fortunately, the SCMS protocol specifies that if no copy inhibit bits are written into your master then they won't be added in downstream copies.

I can't guarantee that every consumer machine on the market will conform to the letter of the protocol and not add copy inhibit flags downstream, but that's why the AES/EBU alternative exists. This port is excluded from the SCMS copying limitations and was specifically designed for the recording industry. Sound Tools Pro I/O has this type of port. The best way to avoid SCMS is to stay in the AES/EBU domain both for your DAT master and any additional copies you want to make.

Bill Stevens,
Manager, Media & Market Support,
Tascam

Q I want to add transformers in order to split the stage boxes on our existing snakes. Who offers transformers designed for this purpose (ones that are easy to mount in a box)?

Erik Seifer, San Antonio, TX

A There are many audio manufacturers that offer transformers for the purpose of isolating a split of mic signals in a PA snake.

Two companies that have excelled in this field are Whirlwind Music (Rochester, NY) and Jensen (Burbank, CA). Both of these manufacturers feature a full line of transformers. The Whirlwind models are the TRSP-1 (for a single isolated split) and the TRSP-2 (for two isolated splits). These are encased in a complete metal housing with two rivet points at the base for mounting.

Jensen transformers are also very well known in the audio industry. Mounting these units is more involved, requiring special bracing and separate mounting brackets.

A splitting transformer is technically used to isolate the inputs of two or more consoles from each other and to maintain proper impedance matching of the microphone. It allows you complete electrical isolation between the consoles, which greatly reduces frequency loss from impedance loading and hum resulting from interconnecting. It also must be understood that a transformer does not pass phantom power or any type of DC current.

Mic Cardone,
National Sales Manager,
Whirlwind Music Dist.

Q I play an upright German-style tuba in several polka bands and have not yet been happy with the sound quality over the full dynamic range. How should I mike my instrument to get the best effect?

Robert Yates, Eugene, OR

A Capturing the true sound of brass instruments is no easy task. We congratulate you on your perseverance and desire to get it right.

Because of its overall size (causing the player to frequently readjust his playing position), the size of its horn bell which forces considerable amounts of air through it, and its low tonal range, the tuba is more difficult to mike properly than many other brass instruments. Microphones capable of dealing with such power would ordinarily need to have larger diaphragms and sturdy, noise-free housings.

Condenser-type mics would generally be more sonically accurate, but in your situation — performing live and lugging around a large instrument

continued on page 78

EQ PRODUCT VIEWS

MIDI MOUSE

JLCooper's CS-1 Control Station is a mouse replacement for hard disk-based recording systems, sequencers, and multimedia software. It features easy-to-use controls resembling a tape recorder, and a footswitch input that provides hands-off operation. Meanwhile, a optically-encoded jog/shuttle wheel provides precise positioning and simple "scrub" editing. The CS-1 can be used with a number of disk-based recording and editing programs including Digidesign's Sound Tools, Sound Designer II, and Audiomedia software. It can also be used to control a number of sequencing programs such as Opcode's Vision and Mark of the Unicorn Performer. The CS-1 retails for $599. Contact **JLCooper**, 13478 Beach Ave., Marina Del Rey, CA 90292; (213) 306-4131. Circle EQ Free Literature number 153.

ROCKING IN LA

Crest Audio has begun shipping the new **LA Series** of power amplifiers. Based on the technology of the FA Series, the LA amps give musicians quality amplification at an affordable price. All three models in the series are 2U high with power specs of: (LA601) 120 w/ch at 8 ohms, 225 w/ch at 4 ohms; (LA901) 225 w/ch at 8 ohms, 300 w/ch at 4 ohms; and (LA1201) 280 w/ch at 8 ohms, 450 w/ch at 4 ohms. Common features include stereo, bridged and parallel operation modes, relay output protection from DC and short circuits, balanced inputs on 1/4-inch TRS and barrier strips, octal accessory sockets, magnetic circuit breaker/switch and modular construction. Their retail prices are $630 for the LA601, $782 for the LA901, and $1134 for the LA1201. Contact **Crest Audio,** 150 Florence Ave., Hawthorne, NJ 07506; (201) 423-1300. Circle EQ Free Literature number 129.

MIC, LINE & MIXER

Mackie Designs CR-1604 ($1099) is a compact 16-channel mic/line mixer that features a proprietary convertible design that allows it to be easily configured for tabletop console use or rack mount use. It is armed with six studio grade discrete mic preamps with 48v of phantom power. A three-band EQ and seven auxiliary sends are evident on every channel. The CR-1604 offers low noise, exceptional headroom, dual function channel mute, sealed rotary controls, and a Unity Plus fader design. Contact **Mackie Designs,** 3910 148th Ave. NE, Redmond, WA 98052; (206) 885-7443. Circle EQ Free Literature number 100.

SYNTH CENTER

Peavey's **DPM3SE** synthesizer incorporates a number of capabilities into a single keyboard composition center. Among its many possibilities are 16-bit sampling capabilities, a 20,000 note sequencer, a sampling drum machine, two multi-effects processors, and a 3 1/2-inch disk drive. It provides up to one megabyte of custom user samples as well as four megabytes of internal samples. And its MIDI controller features include multichannel send and receive, and a MIDI sample dump send and receive function. Contact **Peavey,** 711 A St., Meridian, MS 39302; (601) 483-5365. Circle EQ Free Literature number 168.

HERE COMES THE SUNN

Pick whichever one you want — the 200 watt **SPL 1225** or the 300 watt **SPL 1226**; **Sunn's** new pro speaker systems will fill the house with song. With their vented-baffle low frequency sections, these boxes deliver clear, high-power bass response, while the "constant directivity" horns with 2-inch drivers, Titanium diaphragms and circumferential ring phasing couples ensure crystalline and ultra-accurate action in the upper register. The 1225 touts a single 15-inch woofer with 3-inch voice coil, and the 1226 has two. Sunn's woofers bear up in high power situations thanks to their ribbon wire voice coils and polymide KAPTON voice coil bobbins with cast alloy baskets. Both are biamp-ready, with internal third order Butterworth crossover networks (set to 1250 Hz). All you have to do is listen. Contact **Fender Musical Instruments,** 1130 Columbia St., Brea, CA; (714) 990-0909. Circle EQ Free Literature number 99.

R-DAT ATTACK

Yamaha has introduced the DTR2 Digital Audio Tape Recorder, which incorporates Delta-Sigma A/D conversion for increased R-DAT performance. It is equipped with four sets of I/O connections including S/PDIF and optical digital I/O, -10 dB RCA, and balanced +4 dB XLR analog connectors. Switches on the unit's front panel allow selection of digital or analog inputs, and 44.1 kHz or 48 kHz sampling frequency. The DTR2 also provides that Start, Skip, and END IDs can be written and erased, and Start IDs can be automatically entered and renumbered. It lists for $1495. Contact **Yamaha,** P.O. Box 6600, Buena Park, CA 90622-6600; (714) 522-9011. Circle EQ Free Literature number 198.

MASTER MIXER

The **MM-508** from **Ashly Audio** is an eight-input stereo mic/line mixer for recording or live sound reinforcement. The microphone preamp section offers superior audio and noise performance as a result of technology usually found only in more expensive studio recording consoles. Each input offers extremely versatile three-band equalization with Mid-Band sweepable between 140 Hz and 8 kHz , and phantom power is provided for condenser mics. There are two Aux sends per channel, In/Out patch points, and both balanced XLR Microphone and 1/4" phone jack Line inputs. The MM-508 also has two Master Aux Sends and two panable Master Aux returns. The MM-508's retail price is $1499.99. Contact **Ashly Audio,** 100 Fernwood Ave., Rochester, NY 14621; (800) 828-6308. Circle EQ Free Literature number 109.

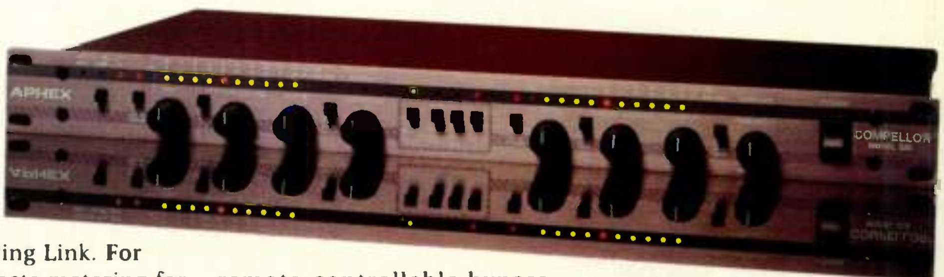
ACCESSORIZE YOURSELF

Ampex has announced a comprehensive line of accessories incorporating both old and new products. This array of audio and video add-ons includes maintenance products, open reels, reference tapes, storage systems, open reel components, cassettes and cassette components and labeling systems. The open reels, which come in two or three windage hole configurations, range in tape width from 1/4-inch to 1-inch and 10 1/2-inches to 14-inches in diameter. The reference tapes are available in the same widths. Contact **Ampex**, 401 Broadway, Redwood City, CA 94063; (415) 367-3889. Circle EQ Free Literature number 98.

PERFECT PERCUSSION

E-mu Systems has introduced its percussion sample playback device called "Procussion" emphasizing that it is more than just a drum machine. This 16-bit, single rack space unit features over 1000 drum and percussion sounds (500+ sounds in both ROM and user-programmable RAM) all of which can be organized into 128 kits. The unit, which E-mu has labeled the "Maximum Percussion Module," sells for about $995. Contact **E-mu Systems**, 1600 Green Hills Rd., Scotts Valley, CA 95067-0015; (408) 438-1921. Circle EQ Free Literature number 139.

IT'S COMPELLING



Aphex Systems has announced a new version of the Compellor compressor/leveler/limiter, the **Model 320.** It features dual monaural circuitry that can be linked in two different ways for stereo operation: with a Leveling link or a Compression and Leveling Link. For simplified operation, a single button selects metering for input, output or gain reduction. The new versions offer a number of new features including reference level switching from the rear panel, Leveling Speed switchable from the front panel, a Peak Limiter switchable from the front panel, remote-controllable bypass relays, and improved I/O circuitry. The Model 320 Compellor is priced at $1350. Contact **Aphex Systems,** 11068 Randall St., Sun Valley, CA 91352; (818) 767-2929. Circle EQ Free Literature number 97.

WIRELESS FUSION

Shure Bros. has announced a new addition to its wireless product line, the **WA400** amplified antenna distribution system. The model is a two-input, eight-output amplified antenna distributor that permits the use of two antennas with as many as four diversity wireless systems or eight non-diversity systems. The WA400 is rack-mountable and it operates with all Shure wireless systems as well as with most other wireless systems. The system is priced at about $584. Contact **Shure Bros.**, 222 Hartrey Ave., Evanston, IL; (708) 866-2200. Circle free literature number 180.

INTELLIGENT FX

Rocktron has introduced **Intellifex,** a 24-bit DSP effects processor with a 64-times oversampled Sigma-Delta A/D converter. This processor utilizes three 16-bit data converters for optimum linearity and a 24-bit data path throughout the unit and its memory circuits for a superior dynamic range (typically 104 dB). These features make stereo delays, reverbs and pitch shifting virtually noise free. It lists for $1149. Contact **Rocktron**, 2870 Technology Dr., Rochester Hills, MI 48309; (313) 853-3055. Circle EQ Free Literature number 174.

DOLLARS AND SENDS

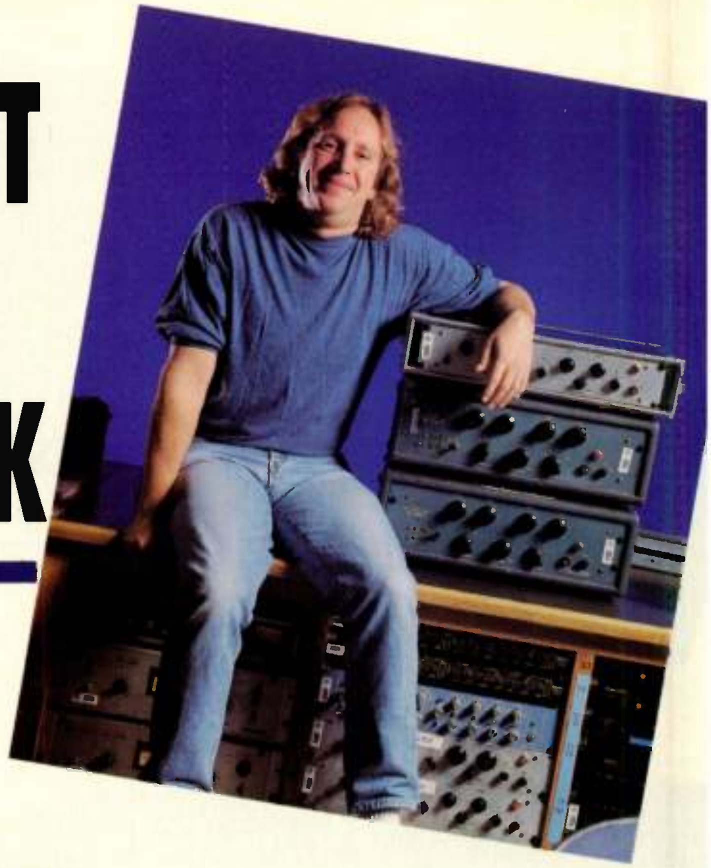
Sendit Electronics is offering two packages that will provide a simple, inexpensive way to add extra effects, cue or stage monitor sends to your mixing console. The first, **Summit Type A,** offers a total of 16 inputs — more inputs than any other line mixer of the same size. By using the on-board output level controls of the effects as return levels, the unit provides a single stereo output to a mixer's aux or effects inputs.

The second unit, **Summit Type B,** allows you to tie two mixers together by giving you one master output control for each mixer's master outputs. Contact **Sendit Electronics**, 544 East Tujunga, #103, Burbank, CA 91501; (818) 841-1078. Circle EQ Free Literature number 176.

UP DATE

News & Notes From The World of Modern Recording & Sound

MEET MY RACK



Engineering ace Cherney with his rack — the first participant in this all-new EQ "Meet My Rack" series.

Given the artists he works with — **Bonnie Raitt, Iggy Pop, Bob Dylan, Elton John** (the list goes on) — mixologist **Ed Cherney** knows what it's like to work in a studio loaded with the best technology has to offer. He also knows that it's sometimes still difficult to capture music the way it was originally envisioned. Cherney's solution is to B.Y.O.R.: bring your own rack.

He's put together an assortment of outboard gear to help him, as he says, "create a space for music to live in." Here's a guided tour:

"Having my own outboard rack enables me to work anywhere, and allows me to maintain some consistency especially since outboard gear differs from studio to studio. My rack gives me peace of mind, and it helps me save a lot of time for my clients. But it is quite expensive to put together," says the veteran engineer.

"I compare my rack to a mechanic's tool box. I carry it around and it helps me solve problems and do my job. I've been carrying around the setup I'm using now for about a year, and I've had some kind of setup for

What About VITC?

Timecode users who come across "VITC" (Vertical Interval Time Code) might think that there is another type of timecode in addition to SMPTE (Drop Frame and Non-Drop Frame), and film timecode (a 24 frame format that has never been officially standardized.) But VITC isn't a new timecode; it's a new way of recording timecode on a VTR or VCR.

Timecode is normally recorded on video using an audio channel or a dedicated track. In either case, the track used for timecode is located along the length of the video tape, and the timecode is therefore referred to as "longitudinal timecode", or LTC for short.

LTC on a video tape has one significant disadvantage: if the tape isn't moving lengthwise, there's no timecode output. In other words, you can't read LTC in still-frame mode. In fact, timecode readers can't read LTC from VTRs in jog or shuttle modes below a certain tape speed, generally about 1/4 of normal play speed.

VITC was developed to get around this limitation by encoding the timecode data and recording it in a portion of the video signal that isn't normally displayed on-screen, namely the "vertical interval." It's the part of the signal you see as black bars if your TV's vertical hold is off.

By encoding timecode in this unused portion of the video signal, it can be read whenever there is a video output, including still-frame and very slow jog/shuttle speeds.

VITC has several disadvantages. You need specialized hardware to extract it from the rest of the video signal, and many synchronizer systems can't handle it directly. Even if you use a VITC-to-LTC translator, some systems will still have problems because of the unusual sequences of frame numbers. Since VITC is encoded in the video signal there will be one frame of VITC for every video frame, 29.97 times a second, regardless of how fast or slow the frames are advancing. In still-frame, for example, the same VITC frame number will be repeated indefinitely. This could never happen with LTC, and a LTC-type synchronizer won't know what's going on. So be sure to think through your entire application and what you'll be using before you opt for VITC.

— Fred Ridder

about four years."

So what's his rack really like?

"My rack is based on vintage tube type equipment for compressors, EQs, and mic pre's.That's not to say that I neglect new technology — the MIDIMan interface box and higher tech digital reverbs are also part of my rack — but the accent is on selected pieces of vintage gear. A lot of it is older, discrete class A tube compressors, limiters and equalizers. The rack allows me to inject warmth and humanness in music that otherwise would have sounded cold and harsh."

With two Neve 1073 mic preamp and EQ modules, Pultec EQP1A3 programming EQs, two Eventide SP2016 digital reverbs, Teletronics LA2 tube compressors, a Fairchild limiter, a Publison 90 Infernal machine, a dbx 160 compressor/limiter, a BBE Sonic Maximizer, a Prime-Time II digital delay line, and a MIDIMan Interface, Cherney's rack is a force to be contended with.

What does he have to say about his assortment of music tools?

"I like the Neve 1073 modules because the high end frequency boost is so smooth (shelving at 12.5 kHz), and they offer a lot of headroom without much distortion. I use them a lot with B&K's 4011 mic; together the mic and the modules provide a unique blend of high tech and classic sound."

"The Pultec EQP1A3s have a sweet high end with a punchy low frequency boost, and the Eventide reverbs have a particularly nice small room sound," he says.

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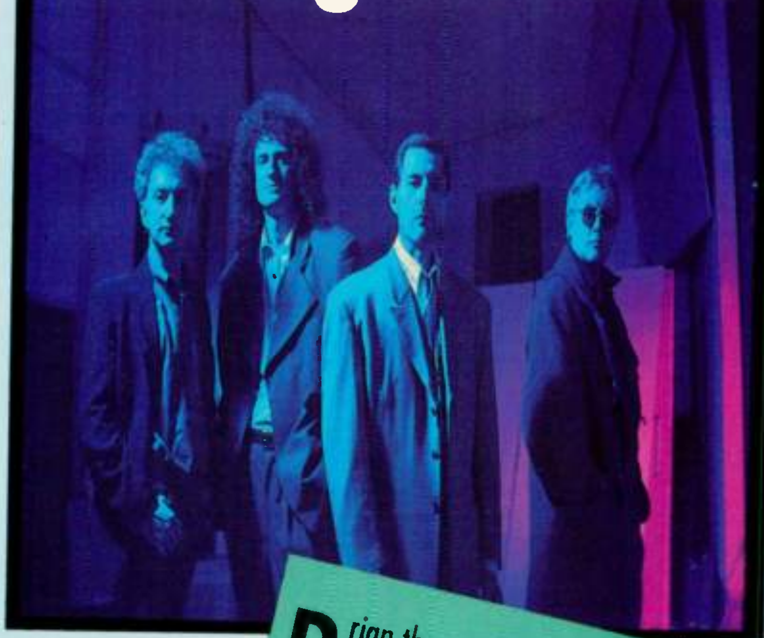
Tracking Innuendo



"First we overlaid a lot of the electric guitar parts in a lot of takes."

Brian then worked out all the parts, playing them on a keyboard and getting all the notes in order.

Innuendo — the title-track of Queen's new album and its first-released single — may have been recorded at Switzerland's Mountain Studios and overdubbed at London's Metropolis Studios with top-dollar gear, but the engineering tricks that went into it can work in studios anywhere. Noel Haris, who engineered the project with band co-producer Dave Richards, explains how the tracks went down:

"The song divides into three basic sections: A military drum-type passage, a calmer flamenco guitar piece, and then the more familiar electric guitar and choral style. The backing track has the main body of the drums and acoustic guitar solos, and as these were returnable, the overdubs — bass, synths, some more drums, and lead and backing vocals - were then performed separately by the musicians.

"First we overlaid a lot of the electric guitar parts in a lot of takes. Brian (May) likes to record scads of guitar material and comp in between them. He's got two Vox AC30's — one with a straight signal, one with a slightly delayed chorus effect that he gets from some old MXR delay units — and for the miking each had a 57 and a 421 placed very close in front, and angled slightly away from the cone.

"We put them up in front of the speakers, cranked the amp volume right up, and Brian would listen through headphones to the return of the mics, and then he'd guide them towards where the hiss was. For the overall ambience, we placed an 87 just a bit further back. He was standing on a carpeted floor, and the amps were facing a wooden surface that I had draped another piece of carpet over, so it was quite dead.

"For the bass overdubs, John Deacon mapped out the track and then played each of the three sections separately, using a DI. The drum overdubs, on the other hand, simply consisted of fills, snare and a bit of top-kit - cymbals and stuff - but this didn't really amount to very much.

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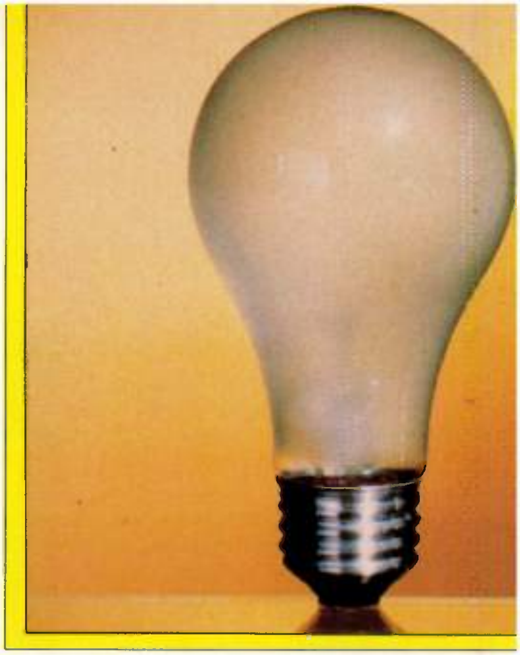


MIC MATCH

Identify these vocal microphones by their pickup patterns: c = cardioid, s = super-cardioid, h = hyper-cardioid.

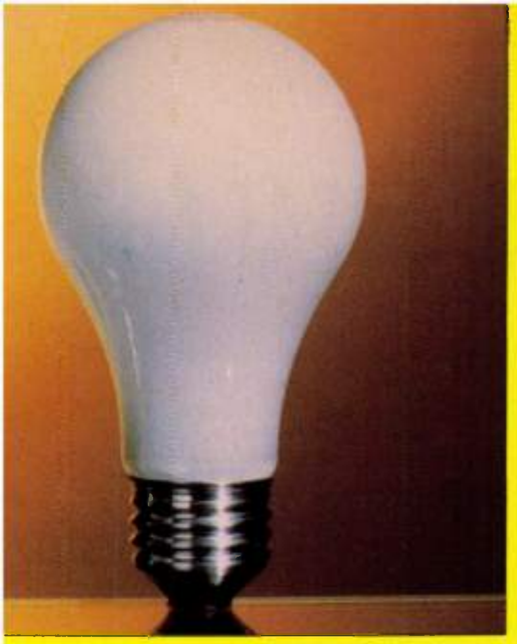
Model	Pattern	Model	Pattern
EV ND257	___	AKG D321	___
EV ND357	___	AKG D330	___
EV ND457	___	Audio-Technica 4031	___
EV ND757	___	Audio-Technica 4053	___
EV ND857	___	Audio-Technica ATM41A	___
EV PL80	___	Audio-Technica ATM63	___
Beyer M-500	___	Audio-Technica PRO5	___
Beyer M-69	___	Peavey PVM38	___
Beyer M-88	___	Peavey PVM380N	___
Sennheiser MD413	___	Peavey PVM45	___
Sennheiser MD416	___	Peavey PVM535N	___
Sennheiser MD421	___	Peavey PVM580TN	___
Sennheiser MD427	___	Shure Beta-57	___
Sennheiser MD429	___	Shure Beta-58	___
Sennheiser MD431	___	Shure SM57	___
Sennheiser MD441	___	Shure SM58	___
AKG C535	___	Shure SM85	___
AKG D310	___	Shure SM87	___
AKG D320	___		

Answers: EV ND257 (c), EV ND357 (s), EV ND457 (h), EV ND757 (s), EV ND857 (s), EV PL80 (s), Beyer M-500 (s), Beyer M-69 (h), Beyer M-88 (h), Sennheiser MD413 (c), Sennheiser MD416 (c), Sennheiser MD421 (c), Sennheiser MD427 (s), Sennheiser MD429 (s), Sennheiser MD431 (s), Sennheiser MD441 (s), AKG C535 (c), AKG D310 (c), AKG D320 (h), AKG D321 (h), AKG D330 (h), Audio-Technica 4031 (c), Audio-Technica 4053 (h), Audio-Technica ATM41A (c), Audio-Technica ATM63 (c), Audio-Technica PRO5 (c), Peavey PVM38 (c), Peavey PVM380N (c), Peavey PVM45 (h), Peavey PVM535N (c), Peavey PVM580TN (h), Shure Beta-57 (s), Shure Beta-58 (s), Shure SM57 (c), Shure SM58 (c), Shure SM85 (c), Shure SM87 (s)

LIGHT-BULBS FROM HELL

And Other Wild Suggestions

Why should life in the studio, club or tour be any harder than it has to? Every day, the world of sound shoves situations on us: frustrating situations, situations that take time to cope with. Making problems, setbacks and the wrong kind of sound go away takes creativity — and after years of experience, you've got a great collection of practical tips and off-the-wall solutions. Here are some of mine:



■ Learn to solder well. Aquiring this talent is very easy, takes very little time — and you'll use it over and over again.

■ Trying to fatten up a snare? Just about everybody has taped a wallet to one, but other common items can work as well. Try household sponges (dampened just a little bit to soften them up) or rectangles of foam rubber in different sizes. We have one report of an engineer (who shall remain anonymous) getting great results with a sanitary napkin.

■ Keep a supply of those little gray AC adapters you see at the counter at your local hardware store. They can be your best friends when it comes to eliminating ground loops and their annoying hum or buzz.

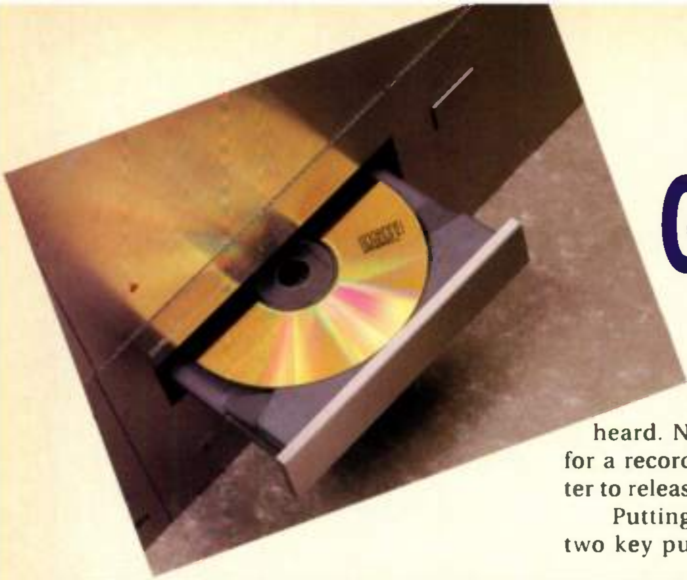
■ Have you ever taken the masking tape labels off your console or other gear, only to find it covered with sticky, gummy adhesive? Nuke it with cigarette lighter fluid and paper towels. Just patch test it first to make sure it won't destroy your finish. This also works on XLR connectors and cables, too.

■ Ever used a "power soak" to get the distortion out of (or is it IN to?) your guitar amp without having to deal with the ear-splitting level? A 60 watt light bulb wired in parallel with your speaker cabinet produces a similar effect: it sounds like it's from hell. And it's always fun to watch the light intensity change as you play.

■ A pair of PZM (Pressure Zone Microphones) taped to the walls around a drum kit as a stereo pair (two individual inputs and tracks) and panned make neat "room mics" and can create great ambience. Experiment with sticking them on the ceiling as overheads or to the sides. Just watch out for wall vibration.

■ Sometime during the next century, perhaps all the audio manufacturers in the world will decide once and for all if the "hot" pin of an XLR connector will be pin 2 or pin 3 (HAH!). Until that unlikely moment, it's a good idea to keep a few short cables around that are deliberately phase-reverse and clearly marked as such, so you don't use them when you don't need them.

—Kevin MacDowall



GOOD TAPE, GREAT CD

You've labored long and hard recording the hottest tape you've ever heard. Now, rather than shopping for a record deal, you decide it's better to release your own CD.

Putting out your own disc serves two key purposes: it enhances your attractiveness to major labels and gets your music out to your fans and the general public in the fastest way possible. But what are the secrets that make a great tape into a great CD?

We asked two of the leading U.S. CD production service companies — Digital House Ltd. and Disc Makers — to tell us the key points.

1 Before sending your master tape anywhere, make a safety copy. This guarantees that you'll always keep your own original master tape safely stored away — safe from all types of potential catastrophes, especially lost overnight deliveries.

2 Always supply the CD manufacturer with a DAT copy of your master. If you must supply an analog master, faster and wider is better (half-inch at 30 ips beats quarter-inch at 15 ips).

3 Record your DAT at the 44.1 kHz sampling frequency, the same rate at which all CDs are recorded. This allows for a direct digital-to-digital transfer and eliminates the costly sampling rate conversion process.

4 Listen to the DAT copy of your master tape to make sure it's accurate. The CD mastering house shouldn't have to enhance or alter anything.

5 Avoid editing charges: make sure your songs are in the right order.

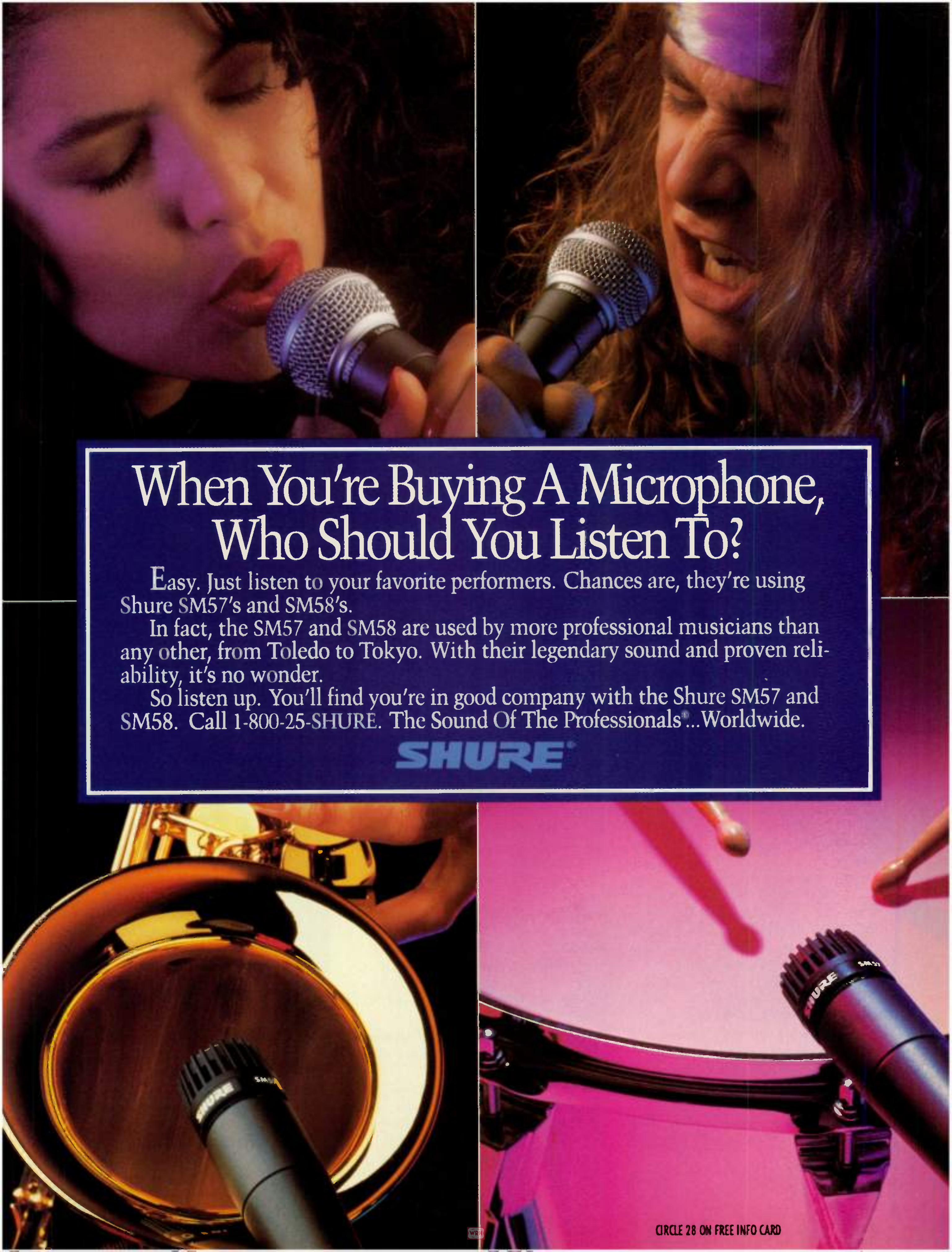
6 Make sure you leave three to five seconds of silence between each cut. You can treat leader creatively if your program material warrants it.

7 Always include an accurate cue sheet, including the titles and lengths of the songs, in the form of a continuous digital time log. This log starts at the beginning of the tape (00:00:00) and runs to the end of the program.

8 Make sure your levels don't peak above zero. On the Sony PCM-1630 Digital Audio Processor, a level above zero results in a mute.

9 Check that your print order matches the order of your CD. This eliminates the unnecessary costs of tape editing, changing your program's order, or reprinting your CD booklet.

10 Always get feedback and a report on how your tape shapes up with clients. This will help you prepare better for your next release.


When You're Buying A Microphone, Who Should You Listen To?
Easy. Just listen to your favorite performers. Chances are, they're using Shure SM57's and SM58's.
In fact, the SM57 and SM58 are used by more professional musicians than any other, from Toledo to Tokyo. With their legendary sound and proven reliability, it's no wonder.
So listen up. You'll find you're in good company with the Shure SM57 and SM58. Call 1-800-25-SHURE. The Sound Of The Professionals®...Worldwide.
SHURE®
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CIRCLE 28 ON FREE INFO CARD

Marquee de Sound

For clubs electric, Bondy builds rooms that rock.

CLUB: Marquee, New York City.
CLUB OWNERS: John Granite, Bob Ellis.
SOUND SYSTEM DESIGNER: Steve Bondy. His systems can also be heard, or shall we say experienced, at several New York City hot spots including the Limelight. He designs and installs systems, leases the equipment to clubs, and is responsible for hiring the engineers to maintain and operate the systems. He is currently installing a system at The Warehouse also in New York City.
HOUSE ENGINEER: Bryan Christiansen
DESIGN CONSULTANT: Chris Tso, Manny's Music, New York.
INSTALLATION CREDITS: Max's Kansas City, Great Gildersleeves, Irving Plaza, The Big Kahuna
RECENT MARQUEE ACTS: Deborah Harry, Circus of Power, London Choirboys, Dred Zeppelin, Joe Jackson, Black Crowes, Blues Travelers, Paul Young.
DESIGN NOTES: "When I'm setting up a room I use pink noise and tone burst to flatten out the room as much as possible. Then I try to notch it out carefully. I check out a system weekly, and I also try to check out the system after every big show. I put a grille over the EQs so that no one except for truly qualified engineers can make adjustments," says Bondy. "Placing the board in a sweet spot is not always an easy trick in a club, but it's important. When I need to determine whether to hang or stack speakers I use a simple rule of thumb: for short-throw, hang 'em; for long-throw, stack 'em."
CONSOLE: Soundcraft Series 500 with 36-input modification.
HOUSE ELECTRONICS RACK: Yamaha 2031 stereo EQ, dbx 166 stereo compressor/limiter, Furman 524B stereo 4-way active crossover with bandpass limiter; Submaster patches: three Rane ME15 15-band stereo EQs, three Rane DC24 stereo compressor/limiters, two Gatex 4-channel gates (8-channel gating).
EFFECTS RACK: Eventide 910 Harmonizer, Eventide FL201 Flanger, Yamaha Rev 7 digital reverb, Yamaha SPX90 digital reverb, Yamaha R1000 digital reverb, two Roland SDE3000 digital delays, DeltaLab 1024 digital delay, DeltaLab 4096 digital delay, Valley People compressor/limiter.
MONITORING ELECTRONICS: Yamaha 2408 monitor board, six Rane GE27 graphic EQs, three Furman 324B stereo 2-way electronic crossover with bandpass limiters, two Crown PSA2 monitor amps, four Crown DC300A monitor amps.
MONITORS: Four 2-way Stage Wedges (JBL proprietary box with JBL 2225 15-inch and 2445 2-inch drivers custom made by Bondy), two Cross Stages (each with 2 2225 15-inch and a 2441 2-inch drivers on a JBL 2395 stand plate custom made by Bondy), Drum Box (with 2 2225 15-inch and 2441 2-inch drivers and a 2390 horn), Midlantic Hardware, Hubble Twist Locks, all monitors mounted on JBL 2311 Throats.
MAIN SPEAKERS: Eight custom-made Bondy Trapezoid-style proprietary boxes modeled after JBL Concert Series each with 2 2225 15-inch and one 2441 2-inch driver and 2381A horns, 4 W-bins custom-made by Bondy each with Gauss 8840 18-inch drivers.
POWER AMPS: Five Crown Microtech 1200LX power amps operated on 4 ohm loads.
MICROPHONES: Eight Shure SM58s for vocals, four Shure SM 57s for guitars and snare, six Sennheiser MD421s for tom toms, two Electro-Voice PL20s for kick drums, three Audio-Technica ATM-33Rs for high hats, two AKG D12, six IMP 2 passive direct boxes.
RACKS: Rack within a rack design with foam insulation and Allen key coffin locks.
ACCESSORIES: Atlas bottom mic stands, Beyer top mic stands, Latin Percussion drum claws, Belden cable, Switchcraft plugs, Hubble connectors, Carol speaker cable, Whirlwind 36-channel snake with splitter, Furman

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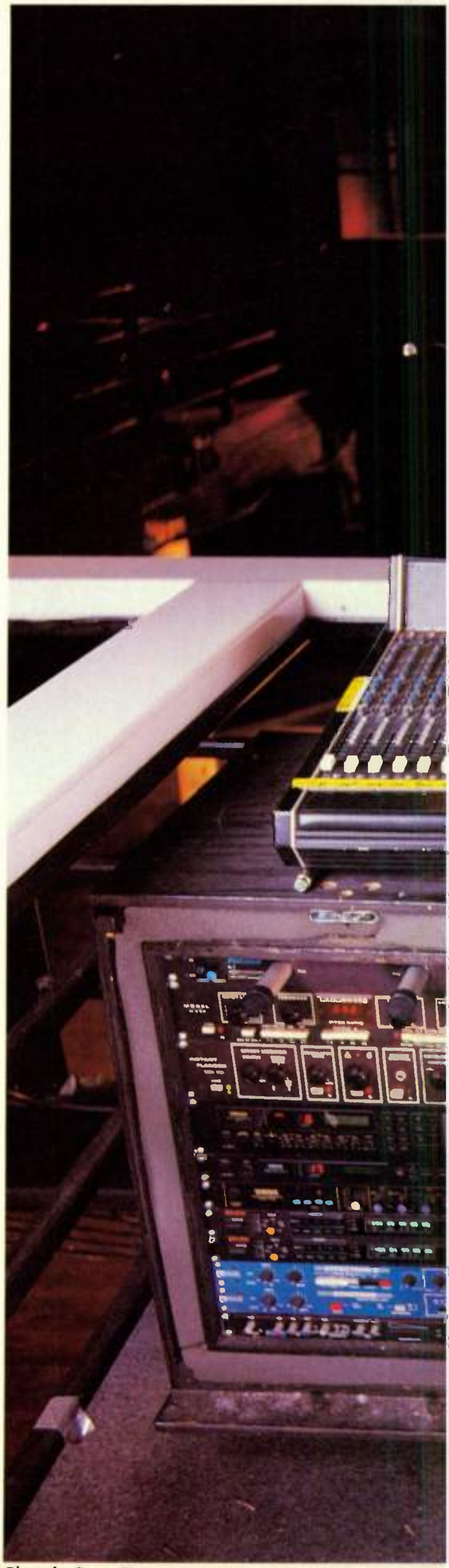
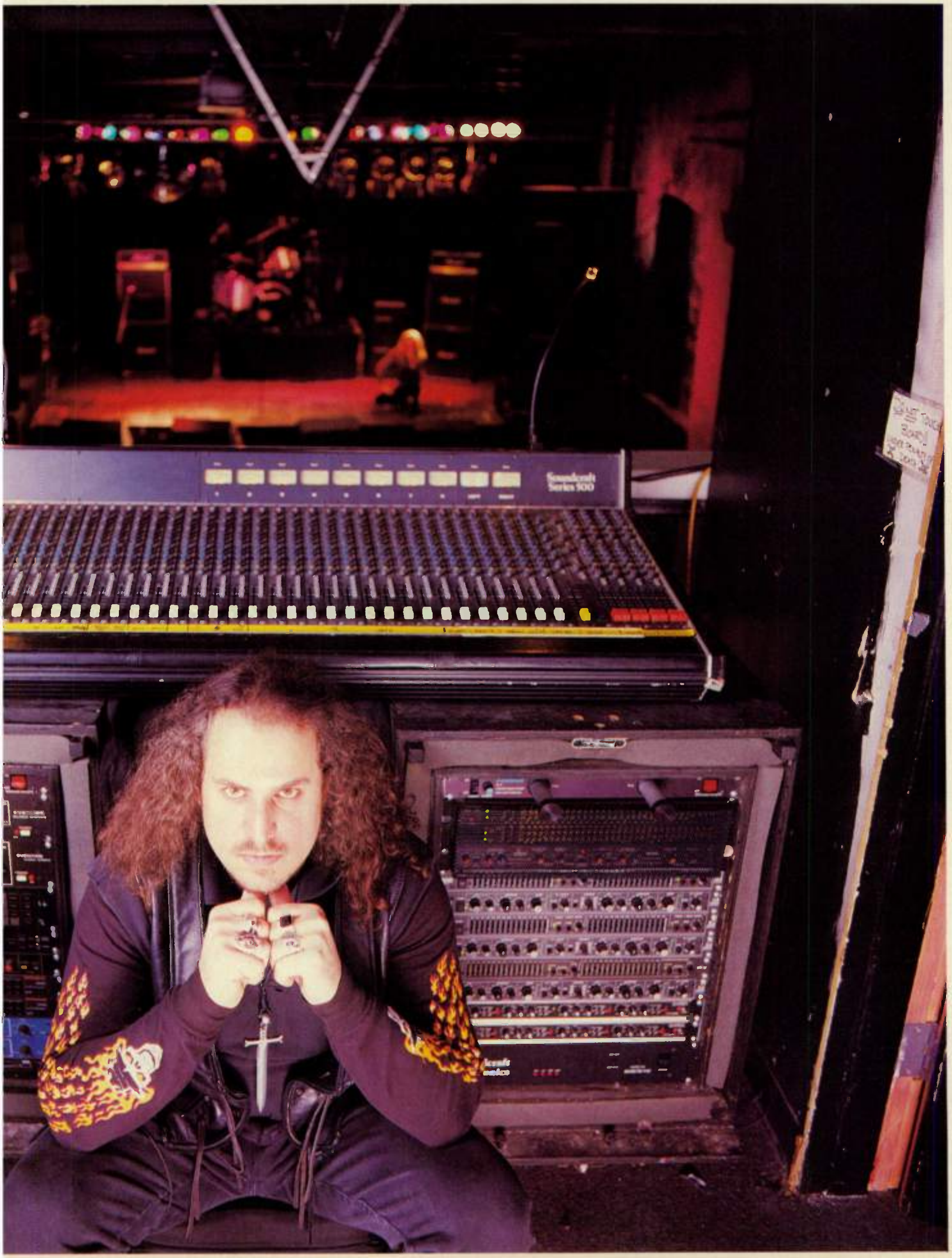


Photo by Peter Monroe


Soundcraft
Series 500

The most powerful microcomposer ca.1761

The most powerful microcomposer ca.1991

You see that guy up there with the funny looking clothes on? That's Wolfgang Amadeus Mozart. And while he was a pretty remarkable little composer in his day, we've got a pretty remarkable little composer ourselves. Namely, the Roland MC-50 MicroComposer.

This remarkable new dedicated hardware sequencer has eight Phrase Tracks, each of which can record data from any of 16 MIDI channels, to play back a total of 128 different parts. Mozart, bless his soul, could only play one part at a time.

Nor did our diminutive friend have a 3.5″ floppy disk for storage, an advanced editing system with microscope editing, an intelligent tape synchronization function, a Super-MRP Performance system for chaining songs together for live performance, an operating system in internal ROM so there's no boot-up time, and 40,000 notes in Internal memory.

But, there was one area in which Mozart shined. As a gifted composer, he could just sit down at the piano, take out his quill pen and immediately hammer out timeless pieces of music. And, he could do it all himself.

As remarkable as the Roland MC-50 is, it does need someone to bring out its full potential.

Which is where you come in.

Roland®

Roland Corporation US, 7200 Dominion Circle, Los Angeles, CA 90040-3696

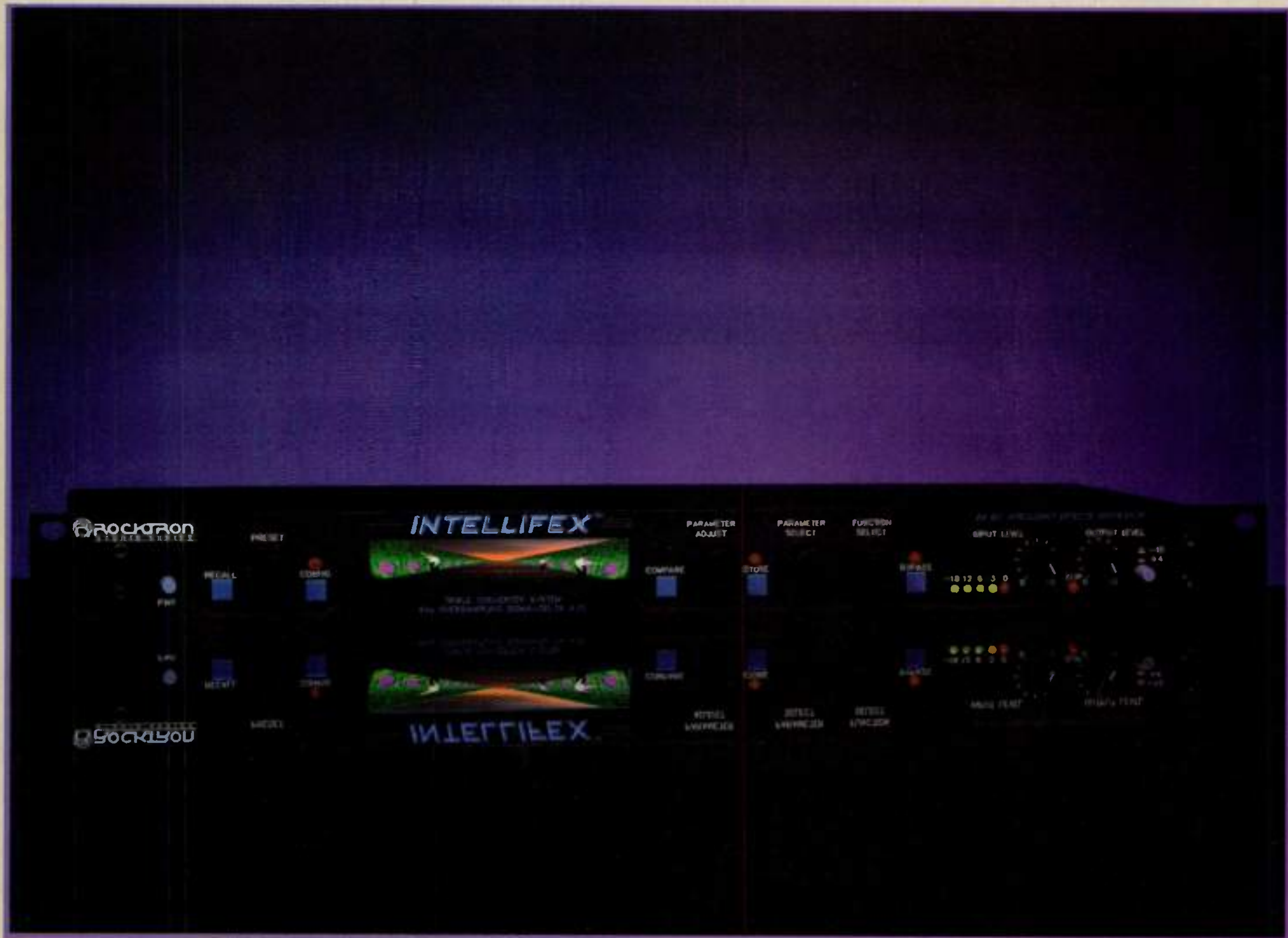
CIRCLE 25 ON FREE INFO CARD

Room To Choose

STUDIO: Reflex Productions, Calabassas, CA.
OWNER/DESIGNER: Greg Edward.
CREDITS: Produced for Corey Hart, Jefferson Airplane, Hunters and Collectors, and Adam Schmitt. Engineered for John Cougar Mellencamp and R.E.M. Music production for the feature films "Footloose," "Streets of Fire," and "Masters of the Universe."
DESIGN NOTES: Studio is isolated on a concrete foundation, has a basic 2x6 construction configuration, and triple pane glass.
CONSOLES: Yamaha DMP7, Yamaha DMP11, Hill MultiMix.
TAPE MACHINES: Fostex E-16, Fostex D20 DAT recorder with Timecode, Studer A721 cassette deck.
MONITORS: JBL Control 10s.
OUTBOARD GEAR: Yamaha SPX90 II, API 560 graphic equalizers, Yamaha SPX50D guitar effects processor, Rane parametric equalizer.
KEYBOARDS/SYNTHESIZERS: Korg M1, Roland D-50, Roland Alpha 2, E-mu Proteus XR, Roland D-10, Casio CV1, Yamaha DX7, Linn 9000 drum machine, Alesis HR-16 and HR-16B.
MICROPHONES: Schoeps CMC5, Neumann TLM175, AKG C-12A, Shure SM 57.
MIDI: Custom 386 computer system with Twelve Tone Systems "Cake Walk" sequencing software, Turtle Beach Soundstage software and hard disk recording system, Alesis sequencer, Fostex 4050 SMPTE-to-MIDI synchronizer, Yamaha MSS1 SMPTE-to-MIDI converter.
OTHER EQUIPMENT: Fairlight CBI video system, two Fostex 4030 synchronizers, JVC CR-8250 video deck, Sony XBR television monitors.
EQUIPMENT NOTES: "I've put in only equipment that fulfills my objectives, and that allows for excellent communication between my 16-track analog setup and my MIDI. It's very fluid. For example, the Fostex E-16 and the D20 let me bounce back and forth between analog and DAT; I can download tracks easily if I have to work on them elsewhere. And while both machines are convenient, they allow me to move tracks around and still maintain the highest quality sound. The (Yamaha) DMP7 is another great work tool — especially when I'm doing a final mixdown — because of its fantastic automation capabilities."

Photo courtesy of Fostex

WHO NEEDS ANOTHER DUMB PROCESSOR?



At Rocktron, we think the current trend to stuff as many effects into a box as possible at the expense of tone and sound quality is, well, sort of dumb.

That's why we've designed our new **Intellifex**™ effects processor with a decidedly "old" idea in mind: that the purity of the original signal is as important as the purity of the effects that process it.

Intellifex is a true 24 bit intelligent DSP effects processor with a 64 times oversampled "high purity" Sigma-Delta A/D converter. And while most processors squeeze the entire signal through a single converter, Intellifex uses *three* true 16 bit data converters for maximum linearity and a 24 bit data path throughout the processor and memory circuits to maximize the dynamic range.

Due to this advanced signal path technology, the stunningly beautiful sounds of our unique eight tap chorusing, stereo delays, reverbs and pitch shifting are virtually noise free. And thanks to our new software programmed *digital* HUSH™ circuitry, this absolute sonic integrity will not be compromised by typically noisy input signals.

So if you're looking for an effects processor that will satisfy your need for higher intelligence, audition Intellifex at your local Rocktron dealer.

Who said brains can't be beautiful?

ROCKTRON
STUDIO SERIES

Rocktron Corporation 2870 Technology Drive, Rochester Hills, MI 48309 (313) 853-3055

CIRCLE 24 ON FREE INFO CARD

Really Re- Re- Reverb

I usually hate reviewing equipment because it costs me so much money. If I review something that I like then I have to have one of my own. Techno-lust is not curable. This time, however, I am ahead of the game. I have owned a Roland R-880 reverb for about two years now, and I use it on every album I mix.

The Roland R-880 is a digital reverb featuring 16 bit A/D, 18 bit D/A and 28 bit internal signal processors. The unit has greater than 90dB dynamic range and less than .015 percent total harmonic distortion. The Roland R-880 can be used for reverb, non-linear effects, digital delay, digital domain EQ, chorusing and digital domain compression. Coaxial and optical digital I/O allows digital domain processing between two DAT machines at either 48 kHz or 44.1 kHz.

The control center for the Roland R-880 is the GC-8 Graphic Controller. This remote plugs into the main box with a cable that looks like a MIDI cable with extra pins in the connector. Multiple 880s can be cascaded together and controlled by a single remote. The controller uses a back-lit LCD display with five soft knobs under the display for communication with the user. To the right of the display are the numeric input keys as well as a row of dedicated keys for data input. There is a slot in the controller for program uploading to the reverb as well as RAM cards for storing your favorite reverb settings.

The only thing negative I could possibly come up with about the controller is the fact that there is a barely audible 400Hz tone that comes from the DC-DC converter that supplies the power to the light. Roland told me that they would look into providing a switch to turn off the display light for those people who are bothered by the whistle. I have had my unit for two years, and there has only been one occasion when the whistle was noticeable.

SOUND OFF

Now down to the real nitty-gritty of a reverb review. What does it sound like, you ask? Well, I'm sorry, I don't have enough room left this month, more about that later. Just kidding....

The first time I heard the unit was at a demonstration at a studio in Nashville. I was very impressed with the unit and asked if I could use it on the album that I was currently mixing. Roland loaned me the unit for four days and I put it through its paces.

The most important part of a reverb's sound to me is its transparency. I really need a reverb that is able to provide natural ambience for acoustic instruments. If you can get a reverb that will accomplish that task, then the heavily processed sounds will be fairly straight ahead.

I went through all of the reverb settings that came with the unit, and found a few that I liked, but I knew that the only way I would be really happy would be to program some reverb settings of my own. Time to hit the instruction books.

THE DEEP

The Roland R-880 reverb lets you get as deep as you like into the programming. You can just change parameters available in existing reverb programs if something is close to what you want, or you can go all the way to the algorithm level and build an effect from the ground up. They give you a little work bench where you can lay out all of the component parts of the system and then interconnect them any way you want. How about an infinite feedback loop with no inputs or outputs? Not very useful, but you can do it if you want to.

With this system of building up reverbs, you can route a dry signal in one of the inputs and out of the outputs with no processing. This signal can also be split off and mixed in with the processed signal from the other input at any point along its path.

So you want the left input to go through a chorus and then to the reverb section, but you would like the right input to bypass the chorus and go into the reverb, maybe with a little compression along the way? No sweat.

After learning the basics of programming the R-880, I sat down for

How I made 11 albums with the Roland R-880.

ROGER NICHOLS

Rear view: R-880. Roland has some free upgrades to the machine, including an improved user interface for the compressor and gate, an added internal memory protect function, improved muting and streamlined functions and menus.

about a week and entered my favorite reverb sounds. Some of them were easily obtained by changing the parameters supplied for existing reverb settings while others were constructed from the ground up.

The detail with which you can construct a process is amazing. You can select the size and shape of the room, the number of early reflections as well as the time and amplitude of each individual reflection.

You can have two different choruses going at the same time, each with its own depth and rate. The built in digital domain parametric EQ allows you to tailor the effect to your every need.

A nice feature in the reverb program is the ability to have a "sub-reverb," that is, a second reverb that is fed from the same input. This second reverb can have its own pre-delay, level and decay time. This works well if you want to have more than one reverb sound on a vocal, for instance, without taking up more than one reverb send on your console.

I really need a reverb that is able to provide natural ambience for acoustic instruments.

All of these features plus the fact that the Roland R-880 has optical and S/PDIF digital inputs makes it quite a remarkable machine.

I have given copies of all of the reverbs that I programmed to Roland and asked them to give the programs away with the machine.

I think the Roland R-880 will do well for you. After using it on eleven album projects it has done very well for me.

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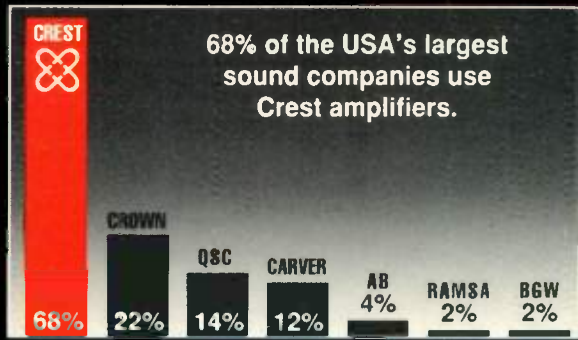

Totals are over 100%; some respondants mentioned more than one brand.

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CIRCLE 10 ON FREE INFO CARD

EQ — The Circuit

With so many MIDI sound generators now featuring multiple outputs, it seems like there are never enough mixer inputs. One solution is a line mixer, so you can submix multiple instrument outs and send them to your main mixer. Unfortunately, line mixers usually have minimal EQ, which definitely limits your tonal options — but not if you build the MIDI Studio EQ described here.

Build-it-yourself lab sessions with our West Coast Editor. This issue: building the MIDI Studio EQ.

CRAIG ANDERTON

For about $25 in parts cost per channel, this deceptively simple circuit can handle most of your EQ needs with a lot of sonic finesse. Features include a boost/cut control, relatively low-Q bandpass response for a "musical" tone, low noise, and your choice of any one of three low shelf (LS), six bandpass (BP), or three high shelf (HS) responses.

There are several ways to integrate this MIDI Studio EQ into your recording rig. Perhaps the most useful is to build as many as will fit in a rack enclosure, mount the input and output jacks on the front panel, and patch these into your mixer's insert points (or between an instrument output and mixer input) as needed. But the MIDI Studio EQ requires only two controls, so if you install a miniature rotary switch in it, you can substitute it for the bass/treble controls found in low-cost mixers. You can even build it inside a synth if you want more tone control.

Check out the schematic and follow along. S1, a two-pole, 12-position rotary switch, selects different capacitors (C1-C17) as appropriate to tailor IC1's feedback loop for the desired type of filter response. Table 1 shows the response type and frequency for each switch position.

The circuit itself is a classic shelving EQ circuit. Combining high and low shelf response provides the bandpass response.

IC1 can be virtually any internally-compensated op amp. However, I'd recommend using a dual op amp and building two EQs on one board. The

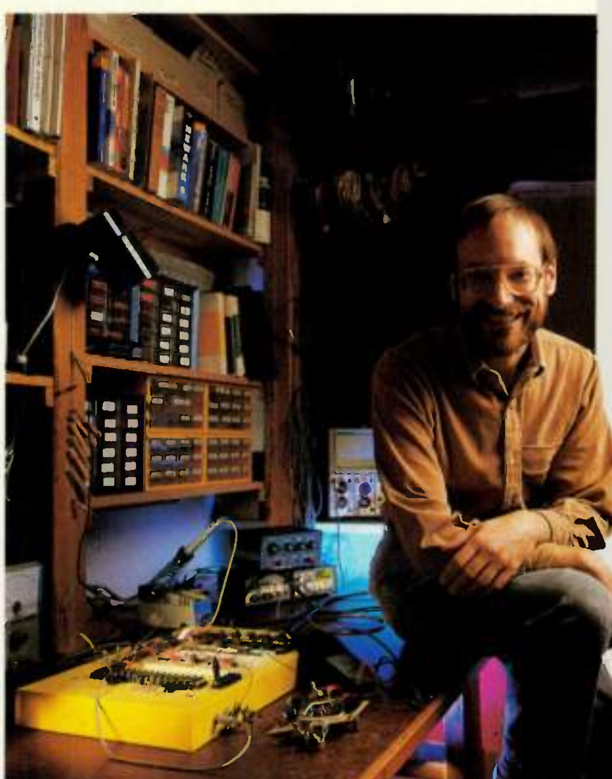
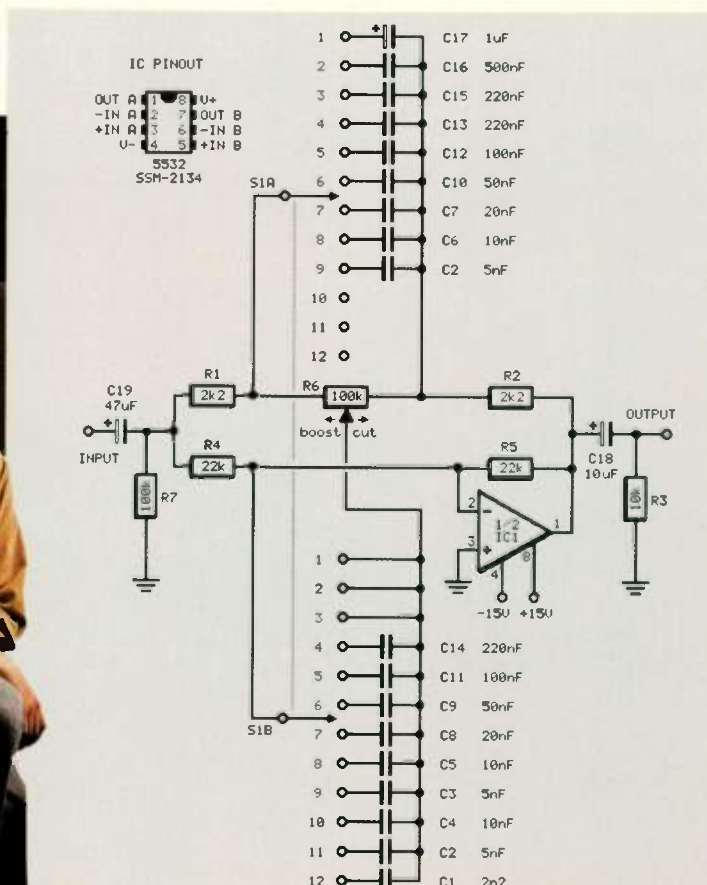


Photo by Ed Aiona



STUDIO EQ SCHEMATIC

bipolar NE5532 or SSM-2134 dual op amps (see schematic for pinouts) are perfect for this application. FET-input op amps like the TL072 also work well.

The schematic parts values are in international notation, used by virtually all countries except the U.S., because it tends to reduce artwork errors. The parts list gives values in both American and international notation. Just remember that 1 nanofarad (nF) = 1000 pF=0.001 μF and you'll be fine.

SWITCH #	RESPONSE	FREQUENCY
1	LS	50 Hz
2	LS	100 Hz
3	LS	200 Hz
4	BP	100 Hz
5	BP	280 Hz
6	BP	530 Hz
7	BP	1 kHz
8	BP	2.3 kHz
9	BP	4 kHz
10	HS	1 kHz
11	HS	2.5 kHz
12	HS	5 kHz

TABLE ONE

FINDING PARTS

All parts are commonly available except for the rotary switch. However, Newark Electronics (Chicago, IL 60640 with distributors in most major metropolitan areas; tel. 312/784-5100) stocks several suitable switches such as the Oak 399421K, Oak 39947JC, Mepco/Centralab PSA205, etc. These all list for about $15 each.

NE5532s (and some nifty rack mount enclosures) are available from Sescom, 2100 Ward Drive, Henderson, NV 89015 (tel. 702/565-3400). For SSM parts, contact Precision Monolithics, PO Box 58020, Santa Clara, CA 95052-8020 (tel. 800/843-1515).

The capacitor pairs used for bandpass response (switch positions 4-9) should be matched. In any case, do not use ceramic capacitors for capacitors C1-C17 as their values vary greatly with temperature; use high-quality mylar or polystyrene types.

BUILDING BLOCKS

Wiring to the switch is the hardest part. I cheated and soldered one lead of each capacitor directly to S1's lugs, brought the other leads out in parallel with the switch shaft, then soldered these leads to a "ring" of bare wire that connected to the main EQ circuit board. Keep connections between S1 and the board as short as possible.

The power supply can range from ±9V to ±18V. When driving several op amps, solder a 100 nF (0.1 μF) bypass capacitor from each op amp's V+ and V- connection to ground.

LIVING WITH IT

The MIDI Studio EQ has a relatively low input impedance and will load down high-impedance outputs, although most synthesizers and signal processors can drive this circuit satisfactorily. If there are any unforeseen problems (and in this life there often are), just patch a preamp, compressor, unity gain buffer, or the like in between the signal source and the EQ. By the way, it should also work admirably with insert jacks, or wired between the mic preamp output and the subsequent stage.

Good things do sometimes come in small packages. Don't let the simplicity of this EQ fool you. This circuit provides a lot of sonic power with little noise or expense.

Parts List

CAPACITORS

(18 or more working volts, mylar or polystyrene preferred except as noted)

C1	2n2 (2200 pF)
C2, C3	5n (5000 pF)
C4-C6	10n (0.01μF)
C7, C8	20n (0.02μF)
C9, C10	50n (0.05μF)
C11, C12	100n (0.1μF)
C13-C15	220n (0.22μF)
C16	500n (0.5μF)
C17	1μF (tantalum or mylar)
C18	10μF (tantalum or electrolytic)
C19	47μF (tantalum or electrolytic)

RESISTORS

(5% tolerance, metal film preferred for fixed resistors)

R1, R2	2k2 (2.2k)
R3 10k	
R4, R5	22k
R6	100k linear taper pot
R7	100k

OTHER PARTS

IC1	NE5532 or equivalent (see text)
S1	2-pole, 12-position rotary switch
Misc.	IC socket, perfboard, wire, solder, knob, power supply, etc.

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- Mirror-image (left & right) pairs
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- Switch-selectable response modes (equalized & reference)
- Acoustic foam blanket reduces baffle reflections
- Impedance 8 ohms

PRM™ 312A

- Mirror-image (left & right) pairs
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- Acoustic foam blanket reduces baffle reflections

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CIRCLE 23 ON FREE INFO CARD

Peavey Electronics Corporation

Setting The Stage

For a band to perform at its very best, it needs a simple stage accommodation — it needs to hear what it is playing. As a result, successful stage monitoring often makes the difference between a great and mediocre live performance. When the monitors are set just right, everything is in the right perspective. When they're not, your band can't hear the right things at the right levels, and to make matters worse, they can't hear all the vocals.

Sorry, turning things up doesn't help. So what can you do? Make stage monitoring a sound system priority, that's what. You also have to evaluate the type of equipment available. And you need to ask the band members what they prefer. Then you explain a few simple facts:

Monitors are there to help the band hear so it can sing on-key and jam in harmony. The monitors are not there to let them know what the audience is hearing, nor what the mixing engineer is doing with the sound over the PA (if they can't trust their sound engineer, who can they trust?).

HARD FACTS

Equipment makes the whole thing go round. Monitors, microphones, equalizers, power amplifiers all play their respective roles in proper stage monitoring. But first things first. Consider this steadfast stage monitoring priority:

In order to be heard, monitor speakers, no matter how good, require enough amplifier power to overcome the stage volume (at the musician's ears) by at least 6 dB.

Fortunately, with a little basic mathematics you can figure this out for yourself. Pull out the calculator and follow along carefully.

Stage monitors are usually rated by the simple formula: dB SPL x one watt x one meter distance. If the distance from the monitors to the performer doubles to two meters, and if you're maintaining constant power of only one watt per channel, your SPL drops by 6 dB.

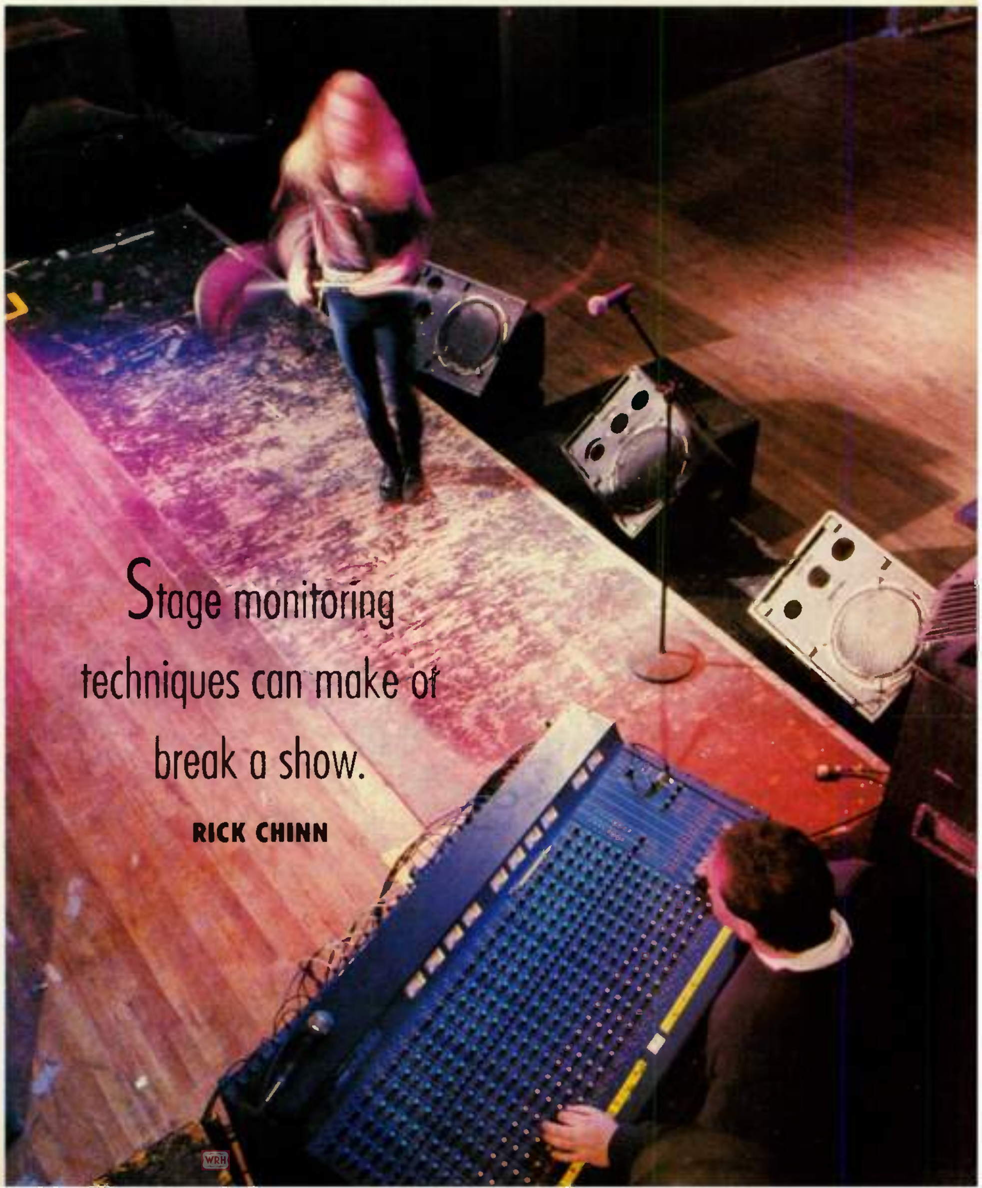
What if you increase the power? Now another series of formulas hold true: Increasing your power to 50 watts increases the SPL by 17 dB, 100 watts increases it by 20 dB, and 200 watts increases it by 23 dB.

Thus, if your stage monitor specs say it delivers 95 dB at one watt and one meter, at two meters and 100 watts input power, it will deliver 109 dB SPL.

HARD WEAR

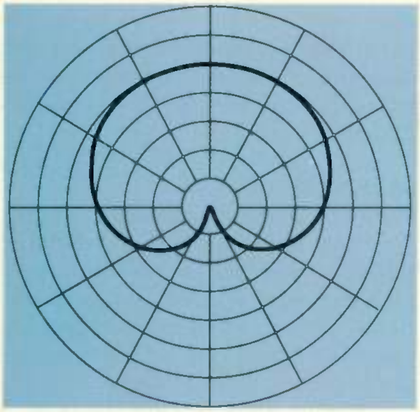
Once you've done your basic figuring, you have a general idea of how loud things have to be. The next step is to pick the right monitor types: traditional wedges, small frontal speakers, powered speakers, or even headphones (more on this later).

Wedge monitors have improved a great deal in the last few years. Horns are the norm for efficiency reasons, and constant coverage horns provide unchanging timbre as the performer moves off-axis. For side-fills, smaller frontal speakers like the E-V SH1502 or the Peavey 115H work well if you

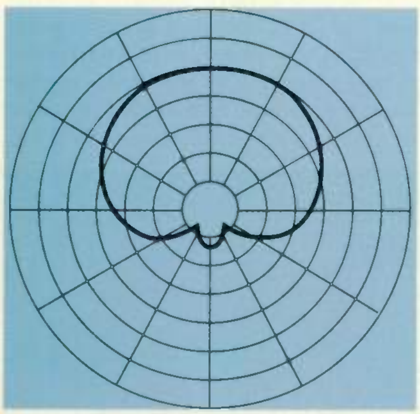


Stage monitoring techniques can make of break a show.

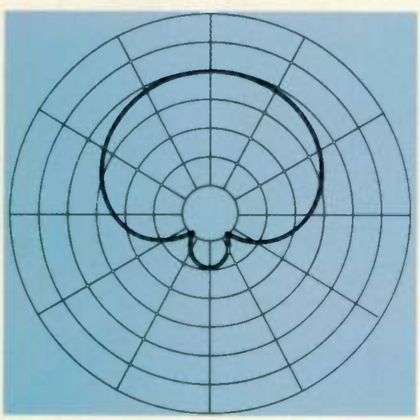
RICK CHINN



CARDIOID PICKUP PATTERN



SUPER-CARDIOID PICKUP PATTERN



HYPER-CARDIOID PICKUP PATTERN

Above: The pickup pattern of on-stage microphones should dictate how you set up the monitors (or vice-versa). Note that cardioid mics will work with a rear-placed speaker, while the super-cardioid and hyper-cardioid varieties are more suited to side-firing placement, helping the artist experience a stereophonic effect when listening. Right: Peavey wedge monitor, model HDH-M. Opening Page (35): Photo by Peter Monroe.

put the horn at ear level. And don't overlook personal monitors like the Galaxy Hot-Spot or powered speakers from TOA or Roland; you can put them closer to the performer's ears, which is the same as increasing the output capability. Powered speakers can operate from a mixer output, or they can bridge an existing monitor speaker without loading. Their lower output also causes less monitor leakage into the PA mics.

Then consider the next hardware variable in the stage monitoring equation: power amps. Remember that solid state power amplifiers deliver more power into lower impedance loads. So maximize the performance from your amplifiers by loading at their minimum load impedance.

If this were a perfect world, a 100 watt/8 ohm amp would deliver 200 watts/4 ohms and 400 watts/2 ohms. But this is not (sadly) a perfect world; most 100 watt/8 ohm amplifiers deliver around 150 watts into a 4 ohm load and most will not drive 2 ohm loads at all. Be sure to check the amp's spec sheet before using it, since loading an amplifier below its minimum load impedance is like begging for a repair bill.

Lavaliers, hanging choir mics, or performers that won't stay on top of their mics are leading contenders for monitor disasters.

WHAT'S GOING ON?

If you're having monitor troubles, or if your band can't hear their monitors, then it's time to ask some questions:

Q. Is the coverage angle of the monitor right for your application?

A. The monitor's coverage angle (dispersion) determines how it disperses its sound output. Monitors with horns tend to be more directional than other types. If your performers move around a lot, you have three choices: make do with less-than-perfect monitors, use side-fill monitors, or use a floor monitor with wide dispersion. If you use a model with a narrower dispersion angle, the performer can't move around.

Q. Have you taken mic pickup patterns into account?

A. You're supposed to put the monitor behind the mic, right? Maybe. This is true for a cardioid microphone, but if your mic isn't a cardioid, then behind isn't quite right.

The diagrams to the left tell the story. They are polar patterns, showing how well a mic picks up sounds originating at various angles relative to its front (0 degrees is the front and 180 degrees is the rear). Notice that the pickup at 180 degrees drops dramatically for the cardioid pattern, which is why you put the monitor behind the mic.

However, many of today's popular vocal microphones have super-cardioid or hyper-cardioid patterns. For super-cardioids, the maximum rejection occurs at 120 degrees and for hyper-cardioids the maximum rejection occurs at 135 degrees. Remember that there are two maximum rejection points, and that there is pickup at the rear due to the rear pickup lobe. These mics work better where the performer wants two monitors spread on each side of the mic stand.

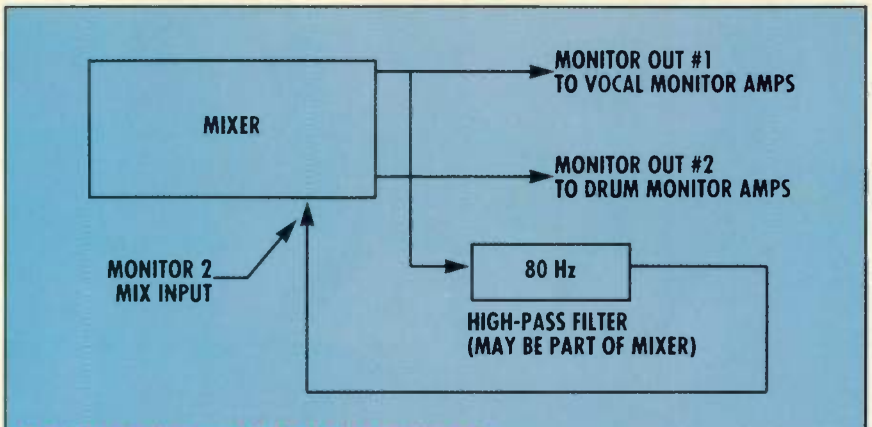
Q. Is the problem the source-to-mic distance or mic technique?

A. One of the primary rules of good stage monitoring is getting the microphone as close to the source as possible. This means that lavalier mics, hanging choir mics, or performers that won't stay on top of their mic are top contenders for monitor disasters.

A novel solution to keeping the microphone close to the source is a boom (head-worn) microphone. Coupled with a wireless transmitter, this may be the ideal solution for someone who wants to perform live, yet still have mobility and the use of their hands. Check out Madonna if you have any doubts.

Mic technique is important; when singers take a bow, they should turn off their mic or point it away from the monitor (the resulting feedback may temporarily deafen you or it may even blow up the monitor).

By the way, when you only have one monitor mix, all vocal microphones should be the same make and model. Different mics have different frequency responses, which causes problems when you try to find one



Equalizing vocals separately for the drum monitor mix.

monitor EQ setting that works for them all.

Q. Are you dealing with bad or inadequate equipment?
A. Bad equipment and good monitors aren't synonymous. Consider: Guitar speakers in a wedge-shaped cabinet don't make good monitors. Adding a horn and a crossover may or may not help, depending on the crossover frequency. Insufficient amp power means that your monitors simply won't be able to keep up.

Mics with inadequate polar patterns cause feedback because they lose their ability to reject off-axis sounds at higher frequencies.

Q. Are you using equalization correctly?
A. For most rock shows, octave-band equalizers are sufficient. They're easy to set by ear, and they do enough to make the difference — usually.

One-third octave equalizers divide the spectrum into smaller pieces for more precise work. With experience, you can learn to set them by ear, but a real-time analyzer helps you get the right setting much faster and more reliably. You don't need to use pink noise with the analyzer; instead, use the analyzer to tell you what frequency band is causing feedback and reduce the setting of that slider.

If you equalize monitors by turning them up until they ring and fiddling with your equalizer to find the offending band, stop when you can hear two or more frequencies ringing at once. Going further will only make the monitors sound worse. You can use a little boost to compensate for frequencies you can't hear. If this causes feedback, then you can turn down the overall level or reduce the amount of boost.

If you can, eliminate the low frequencies below around 70-80 Hz from the monitors. This saves power, and stops boominess.

Q. Is your stage volume simply too loud?
A. The band needs to be loud enough to hear themselves, and perhaps loud enough to carry the room (especially if they don't run their instruments through the PA). Your monitors have to work against this loudness to be heard.

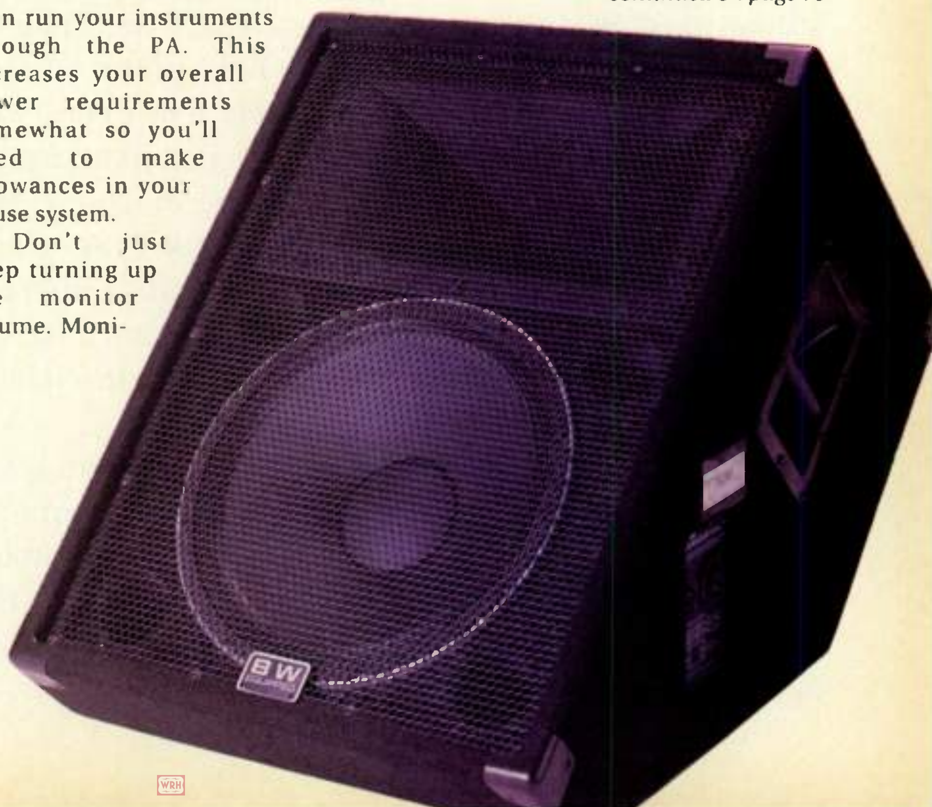
Always try to minimize your stage volume. Use small instrument amps and turn them up just enough so that the performers can hear themselves — and no more. Point guitar amps at the guitarists' ears if necessary. If you need more SPL in the audience area, then run your instruments through the PA. This increases your overall power requirements somewhat so you'll need to make allowances in your house system.

Don't just keep turning up the monitor volume. Monitor signals usually bounce off the walls before your audience hears them, which puts them out of time with your stage sound and your mains. This adds up to musical mush, with the vocals sounding boomy and boxy. In clubs, minimize monitor backwash into the audience by slightly notching the response of the mains around 250 Hz. If you can do this for vocals only, so much the better.

Q. Have you set up a good monitoring system for the drummer?
A. Often, it's the drummer who complains he can't hear his monitors. Consider: The drummer sits inside his instrument. Drums are loud. What's more, they may require a different monitor mix than the rest of the band because of stage placement. And, of course, singing drummers have to combat severe leakage into their vocal mic (and you thought Phil Collins had it easy?).

The straight-ahead solution is a big, bi-amped side-fill type of speaker that can be aimed at the ears. However, this often results in vocal leakage into the mics, and may make things too loud out front. What's more, if you high-pass the mix for the vocal monitors, this isn't going to work for a drum monitor. For one, if you add kick drum to the drum monitor mix,

continued on page 70



I DO IT MY WAY

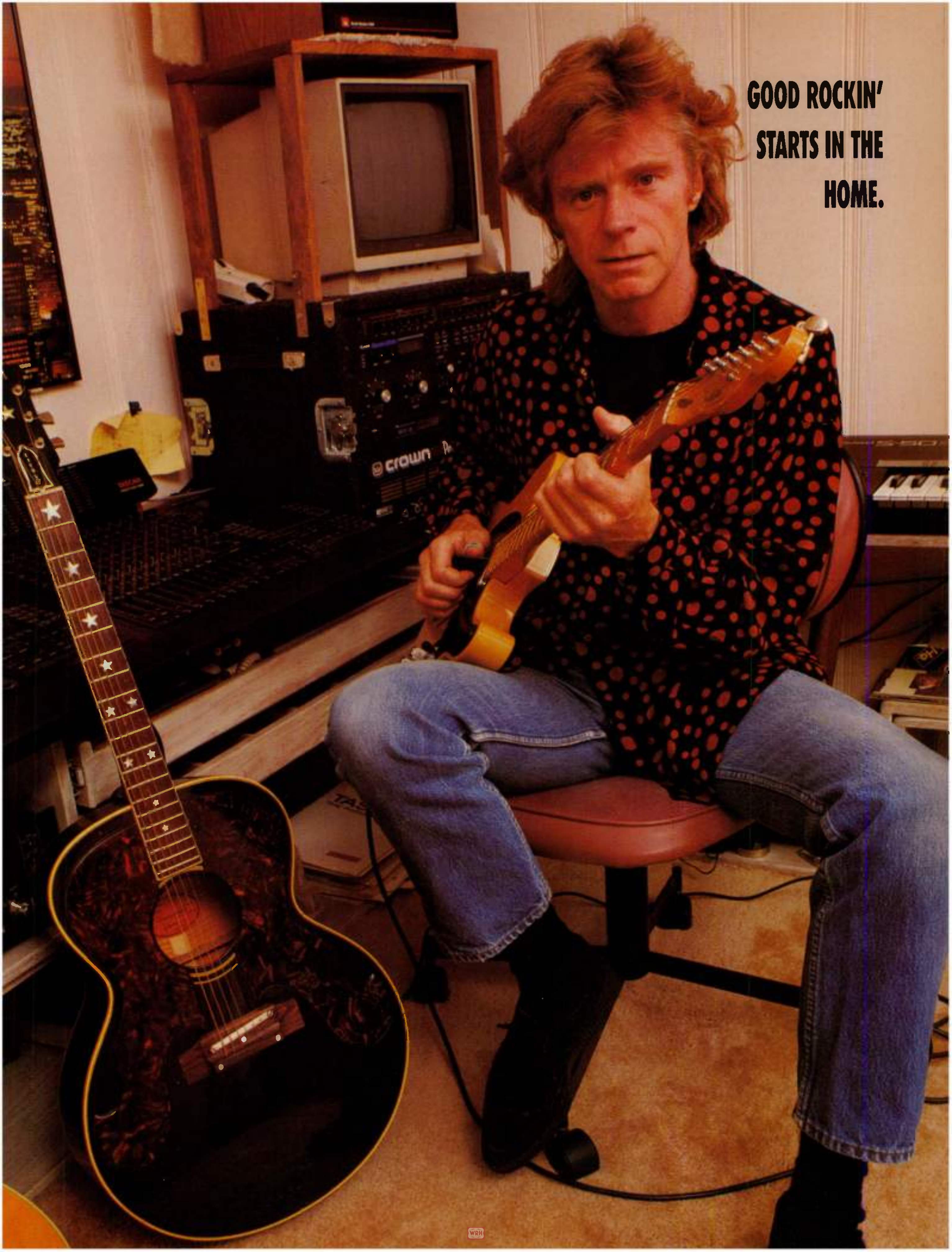
BY DAVE EDMUNDS

The first record I made was in 1968. Four-track was the prevailing technology then. We went in at 9:00 in the morning, and by noon we'd cut and mixed an A side and a B. I didn't realize you could go back and play the song a second time! Maybe things would have taken longer if I had.

Anyway, the record, "Sabre Dance" actually got to Number Two in England. Then in 1970, I did "I Hear You Knocking" on an eight-track and it sold three million copies. I've always retained a certain fondness for the studio tricks we used in those days. There was virtually no outboard gear. All we had were EMT plates, a compressor and maybe some external EQ. For delay, we'd wedge a cigarette packet between the record and playback head on an EMI TR-90 mono tape machine. This would increase the length of tape between the record and playback heads. What made the sound so interesting was that it was constantly changing, because the distance between the heads would vary as the packet started to crumble.

I always keep experiences like this in mind when I'm working in my project studio — just to maintain a sense of perspective. I'm actually a newcomer to personal recording. Although I've always played all the instruments on most of my records, I always avoided setting up my own studio — mainly because of the cost.

Until now that is...



GOOD ROCKIN' STARTS IN THE HOME.

I TEND TO GET VERY IMPATIENT IN STUDIOS. THE MORE SOPHISTICATED THEY'VE GOTTEN OVER THE YEARS, THE LONGER IT TAKES TO DO THINGS; THAT DRIVES ME CRAZY.

GRANDEUR FOR 15 GRAND

For about $15,000 I've been able to get just about everything I want in a personal studio. My setup is based around two Tascam 688 MidiStudios. I bought one and liked it so much I had to get a second. The 688 is perfect for me because it's an eight-track cassette recorder and MIDI controller mixing board all in one. All patching is digital, which means the end of conventional patch bays. And I'm all in favor of that.

I tend to get very impatient in studios. The more sophisticated they've gotten over the years, the longer it takes to do things; that drives me crazy. I'm a rock and roller. I don't plan everything out meticulously in advance. I get an idea and think, "I want to do this now." And with the 688 and a studio the likes of which I have here, it's just that instant.

My two 688s are linked via a Tascam MIDlizer. Drum sounds are provided by a Roland R-8 drum machine that I trigger with a Kat controller and some custom pedals. Keyboard sounds are provided by a bunch of instruments, including a Roland S-50 sampler, E-mu Proformance and an old Korg CX3. The MIDI gear is driven by an Alesis MMT-8 sequencer, and all lashed together via a Digital Music Corp MX-8 MIDI patch bay. I generally record vocals on an AKG C-12 microphone, through an old UREI 1176 tube limiter. That's my pride and joy. It makes vocals sound so good I generally don't even have to EQ them. I've got some other good mics and outboard processors too, plus a Tascam DAT machine that I mix down to ... I've got everything I could want.

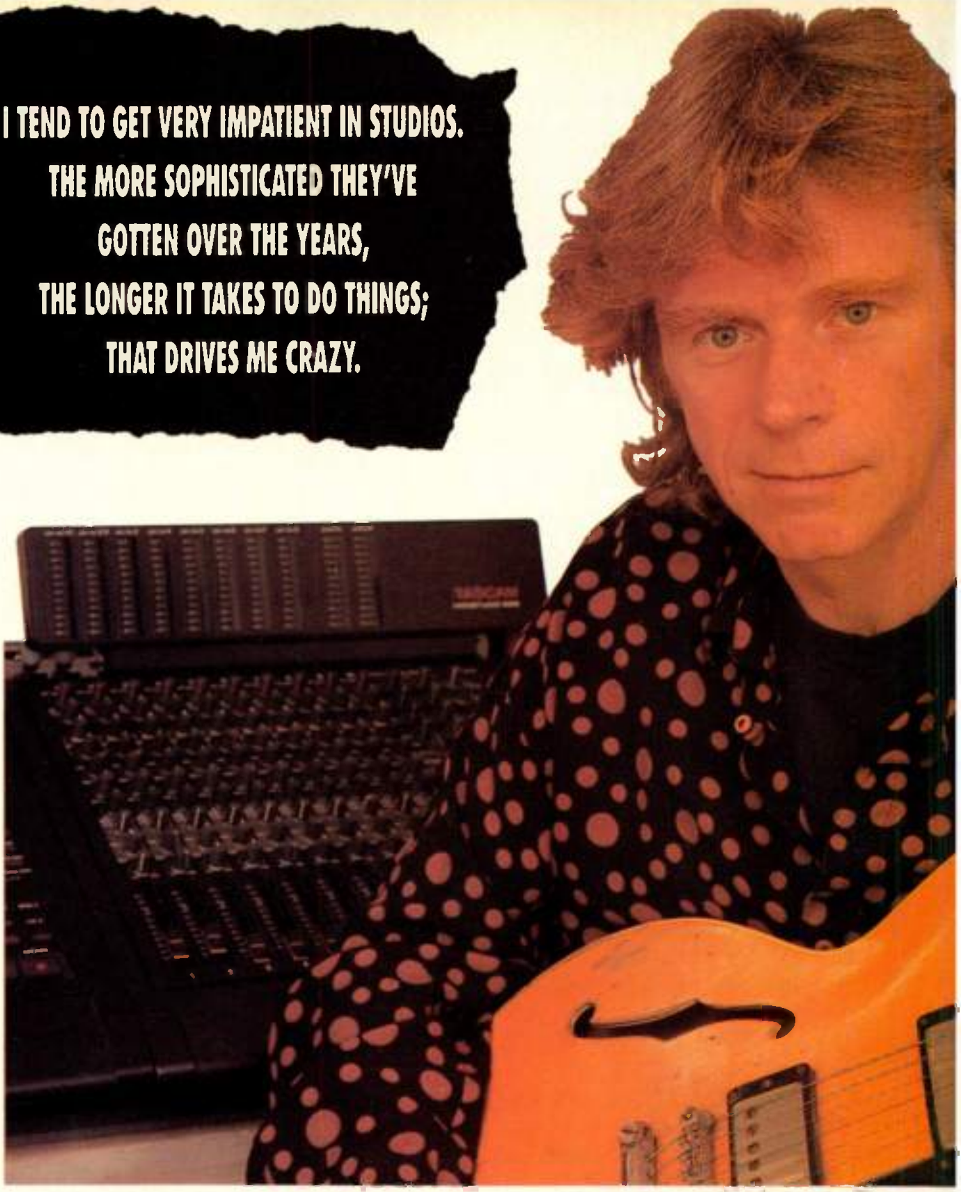
With the 688, I can store a separate patch in memory for recording every microphone and every keyboard I've got. I keep everything plugged in all the time. So I can say, "Okay, what next? Bass? Ah, go to Scene 14." And Scene 14 will have bass routed to the appropriate tape channel. I even made up a wall chart to help keep track of which instrument is in which scene. The 688's "Dual" mixer section gives me enough inputs for my MIDI gear. It's like having a separate keyboard mixer, independent of the tape machine mixer. In all, then, I've got 16 virtual tracks and 14 tape tracks (eight on each 688, minus one track per machine for sync code).

DOING THE DRUMS

I find the best way to start recording a song is just to get something going on the drum machine. I gradually put in the drum fills after I've got some other instruments on and I know what I want. So I'll program a selection of drum fills and then just pop in whichever ones I want wherever I want them. What doesn't work is doing a complete drum part first, with fills and everything. Because what you imagine will be a good drum track — and what may sound great all by itself — usually turns out to be horrible once the other instruments are on.

Back when I started recording on eight-track, we'd put the whole drum kit on one or two tracks. Even on 16-track, we'd just balance the whole kit out in stereo and then take the snare and bass out and put them on tracks three and four. I used to love that way of doing it. Today it's not unusual to use up 12 tracks on the drums. But I can't help thinking that's a tremendous waste. It can cause problems too, because all you're doing is putting off decisions about the drum sound until later on.

So in my personal studio, I can go back to the good old days. So far, what I use for my drum machine is just two channels on the Dual. Should I need to isolate a kick or snare, I can do that on the Dual as well. But the nice thing about modern drum machines like the R-8 is you can balance the entire kit right inside the machine.



Photos of Dave Edmunds by Ed Colver

GETTING GOOD GUITAR

I used to go for D.I. guitar sounds all the time, but now I find I don't like direct guitars any more. I got tired of struggling with them, basically. So now I've got this old, tweed '53 Fender Twin guitar amp in my project studio. I've tried it in a lot of different angles in the room, with various mics and various distances. And what I've come down to is the Shure SM 57 — that old favorite, great all-around mic.

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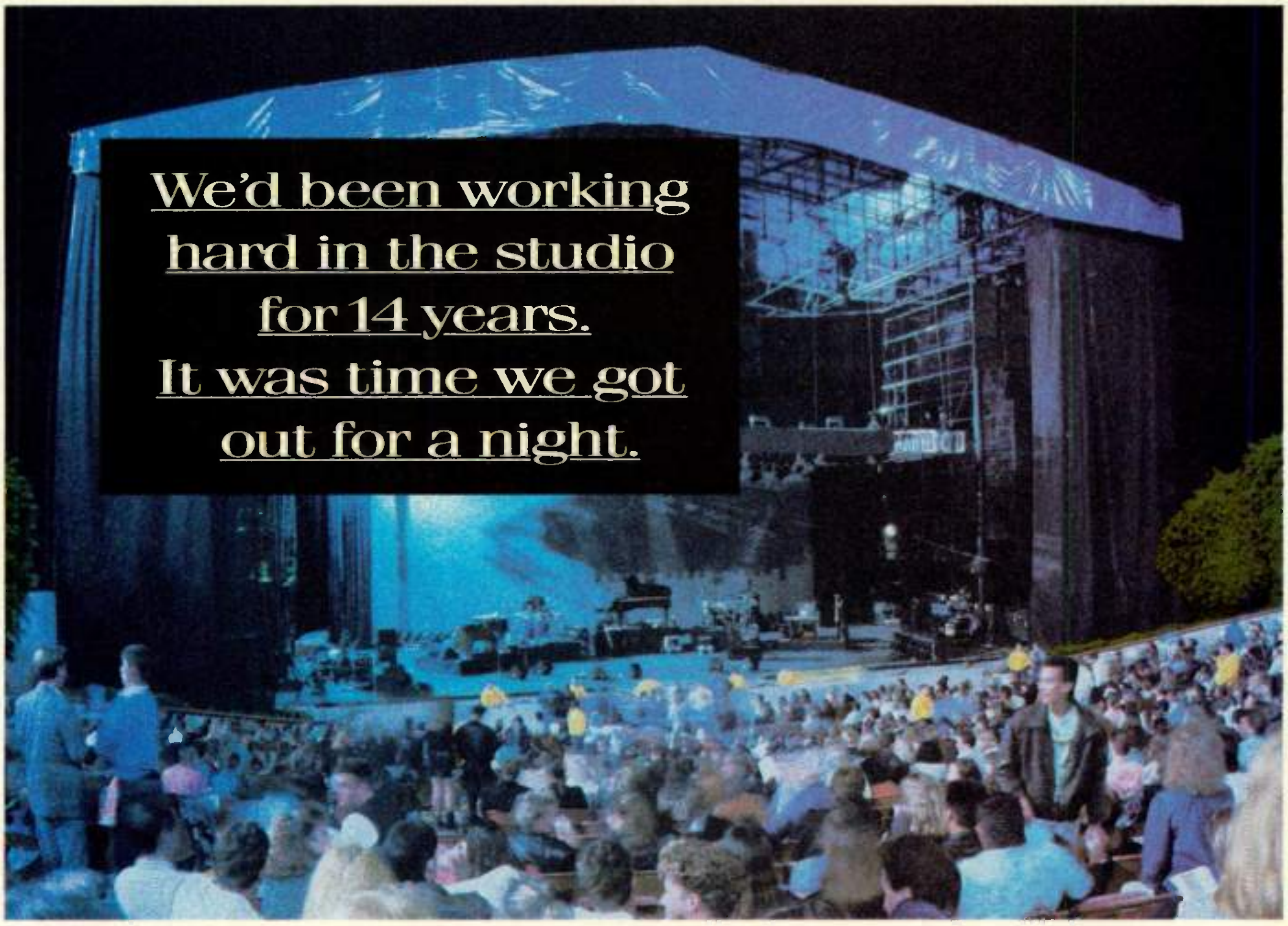


Spending years on end cooped up in small, dark rooms with a bunch of engineers takes certain special qualities. Durability, for one. We've always been known for that. Of course, clear, uncolored sound quality doesn't hurt, either. Or hand-assembled components, with gap precision to plus or minus one-millionth of an inch.

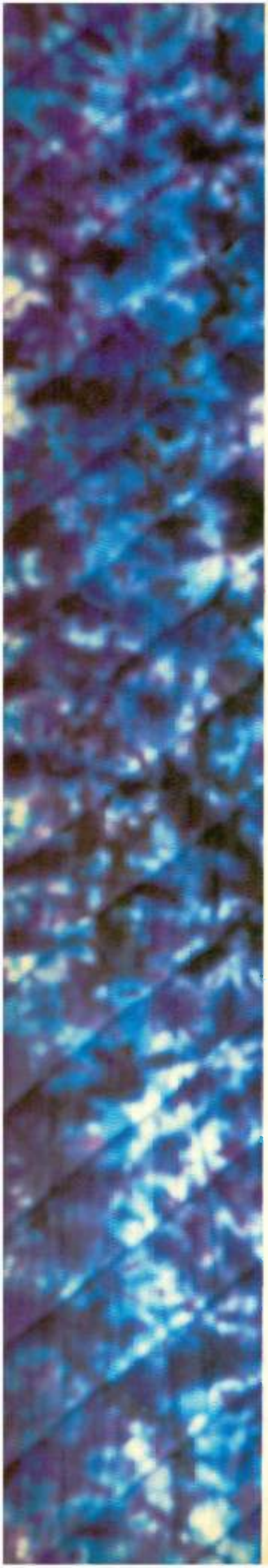
These features got TAD speakers into studios like Record Plant, NOMIS and Masterfonics. And the same features are now getting us out of them.

See, we had this funny idea that if TAD could make music sound terrific in a small room, we could make music sound terrific in a huge arena. And every outing we've had with Maryland Sound has proved us right.

Not that we won't still work our woofers off in studios from L.A. to London all day. But, at night, we'd like to get out and jam more often.

TAD Technical Audio Devices®

Pioneer Professional Products Division



RAISING THE DEAD

The Grateful Dead are always in a state of critical mass. That's been our life story.

When we were first starting out — putting up gigs in nightclubs and eventually graduating to larger and larger halls — we began to realize that not only had the venue doubled and tripled in size, but so had the difficulties of getting quality sound. Moving the music all around a stadium or theater and keeping control over its acoustic precision was a challenge that really got me thinking ...

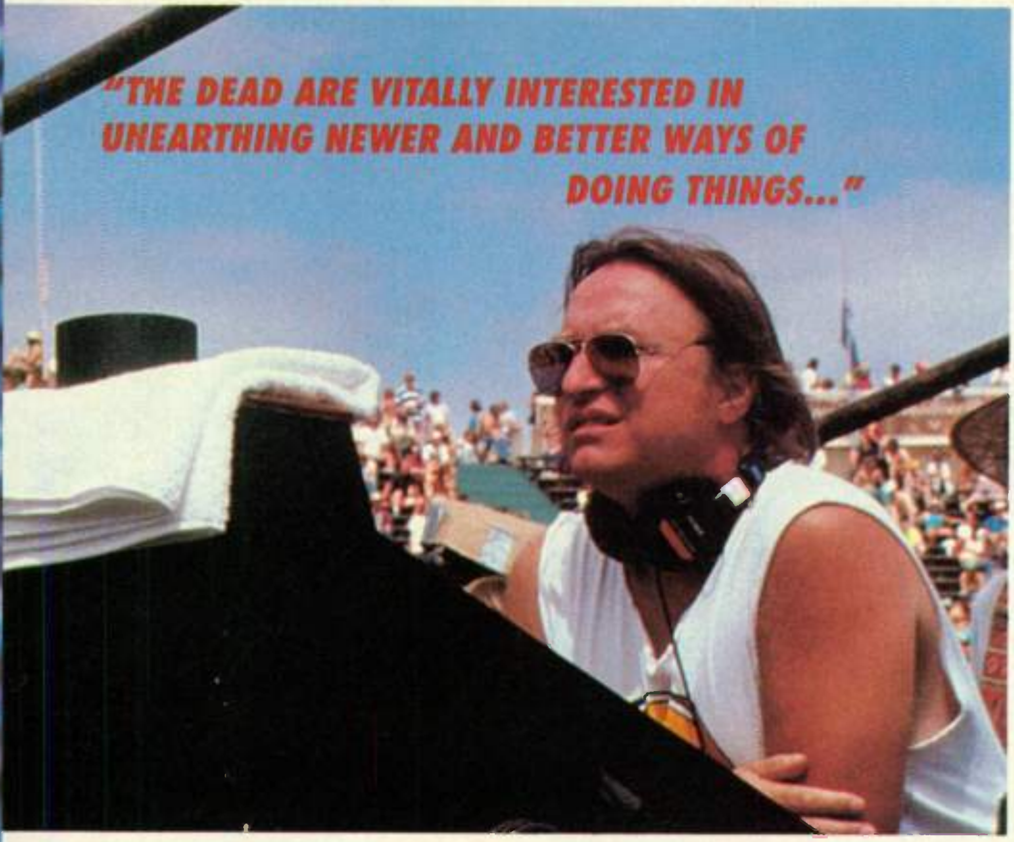
What is live sound in an ideal world? We started out with a paradigm, a dream model of what listening might be like in an infinitely large hall. You would experience the sonic power of an endless sound field, yet still feel the same intimacy and warmth you get from sitting in your living room listening to a killer stereo (or in a studio control room listening to monitors, for that matter). Though it seemed impossible, that was our goal from the beginning.

What the Dead have achieved in live sound quality has come from relentless pursuit of that goal. We never stopped believing that it's possible to create super high quality sound even in big venues, even way back in the days when no one had ever come vaguely close to it. Over the years, many, many people told us to forget it, but the more that people told us it was impossible, the more dogmatic about it we became.

When we began to assess the problem in the late 1960s — I mixed the first concert with the Dead in 1966 at the original Fillmore in San Francisco — we began to become aware of what it would really take to realize this dream.

BY DAN HEALY


SONY
GRATEFUL DEAD
'90 '91
COAST
new year's eve



It was around 1970, in the pursuit of eliminating distortion, that a major problem — intermodulation distortion — was discovered. This came up when you mixed all the instruments and the vocals together. Some test instruments of the time, Hewlett-Packard and Tektronics distortion analyzers, were used to store sound digitally, and then subtract it back out of the composite, to see how much one individual signal interacted on the other — which turned out to be immensely. These ideas were later developed in the Wall of Sound.

THE ALTEC A-7 SYSTEM

In those days we used a three-way system made of standardly-available Altec equipment. They were mainly A-7s that we began modifying, using other brands of loudspeakers, experimenting with tuning, redesigning the cabinets slightly and building it out of better materials. I got an A-7 where it could cough up three times as much sound as the stock item with much higher sound quality. Since all the equipment the band was using — instruments and guitar amps and not just the sound system — was really from the dark ages, we innovated hop-ups for the guitars and guitar amps. I was spending my time rewiring guitars and winding pickups.

We finally got to the point with the Altec system where we realized there wasn't anything we could do to improve it. Altecs used horns, which most systems do, but at the time there was very limited knowledge in the mechanics of horns. Today's horn speakers are far more accurate than their primitive forebearers. The "high tech" horn speakers of the 60s were primarily designed for more utilitarian purposes than music reproduction; they were meant for public address systems and foghorns. Their acoustic accuracy was more suited to an air-raid siren than it was to a speaker for a rock concert.

With horns, clean sound wasn't that a big consideration — loudness was everything. Trying to get decent quality out of horn systems tended to generate a phenomenal amount of distortion. It wasn't really possible with the tools that we had to go further in horn design, and I really didn't know that much about it. No one really did. It was stuff that had yet to be uncovered.

THE TIE-DYE SYSTEM

The rest of the 1960s and the first year or so of the 1970s were spent developing the Dead's first model sound system. It was called the Tie-dye system, because it had tie-dyed grille cloths on all the speaker cabinets.

The Tie-dye system was a forerunner of the Grateful Dead's "Wall of Sound." Its primary feature was that it had cone radiator speakers rather than horns for a smoother sound. It was a better polishing of a better wheel, but at the same time it offered me the opportunity to learn a lot about what was wrong with loudspeakers. Cones offered the lowest possible distortion and the smoothest transition between the lows, mids and highs. Their problem was that they couldn't project and get loud enough. At the time, we had JBL and some other manufacturers were working with us, designing speakers to our specs.

THE WALL OF SOUND

By the time 1971 rolled around, a small San Francisco sound company called Alembic began building the legendary Wall of Sound. Principal designers of the system were Ron Wickersham and John Curl, with conceptual guidance from Owsley Stanley.

The Wall of Sound was a study in absolute elimination of point source and inter-harmonic distortion. The main theme was to generate a separate sound system for each and every instrument and voice on stage so that no two things went through any one loudspeaker. By 1974, the Wall of Sound was up and running, and the Dead toured with it for two years. It was very painstaking to operate; very difficult and very costly. But the results yielded tremendous amounts of information about sound reinforcement that no one had yet uncovered in the audio field.

It was a model that was to answer a whole set of questions above and beyond the research anybody had done. It generated answers which were incorporated into Don Pearson's Ultra Sound system: the criteria for extremely low distortion, extreme signal-to-noise, extreme flexibility, and the ability to set it up and array it in different places.

It was a very extravagant attempt, and we actually pulled it off. Everything was stereo, all built into one large array, with hundreds of little cabinets. We also owned all our own scaffolding because the system couldn't support itself. It had to be custom fitted for each theater to give the proper dispersion of sound.

With the Wall of Sound, we unraveled the mysteries of intermodulation distortion. It was good for fairly large stadiums. In larger situations it worked, but it wouldn't really go loud enough. But the quality of it was extremely sweet. When it was clean and good, it was beyond your wildest dream.

Its imaging was actually perfect, because each bandmember's sound system was right behind him on the stage, so it maintained faithful imaging, but it lacked the ability to project. In 1976, when the whole American economy went out the window, the expense of moving the sound system increased astronomically, and we retired it. Now it's in rock and roll heaven. We gave most of it away, which amounted to hundreds of little speaker cabinets, so we outfitted probably half the beginner bands around the Bay Area.

OVER TIME

It was around that period that the Dead became really obsessed with the importance of time alignment. We were probably the first people in all of the audio world to start talking about it. In a way, we had the first time aligned system with the Wall of Sound. Time align, a trademark of E.M. Long Associates, was an amazing audio concept that Dead audiences quickly appreciated.

Later on, a fellow named Jeff Cook, from Sound Storm in Santa Barbara, who specialized in filters and networks, began to help us out with the time alignment problem. With the time delays between the different parts of the system, I could tune out all time weirdness and make it totally smooth. That was the beginning of recognizing that in the audio dispersion world, time is as valuable a commodity as amplitude.

Now we're using a system that evolved from the radical horn designs of John Meyer, a Meyer MSL-3 based system from Don Pearson and Ultra Sound. We're now working on a much more refined version of time alignment that can time align along the sides of a venue, not just out the front. I've got the stereo imaging out to 170 degrees now.

It doesn't stop there. Coming over the horizon is the implementation of fully digital audio in a live con-

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SPACE JAMMING WITH THE GRATEFUL DEAD

In a live Dead show, spontaneity is everything. We experiment in what we do, and we stumble onto new ideas. We don't even make set lists, we play what we play as we go.

The way it works is that the band supports me and has faith in my knowledge. I've pretty much got a free hand to do whatever I want with the sound. The idea is that we feed back and forth between the band and the mixer. The Dead are as vitally interested in unearthing newer and better ways of doing it as I am.

There are times when I intentionally go beyond the limits of pure sound with a particular effect in mind. I don't try and do the same thing every night. Every show is different. I tend to play it amorphically; whatever strikes me. I try not to get any song to fall into any one groove. It's all different. I approach it like the news every night.

I alter things like textures of echoes and delays, and effects on the voices and instruments. I try and create new and original feelings. For instance, Jerry will be singing a song and I'll get an idea — a flash — to try a new texture on his voice. So I'll put the texture on his voice and he'll hear it. I have a great rapport with the band so if they pick up on it they'll sing into it for me. That will enable me to take another step and then they'll take another step — even though I'm 125 feet away at outdoor shows I have as good a working rapport with them as if I were standing right beside them on the stage. Over the years we've all learned how to read each other.

The Space Jam

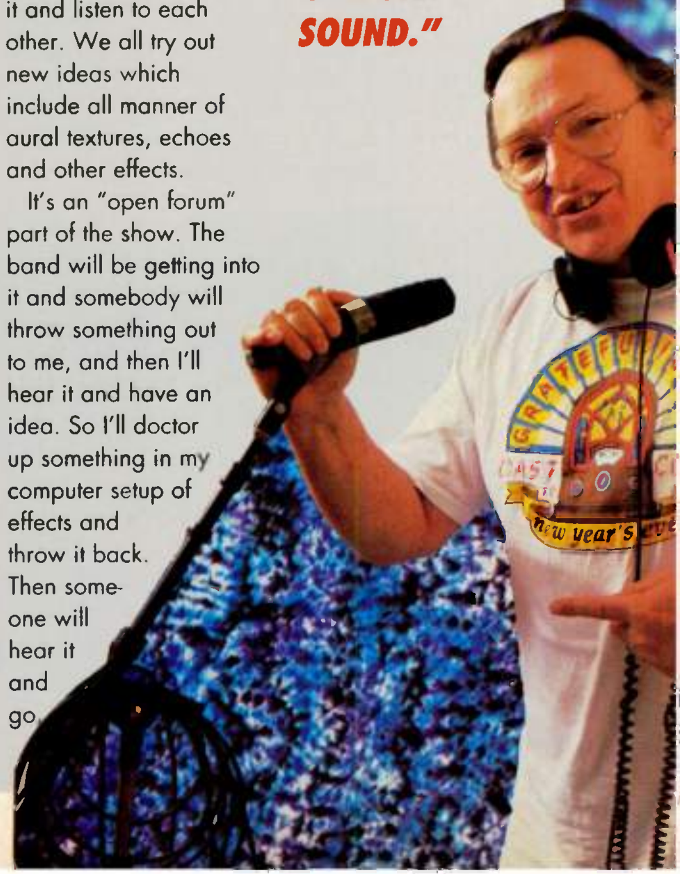
I personally view one part of the show, what we call the Rhythm Devils/Space Jam part of it, as the new music. That's an area of the show that's like a workshop, only we do it in front of the audience, with the audience. The idea is that we all just jump into it and listen to each other. We all try out new ideas which include all manner of aural textures, echoes and other effects.

It's an "open forum" part of the show. The band will be getting into it and somebody will throw something out to me, and then I'll hear it and have an idea. So I'll doctor up something in my computer setup of effects and throw it back. Then someone will hear it and go, "Wow, that's far out," and then they'll lay something else on it. It all happens in real time and none of it is ever talked about except maybe after the fact.

I'm trying to create something so that if you buy a ticket and come to the show, and you're enjoying the music, and I add in some incredibly desirable and fun sonic effects that go along with it, you go, "Hey wow! Listen to that. This sound's coming from inside me; this one's coming from over there."

The object really is to create more trips for everyone.

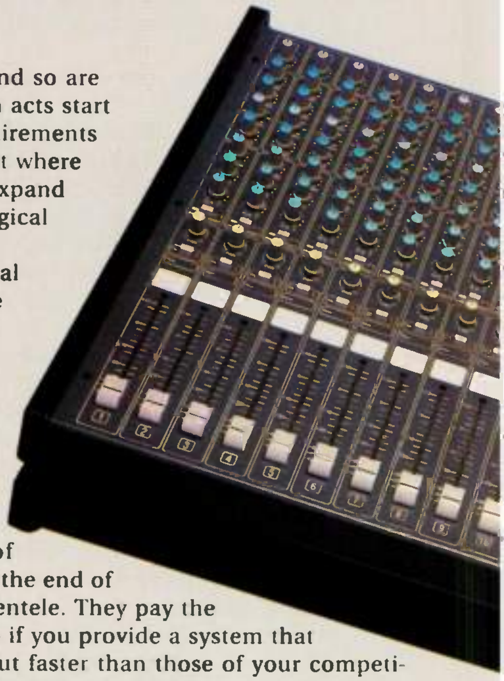
"GO BEYOND THE LIMITS OF PURE SOUND."



Photos by Susana Millman

SELLING

HOW TO START YOUR OWN LIVE SOUND COMPANY



So you put up shows. You're good, and so are your client bands. More out-of-town acts start calling you to fill in their sound requirements as they pass through. There comes a point where you feel limited in the amount you can expand locally, and touring becomes the next logical step.

Welcome to the world of professional live sound production. Now all you have to do is figure out how you're going to change your life and your electronic livestock to meet the growing requirements of your trade. And doing that right takes more than buying a truck, a few more speaker cabinets and an answering machine for the office.

This is the most important piece of advice: Remember who pays the check at the end of the day. Promoters will be your major clientele. They pay the local crews in each venue by the hour. So if you provide a system that loads in, sets up, tears down and loads out faster than those of your competitors, you just might move to the top of the promoter's list for the next tour.

How do you do it? Start by evaluating your equipment. If you've got a good base, build on it; if not, auction it all off and start from scratch.

LOUDSPEAKERS & MONITORS

In loudspeakers, you ideally want those smaller and more powerful systems that you can configure to suit almost any venue. These systems take up less room in the truck, go together faster, and cover the audience more evenly than many of their predecessors.

Products (from left to right): QSC EX series amps, E-V BK-1242 stereo mixer, Celestion SR Series PA; Yorkville Pulse PA (page 48); Peavey rack (page 50).

Listen to as many speaker systems as you can and talk to the techs that use them. Keep in mind that more efficient speakers take less space and power. Also, some systems are designed to stack and be focused securely or to fly from rigging points above the stage, as required. These are major selling points when discussing a prospective tour booking.

Of course, different kinds of music do require different kinds of speakers. High volume rock and roll usually needs a punchy, efficient system that doesn't get muddy at high sound pressure levels. Other high volume music acts, notably reggae and rap, may also require additional low frequency loudspeaker systems. More moderate volume acts often need a great deal of clarity, since the vocals have to

SOUNDS

WADE McGREGOR

be distinct even with a large orchestral arrangement backing them up. With vocal intelligibility a primary concern, loudspeaker arrays must not interact destructively, that is, cause an uneven distribution, especially of the mid-band frequencies, across the audience. Their dispersion must be well behaved off-axis so reflected sound doesn't adversely affect the direct sound from the loudspeakers.

These systems usually require an excellent signal-to noise ratio so that the dynamic range of the performance isn't masked by hum, buzz and hiss from the sound system. This can demand signal-to-noise ratios approaching 100 dB from the sound system.

The same can be said of the artist's monitor loudspeakers. The monitor loudspeakers' sound may leak into the open mics on stage as well as be reflected from stage walls and be audible to the audience.

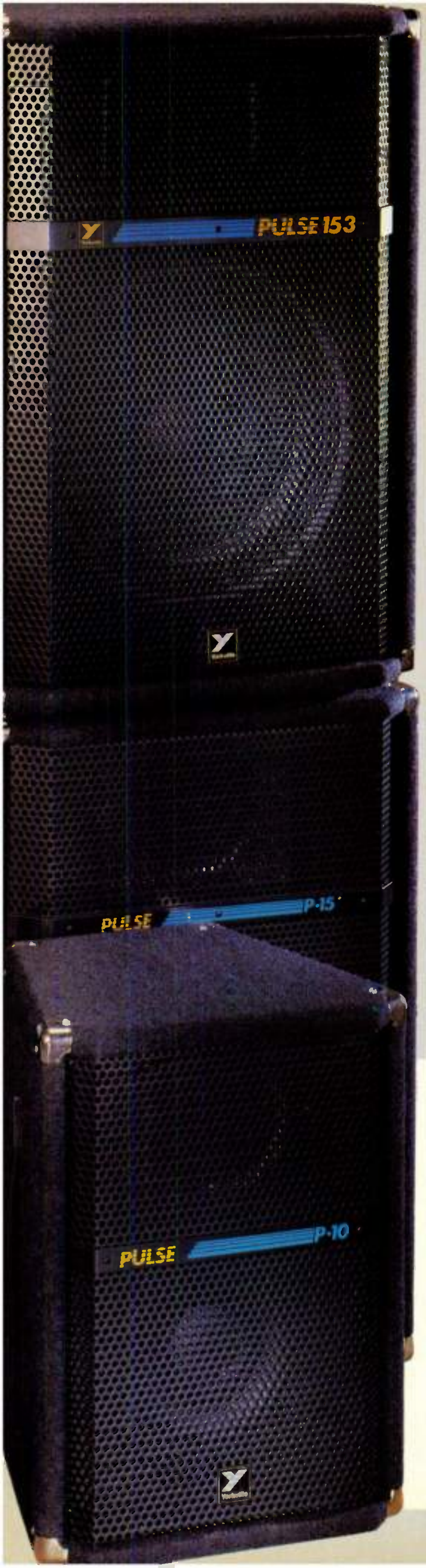
When a band is playing the bar circuit, they often find themselves on stages so small that they can hear almost every monitor wedge on the stage. This makes the use of more than two or three separate monitor mixes arguable, but when they begin to play bigger clubs, and soft seaters (auditoriums and concert halls) they find themselves spreading out across larger stages. Now they may need a separate monitor mixer with separate mixes for every member of the band and two more mixes for the side fill monitor.

Monitor loudspeakers differ in requirements from the FOH loudspeakers. The monitor loudspeaker can be custom built as long as it fulfills the needs of your client.

HOW MANY SYSTEMS?

How much to get? Determining the number of loudspeaker systems (cables, crossovers, amplifiers, sub-bass, bass, mid and high frequency components) required to cover a venue can be very complicated. There are a few rules that can help in creating a method for converting seat count and venue size into sound system requirements, but the style of music will also have to be factored into this calculation. Determine the efficiency of the loudspeaker system (dB SPL per watt) and you may, as an average, calculate a 3 dB increase in acoustic power output for every doubling of loudspeaker systems. If 10 loudspeakers produce 119 dB, then it will take 20 loudspeakers of the same type to produce 122 dB (all other things being equal). Horn-loaded loudspeaker systems couple together quite differently than front-loaded loudspeaker systems. Many systems use combinations of the two.

Test the system's limits at your first opportunity. You'll be the best judge of how much it takes to cover a specific venue. Find out how loud it can comfortably go in various combinations; it will pay off later.



POWER & AMPLIFIERS

Once you've picked a loudspeaker system, it's time to pick power amps that match their power and drive capabilities. Amplifiers vary in terms of minimum impedance they can drive at full power, input sensitivity, overload and fault protection, weight per watt, and rack space required. Consider these factors when building a touring system; each one will have an impact on the long-term cost.

Many of the loudspeaker systems available today have one form or another of processing included. These may take many forms but usually include some form of driver protection. Some of the more common active crossovers also include limiting. If you are going to build a system that doesn't, then consider including limiters in the power amp rack. This will allow the system to sustain some pretty awful faults upstream of the amp rack without causing driver failure. A system that has acoustic output throughout the show always sounds better than one that has had the drivers destroyed (by a phantom powered mic with a bad cable) during the first song of the night.

Power distribution for the system may be a new concept if you have been running systems that could be powered from a couple of 15 amp wall outlets. Larger sound systems require power distribution from three-phase 60 to 400 amp disconnects. Get acquainted with the electrical codes in your area and take a look at a number of 'distro boxes' in use by other sound companies. Learn from their mistakes as well as their good ideas. You might want to consider building up modular power distribution boxes so that you can configure systems to suit various sizes of venues and still be able to split your inventory between smaller jobs when it is required.

> You may find it hard to substitute that custom rosewood console for the Yamaha PM3000 they've asked for.

Also consider building the amplifier racks in modules that you feel will cover 300 to 500 people per rack. It makes it easier to send out for the right size of system without having to reconfigure the racks for each tour.

MIXERS, MICS & PROCESSORS

When a production company deals strictly with local clients, the right price may be all that is required. When doing business with people who judge the quality of your equipment on the basis of models already familiar to them, then it's not just the functionality that counts, it's the name.

You may find it hard to substitute that fabulous, custom-built, 45-channel, rosewood mixing console for the 40-channel Yamaha PM3000 they've asked for, even though it may have 6-band parametric EQ on every channel and 24 auxiliary sends. The band's technicians like to feel at home on mixing consoles and develop a familiar pattern on common mixers. This, of course, applies to the monitor mixing console as well. The mixers are probably second only to loudspeakers in selling your system to prospective clients.

A similar case can be made for

Play It Safe

Preventative PA Medicine

1. Rig It Right. Have your rigging professionally designed and use a rigger who is certified and insured.

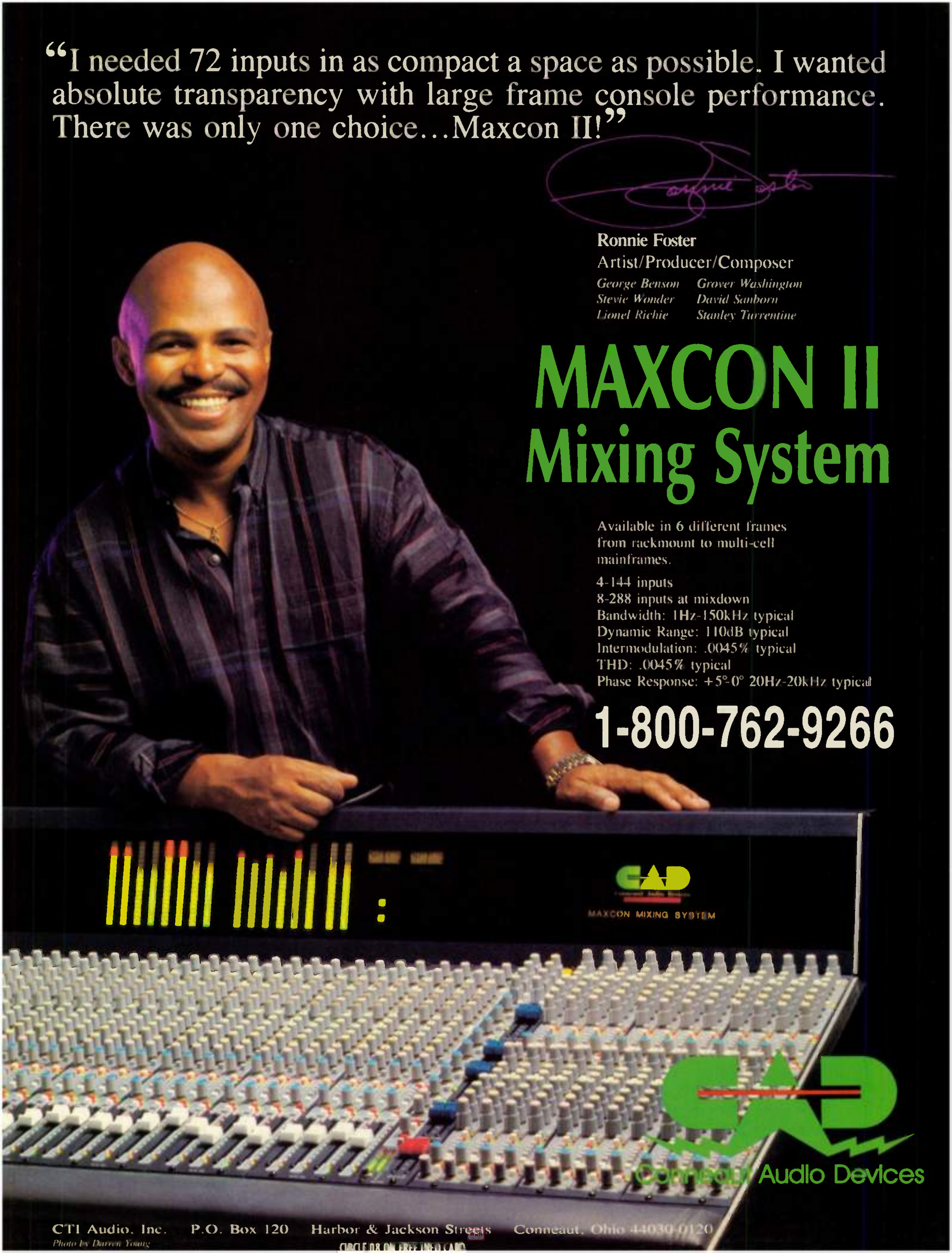
2. Power House. Carry an AC power distribution system by a licensed electrician and hire a licensed electrician to connect your AC system to the house AC.

3. Ground Out. Beware of old, two-wire guitar amps with "hum" switches. These can put 115 VAC between the guitar strings and a PA microphone. Modify, isolate, ground or otherwise neutralize these hazards.

4. Plug Your Ears. Use ear plugs when mixing and encourage the performers to use them too. Ear plugs work like sunglasses: At first you don't hear accurately but after a while your brain adjusts. Keep the SPL within reason, no matter what.

5. Safe & Secure. Avoid stage hazards by taping down stage cables and securing speaker stacks, including guitar amp speakers. Check out facility security and crowd control.

Thanks to Panasonic Pro Audio Systems for their help in preparing these safety tips.



"I needed 72 inputs in as compact a space as possible. I wanted absolute transparency with large frame console performance. There was only one choice...Maxcon II!"
Ronnie Foster
Artist/Producer/Composer
George Benson
Grover Washington
Stevie Wonder
David Sanborn
Lionel Richie
Stanley Turrentine
MAXCON II
Mixing System
Available in 6 different frames from rackmount to multi-cell mainframes.
4-144 inputs
8-288 inputs at mixdown
Bandwidth: 1Hz-150kHz typical
Dynamic Range: 110dB typical
Intermodulation: .0045% typical
THD: .0045% typical
Phase Response: +5°-0° 20Hz-20kHz typical
1-800-762-9266
CAD
MAXCON MIXING SYSTEM
CAD
Conneaut Audio Devices
CTI Audio, Inc. P.O. Box 120 Harbor & Jackson Streets Conneaut, Ohio 44030-0120
Photo by Darren Young

processing gear. Extremely useful effects units that give the ultimate death flange won't get much more than a sigh over the phone from prospective clients if they don't recognize the model. Research the requirements of the client you are planning to work with. If you can supply familiar equipment, half the work of selling them on your services will be done.

Take the same attitude with your mics. Reconsider your mic complement not in terms of what you think is the best sounding mic for the job, but rather what your clients will expect to have available. There will always be people who ask for the unusual, but the likelihood that any of your competitors can supply a Calrec Soundfield for a drum overhead is also pretty small.

TIPS AND TRICKS

Of course, when you're running a sound production company, no matter how small, there are a million things to keep on top of. Here are a few tips to get you started:

- Keep your system interconnections as simple as possible. Color code all the cables and their receptacles so that you can quickly instruct local crews on connections. Segment the inside of road cases so that cable types aren't hopelessly mixed up when you start the next setup. When you start to work in venues that require larger systems, setup will become a job of supervising the local crew and instructing them as briefly as possible, so that things progress efficiently. In these situations you will encounter crew members with a variety of skill levels, from people that have never touched a mic cable before, to those that have skills equal or superior to your own. Keep it simple!
- Hire good people. It's not easy loading all the equipment you own into a truck and handing the keys to someone you may not see again for four or five weeks. Your workers will have to manage the truck and sound equipment on their own as well as get along with the band's techs, the crew in each venue, the road manager and also return ALL the equipment intact. The requirements of your staff and their number will vary from one tour to the next. Some acts will tour with their own technicians for FOH (front of house) mixing and maybe even mixing monitors, while others will not. Your staff will have to be capable of any of these functions as well as being organized when supervising local crews, setting-up, troubleshooting and tearing-out the sound system.
- A related tip: Make sure there's a technician on stage to react to the unexpected changes

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Technical Rider Tips

The contract between the concert promoter and an artist will usually have an addendum attached called the technical rider, commonly known as the Tech Rider or simply the Rider. The Tech Rider contains the specifications for the staging, lighting, sound and other requirements needed to fulfill the contract and perform the show.

Compliance to the letter is required by many artists, while a few artists will use these as wish lists. Assume the former. The promoter has signed a contract stating that complete compliance with Tech Rider is acceptable and will be carried out. Any changes to the Tech Rider have to be approved by both parties. The artist's management will need to approve substitutions and the promoter will need to know if there are unforeseen costs.

The best Tech Riders are specific, mention typical models of equipment they expect and note models that cannot be substituted. They will also include a stage plot and a mixing console input list. They may also include monitor mix information and sound check or rehearsal times.

Where a Tech Rider is vague, try to read the real needs of the client into the specifications. Often the vague or conflicting specifications will leave many unanswered questions.

They may specify equipment by model number to give you an example of the quality of devices they require. Occasionally a Tech Rider will mention microphones or effects units that are hard to find. If you're lucky, the artist will provide them and simply have forgotten to strike them from the Tech Rider. Unfortunately, you cannot rely on it. Some Tech Riders will mention every brand of beer and mineral water as well as the pattern the deli plate is to be arranged in, but will cover sound requirements with a simple "professional quality sound system" required. It is best to try and get more detailed information.

Telephone calls are cheap compared with trying to find a 40-channel mixer at 4:00 in the afternoon, when the performers arrive at the venue and are panic stricken because a 16-channel mixer was all that you had set up. The phone call to the artist's tour manager, a few weeks before the first show, will really pay off in the long run. When this isn't possible, then strict compliance to the details of the Tech Rider will need less explanation on the day of the show than trying to excuse your substitution to an angry artist.

— *Wade McGregor*

NAMM-ARAMA

Not just anybody gets into the NAMM show. Rock endorsers, groupies and, oh yeah, music dealers. If you weren't there — we were there for you. The following pages list what's hot in sound and recording at your local pro audio shop. Use the EQ Free Lit. Numbers to get information right from the source.

— *The Editors*

LIT #	BRAND	MODEL	PRICE	PRODUCT TYPE
101	ADA	T100S	$950	guitar amp
102	AKG	C406	$229	mini gooseneck condenser mic
		C407	$159	mini lavalier condenser mic
		C580E	$189	gooseneck condenser mic
		V6HP	$429	headphone amp
103	Alesis	RA-100	$349	power amplifier
		3630	$299	compressor/limiter
104	Allen & Heath	Spectrum series	$8900-$14900	console
		Vision series	$1400-$2400	live mixer
		Scepter Monitor	$2800	matrix mixer
105	Alpha Audio	Azonic	$2-$5/sq.ft.	acoustical foam
106	ART	MDC 2001	$499	compressor/limiter/de-esser/expander
		X-11	$129	MIDI foot controller
107	Artefx	SB-2300	NA	headphone cue system controller
108	Artist Systems	3000	$849	loudspeaker
		4000	$1349	loudspeaker
109	Ashly	MM-508	$1499	stereo mixer
		MM-106	$599	microphone mixer
		LM-308	$499	line level mixer
110	Atari	MIDI Translator	$5500	—
111	Atlas	Mic Stands	NA	—
112	Audio Logic	AL266	$500	dual compressor/limiter/gate
		AL440	$400	quad noise gate
		X23	$400	stereo 2-way/mono 3-way crossover
		X34	$475	stereo 3-way/mono 4-way crossover
		X22	$425	XLR stereo 2-way precision crossover
		X32	$475	XLR stereo 3-way precision crossover
113	Audio-Technica	PRO 10HE	$125	mic
		PRO 25	$139	hypercardioid mic
		PRO 35R	$135	remote-power condenser mic
		ATM series	$208-$250	mics
		PRO 37R	$159	cardioid condenser instrument mic
114	Audix	D-1	$229	drum mic
		OM-7	$359	concert dynamic mic
		HRM-3	$799/pr.	studio monitor
115	Bag End	TA-15C	$896	loudspeaker
116	Beyer	TG-X 480	NA	wireless mic
		TG-X capsules	NA	retrofit capsules for SCM 186 wireless
117	Big Noise	Cadenza	$199.95	graphic sequencer
118	BGW	200	NA	power amp
		6500T	NA	power amp
		350 series	NA	power amp
119	Biamp	Newport	$2599-$2999	4-submaster portable mixer
		Olympia	$3500-$4800	4-submaster mixing console
		Columbia	$5999-$10999	8-submaster mixing console
120	BBE	651	NA	3-channel preamp
121	BSS Audio	EPC 780	$3600	power amp
		DPR-404	$1399	4-channel compressor/de-esser
		DPR-901	$1429	equalizer
		TCS-803	$1695	time corrector
		TCS-804	$2995	time corrector

LIT #	BRAND	MODEL	PRICE	PRODUCT TYPE
122	CAD	Equitek II	NA	condenser mic
		Maxcon II	NA	mixing consoler
123	Carver	PT-2400/PT-1800	NA	amps
124	Carvin	FET1000	$679	power amp
125	Celestion	Celestion 5	$450/pr.	studio monitor
		SR series	$199-$399 ea.	PA system
126	Coda Music	Finale PC 2.0	$249	music notation software
127	Community	RS series	$636-$2825	loudspeakers
128	Cool Shoes	Sound Globs 3.0	$199	composition program for PC
129	Crest Audio	FA series	$738-$1674	power amps
		LA series	$630-$1134	power amps
130	Crown	Macro-Reference	$3500	amplifier
		CM-30	$215	condenser mic
		LM-300	$247	miniature gooseneck mic
131	dbx	363X	$269	dual noise gate
132	Denon	DM-4000	$2000	dual transport CD player
133	Digidesign	Studio D	NA	music production system
134	Digital Designs	DD6-S	$336/pr.	studio monitor
		DD261-08	$740/pr.	oak studio monitor
		DDA420	$1000	D/A converter
135	Digital Dynamics	Series III	NA	software for ProDisk-464 system
112	DigiTech	GSP 7	$500	guitar effects processor and amp
		GSP21 PRO	$800	guitar signal processor
		DSP16	$300	effects processor
		DSP256XL	$440	digital multi-effects processor
		TwinTube	$650	tube guitar preamp
		RDS8000	$350	8 second digital delay
136	Dynaware	Ballade	$195	MIDI sequencing software
137	Echo Audio	EASE III	NA	speaker enclosure
138	Electro-Voice	RE27N/D	$750	cardioid mic
		RE38N/D	NA	cardioid mic
		FM-12C	$500	floor monitor
139	E-Mu	Procussion	$995	sound module
		Emulator 3X	$7000	expander unit for Emulator 3
140	Ensoniq	EPS-16 PLUS	$2495	digital sampling workstation
141	E3MC	2000 series	NA	loudspeaker
142	Eventide	H3000KS	$4590	digital sampler
143	Fostex	G-24S	$14,500	tape machine
144	Furman	AR-PRO	$1749	line voltage regulator
		AR-230	$599	line voltage regulator
		DJM-8	$749	DJ production mixer
145	Galaxy Audio	Pro Spot 2	NA	loudspeakers
		Cricket	$300	polarity tester
146	Gemini	MB150	$120	speaker system
		MX-9300	$480	audio mixer
		PVX-160	$588	power amp
147	Groove Tubes	Trio	$1300	preamp
		D75	$1400	stereo tube amp
148	Hafler Pro	Pro 5000	$1200	power amp
149	Hughes & Kettner	ATS 100	$1295	tube system guitar amp head

AKG V6HP

Alesis RA-100

Artefx SB-2300

Carver PT 2400

Denon DM-4000 dual transport CD

DigiTech 21 Pro

JBL Control Micro and Control SB Micro

JLCooper CS-1 Control Station

Korg A5

Parasound HSCA-1200

LIT #	BRAND	MODEL	PRICE	PRODUCT TYPE
150	Innovative	VT-MP 2	$1395	mic preamp
		VT-DB-2	$399	direct box
151	JBL	SR4700 series	$795-$1495	loudspeakers
		Control Micro	$160/pr.	reference monitor
		Control SB Micro	$195 ea.	subwoofer
152	Jeanius Electronics	RD-T	$249	Russian Dragon
153	JLCooper Electronics	Sync-Link	$199	MIDI interface/synchronizer
		PPS-2	$169	synchronizer
		CS-1	$599	control station
154	KABA	RTDS-4TM	$1697	4-track duplication system master
		RTDS-4TS	$1768	4-track slave
155	Klipsch	kp-320	$883	2-way loudspeaker
		kp-115-sw	$379	loudspeaker
156	Korg	Wavestation A/D	$2499	rack-mountable keyboard
		A5	$350	multi-effects processor
		A1	$1999	multi-effects processor
157	Kurzweil	K-2000	NA	keyboard
		1200 series	NA	keyboard
		Grand Mark III	NA	keyboard
		EGP-K	NA	portable keyboard
		MS-1A	NA	micro sequencer
158	Lexicon	LXP-15	$1050	multi-effects processor
		300	$4795	digital effects system
159	Mark of the Unicorn	7S	$595	MIDI mixer
		Performer 3.6	$495	sequencing software
160	Midiman	FineLine	$350	line mixer
		Syncman Pro	$600	synchronization bank
161	Nady	650-VHF HT	$660	rack mount wireless system
		1200 VHF HT	$1700	wireless mic system
162	Numark	DM1475	$650	DJ preamp/mixer
163	Opcode	OMS	$495	MIDI operating system
		Galaxy Plus Editors	$379	universal librarian program
		MAX	$395	graphic programming environment
		Track Chart	$179	studio management software
		Vision 1.3	$495	sequencing software
		Studio Vision	$999	sequencing/recording software
164	Optical Media Int'l	ProCDP SCSI	$1395	CD-ROM player
165	Panasonic/RAMSA	SV-3900	$2100	DAT recorder
		SH-MK390	$400	SV-3900 required remote
		SV-3700	$1599	DAT recorder
		SV-255	$2700	portable DAT recorder
		WS-A500 series	$650-$1000	loudspeakers
		WRS840	$38,500	mixing console
		WM-S2	$170	mini condenser mic
		WM-S5	$280	mini condenser mic
166	Parasound	HCA-1200	$775	amp
		P/FET-900II	$425	preamp
167	Passport	AudioTrax	$199	MIDI desktop recording studio
		Master Tracks Pro	$395	sequencing program
168	Peavey	Pro-Fex	$800	stereo multi-effects processor
		DPC 750	NA	digital power amp
		Protege	NA	sound reinforcement console
		DPM 2	NA	digital keyboard
		DPM 3SE	$2300	digital synthesizer
		DPM SP	NA	sample playback module

LIT #	BRAND	MODEL	PRICE	PRODUCT TYPE
169	Pro Audio Sys.	RS-2	$2400	loudspeaker
		TOC Stage Wedge 2	$2900	loudspeaker
170	Prosonus	PD3	$70	drum/bass CD
171	QSC Audio	EX 2500	$1898	power amp
		EX 1600	$1498	power amp
		EX 1250	$1198	power amp
		EX 800	$948	power amp
172	Rane	Flex FPL 44	$349	program limiter
173	Rockman	PGE 2	$495	MIDI graphic equalizer
		Smart Gate	$190	noise gate
		MIDI Octopus	$260	MIDI controller
		Guitar Ace/Bass Ace	$99	headphone amp
174	Rocktron	Intellifex	$1149	24-bit DSP
		HUSH 8X	$299	single-ended noise reduction
		PRO Q	$629	stereo guitar enhancement system
		Patch Mate	$749	guitar rack switching system
		300G	$419	compressor/limiter
175	Roland	DM-80	$5495-$7695	hard disk recorder system
		S-770	NA	operation software for S-770 sampler
		SBX-1000	$3495	MIDI cueing box
		JD-800	$2895	synthesizer
		S-750	$4995	digital sampler
		SC-55	$795	sound module
		SB-55	$695	sequencing/playback device
		RSP-550	$1275	multi-effects processor
176	Samson	Concert Series II	$500	wireless mic system
		Hartke 412 TP	$899	guitar cabinet
		Hartke 7000/3500	$999	rack mounted amps
		Hartke 210B/1155B	$999-$1150	combo amps
177	Scorpion Systems	sYbil 2.5	$299	music software
178	Sendit	4X3	NA	effects return mixer
		Summitt A/B	NA	stereo effects return mixer
179	Sennheiser	MD 422	$535	cardioid mic
180	Shure	VP88	$995	stereo mic
		WA400	$583	amplified antenna dist. system
181	Singular Solutions	A/D64x	$1295	analog/digital interface
182	Soldano/Caswell	X99	NA	motor control MIDI preamp
183	Sonex	PSP-7	$234	1/8-inch thick barrier
		PSP-1	$274	1-inch thick acoustical foam
184	Sony	ECM-510	$368	electret condenser mic
		ECM-530	$328	electret condenser mic
		MU-L021	$799	compressor/limiter
		MU-EQ041	$699	parametric equalizer
		MDR-7506	$109	headphones
		MDR-7504	$89	headphones
		MDR-7502	$49	headphones
		PCM-2700	$2999	DAT recorder
		CDP-2700	$1299	CD player
		DPS-D7	$1149	digital audio hyper delay
		DPS-R7	$1249	digital audio hyper reverb
151	Soundcraft	Delta Monitor series	$13,300-$19,850	live mixing console
		Spirit series	$2395-$3495	PA mixer line
		Delta 8	$4290	mixing console
		Venue series	$11,550-$23,620	PA mixing console


QSC EX 2500

Roland SAX-1000

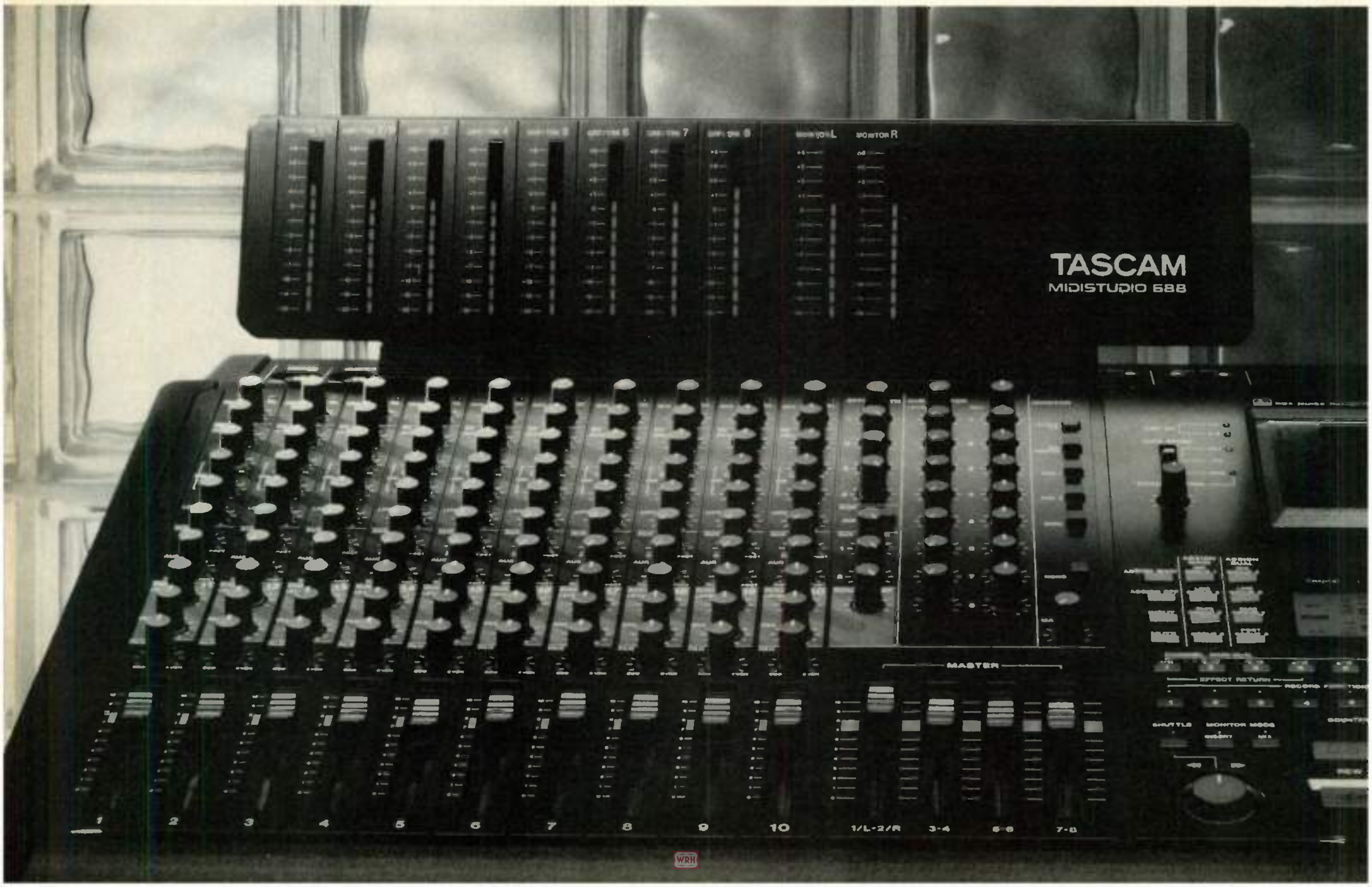

Sennheiser MD 422


Shure VP88

Sony ECM-510

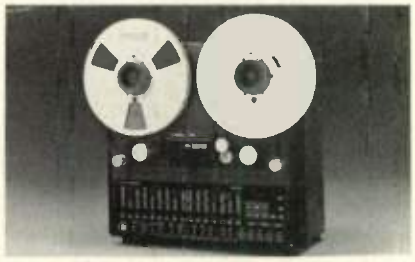
Please note: *this is a sampling of new gear exhibited at the Nat'l Association of Music Merchandisers Show. Information was provided by manufacturers; prices are suggested retail.*

LIT #	BRAND	MODEL	PRICE	PRODUCT TYPE
185	Soundcraftsmen	PRO-EQ 22	$349	10-band graphic equalizer
		900X2	$1599	power amp
		PM860	$599	multi-channel power amp
		RA6500/7500 series	NA	amps
186	Sound Support	112/212/412	$140-$280	speaker enclosures
		ER series	$136-$315	effects racks
		AR series	$158-$515	amplifier racks
187	Soundtracs	Sequel	$29,000-$39,000	sound reinforcement console
188	Spectral Synthesis	SynthCARD	$4995	DSP system
		SynthEngine II	$325	sampling software
		AudioVision	$495	soundfile/sample editor software
		Digital Studio	NA	hard disk recording/editing
189	Steinberg	Niche Mix Automation	NA	—
		Niche Audio Control	$479	—
		Cubase 1.8 Mac	$579	sequencing program
		Avalon 2.0	NA	sample editing software
		Synthworks	NA	editor/librarian program
190	Studiomaster	Powerhouse series	$2095-$2595	powered mixing consoles
		Gold Series	$1100-$5675	mixing consoles
		Trackmix series	$9750-$11095	recording consoles
		Diamond series	$630-$1050	mixers
		UFEX series	$475-$650	audio processors
191	Symetrix	S64E	NA	expander/gate



LIT #	BRAND	MODEL	PRICE	PRODUCT TYPE
192	TAC	B2	$4680-$10320	compact mixing console
		SR6000	$59500	40-input mixing console
193	Tascam	MSR-16S	$8499	16-track recorder
		Porta 03	$329	portable recorder
		424 PortaStudio	$549	portable recorder
		488 PortaStudio	$1599	portable recorder
		2516	$2999	mixer
		2524	$3999	mixer
194	Technics	SM-U2	NA	group lesson controller
		SX-PR3	13000	digital ensemble
195	TOA	MR-8T	$1895	8-track recorder
		F-500/F-600	$429-$749	foreground speakers
		ME-AV series	$158-$869	studio monitors
196	Turtle Beach	Soundstage 1.1	NA	editing software
197	Vestax	MR-44	$749	4-track recorder
198	Yamaha	P120	$445	monitor amp w/graphic equalizer
		DTR2	$1495	R-DAT recorder
		EM1620	$795	powered mixer
		EMP100	$345	multi-effects processor
		QY10	$399	music sequencer
		RY30	$595	programmable rhythm synthesizer
199	Yorkville	AudioPro 1200	$1249	power amp
		YSM-1	$280/pr.	nearfield studio monitor
		Elite MX-2000	$1949	PA loudspeaker
200	Zoom	9030	NA	instrument effects processor
		9010	$1999	sound processor
		8050	NA	advanced foot controller

Soundcraft Spirit



Tascam MSR-16S

Yamaha EMP100

The $5,000 Dream Rack

Setting out for effects heaven with five grand and a dream.

J.D. SHARP

IT'S A MIGHTY hard task to take a budget of $5,000 and come up with the ultimate effects rack. The hard part is that all the significant products on the market deliver excellent performance. So the issue comes down more to features and the subtle nuances of sonic character rather than specifications on a sheet of paper.

This requires careful listening before buying anything, and making sure an effect's character is something you'll be happy with for more than a couple weeks.

Even with all the great equipment, sooner or later it comes down to making choices, and that's what we've done to come up with the $5,000 effects rack.

Several very desirable effects were eliminated sheerly on price: these include the Eventide Ultra Harmonizer S, Lexicon's new 300 unit, and Roland's R-880 Digital Reverb. It just didn't seem to make sense to spend half or more of the budget on one unit — fabulous as they all are. Today's mixes cry out for multiple effects and multiple ambiences, and on a $5,000 budget, this is best achieved with an array of digital processors rather than one killer unit.

SOUND OFF

Just what kind of effects are we after? Right at the top, we've gotta have several digital reverbs, with at least one unit that has performance good enough to make you cry...something that makes a vocal sound big and a snare drum enormous. This can be supplemented by one or more high quality but lower-cost units. We also need pitch transposition both for subtle detuning effects as well as correction of off-pitch material. Delay and modulation effects such as slap echo, chorus, phase, and flange are certainly necessities. Ideally we'd like to get a good Leslie sound. Any decent studio should consider a 'guitar amp simulator,' something you can plug into and come up with searing guitar sounds without having to blow out the windows in your living room. And finally, we favor a vocoder, for mind-boggling voice-within-music effects that are useful for pop music as well as ad, jingle and soundtrack applications.

Several of our needs can be answered at once by two recently introduced products. Lexicon's LXP-15 ($1,050) and the Zoom 9010 ($1,999) can each play many roles. From the LXP-15 we get Lexicon's legendary reverberation algorithms, along with high-quality pitch shifting and delay. The Zoom has the advantage of incorporating four inputs and four outputs; depending on the effect selected, the 9010 can act as up to four independent units. It offers reverb, chorus, delay, pitch shift, compression and phasing, all with a 44.1 kHz sampling rate and 20 kHz bandwidth, as well as complete programmability all around. Both of these heavy hitters can play multiple roles in the studio. Our ideal rack calls for one each.

Our choice for a supplemental reverb unit is a toss-up between the DigiTech DSP256XL ($439) and the Alesis Quadraverb ($495). The Quadraverb is (in my view) a better multi-effect, but if the emphasis is purely on reverb we give the nod to the DigiTech. They're both great. Select one or the other and get on with it.

Although pitch transposition is incorporated in both our LXP-15 and the Zoom, it's important enough to warrant its own rack unit. Here there's little competition for the DigiTech IPS-33B Super Harmony Machine ($899). Aside from being smart enough to generate up to two harmonies from a single note and have them fall on the correct note in up to 41 different pre-programmed scales, the IPS-33B also allows automatic cor-

Photo by James Martin

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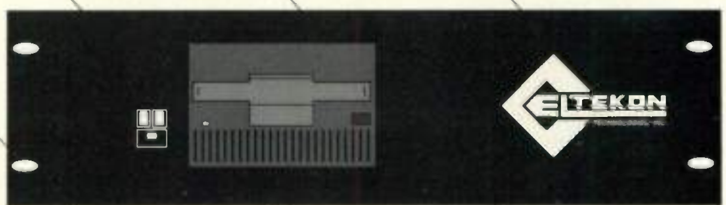

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rection of out-of-tune material, an ability shared only with the far more expensive Eventide Ultra Harmonizer. This feature alone is worth the price of admission. In delay mode, up to 1.5 seconds are available, and this can even be added to the harmonies.

BE EFFECTED

Guitar amp simulation is a hot effect area, and there are plenty of choices. One of our favorites, the Boss GL-100 (listed at $450), seems to be on the way out, and it's a shame, since it's one of the only solid-state units that seems to produce a distortion that's completely satisfying even to tube fanatics.

If you can't find it, the A.R.T. Power Plant ($329) is an excellent and low-cost substitute. These are both completely non-programmable, analog units with no effects and just basic EQ. For a more modern, highly effected sound from within one unit, the current favorite seems to be the Korg A-3 ($1,299). It can apply just about any conceivable twist to guitar signal except a convincing tube distortion, but the Power Plant or GL-100 will take care of that. If you can't live without tubes, try a Real Tube preamp instead. We say check them all out and choose your favorite; our rack will contain an A-3 and a GL-100.

We've gotta have at least one reverb with performance good enough to make you cry...Something that makes a vocal sound big and a snare drum enormous.

Roland's recently-debuted Boss SE-50 ($550) is a unique multi-effect that picks up our last two requisite effects, Leslie simulation and vocoding. Aside from being a very credible reverb, chorus, etc., the SE-50 offers a Leslie effect with MIDI control over the 'top' and 'lower rotors;' you can use two different controllers to speed up and slow down the two rotating effects separately.

The built-in vocoding effect is quite impressive, and is truly a bargain when you consider that the cheapest vocoder available until now costs more than the SE-50 — which does tons more. The SE-50 also contains an unbelievable multi-tap delay routine, capable of some original and explosive effects.

Our completed effects rack comes in under $5,000 with some judicious shopping, and should provide enough horsepower and high performance to tackle any type of task with ease.

COMPANY	LIT #	PHONE
ART	106	(716) 436-2720
Digi Tech	112	(801) 268-8400
Eventide	142	(201) 641-1200
Korg	156	(516) 333-9100
Lexicon	158	(617) 891-6790
Roland	175	(213) 685-5141
Zoom	200	(415) 873-5885

J.D. Sharp *is EQ's Systems Workshop columnist. When he's not writing he can be found configuring systems like this at San Francisco's pro audio dealership Bananas-at-Large.*

THE WORLD'S FIRST MASTER SYSTEM CONTROLLER

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- Balanced XLR inputs/outputs
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- Limiter link/voltage control jack
- Stereo auto-detect circuit
- Switchable detector loop
- Stereo gate key input
- Full stereo processing
- Threshold activation LEDs for all functions
- Parasweep integrated controls simplify operation
- Actual output monitoring limiting circuit
- Unsurpassed audio specifications

The MDC 2001 is a revolutionary new product that can improve performance in any audio application in which it is used.

In live sound, the compressor can smoothly and transparently control the musical dynamics. The gate and expander can totally eliminate stage noise, gate off microphones for higher gain before feedback, and eliminate drum and amp bleed. The de-esser will knock out excessive sibilance protecting your drivers and balancing the material. The all new exciter circuit brings out the upper harmonics for sparkling clarity, even at high volumes. The final peak limiter can be set to the maximum input voltage of the amps to prevent your system from ever being overdriven.

In the studio, the MDC can smooth out the dynamics of voices and instruments, eliminate background noise for much cleaner recordings and remove harsh "s" sounds before they saturate the tape. It can also add back in all the natural harmonic brilliance that is lost for recordings with crystal clarity. The peak limiter will prevent the tape from overloading and is essential in digital mastering. And because the MDC is fully stereo, all the left/right balance stays intact.

In churches, theaters and fixed installations, the MDC is the one piece of gear that will provide level protection, noise gating, de-essing, signal enhancement and dynamic control in full stereo in a single rack space. All functions are independently controllable and may be taken out of the chain if desired. Ducking, keying, and linking jacks are provided, and the audio performance of each individual processing circuit is world class.

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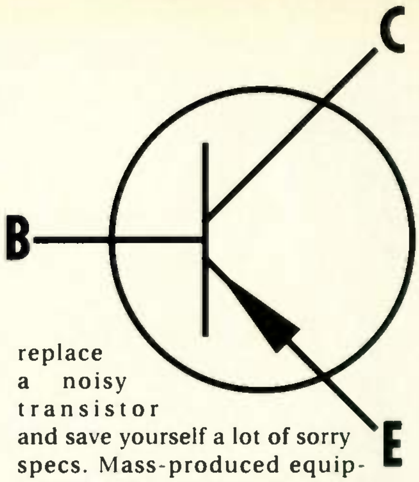
Sound Micro Surgery



Cutting noise off at the root just takes a little electronic know-how.

MARK BERNARD &
MICHAEL BIRNBAUM

THE BOTTOM LINE is this: noise is the enemy.

But no matter how carefully you buy your gear and how well you wire it all up, you're probably still going to hear some. Hiss. Hum. The ugly underbelly of a bad S/N ratio.

You can fight back. Many problems you may have thought were inherent in your studio gear — including sonic performance, noise and general reliability — can be remedied with relative ease and economy.

The techniques illustrated here apply to a wide variety of gear ranging from a '50s organ to a '90s mixer or preamp. All you need is patience and some decent soldering chops.

POT GRIME

Switches and potentiometers are the most unreliable electrical components, and treating them with care can be a cheap and easy way to clean up your sound. The major enemy here is grime buildup between their electrical surfaces.

Both switches and pots use an operator-controlled moving wiper that contacts a stationary surface. Normally, their wiping action keeps them fairly clean, but in time, accumulating dust translates directly into noise. Your first step — and often the only step needed — is spraying a cleaning agent into the contacts. But remember that if there's no opening inside the device, you might have to take it apart.

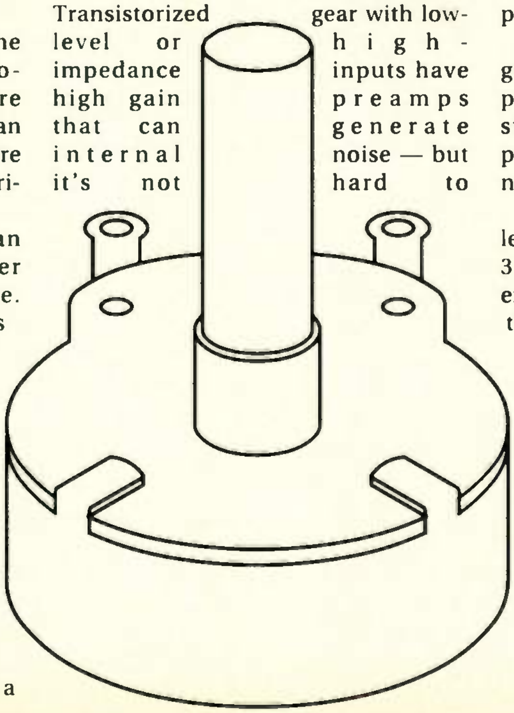
Disassembling a switch or pot isn't too difficult. They're usually made of a steel or aluminum stamped piece attached to a flat fiber or plastic base that holds most of the electrical parts. Just bend out the metal tabs that wrap around the base (see Fig. 1).

Clean the potentiometer's resistive material with a swab dipped in 99 percent pure alcohol (refer to Figure 2). If some of this blackish material has worn all the way through, try bending the contact wiper(s) that touch it so they wipe an as-yet unused surface.

Before you put it back together, apply a thin film of contact grease on all the contact surfaces and any sliding or turning parts. This protects the contacts from wear and restores the original smooth feel. Try GC Electronics TunerLub or Cramolin Paste. Lubrication products in spray cans aren't viscous enough to bring back the fluid feel and protect surfaces for long. And don't use grease or oil that isn't specifically intended for electrical contacts.

PREAMP TRANSISTORS

If you're getting a lot of noise from your transistorized gear (like your mixing console, outboard preamp or active direct box, for example) the problem might be in the transistors. Transistorized gear with low-level or high-impedance inputs have high gain preamps that can generate internal noise — but it's not hard to replace a noisy transistor and save yourself a lot of sorry specs. Mass-produced equipment hardly ever contains handpicked preamp transistors, and by taking time and spending a little money, you can clean up a hissing front-end.



Beware: some designs are inherently noisy, and the most careful work may not produce significant results. In a multi-input application, check to see if noise levels vary from channel to channel. If they do, the gear is a good candidate for this upgrading technique.

First, unplug any input to the preamp and crank the gain trim (if any) all the way up. Then adjust other levels down the line so you get a meter reading of the noise the preamp is producing. Using an analog VU meter (or an oscilloscope) over a bargraph meter provides better resolution. If you can't get a visual meter reading, you won't be able to make an accurate comparison. In this case, try additional amplification through an insert point.

But make sure that the noise you get is predominantly from the preamp, and not later amplification stages. You can check this by temporarily lowering the preamp gain and noting if most of the noise goes away.

Transistors have three distinct leads, denoted as E, B, and C (see Figs. 3 and 4). If the replacement isn't an exact match of the original, make sure that it has these leads identified and that you know where each of them goes in the equipment. This is a situation where a schematic can be essential.

Locate the preamp transistor on the schematic.

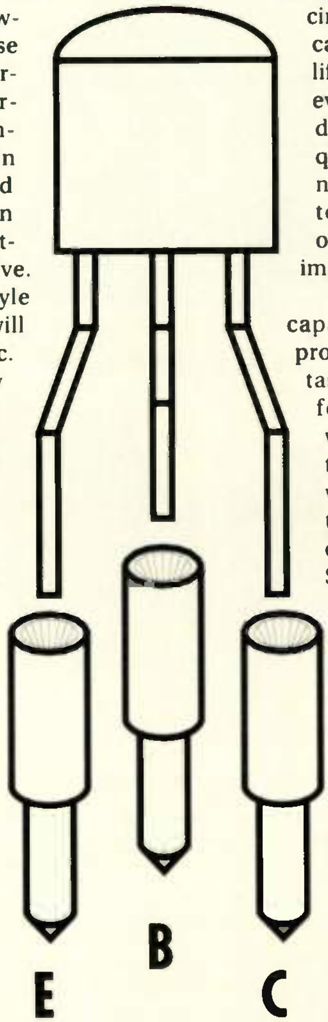
▲ **Fig. 3: A PNP transistor as it would appear on a schematic. The arrow would point outward for an NPN transistor.**

◀ **Fig. 1 : Potentiometer Tabs. Linear pots and switches have similar tabs.**

It will probably be on the left side of the diagram, closest to the low-level or microphone input that it amplifies. Usually, right near it will be any gain trim, line/mic switches and pad controls. Once you've located it on the schematic, it should be easier to find on the circuit board. Transistor part numbers and adjacent parts can help to clue you in.

NPNs AND PNPs

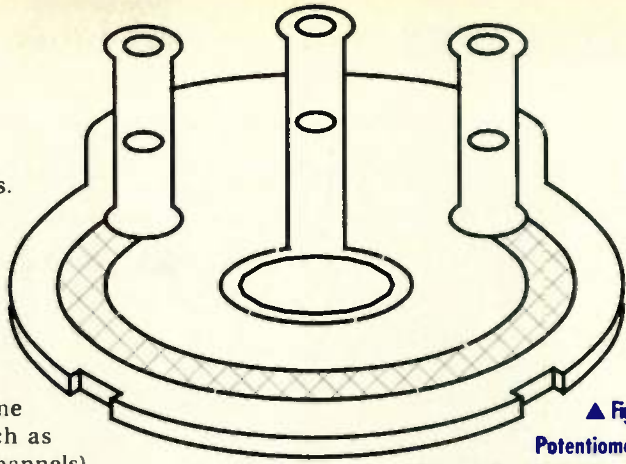
Get your hands on at least 10, if not 100, new transistors. Make sure they're as closely matched to the original part as possible. Some cross-referenced "Equivalents" are inappropriate in this low-noise application, because they are only general-purpose replacements. Furthermore, note that common transistors come in two varieties, NPN and PNP, depending on whether their supply voltage is positive or negative. The first letter of the style tells how the transistor will be drawn on a schematic. An NPN has the arrow pointing out; a PNP has it pointing in (see Fig. 3). These two types of transistors are not interchangeable.



Remove the transistor and replace it with a socket to facilitate easier exchange. High-quality machined socket pins give the best results (see Fig. 4). Separate pins are better than a single socket; they allow the utmost in flexibility on crowded PC boards.

Try each new transistor and mark it with the resultant meter noise level for future reference. When you're done, plug the quietest transistor back into the socket. If you didn't use a machined socket (or if you want the highest reliability), remove the socket and solder the transistor directly. Even the best socket eventually develops dirty contacts.

You can use other transistors that may have had good noise figures for other preamp inputs of the same equipment (such as different mixer channels).



▲ **Fig. 2: Potentiometer base with resistive deposit. Linear faders are similar, but have resistive material in a straight line.**

◀ **Fig. 4: A transistor being inserted into machined socket pins. Not all transistors have E, B, and C in the same order.**

CRACK CAPACITORS

Capacitors couple tube or transistor stages, passing the higher audio frequencies while blocking the DC voltages that constitute proper circuit biasing. Worn-down capacitors can make your life a living hell — causing everything from noise and distortion to lost low frequency response. Fortunately, replacing capacitors is inexpensive and often results in great audio improvement.

Figuring out which capacitors are causing you problems (if any) is a tricky task. You can check them for capacitance value with a capacitance tester, but such a device won't always reveal subtle shortcomings that can degrade your audio. Some capacitors that are over 15 years old, and most that are over 25 years old, will have perceptible degradation. The best way to check such a capacitor's state is to replace it and see if the circuit response improves.

Every capacitor along the main audio path is directly carrying the entire signal. If a capacitor has a reduced value here, it will limit the low frequency characteristics of the gear. If it's leaky, it can cause noise and distortion by upsetting the biasing of adjacent stages.

Replace capacitors with the same or higher voltage rating (labeled as WVDC). Too high a voltage (more than double the original rating) may increase leakage and can audibly affect the unit's performance. Use polypropylene, polyester (mylar), or tantalum capacitors instead of ceramic types, and choose military grade whenever possible.

Before you try to undertake any electrical surgery, consider this: you clearly shouldn't rip up your gear if you don't know what you're doing. It never hurts to open up an old unit and experiment a little before you start messing with your bread and butter. Also realize that you might make a bad problem worse if you screw up, and you'll certainly void a warranty forever by opening any piece of studio gear.

So — you're on your own. Once you get the hang of it, though, it's a piece of cake.

SOURCES

Here are good catalog sources for all types of electrical components, providing detailed specs (i.e. dimensions and thermal drift) in their catalogs, and low minimum order amounts.

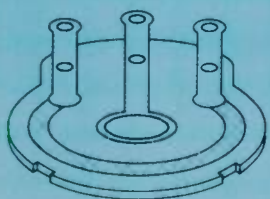
Mouser Electronics, (800) 346-6873.
Digi-Key Corp., (800) 344-4539.

EQ contributor and guitarist **Michael Birnbaum** *owns and operates AppleHead Sound in Bearsville, New York;* **Mark Bernard** *is an acoustics research engineer with Woodstock Percussion.*

Chips 'n' Dips

One of the easiest ways to increase sonic performance in your pieces of older gear is to replace general-purpose, low performance op amps ("chips") with higher performance models. It's not too difficult and a little work goes a very long way. Best of all, it won't cost you a fortune.

One of the most common chips is the 1458 (also known as the 1558 or 5558) dual op amp. This chip contains two compensated op amps in an 8 pin DIP (Dual Inline Package; see Techniques: MIDI Studio EQ, on page 32).

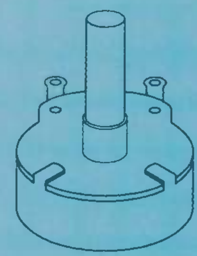


The 1458's drawbacks are low slew rate, comparatively high noise levels, and an inability to drive 600Ω loads.

If the chip that you're planning to replace is in a socket, that's great. Just pull it out and plug in a Signetics NE5532 (a pin-for-pin replacement for the 1458 but with much better slew, noise and drive characteristics), making sure the package notch or dot that indicates pin 1 faces in the same direction as the 1458's notch.

Desoldering an old IC and replacing it is much more difficult than doing a simple plug-in operation; obviously, it is a serious mistake to attempt desoldering unless you know what you're doing. After desoldering the chip, think ahead — install a low-profile socket and save yourself a lot of future headaches.

If you're one of this world's truly picky audiophiles, try out the SSM-2134. This chip is a high-performance replacement for the NE5532. It's expensive, but critical applications like mic preamps justify the few extra dollars — and you're sure to hear a significant difference.



If you're worrying about how all this electronic wizardry will affect your power draw, stop. High-performance op amps tend to draw a bit more current, but that shouldn't be a problem unless your gear's power supply is woefully underrated.

A last word of advice: if you're only going to replace a few chips, start with those high-gain circuits like preamps and filters.

— Craig Anderton

News & Views

Sound bites from the world of pro audio.

THE EDITORS

Roland has formed a new Pro Audio/Video Division, headed up by Curtis Chan, formerly of Ampex. Three new products are slated for delivery this year: a Roland Sound Space (RSS) Processing System that enables the reproduction of a three-dimensional aural environment from a stereo playback system, a self-contained Digital Multi-track disk-backed recorder (DM-80) in four- and eight-track configurations, and the SBX-1000 MIDI Cueing Box that can store up to 32 separate songs internally. For more information, contact Pro Audio Division, RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040; 213-685-5141.

Opcode's "The Book of MIDI," is an interactive HyperCard Stack that teaches about MIDI, music, computers and synthesizers. It runs on a Mac Plus or above with HyperCard 1.2.5 and above. It costs $39.95. The company has also bowed The MIDI Translator, a MIDI interface with one MIDI in and three MIDI outs.

Sound Concepts has just bowed a new videotape on setting up and using sound reinforcement equipment. The 75-minute VHS video, for $39.95, covers the individual SR requirements for every bandmember. But it also tells how to check a venue, choose speakers, mics and monitors, and how to put it all together. It's narrated by David Scheirman, president of **Concert Sound Consultants**, and a live sound engineer/designer who has worked with Linda Ronstadt, Willie Nelson, Luther Vandross and Al Jarreau. For more information, write to Sound Concepts, P.O. Box 831, Julian, CA 92036.

Studio designer/architect **John Storyk** will be teaching a course on the new **Techron** TEF 20 at the Full Sail Center For the Recording Arts in Winter Park, Florida. The TEF analyzer is a qualifying device used in design and manufacturing. For information, call 407-679-0100.

Emmy-award winning sound reinforcement specialist Bruce Burns has recently expanded his company, **Burns Audio**, with the opening of a third facility in the Washington, DC area. Burns has provided sound tech for numerous Grammy shows, the Academy Awards, the Golden Globe Awards and the Democratic National Convention. His new office is located at 4588 Eisenhower Avenue, Alexandria, VA 22304.

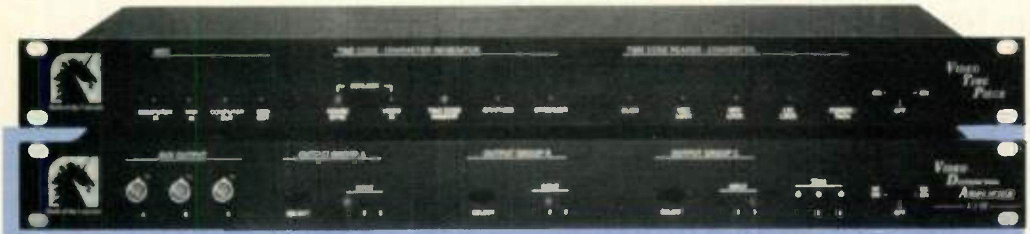
Learning music theory? Try **Bradley's Slide Rule** — for under $20, it teaches chord structure, scales, modes and transposition. Write to P.O. Box 90659, San Jose, CA 95109.

Seven-time Grammy award-winning **Manhattan Transfer** member Alan Paul has recently built a music studio in the L.A. home where he's been creating new material for the group's upcoming album (CBS). The studio features a plethora of **Panasonic/Ramsa** gear, including a WRS-T820B mixing console, an SV-255 portable DAT recorder, an SV-3500 pro DAT recorder, WP0220 and WP9055 amps, along with an SL-4700 compact disc player. The recording, featuring songs nearly all written by the Manhattan Transfer, is due out this spring.

Motley Crue has recently gotten a Sound Tools system and DECK multitrack software from **Digidesign**. The group is working on a greatest hits album, "Decade of Decadence," due out (tentatively) in September on Elektra records. Write Digidesign at 1360 Willow Road, Suite 101, Menlo Park, CA 94025.

The **University of Iowa's** summer Seminar in Audio Recording (June 10 - 21) promises to be hot this year. Bob Ludwig, VP/Chief Engineer of **Masterdisk** (and EQ contributor) will be guest-lecturing, as will Jerry Bruck, President of New York's **Posthorn Recordings**. Contact Lowell Cross, Recording Studios, School of Music, University of Iowa, Iowa City, IA 52242; 319-335-1664.

Window of Opportunity



▲ Reasonably-priced units (like Mark of the Unicorn's Video Time Piece, above) can help you get the job done with two video decks; even consumer-quality decks do fine.

If you're working on an audio-for-video project, window burn can help you score.

PAUL D. LEHRMAN

WHEN PROS EDIT video, there's something besides just the picture on the screen they watch. Somewhere — on the bottom, on the top or in a corner, in large characters or small — there's an eight-digit number changing faster than the eye can read it.

No, it's not SMPTE timecode; it's "window burn," also called "burn-in," or "vis" (as in "visual") code. SMPTE timecode is actually an *audio* signal recorded on an audio or special "address" track of a videotape. It tells your location on the tape — in hours, minutes, seconds and 1/30ths of a second, or "frames." Window burn is a visual representation of this code.

Whether you're assembling edit lists for the cuts in a video program or producing music or effects tracks for a video or film, if you're using a MIDI sequencer or hard-disk audio system locked to videotape to score film, window burn can make your life immeasurably easier. That's because SMPTE can only be read at speeds close to normal. The LED or on-screen computer reader gets stuck on the last frame that was played at normal speed, not the current frame. Trying to grab an individual frame number to determine the exact starting point of a cue can make you look like Charlie Chaplin on the assembly line in "Modern Times."

Enter window burn. By putting the frame number into the video picture itself, it makes sure that whenever you can see a picture (even in stop, shuttle or step-advance), you can see the frame number. Producing a window burn involves dubbing the tape while inserting the window into the video image. (You can't put a window onto an existing tape any more than you can overdub audio on a mono tape deck.)

Reasonably-priced units (like Fast Forward Video's F30 and Mark of the Unicorn's Video Time Piece) can get the job done admirably. You still need two video decks, but they needn't be expensive: you're only making a "workprint," not a video master, so even consumer-quality VHS decks can do the job just fine.

BURNING QUESTIONS

If the tape you're copying already has SMPTE timecode on an audio or address track, the procedure is simple: Plug the video output (not the "to TV" jack) and audio or timecode output of the source deck into the SMPTE generator's inputs, and connect the generator's timecode and video outputs to the appropriate inputs on the copy deck. Tell the generator you want to do a burn-in, and tell it (if it gives you a choice) where on the screen to put the numbers and what color to make them. Remember that the bottom of the screen often distorts in PAUSE and SLOW when you're using a low-end video deck.

Also tell the generator you want to "regenerate" the SMPTE timecode (the audio signal), laying down a fresh copy of the same numbers as are coming into it, since the timecode signal is highly susceptible to distortion. Make sure any noise-reduction on the audio is switched off.

Start the copy deck in RECORD, and then start the source deck in PLAY. Make sure there's at least 15 to 30 seconds before the picture starts, where you only are recording timecode. This will allow your sequencer later on to locate and lock to the code in a comfortable amount of time before the first cue.

Sometimes you'll get a tape that has no timecode on it. With a good SMPTE generator, it's easy to create your own. Hook up the video output, not the audio or timecode outputs, of the source deck to the generator, and the outputs of the generator as described earlier.

Tell the generator to "genlock" to the incoming video signal, and to generate new code — both audio and window burn — based on the timing of the sync signals contained in the video. This makes sure the timecode numbers and the video signal itself are in perfect sync. If you just let the generator stripe the code using its own internal clock (this is called "freewheeling"), the video and the code will drift away from each other, and your music and sound cues will end up being inaccurately placed.

You'll also have to tell the generator the starting time for the new code. Standard practice is to use a non-zero hour mark — 01:00:00:00 or 02:00:00:00. Don't use 00:00:00:00, because you may get in trouble if you rewind back before the beginning. Again, try to leave plenty of time before the picture starts.

Window Burn Checklist

✓ Have you thought out where you want the window burn to appear on the screen? Usually, the very top and bottom of the screen aren't good choices.

✓ Have you chosen an appropriate color for the numbers in the window? While it might seem self-evident, the background color can make a big difference. For example, if you've got a lot of "deep space" footage, black would be a terrible choice.

✓ Have you told the timecode generator to refresh the code? This is an extremely important point, since the time-code signal can very easily be distorted. If you're not using refreshed code, chances are your cues will be off-time later in the editing process.

✓ Have you switched off any noise reduction on the audio? This signal-changing technology can have a detrimental impact on the purity of your timecode.

FREEWHEELING

If you have a tape that is a rough edit, with no timecode and with starts, stops, and dropouts in the video, you can't genlock it; the video sync keeps getting interrupted. Here you're better off freewheeling the timecode generator. The code will drift away from the picture, but that won't matter much — the final edit, which you'll receive later on, will have different timings from this rough edit, and you'll have to tweak the timings of all the cues anyway.

> Trying to grab an individual frame number to determine the exact starting point of a cue can make you look like Charlie Chaplin on the assembly line in "Modern Times."

When you burn-in to discontinuous video, because you aren't genlocking, you may find an occasional frame with no number or two with the same. Again, because it's a rough edit, this won't matter too much, and the window burn will still be a valuable guide to laying down your sounds.

Occasionally, you may get a tape where the numbers in the window don't match the code, but are offset by a fixed amount. Different head arrangements in video machines can cause this. Try to figure out what the offset is, and take it into account when you make all your calculations.

Remember that window burn is only necessary for workprints. The original video master will *never* have numbers on the screen, and the final audio master which you deliver to the director or music editor doesn't need them either — as long as you've got good SMPTE timecode on the tape, and the music is locked to that timecode, the editor will have no trouble getting your music in perfect sync.

One other thing — make sure you've also given him or her a written list of all your cues, with their SMPTE starting times.

Massachusetts-based **Lehrman** *has reported on desktop audio-for-video and making a great sounding video demo for EQ.*

MONITORS

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then the 80 Hz high-pass filtering that works so well with the vocals takes the guts out of the kick sound. A workable compromise is to high-pass the vocals first, then add them into the drum mix. Of course, your mixer has to be able to create two separate monitor mixes.

Headphones are often overlooked as a solution to monitor trouble — for drummers and everyone else. Consider the benefits: freedom from feedback, ridiculously high volume capability without spillover, and extreme high-fidelity. Those little sound buds that you insert into your ears and plug into your Walkman can work quite well as inconspicuous stage headphones. However, you can't just plug these phones into your monitor system's power amp without some way of reducing the power delivered to them (remember that the average Walkman puts out a few milliwatts, yet they can achieve levels exceeding 110 dB SPL).

Q. Are you giving the band the best possible monitor mix?

A. Giving everyone his own mix seems ideal. Everyone can control what he hears without affecting the others on stage. On large stages, this works fine. On the postage stamps that often pass as stages in most nightclubs, everyone hears the other guy's monitor.

You've got to fix the problem in the mix — often adding more mixes. For a band that does its house mix onstage, a separate monitor mixer may be as simple as a rack-mounted unit that connects to the mixer's channel insert jacks. But remember that you'll need separate power amps for each mix (or powered speakers). A side benefit is that you can now equalize each mic's sound in the monitor mix. Be careful: excessive EQ can cause a feedback problem.

It takes time and experience to master the fine art of stage monitoring. Try to convince your band how much it means to the quality of their performance and make sure they give you sufficient time to get it right.

Of course, there are a range of sound measurement tools that can help. A real-time analyzer allows you to visualize what you are hearing — a decided bonus. In the end though, you've got to depend on your own hearing.

So take care of your ears. They're the best sound analyzers you've got.

HEALY

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cert environment, from the microphone to the loudspeaker. As far as I'm concerned, the sooner it comes the better. Once you realize the immense magnitude of what you can do digitally, you can't go back.

There are some other transducer concepts out there that bear looking into: flame speakers and ion speakers. I've heard them all. I've experimented with a limited basis on all of them; I've heard them all make sound. The ion speaker impressed me the most in terms of the purity and quality of its sound. It's just unbelievably accurate, since it works by ionizing the air molecules. By exciting the air, you expand and contract the molecules, thus causing motion, which then becomes acoustic energy or sound.

The ion speaker impressed me the most in terms of the purity and quality of sound. It's just unbelievably accurate.

The ion speaker does the same thing as a loudspeaker, but without mechanical motion. Today's loudspeakers are slave to the movement of the cone, the paper, the shape and size of it, and all the things you have to do to make it work as it exists. Again, it's moving parts. These innovations are eliminating motion and the signal loss that goes hand in hand with it.

We're experimenting with transputers now, a concept that is a form of parallel processing where you can do much greater amounts of processing in shorter periods of time. It's the future. It enables us to digitize the signal and keep it impeccably clean and undistorted.

With transputers, I no longer have to use audio networks to manipulate the time alignment. I simply tell the computer to perform the delays at the correct frequencies. The result is incredible control over the live sound. It's the cracking open of a whole new idea.

The whole audio network, and the way audio's been dealt with in the past, is all going to be pretty much passe.

I can't wait to hear it.

CHERNEY

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"I like to use the (Teletronics) LA2 on electric guitars because I can get a good rough edge on them, and the Fairchild for compression on the stereo program (when needed) to get a mix to be apparently louder overall.

"The Publison Infernal Machine works well for sampling (it has 20 seconds of full bandwidth memory) to correct an instrument's time and pitch for flying in vocals or other performances.The dbx compressor/limiter helps me take care of unwanted peaks, and it sets up very quickly and easily making it perfect for vocals."

Cherney claims that the BBE is "the best high frequency exciter around. I may use it to imply air around a track; it works especially well on synthesizers."

About the MIDIMan Interface he says, "It allows me to reinforce MIDI'd keyboard tracks at a later date if the need arises. I can record MIDI information on tape in real time so I can quickly and easily record a performance and later enhance it if necessary without the hassle of running a sequencer."

Songs are alive, and not all creatures require the same things.

Just because Cherney has built such a smoking rack doesn't mean he feels the compunction to overuse it. "I only use a piece when I need it, otherwise it stays in its case," he says. "I've learned to overcome my first inclination — to come into a studio, plug everything in, and use it all. The rack is there to back me up. There really is no magic way to use a rack, the magic is in your perception of what the music is saying, and of having some idea of what you are specifically trying to do to make that perception a reality. With the kind of recordings I try to make, it's the song and the performance that must speak in the most compelling way — all of the gear and hard work should be invisible. You have to remember one thing: nobody ever leaves dancing to the equipment."

"I try to treat a song as if it were a living thing, and not all creatures require the same things."

—*Jon Ruzan*

INNUENDO

continued from page 19

"Vocalist Freddie Mercury can hit loads of harmonies, so there was lots of triple tracking. Brian worked out the parts, playing them on keys and getting the notes in order to map the track out, and then in the end, on top of the lead vocal, six backing vocals were compiled from about 24.

"Freddie stood near the back of the control room. Closer up, we had a pair of Yamaha NS10's upside down on either side of him — sitting on the keyboard stations at a slight angle - with the tweeters at waist level so the high-end didn't spill onto the Shure SM 85 mic.

"We couldn't set up an 87 or a B&K or something like that, because there would have been just too much spill. The 85, however, sounded great with a slight bit of compression and flat with no EQ on the desk.

"*Innuendo* is a very busy track, and so having recorded everything cleanly there was then one hell of a mix on there. It took patience, but it worked out fine in the end!"

—*Richard Buskin*

Miking Cobham's Kit

Craig Bishop's secret recipe for Billy Cobham's trademark drum sound.

DAN DALEY

WITH THE PROLIFERATION of drum machines on the scene, the art and craft of drum miking takes on new dimensions. Perhaps even more so than any other acoustic instrument, drums seem to fascinate engineers because of their huge dynamic range, their dependency on microphone choice for sound and their ability to eat up so many microphones at once.

Over the years, certain familiar mic combinations have evolved for drum kits — an E-V RE 20 on kick drums and a couple of Shure SM 57s on the snare. But every engineer has his or her own particular favorite combinations. The example we'll be looking at here is definitely a highly evolved one.

Billy Cobham has attained legendary status, from his early days with Mahavishnu Orchestra, Miles Davis and on 22 solo records, Cobham has helped redefine both drumming and drum sounds. The engineer on Cobham's latest project, a combination of sessions for a new CD release and an instructional video, is also a veteran. Craig Bishop spent several years as senior engineer at Lucas/McFaul, a major commercial production house in New York, where he worked with the creme de la creme of session drummers including Steve Gadd, Anton Fig, Steve Jordan and Alan Schwartzberg. Bishop is now a partner in another production company, Reel World Productions.

Last December, Bishop began recording Cobham at Manhattan's new Brielle Music studios. Cobham's kit was somewhat slim — a snare, two kick drums, four rack toms, two floor toms, a high-hat, ride and crash cymbals and a couple of effect cymbals.

KICK & SNARE

In trying to achieve Cobham's desire for a wide open bass drum sound, Bishop placed an AKG D-12E on each kick. Most of the front skin on the kick had been removed and Bishop moved his hand around in the barrel as Cobham played, looking for the point of maximum air movement to aim the mic at. "I like the D-12E for kick drums because it has a nice edge and can handle tremendous sound pressure levels," explains Bishop. "It has an extremely wide dynamic response; it covers everything from a feather touch to pulverizing crescendos." Like many engineers, Bishop grew up with the E-V RE 20 as the standard kick drum mic, but now feels it doesn't have the depth to capture the high-impact kick drums of contemporary music.

The snare was next, and it got a somewhat more conventional approach with a Shure SM 57 and an AKG 451. The SM 57 was on a stand placed directly opposite Cobham's seat a couple of inches from the rim and aimed at the center of the skin at a slight angle. The 451 was placed on a gooseneck several inches over toward Cobham's left and aimed at a more acute angle at the edge of the top skin. Bishop's reasoning is that the SM 57

handles the fatness of the snare while the inherently brighter 451 captures its crispness and attack when sharply aimed. "I sometimes use the standard 'one on top, one on bottom' approach to snare miking, but after I experimented with this arrangement I found it works better in highly dynamic situations," says Bishop.

The choice for the toms was based on what was available in quantity. There were only two Sennheiser MD421 mics available, so he placed them on the two floor toms and used four of the more abundant SM 57s, one for each rack tom, reasoning that he wanted to keep the mic types consistent for each type of tom. The floor toms got the MD421s on booms and at a slight angle. The SM 57s were placed at 12 o'clock above the double-skinned rack toms and at an angle slightly more acute than was used on the floor toms, and about two inches from the rims. At that distance and angle, Bishop says, he gets more definition without sacrificing the dynamics of the barrels. Bishop does occasionally mic toms from both above and below, but when doing so he cautions to be aware of phase cancellation, which occurs when each mic picks up the same frequency. To avoid this, either re-aim one of the mics or use the phase reverse function on the console input module.

> Bishop moved his hand around in the barrel as Cobham played, looking for the point of maximum air movement.

CYMBALS

Cymbals are another area where phase problems can crop up. "You'll notice it if you mike them too closely," Bishop explains. "It's a function of their movement; cymbals not only vibrate, but they also wobble when hit." That changes the distance from the mic, slightly but enough to sometimes produce the hollow sound characteristic of phase cancellation. Bishop suggests that backing off on cymbal overheads will avoid the problem as well as providing a bigger sound. On Cobham's session, he used a pair of AKG 414 mics on booms positioned behind Cobham and reaching inward over his shoulders. They are then angled down towards the kit so they rest about seven feet off the floor, their baskets at a 45-degree angle relative to the floor with an arc of 90 degrees between them.

The high-hat also had a 414 on it, and because by this time Cobham's kit had become a jungle of stands and booms, Bishop at first placed the mic underneath the bottom cymbal. After experimenting a bit with the sound and trying to keep it as isolated as possible from the rest of the kit, he found a spot just above the edge of the hat with the sweet spot of the mic pointing away from the kit. "The 414 is a very flexible mic," Bishop says. "By nature it tends to be bright and can handle sibilance well. With a pad on it, it can also handle high sound pressure levels and it has a wide range of (pickup) patterns." Bishop chose the cardioid pattern for this session.

And finally, what might be the most important microphones of all — the room mics. Some engineers pull out all stops at this stage, cleaning out the closet to cover every angle in the room, looking for a variety of reflections to provide a more natural reverb.

For the ambient miking of Cobham's kit, Bishop chose a seemingly parsimonious pair of PZMs taped to the floor three feet in front of the kit and six feet to either side of it. This arrangement is plenty, he says, giving him a combination of low-end resonance and some added brightness as the cymbals reflect off the floor. The PZMs were recorded to a stereo pair of tracks.

Just before going to tape, Bishop checked for phase alignment on each individual track by soloing the tracks one by one. He started with the highest frequencies and worked down through the spectrum. He then recorded the drums with no processing, no gates, no compression, and just a touch of EQ.

"Some processing might be added later in a mix," he explains, "depending on the song. The kick and snare usually get compressed at that point, too, and the snare might also get a gate on it."

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CIRCLE 39 ON FREE INFO CARD

Decks of Glory

How to add another chapter to the tale of the tape.

RICHIE MOORE, PH.D.

THERE ARE TIMES in this life when you want a new tape machine so badly you can taste it.

Unfortunately, if you ain't got the bucks, forget it. But it's not so bad; if your spinner is totally shot, you can go buy a used model and bring it back to life — or you can inject some life — force into your trusted sidekick. Bringing a straggling reel-to-reel back to the land of the living just takes common sense and a couple of parts.

FIRST RITES

The main problem in very old (or very used) machines is mechanical wear and tear. Usually, the tape lifters and the pinch roller are the first to go. The tape lifters are tough to replace, but they're often more scored than the heads. They eventually start cutting into the oxide on the tape and ruining the material. The pinch roller (puck), on the other hand, is made of rubber; it just plain dies in time. If it gets too shiny and hard, it can't pull the tape effectively and causes speed problems and increases wow and flutter.

So replace them — but remember that parts aren't usually interchangeable between brands. Make sure to buy only OEM (original equipment manufacturer) parts. Most pieces of gear are standard in their operation and their components. Major manufacturers such as Ampex, Soundcraft, Studer-Revox, Otari and TEAC all maintain a very good inventory of parts for even their oldest machines. You can order them from your dealer or directly from these companies.

PREVENTATIVE MEDICINE

Keeping your machine clean helps stretch its life — and a good cleaning is always the first step in a restoration job. Swab everything that touches the tape with a cleaner like 'Formula 409' (a real favorite of mine) or ethanol (no water). Some manufacturers sell rubber cleaner and head cleaner that work as well. Don't use anything abrasive or that leaves a residue. Compressed air can also go a long way; buy an environmentally safe variety or rent a bottle and regulator of compressed CO2 or nitrogen from your local gas supply company. Finally, check with your machine's manufacturer and see what kind of oil they recommend for the capstan. Not just any lube will do. It has to be very fine and light.

Once you've got the machine clean and you've insured a smooth tape path, it's time to go inside and mess with heads and transistors. These operations are model-specific (and should only be done once your warranty is expired); we'll look at the TEAC 80-8 and the Otari 5050B - two of the most popular vintage workhorses that exist.

TWEAK THE TEAC

The TEAC Model 80-8 is a great example of a machine that still has a lot of usefulness left in it. Usually it's the heads that degrade its performance most. The original heads are good in design, but have a tendency to wear out after 1800 hours of use. That's about 225 eight-hour days. Gently touch the heads with your fingernail. If you feel a lip or a groove on either the top or bottom where the tape does not touch, chances are the head needs to be refurbished or replaced.

In the 80-8, and similar machines like the Model 36 TEAC and the 3340, it is probably cheaper and more efficient use of time to put on new heads. These can be purchased from TEAC, but I prefer to get them from JRF Magnetics in Greendell, New Jersey or from Saki Magnetics in Calabassas, California. JRF and Saki heads are excellent and can add some transparency to the high end and a little more tightness in the bottom. Their cost versus TEAC can help your bot-

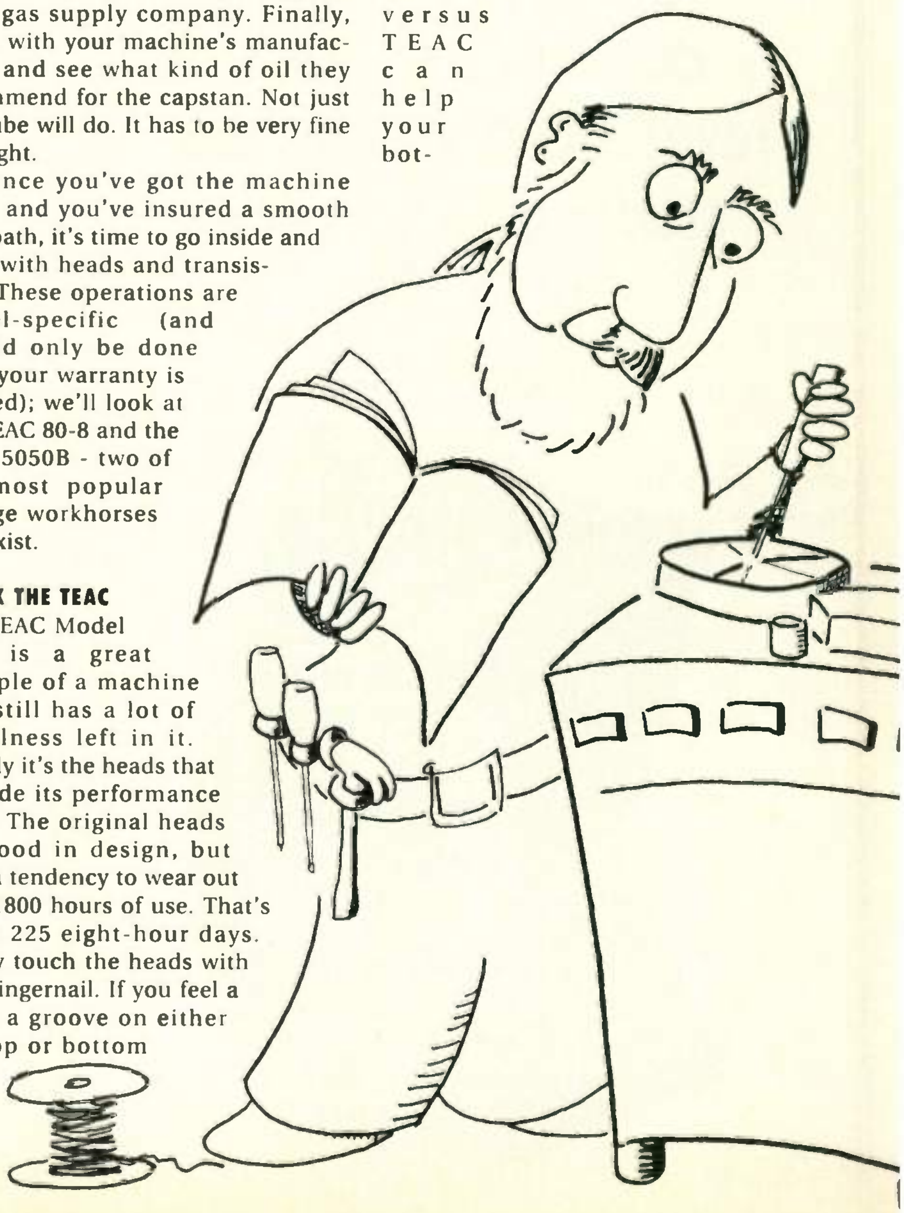

Illustration by Milton Reyman

tom line.

The electronics can be made better with a little work and patience. The original IC's that come with the machine are (2) RC4558 dual op amps per Rec/Play Amp. They are O.K., but for a bit better noise figure overall and a cleaner frequency response, I replace them with NE5532AN op amps in machined IC sockets. An 'AN' denotes a more select, matched, and tested variety of 5532. Besides the sonic improvements, the sockets allow you to be able to change an IC quickly if one should go out. The machined pin sockets are a bit more expensive than the pressure variety, but chips stay seated in them and do not corrode the contacts of the IC.

When you do all this you have a great 8-track that will serve you many more years with increased sonic performance.

There is a truly wonderful modification to the EQ circuits that is possible with the 80-8. With this mod it is possible to get the frequency response a lot flatter and to have more latitude in the adjustment. It's complicated and time-consuming, and requires changing some wiring on the PCB and adding some components (resistors and capacitors). If you want to try it, contact the inventor Greg Hanks (a real technical whiz), at New York Technical Support (914) 238-4171, for information.

When you do all this you have a great 8-track that will serve you many more years with increased sonic performance.

TWO TRACKS

On an Otari 5050B 2-track — also an excellent workhorse — usually the only things that need replacing are the pinch roller, tape lifters, and the heads. You actually have to desolder the heads that are installed and make sure that you put the wires back in the same polarity. This procedure is best accomplished after a good night's sleep — and some thorough technical training. The heads are really excellent from Otari and are fairly inexpensive (they run about $70 apiece).

If you decide to replace the heads, you should have a good shop available. A reliable dual trace oscilloscope is mandatory, as well as metric hex drivers and #1 and #2 Phillips. Beware of the driver: Phillips can damage Otari parts. For the best fit, call Otari for their screwdriver set (part number ZA-53A) and hex-key set (ZA-51A). Demag the tools before you start. Otari sells an ink (lovely red) that you apply to the heads and then run the tape over to check the physical alignment. Great stuff, this Japanese postal ink (even if the box is in Japanese).

The only real improvement to the electronics that should be made is the replacement of the rec/repro relays on the motherboard. There are two relays marked R301 and R401 that are usually manufactured by Omcron, but I don't recommend them. They aren't sealed and the contacts wear quickly. I replaced them with an IDEC or Aromat gas-filled, bifurcated (two contacts), sealed, 24-volt relays. This relay will fit into a modified 16-pin DIP socket, but they last a long while. If your 5050 exhibits drifting and wavering tones, or is a little noisy, the relays are probably at fault. They cost about $7 each and are easy to install. Otari also sells sealed relays (number RY2DC087). Remember that some soldering finesse is required to do this operation.

These and other tweaks can keep old equipment in use for years with much satisfaction. If you think that your equipment is not performing up to snuff, maybe a few parts can bring back its days of glory.

EDMUNDS

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So my approach to guitars in my project studio isn't really very different than what I'd do in a commercial studio. When I produced the last Dion album for Arista, all the guitars were recorded using an old '50s tweed Twin. I stuck it in an iso booth and didn't even change the settings. I had the guitar in the control room, and to get a new sound I'd just switch pickups or tweak whatever I had to on the board. Probably the hardest thing about getting a good guitar sound in a personal studio is that you can't monitor it in a separate control room like you can at a commercial facility. But on the positive side, once you find an amp, a mic and an angle that make the guitar sound great, you can just leave it there forever.

I would definitely consider cutting a master at home.

I generally don't print separate room ambience tracks for guitar or drums, by the way. I always get ambience from a device. My main reverb is an Alesis Quadraverb. In situations where I'm using an effects device for just one track — as opposed to using it for a general ambience on several tracks — I'll usually just print the effect. That's smarter than using up a whole effects send just for one track. And it's good to commit to a particular effect on a particular sound. Lock it in rather than putting off all your decisions until you get to the final mix and you're so confused you need a computer to sort things out.

BEYOND DEMOS

I'm currently looking at ways of transferring all my virtual and tape tracks to digital multitrack. So if I need to go into a commercial studio with a project, I can do it without moving every single piece of gear. That's the general plan for the album I just started working on. I'll take things as far as I can and then bring it all into a commercial facility. But then again, who knows? I originally assembled this project studio just to do demos. But I'm rapidly finding its usefulness extends far beyond that. For a rock and roll song, I would definitely consider cutting a master right at home.

EQ&A

continued from page 13

— it may be better to stick to the dynamic transducers for that bit of extra ruggedness they so often provide.

You will also need to pay attention to the microphone's directional characteristics. Sticking to a fairly tight pickup pattern such as hypercardioid will cut down on leakage from the wall of brass instruments around you in a polka band. Further, try to get a mic with a flat frequency response, and pay special attention the the mic's proximity effect. Placing certain mics inside the bell of a tuba will result in a rise in the low frequencies: something you generally don't want from a tuba. Try working with a mic stand (or extended gooseneck attachment) and the mic at least 18 to 24 inches away, about one-third off center. However, if you must move around to that polka beat, understand that quality will be compromised as a result.

Before miking any instrument (or vocalist) it's necessary to have a thorough understanding of that instrument's tonal range. In addition, we also need to know the musical style being played and the capabilities of the player. The tuba has a fundamental range of about 40 Hz - 2.5 kHz. Overtones will carry its range slightly further, and bass tubas will reach down even further.

Still, with such a limited range, it doesn't make sense to mike the tuba with a mic made for the broader range of a vocalist, piano, etc. Choose mics which have the ability to reproduce (and withstand) the lower end of the spectrum.

Some mics you might consider are the Electro-Voice RE-20, RE-27, PL-20 and ND-308A; AKG D-12 and C-409; Sennheiser MD 421; Peavey PVM 580TN and PVM 520TN. All are depended upon to reproduce the sonic qualities of low-frequency instruments. Nady and Samson will have capsules for some of these mics should you want to go wireless for extra convenience and mobility.

There are undoubtedly other microphones that are capable of faithfully capturing the sound of the tuba, so I invite manufacturers to write to us and share that information with our readers.

Hector G. La Torre,
Executive Director, EQ Magazine

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CIRCLE 43 ON FREE INFO CARD

MARQUEE

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PL8 Power Line conditioner with light on top of house electronics and effects racks.

DISCO SYSTEM: Two Technics SL-1200 MkII turntables, Rane MP24 DJ mixer.

EQUIPMENT NOTES: "I've got three Soundcraft Series 500 consoles in stock because I really like their sound. The best aspect of the Soundcraft board is the way its EQ's sweep — it's really musical. It also has six effects sends, and I use them all. And with its eight subgroups I get excellent separation."

PROJECT TIPS: "The system has to sound good consistently, and it has to be reliable. If it isn't reliable, it isn't worth anything to you. It has to be there for you every night. So try to stick with equipment from brands you're 'comfortable' with, because you want to rely on your gear without nightmares. Nothing is worse than having things go wrong during the middle of a gig. The whole audience turns to you; it's no fun, believe me.

"And be prepared for anything; weird things can happen at clubs," Bondy adds."When you go into a club to engineer or play a gig, trust the house engineer. I can't stress that point enough. The house mixer is the one. He knows the deal with the mix, the volume and the EQ that's right for the house.

Also, make sure that you always call the club in advance and find out beforehand what effects and outboard gear are going to be there. If you need to bring some of your own outboard gear, let the engineer know in advance. Try to check out a club before the show: forewarned is forearmed. And take your job seriously, but not *too* seriously."

> Weird things can happen at clubs. Nothing is worse than having things go wrong; the whole audience turns to you. You have to be prepared for anything.

SELLING SOUNDS

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needed in the monitor mix as well as the entangled lead guitarist. It'll make any sound check or show run more smoothly.

■ Keep staff at home. If you're touring, you must have staff at home in charge of running the business, paying bills and and booking future tours while technicians and sound system(s) are out on the road.

■ Building a network of contacts with other regional companies that carry the same system allows you to hire yourself out for the occasional show in a big venue.

Before building your sound system, consider standardizing loudspeakers, rigging, cables, connectors, hardware, etc. so that you can interface easily with the equipment of your peers in other regions. They may need to rent supplemental loudspeakers and amp racks from you some day and if all your systems are not out on the road, why not?

Equipment doesn't make much money in the warehouse. With a well established network you can bid on tours that may include an odd venue that is larger than you can ordinarily handle.

KEEP ON ROLLIN'

Half the battle is being a sensible troubleshooter, catching things before they fail. It's also important to know how to ship parts overnight because you could never carry a spare for every item in the system. A good deal of these qualities come with experience, of course.

So give yourself a break and don't expect to know everything about everything! It's impossible; people are specialized in this life. Find those helpmates with expertise in the fields where you don't. If you're going to fly loudspeakers, get the assistance of an experienced theatrical rigger. Good connections really are everything.

Make it a special point to meet the promoters that will form your client base. When you get together, sit down and discuss their needs and concerns in depth. Their needs always have to come first. This path will lead you to a more rounded outlook on your business, a few new friends and maybe bring some new business your way as well.

My thanks for the observations and recollections of Jack Jamieson, Singer Hall, Bob Humphreys, Sun Sound and Clive Alcock, Allstar Sound.

The World According To Woram

Everything you have to know but were too scared of physics to ask.

JOHN WORAM

Photo by Peter Monroe

"If you want to break the rules, first you have to know what they are."

I don't know who said that. It doesn't show up in my *Bartlett's Familiar Quotations*. But with or without the proper credit, the quote is a good way to start off this new column, since the subject here will more often than not have something to do with the "rules" of sound recording, or maybe just of audio in general.

If sampling and fancy software are haute cuisine, this column is going to be meat and potatoes. But every now and then a basic meal is the best bill of fare. And the more complex technology becomes, the more those basics seem to matter.

SONIC STONE AGE

Believe me, I know. Way back in the sonic stone age (early seventies), I wrote a book about sound recording. I had just left a chief engineer gig at Vanguard Records and had to get another real job real fast, or find some other way to make a living. I decided to take some time off and write a book.

It stayed in print for many years, and although the basic principles remained valid, the hardware changed almost beyond recognition. By the late eighties, it was time for a completely new book, if only to replace photos of ancient gear. So I decided to do it all over again, this time taking advantage of the by-then ubiquitous personal computer.

In pre-PC days, the line between the recording engineer and the "real" engineer (the one with a degree, a slide rule and a plastic pocket protector) was easy to see, but hard to step over. The academic type spoke his own language and was generally regarded with some suspicion — often deserved — by everyone else. If he wrote a learned paper, in which the conclusion appeared as an equation with six variables, the rest of us would be left pretty much in the dark.

PC: PRE COMPUTER

The PC changed all that. Well, if not all of it, then maybe at least some of it. Given a fair-to-middlin' system and a little programming time, it's reasonably easy to answer all sorts of "what happens if ... " questions. For the purposes of a new book, that meant I could now explain in all the gory detail some of the "what happens if ..." of pro audio. And since you can't answer every possible question in print, I showed some of the math that, in an earlier day, would have lurked invisibly behind the text. To have left it out would have been akin to writing a music theory book without showing the reader those little black things with the stems on them.

Last September, I went out to L.A. for the AES show, and wasted no opportunity to plug my latest opus to all the teachers at the convention's "Education Fair." I'd like to report that I was overwhelmed by the unanimously enthusiastic response to the book, but it wasn't quite like that. Some just loved it and some even said they were using it in class, but others said it was just too complex for the students. To which I would say something clever, like "Hey, I don't make this stuff up. I just cover what needs to be covered." After a few encounters, I could anticipate the response: "Right. You know that, and I know that. But the students don't want that math and physics stuff. They're into recording, not rocket science."

FATHER TIME

And that brings us up to late January, 1991, and a phone call from EQ. They wanted a basic audio column. I say, "That's nice. Keep looking." But those editors don't give up easy. "If you can write a whole damn book about the stuff," I was asked, "how come you can't handle a little column?" So I explain the facts of life. Especially the part about people not wanting to know what they need to know. They tell me to read the December issue, where J. D. Sharp actually mentions (gasp!) technology for musicians. Sharp makes a few observations about owner's manuals, finding many of the latter "useless to anyone without engineering experience ... might as well have been written in Sanskrit."

Well, J. D., I think you're being a bit too kind: some of

If sampling and fancy software are haute cuisine, this column is going to be meat & potatoes. But every now and then a plain meal is the best bill of fare.

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Ensoniq EPS 16 Plus

EQ LAB REPORT

MANUFACTURER: Ensoniq, 155 Great Valley Parkway, Malvern, PA 19355. Tel: 215-647-3930.

APPLICATION: Sampling keyboard with onboard sequencer and modulatable signal processing. Has the sound quality and memory required for flying in vocals and doing general mix-fixing.

SUMMARY: The EPS has been an extremely popular sampler. The 16 Plus cleans up the sound and adds a whole battery of new features.

PRICE: $2,395 (keyboard), $2,495 (rack; includes 2 megs of RAM and four stereo outputs)

FREE LIT NUMBER: 140

ENSONIQ SAMPLERS HAVE always had a reputation for innovation, comprehensive features, and cost-effectiveness, although the tradeoff for the low price tag was sound quality that didn't quite match that of the "big guys." Well, this product is going to do a lot to change the company's image. The EPS 16 Plus retains all the features that made the EPS a best-seller, but adds an overlay of impeccable sound quality, major-league built-in signal processing, and the granting of just about every wish — some major, some minor — on a diehard EPS owner's wish list.

Although the EPS 16 Plus retains its predecessor's prowess on-stage, Ensoniq has now produced a sampler that can hold its own against tough competition in the most demanding studios. Yes, you really can think of it as a digital recorder that just happens to be controlled by something that looks like a keyboard — fly in vocals and do drum replacements without the word "compromise" crossing your mind once. And the EPS 16 Plus is still cost-effective.

Long-time EPS users needn't worry about learning a new instrument from scratch. The logical operating system has been retained and most of the functions are similar — so similar at first glance, in fact, you might think this is just an EPS with 16-bit sound. Further investigation, however, shows that the EPS 16 Plus is the exact opposite of old wine in new bottles: the bottle may look the same, but the wine is something else altogether.

EPS BASICS

For those not familiar with the original EPS, here's a rundown of the basic features on both machines. The EPS allows for eight separate instruments (guitar, flute, drum kit, etc.); each one can have eight layers. Each layer can be multisampled or a single sample. Combining layers produces doubling, chorusing, octave, and other effects; two front panel patch buttons allow any four combinations of layers to be called up in real time. For example, with a lead guitar patch, one patch button could turn off the normal layer and turn on a layer with feedback samples. Instruments can also be doubled, arranged as splits, have individual levels, and so on, with the configuration savable as a performance preset.

The synthesis capabilities are outstanding. Polyphonic key pressure (which provides individual aftertouch data for each key) allows for exceptional expressiveness, and the keyboard has lost the "clack" of the early EPS. The four-pole digital filters (non-resonant, unfortunately) can split into two individual filter sections, allowing for highpass, lowpass, bandpass, and lowpass/highpass combinations; each section can be individually modulated.

The envelopes offer both low velocity and high velocity curves, with interpolation occurring at intermediate velocity values. Modulation follows a "matrix modulation" routing system where different modulation inputs (filter cutoff, oscillator pitch, LFO rate, level, etc.) can be assigned to any of several modulation sources (three envelopes, LFO, pressure, random LFO, three velocity curves, note position, pitch wheel, mod wheel, pedal, external controller, etc.).

Sampling is easy; the fluorescent display shows signal level (although levels must be adjusted at the source since the EPS doesn't have an input level control), and when multisampling, the EPS automatically chooses split points, which can be edited later.

> Although the EPS 16 Plus retains its predecessor's prowess on-stage, it can hold its own against tough competition in the most demanding studios.

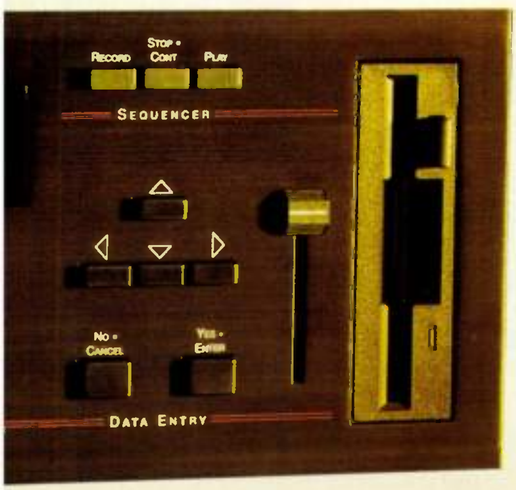
After sampling, there are a ton of looping options, gain normalization, mixing, fading, and other digital fun and games.

If you're not into sampling, the EPS 16 comes with 15 sound disks and can translate Mirage DSK and EPS sound disks.

Interestingly, Ensoniq-generated EPS factory disks are inherently 16-bit and will play back with full 16-bit sound on the EPS 16. Like the EPS, the EPS 16 can load sounds in the background, while you play.

What's new? Aside from 16 bits, there's internal signal processing similar to the Ensoniq VFX synthesizer. Samples can be resampled using effected sounds, thus freeing up the effects to do something else. Each instrument can have its own effect parameters, or a bank of instruments can have a global effect. What's more, the effects algorithms are updatable, so future sound releases can include new effects algorithms and/or combinations.

Current effects include 13 different combinations of reverb, delay, rotating speaker, chorus, flanger, phaser, and for guitar sounds, compression, distortion, and wa. Different "modules" within the effect are routed to different buses/mixers, depending on the algorithm being used, to allow for various blends. Most effects allow one of the parameters to be modulated by a modulation source, thus integrating the effects into the patch.



The sequencer has been enhanced to 96 PPQN resolution, and when constructing a song out of sequences, eight more song-length tracks are available. For those who take the EPS out on gigs and use it as a master sequencer, but edit their sequences on computer at home, version 1.1 system software offers a multitrack recording mode for transferring sequences over in one pass.

The EPS 16 Plus has three different switchable playback rates that also affect polyphony: 78 kHz (7 voices), 44 kHz (13 voices), and 30 kHz (20 voices). The difference in sound quality isn't as noticeable as one might expect. One reverb and one delay effect automatically select 44 kHz.

ACCESSORIES

Like the original EPS, the EPS 16 is expandable — but the accessories really drive up the base price. The OEX-6 output expander ($249.95) provides 6 more outputs for a total of 4 stereo pairs or 8 mono outs. The ME-16 Plus memory cartridge ($229.95) doubles the onboard RAM to 2 megs and is also used with the SP-2 SCSI adapter ($199.95). SCSI is a high-speed bus that lets you hook the EPS up to hard drives, CD-ROM players, and computers (Alchemy 2.22 works fine with the EPS 16 Plus over SCSI). "Flashbank"—batteryless, nonvolatile memory — can also be retrofitted ($349.95 for 512K, $649.95 for 1 meg) to give three storage options: onboard floppy disk, SCSI, or Flashbank.

The Flashbank works fine (though you can't save bank data, only individual instruments), as does SCSI. Speaking of things that work, I was concerned that Mono Modes A and B, which never worked with my Quantar MIDI guitar, would still be problematic, but they work perfectly on the EPS 16.

What's missing? Stereo sampling, although stereo samples can be transferred to the EPS 16 through sample-editing programs. I would have preferred a 4 or 8 meg memory limit. One minor annoyance: the sequencer still can't record incoming program changes (although you can add them manually).

Sound quality (as evidenced by the factory disks), polyphonic pressure, excellent sequencer, signal processing, conscientious MIDI implementation — Ensoniq has been doing samplers/workstations for so long they've gotten really good at it. The EPS 16 is a welcome encore to the EPS that lowers the admission price to high-end sampling.

— Craig Anderton

DigiTech IPS 33B

EQ LAB REPORT

MANUFACTURER: DigiTech, 5639 South Riley Lane, Salt Lake City, UT 84107. Tel: 801-268-8400.

APPLICATION: Generate diatonically correct harmonies according to scale or chord type from monophonic audio inputs. Can also produce parallel harmonies with polyphonic material.

SUMMARY: While specialized, the IPS 33B is novel and useful. It's particularly well-suited to smaller studios and live performance.

PRICE: $899.95

FREE LIT NUMBER: 112

THERE ARE A LOT of multieffects boxes on the market with reverb, other time-delay functions (chorus, flange, delay), and the occasional EQ or fuzz tacked on for good measure. Well, the IPS 33B is most definitely a multieffects box, but it doesn't follow tradition at all. Whereas most multieffects use reverb as their centerpiece, the IPS 33B is at heart a pitch shifter. It does include chorus and delay options, as well as some sophisticated MIDI control options, but the IPS 33B is unlike any other signal processor on the market for under $1,000.

For one thing, it's a vocalist's delight. When you sing into it, up to two independent higher or lower har-

IN REVIEW

mony lines play along. The same thing works with a monophonic instrumental line. These aren't parallel harmonies as found on most pitch shifters, but diatonically correct harmonies that follow major, minor, mixolydian, dorian, lydian, blues, etc. or custom user-programmable scales. It can also arpeggiate harmonies (chromatically or according to a particular scale) or have them fit the notes of a specified chord. There's even a pitch correction option if your singer's out of tune.

In addition to intelligent harmonization, the IPS 33B provides (for each harmony line) delays of up to 1.5 seconds, variable delay feedback, programmable level, chorusing, vibrato, and tremolo. Furthermore, the sound quality is remarkably smooth; since this real-time sampler "knows" harmonic rules, it finds a pitch-related "splice" point.

Its biggest drawback is that intelligent harmonization works only with single-note sound sources like voice, sax, flute, lead guitar, bass, a drum, etc. Lead guitar works very well — if you've ever wanted Allman Brothers-style twin leads from a single guitar, this box will do it — but you have to pick cleanly to avoid hitting more than one note at a time. Chords work with chromatic harmonies, although the sound "warbles" a bit (as we've come to expect from pitch shifters).

BASICS

The IPS 33B is a 1U rackmount device. There are three front panel controls (input level, output level, and data entry/mix), 11 switches for programming, and three displays (32-character backlit LCD for programming, LED program readout so you can see the selected program at a distance, and LED meter for setting levels).

Turning to the rear panel, the unbalanced 1/4-inch phone input jack has a 220kΩ impedance, making it sensitive enough for most stock guitars. There are three 1/4-inch output jacks for dry, left harmony, and right harmony signals; plugging into only the left jack sums all three. There are two jacks for continuous control via footpedal, connectors for the included remotes (hand control and footswitch), and MIDI in, out, and thru. A level setting switch accommodates -20 dB or +4 dB signals.

TOTAL CONTROL

Using the IPS 33B at its most basic level simply involves calling up a preset from the total of 128 factory and 128 user-programmable presets, then specifying the key if the preset uses intelligent harmonization. Selecting a preset can be done in several ways: from the front panel, via the remote footswitch or hand control, or by playing an appropriate note on a guitar neck while holding down one of the footswitches. The same techniques can be used for key-setting, with the additional option of changing key according to whatever note appears at the MIDI in.

Several parameters (pitch bend, input level, output level, mix, tremolo depth, vibrato depth, and vibrato speed) can be controlled via footpedal or MIDI continuous controller commands. Space prevents listing all the programming options, but they're extensive. Programming follows the usual scroll-through-pages-and-adjust-parameters protocol, and is not difficult.

OVERALL

The IPS 33B does harmonies, not miracles, and there are certain sonic limits. Transposing up gives "munchkinization" effects, and transposing down gives a Darth Vader-like boominess. As a result, I generally mix the harmony line at around 35 percent of the straight signal; this blends the two together into one big sound, rather than sounding like two individual signals. When working with female vocalists, whose higher pitch range often suffers most with munchkinization, try rolling off the highs on the harmony outputs with external EQ. Also, each harmony line has an inherent delay (estimated at somewhere around 10 milliseconds) due to the time required to analyze a note prior to generating a harmony.

Several months before being asked to write this review, I had purchased an IPS 33B primarily for vocal use in live performance. Live, the IPS is a superb all-around vocal processor (when not harmonizing, I use the chorus settings to thicken my voice) that can greatly enhance vocals. I sync program changes and controllers to a sequencer, so vocal processing changes on the fly — great fun. If I could afford two, I'd throw one on my guitar as well.

Of course, a singer singing a harmony or guitarist overdubbing a harmony will always sound better (timbrally, at least) than a synthesized harmony, so if you have the tracks, it's better to do an overdub with a real musician. And if you already have a complement of boxes that provide time-based processing, you probably have no further need for echo and chorus effects.

Nevertheless, the IPS 33B can be a lifesaver. If you run out of tracks and don't have space left for a vocal harmony or harmony guitar part, use the IPS during mixdown. Those who sing live over "virtual tracks" into a DAT or other mastering deck can use the IPS to fill out the vocal sound in real time. In a pinch, it's a delay line and chorusing unit.

There are a lot of signal processors out there, but the IPS 33B is a novel and useful effect. It takes some practice to get the most out of it (and a working knowledge of harmony is almost a prerequisite), but the results justify the effort. When it's time to play a gig, I never leave home without it.

–Craig Anderton

WORAM

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them are written in Sanskrit. Or maybe it's Serbo-Croatian.

The sorry state of documentation in general is so well known that not much more needs to be said here. Except for one little thing. How the hell do you explain some tough technical topic (MIDI, digital audio, magnetic recording, you name it) to someone who doesn't want to meet the subject half way? Here's a quickie example: in that same December issue, Tom Lubin says that if you move a mic close to a speaker you'll get a low-end boost, or proximity effect.

MIKING THE UNKNOWN

Well, you can take his word for it if you like (Lubin is a reasonably honest guy, even if we don't know why he really moved to Australia). You can also try it on your own, and you might even come up with the same results. Or you might not. It depends on the mic you use. If you have a bunch of mics, you can try them all, and then try to remember which one gives the best (or worst, depending on whether you need it or not) proximity effect. Then you can go to someone else's place and encounter a completely new set of mics. Which one should you use? If you have unlimited time, you can do a bunch of tests to find out. Or you could know a little something about what to expect before you even pick up an unknown microphone.

Which brings us back to basics, and to this column (they just don't take no for an answer at EQ). Starting next issue, Basics will start looking at some of the dull stuff that anyone who wants to make a long-term living in or around the audio industry needs to know. However, it may not help much with those Sanskrit translations. But if you want to answer your own "what happens if ..." questions, stick around. And ask questions.

John Woram *is the author of the critically-acclaimed industry reference "Sound Recording Handbook," as well as the editor of the John Woram Audio Series from Howard W. Sams & Company. A frequent lecturer, Woram is the former director of the University of Miami Music Engineering Program.*

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"Right there, that bar sounded good." But alas, the next bar came and went with a different slop, and the "feel" was different. We ended up abandoning the second tune until a later date. I am spending my spare time looking for machines to sync up that don't wander or are at least predictably repeatable (is that English?).

The MPC60 is not the only machine that was SMPTE controlled during this ordeal. The bass parts were played with an AKAI S1000 and S1100 samplers. For future reference, there is a 5.1 millisecond delay from the time the MIDI note is sent to the S1000 or S1100 and the sound actually comes out of the sequencer. This was corrected for by advancing the MIDI trigger and delaying the SMPTE.

The birth of a new Donald Fagen album will be worth the effort, I just wish we could keep the gestation period down to something a little more manageable.

Remember that I mentioned a real drummer was supposed to play this sequencer? Well Chris Parker came in to play the drums. We recorded about a zillion passes of Chris playing along with the sequencer. After Chris was safely on a plane back to New York, we took his drum track apart piece by piece. Mind you, the performance wasn't bad, he was just half a millisecond late here, a millisecond early there, you know, the usual stuff. What he played on the intro, bridges and fade were actually kept intact. All we really manufactured were the verses. We took a piece of hi hat pattern from one verse, another piece of hi hat from another verse, some cymbal crashes from somewhere else, a bunch of snare hits from all over the place, and put them all together in a sequence and made them match the drum machine pattern.

The sampler that we used for this part of the sequencing was an Akai DD1000 optical disk recorder. It will trigger sounds from MIDI or SMPTE, and the sounds can be stereo up to 30 minutes long. We triggered some sounds from MIDI notes generated by the MPC60, while other segments were triggered directly from SMPTE event cues. The MIDI delays were about the same as the S1000 family, but the SMPTE firing delays were more like 33 milliseconds. In order to get a sample to come out in the right place, you had to shift the SMPTE event time 2.1 frames earlier. Oh well, what's another couple of increments among friends?

Well we got part of one track done (drums and real Rhodes played by Donald). Maybe by March, when we get back in to do some more, I will forget how painful the increments were. The birth of a new Donald Fagen album will be worth the effort, I just wish we could keep the gestation period down to something a little more manageable. These albums have been known for pushing the outer limits of the envelope. By the time this album comes out it may be about 5 milliseconds ahead of its time.

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This Won't Take Long, Did It?

Labor Pains and digitally recording the new Donald Fagen.

ROGER NICHOLS

Courtesy of Rupert Neve

Women often accuse their husbands of having no idea what it is like to get pregnant and to give birth. They also talk about the fact that after the birth, nature somehow makes them forget about all the pain and discomfort so that they will want to become pregnant and give birth again. Well, I beg to differ with this point of view. Those women never recorded a Steely Dan album. Nine or more months of gestation followed by intense labor (mixing) and finally the birth of each album. A short time period would elapse and we would be ready to jump right into the next one.

I just got back from a month working on the beginning of a new Donald Fagen album that Walter Becker is producing. It's been eight years since the last Donald Fagen album, and I guess I forgot about the surgical precision with which Donald approaches his tunes. Squeezing 110 percent out of the machines and musicians like water squeezed from a stone.

HELL'S BELLS

This last month was sequencing and synchronizing hell. The game plan was that Donald had a sequence of his tunes in an Akai MPC60 (empathy-sixty). The sequence contained a drum part with percussion, a MIDI bass line and a DX-7 Rhodes part. We would print that sequence on a digital multitrack as a guide for a real drummer, and later on other real musicians, to play to.

The first task was to get the machine sequence to "feel" right. Donald didn't particularly like where the Akai placed the beats in a pattern, so he had each output going through a digital delay so that he could move the beats around. That is, he could make the snare drum a little more laid back by adding 7.3 (or whatever) milliseconds of delay to it. If he wanted the hi hat to come earlier, he would move it one increment earlier in the sequence, and then delay it back to where it was originally less the amount of advance that he wanted. Most of the time he was working with increments of one tenth of a millisecond, but once in a while we would have to whip out a delay that allowed changes as small as 20 microseconds.

BITS AND BEATS

After Donald decided where he liked all of the beats to fall, I would then print the drum sequence to tape. Not wanting the already pusillanimous sounding samples to deteriorate any further by going through the delays, we had to come up with a way to get the drum machine locked up with the right offset so that the audio could go straight to the digital multitrack. The easiest way to do it was to run the SMPTE from the multitrack through the digital delay that we were using for that sound. That is, if we were using the Roland SDE-3000 to delay the kick drum by 9 milliseconds, then we would run the SMPTE through the Roland SDE-3000 set at 9 milliseconds and then into the SMPTE IN of the MPC60. SMPTE through delay works just fine. The frequency of a SMPTE signal is about 2400 Hz plus overtones, which is well within the audio spectrum of a good digital delay. As long as you don't clip the level on the delay or have some weird effects like feedback or chorus dialed in, you should be O.K. On paper this whole scheme looked great.

We all know about MIDI slop. Well, we had SMPTE slop. It turns out that when you sync the MPC60 up to SMPTE and record a pattern of beats onto the multitrack, and then go back and sync up the same pattern from the same SMPTE, there is a variation of up to 1.5 milliseconds either side of the center of where the original beat was. This variation is random and non-repeatable. It also didn't make any difference whether you started from the top of the tune or somewhere in the middle, the results were the same. This sync slop was 75 times greater than the increments that we were using to place the beats originally.

STOP THE SLOP

On one of the tunes that we were working on, the slop seemed to average out, and the "feel" of the tune didn't seem to be adversely affected by the sync variations. On another, however, we weren't so lucky. We played the tune over and over again, trying to line up the increments. As the bars would go by Donald would call out

> It's been 8 years since the last Fagen album, and I forgot about the surgical precision with which he approaches his tunes.

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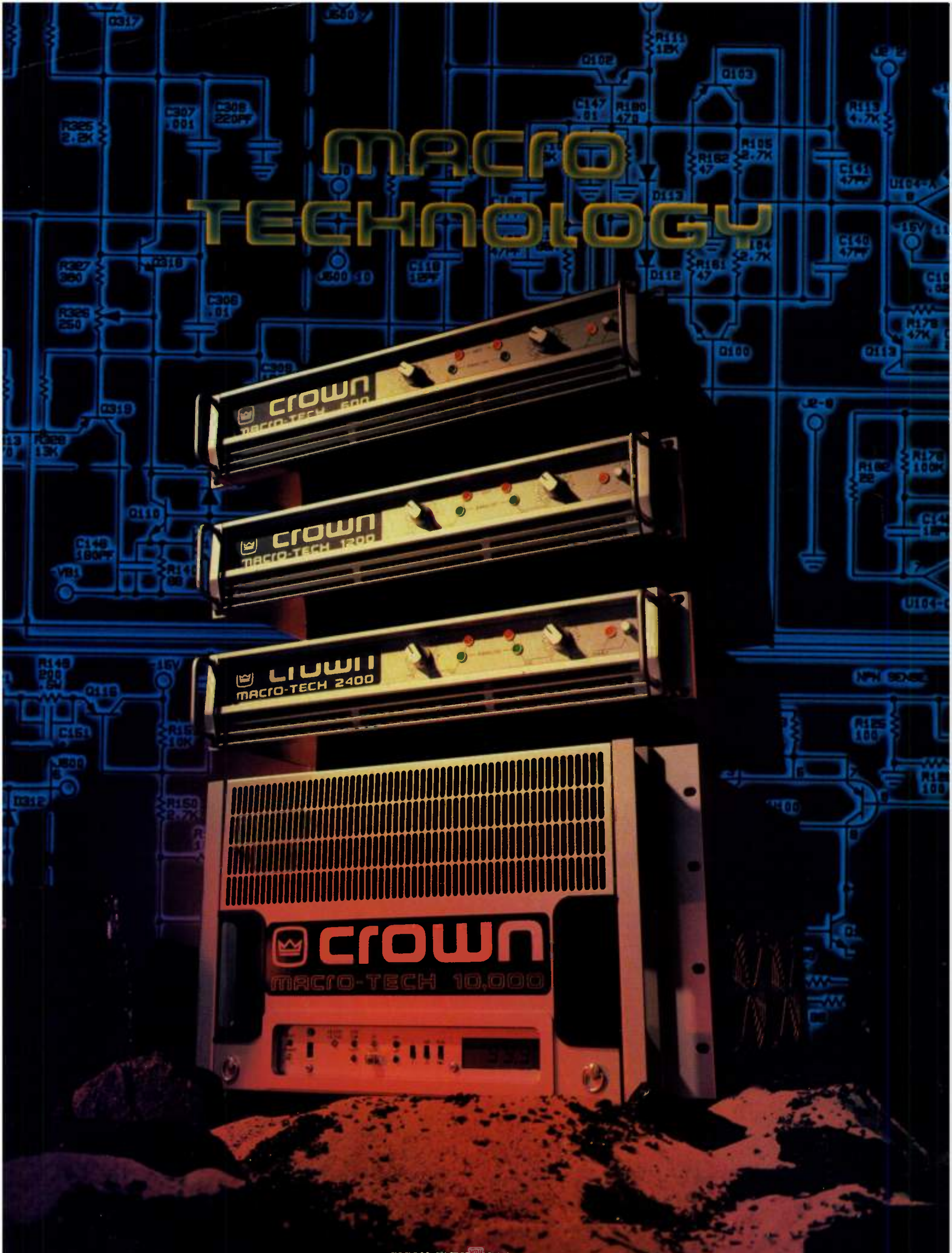
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