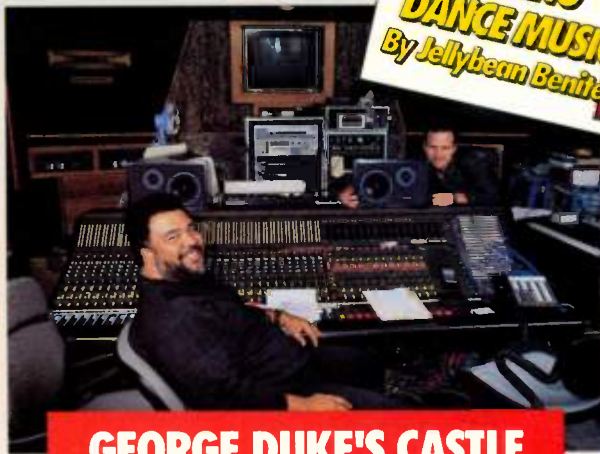
PROJECT RECORDING & SOUND TECH

MIXING DANCE MUSIC By Jellybean Benitez

EQ



GEORGE DUKE'S CASTLE

JUNE 1991 · PSN PUBLICATIONS

TEAR DOWN THE WALLS By Bruce Swedien

EXCLUSIVE PRODUCT REVIEWS
- E-MU PROTEUS
- LEXICON 300
- TASCAM M-25 SERIES

SOUND ON STAGE
- DJ System Design
- Soundman's Tool Kit

PANASONIC PV-3700 REVIEWED

By Bob Ludwig

THE COMPLETE SAMPLER
- SUCCESSFUL SAMPLING by Roger Nichols
- HANDS-ON FOUR HOT SAMPLERS
- SAMPLING & THE LAW

12
0 71896 47438 1
$3.95
Can. $4.95
U.K. £2.50
Display Until 7/91

************************SCF 500
#CRL02561093 8#410588 H3
RON CARLSON M/A 92
ZOOM TAPES-PRODUCTIONS DOME
256 S 1ST ST RR 3 BX 19A#EB03
OGDEN IA 50212-1017

AKG

The Sweet Sounds of Success.

Take a look around the next time you're in the studio. Talk to the most demanding musicians and recording engineers. Ask them what mics and headphones they prefer. You'll find that most of them count on AKG to record and reproduce their music's incredible dynamic range, subtle nuance and emotion.

It's no wonder that 1990 and 1991 *Billboard* surveys found that AKG microphones are the "#1 Most Widely Used Studio Microphones," and that AKG headphones are the "#1 Most Widely Used Studio Headphones."

AKG. We record the sound of success.

AKG Acoustics, Inc.
1525 Alvarado Street, San Leandro
California 94577 USA
Tel: (1) 415/351-3500
Fax: (1) 415/351-0500

© 1991 AKG Acoustics, Inc.
AKG is a registered trademark of Akustische und Kino-Geräte GmbH, Austria.

CIRCLE 05 ON FREE INFO CARD

MORE ARTISTS GO GOLD ON AMPEX THAN ON ALL OTHER TAPES PUT TOGETHER.

Every artist pictured here has earned the prestigious Ampex Golden Reel Award for creating a gold album exclusively on Ampex audio tape. Find out what makes Ampex tape right for your sound. Just call or write for a copy of our new 456 Technical Brochure, and see why Grand Master® 456 is engineered like no other tape in the world.

AMPEX

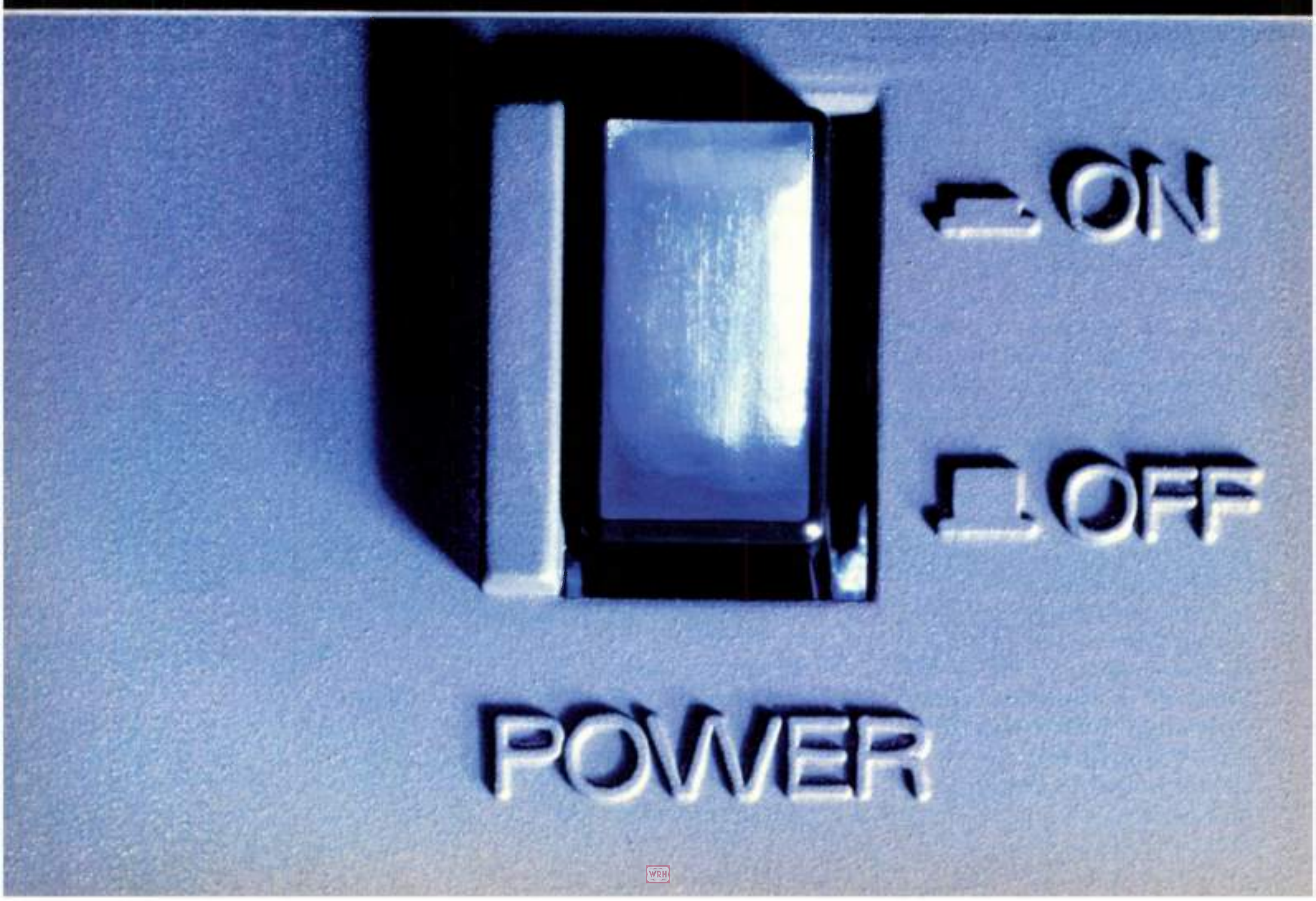
A MASTER OF ENGINEERING

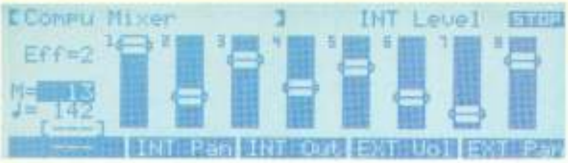
Ampex Recording Media Corporation • 401 Broadway, M.S. 22-02A1 • Redwood City, California 94603 • (415) 367-3809

CIRCLE 02 ON FREE INFO CARD

WRH

Once you've mastered this sophisticated function, dozens more are a piece of cake.





Studio M's Compu-mixer records MIDI volume levels and pan settings for internal and external parts.

If you'd like to produce professional quality music without first obtaining an advanced degree in rocket science, you've come to the right ad.

Introducing Studio M, a remarkable little powerhouse that integrates a sequencer, sound module and Compu-mixer into a versatile music production system. But just because Studio M is engineered to be incredibly simple to use doesn't mean it's simple-minded.

After all, in addition to its seamless integrated design and helpful displays, Studio M is 8-part multi-timbral and offers 30-voice polyphony. So you'll be able to play a lot of sounds at once. And you'll have lots of them to play with, since 220 sampled sounds are built in.

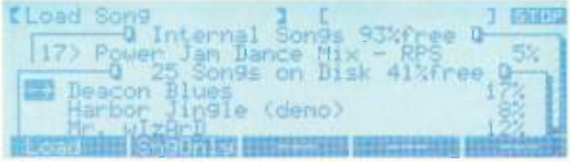
Call up the Timbre Edit display, and you'll have full control over envelopes, LFOs and multi-mode filters.

And whether we're talking about Studio M's great piano, drum and synth sounds—or any of its other sounds, for that matter—they're always easy to access. And even easier to edit. You'll also have built-in effects to add reverb, delay or chorus.

You can forget about MIDI channels with Studio M. To sequence: Pick a track, choose a sound and go.

Then there's Studio M's 16-track sequencer, which will allow you to choose the sequencing style—pattern, linear or loop—and then control up to 136 internal and external parts. When you combine this with Studio M's more than generous 50,000-note memory, we're obviously talking about major composing power.

When you're ready to mix, any adjustments you make using the 8-fader Compu-mixer are stored to provide a powerful auto mix-down function—as you monitor every fader movement on the LCD.



Each song will load along with its proper sounds.

Need more features? To sync to tape players with ease, Studio M offers Tape Sync II. For preserving your creations, the built-in disk drive stores up to 150,000 notes on a single disk. And for adding the newest sounds as your composing needs call for them, Studio M offers two ROM card slots for optional Roland sound cards.

Details, details, details: Studio M's microscope helps you make even the smallest changes to your music.

All of which makes our Studio M the ideal tool for music production at all levels. It serves as a stand-alone music studio or as the core of a larger MIDI system. Providing, of course, that you're up to speed on the function we've highlighted to the left.

Hey, even Studio M can't do everything for you.

Roland®

Roland Corporation US, 7200 Dominion Circle, Los Angeles, CA 90040-3696 213 685-5141

CIRCLE 36 ON FREE INFO CARD

EQ

A PSN Publication
Vol. 2, No. 2
May/June 1991

PAUL G. GALLO
Publisher/Editorial Director

HECTOR LA TORRE
Executive Director

MARTIN PORTER
Executive Editor

J.T. WAY
Editor

JON RUZAN
Senior Editor

CRAIG ANDERTON
West Coast Editor

ANTHONY SAVONA
Assistant Editor

PAULA HEER
Editorial Assistant

MIKE BIRNBAUM, BOB LUDWIG, DR. RICHIE MOORE, ROGER NICHOLS, J.D. SHARP, JOHN WORAM
Contributing Editors

PATRICIA CACCAVALE, JOSH FAGIN
Editorial Interns

MPA EDITORIAL
Editorial/Design Consultants

KATHLEEN MACKAY
National Sales Manager

DANIEL A. HERNANDEZ
Advertising Sales

MATT CHARLES
Advertising Sales

RIVA DANZIG
Art Director

MARK ALHADEFF
Computer Design Consultant

FRED VEGA
Production Manager

Editorial Offices
939 Port Washington Blvd.
Port Washington, NY 11050
Tel: (516) 944-5940, Fax: (516) 767-1745

Administrative/Sales Offices
2 Park Avenue, Suite 1820
New York, NY 10016
Tel: (212) 213-3444, Fax: (212) 213-3484

LETTERS TO EQ

SANGAMO UNVEILED

I am writing in reply to a letter in the April 1991 issue of EQ from some unfortunate who purchased a used Sangamo multitrack recorder. What this machine is (was?) is an FM instrumentation recorder designed to record data from DC to about 10 kHz as most often encountered in scientific research applications. Sangamo Electric is a Japanese company that did maintain American offices, since one of their salesmen once called on me in a former employment situation.

As far as the usefulness of this machine for audio, I wouldn't bet on it. Although the mechanical stability of the transport is excellent, the head assembly is optimized for FM recording and will not like sync reproduce as far as I know. (FM electronics use frequency modulation, instead of the amplitude modulation employed by audio recorders, to extend the low frequency response to DC). The best hope for converting the machine would entail custom installation of audio head stacks (not a cheap proposition) and sixteen audio record/reproduce electronics modules. One might be tempted to try recording audio on the FM electronics, but although the low end frequency response is great, the high frequency cut-off at 15 ips is 10 kHz (intermediate band) and the signal-to-noise ratio is about 45 dB! These beasts are just not designed for audio.

Jay Kadis, Audio Engineer
CCRMA/Music, Stanford University,
Palo Alto, CA

SHARPER IMAGE

Three cheers to J.D. Sharp for "The $5,000 Dream Rack" (April '91). Practical and informative, this is the type of "consumer's guide" article that EQ needs more of.

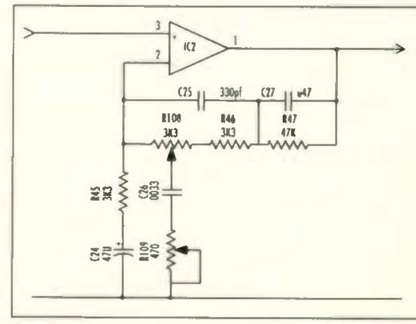
Joel F. Nolte
Arkadelphia, AR

TEAC MOD

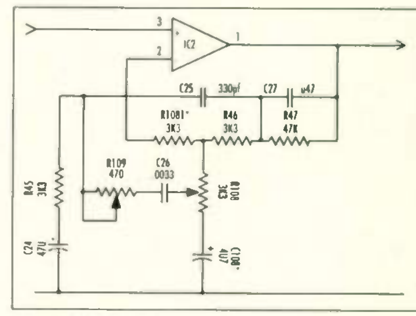
I got so many calls from readers of Richie Moore's "Decks of Glory" (April '91, page 77) about my TEAC 80-8 mod that I couldn't possibly answer them all. So here's the mod for all to read:

1) Remove R108, bend the lead of the pot that goes to R45 straight towards the front, taking care not to break the pot. Use two pairs of long-nose pliers — one for holding, one for bending.

2) Remove R109, and bend the



Original



After Modification

lead that goes to ground in the same manner.

3) Attach one end of a 4.7 microFarad capacitor to the bent lead of R108; this capacitor becomes C108.

4) Attach a 2-inch insulated wire to the bent lead of R109.

5) Re-insert the pots onto the PC board, being sure not to short the bent leads to any old location.

6) Attach the other end of capacitor C108 from R108's lead to the hole from R109.

7) Attach the wire from R109's lead to the hole from R108.

8) Attach a 3K3 resistor (R1081) from the R46 R108 junction, to R45 IC 2 junction.

Note: When adjusting the 80-8, use R109 to affect the 16 to 20 kHz EQ range and R108 for 10 kHz. For optimum noise and headroom, R108

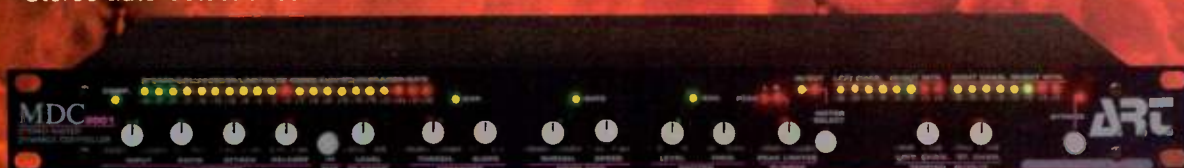
WRITE TO US

EQ wants to dialogue with you. Write to: Letters to the Editor, EQ, 939 Port Washington Blvd., Port Washington, NY 11050. Letters must be signed, and may be edited for clarity and space.

THE WORLD'S FIRST MASTER SYSTEM CONTROLLER

A STEREO COMPRESSOR, NOISE GATE, EXPANDER, DE-ESSER, EXCITER, PEAK LIMITER, CLIPPER

- Dynamics sensitive gating control
- Unequaled audio performance
- Individual isolated processing circuits
- 21 LED gain control metering
- Switchable input and output metering
- Balanced XLR inputs/outputs
- Tip/ring/sleeve balanced inputs/outputs
- 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.

THE SOUND OF PERFECTION

APPLIED RESEARCH & TECHNOLOGY, INC., 215 Tremont Street, Rochester, NY 14608, (716) 436-2720, Telex: 4949793 ARTROC, FAX (716) 436-3942

CIRCLE 03 ON FREE INFO CARD

should be changed to approximately 15 k-ohm.

Greg Hanks
Chappaqua, NY

TIME, TECH & TINKERING

On page 18, "What About VITC?" in the April EQ, Fred Ridder says "...In still-frame...the same VITC frame number will be repeated indefinitely. This could never happen with LTC, and an LTC-type synchronizer won't know what's going on." I am using Mark of the Unicorn's Video Time Piece and their sequencer Performer 3.5 in direct time lock along with a Tascam Midiizer. When my VHS is in still-frame the sequence stays put at the same number frame as does the Midiizer which is reading LTC from the Video Time Piece. In this setup it has been my experience that it works just fine.

By the way, I prefer not to read build-it-yourself hardware articles in EQ. I felt like I was reading a hobbyist magazine. Judging from the types of ads and price of equipment tossed around in EQ, I gather most readers wouldn't be interested in building anything and usually purchase what they need. Using the same space to perhaps interview more industry pros or more articles like Roger Nichols I feel would be more useful.

David A. Roth
Westerville, OH

PRACTICAL REALITY

I truly think EQ's a great magazine. Just last week I passed out 70 sample copies to our Early Spring class and strongly recommended their subscribing. I think more than any other audio trade magazine, EQ addresses the reality of audio recording in the '90s. Practical information — free of the "you're nothing if you don't have a Neve" attitude that is found in other magazines. I honestly feel a strong philosophical bond between our school and your publication.

Jim Rosebrook, Director
The Recording Workshop

SAMPLE THIS

As a musician/engineer/studio owner and longtime Steely Dan fan, I was looking forward to reading Roger Nichols' report on the progress of Donald Fagen's new solo album. However, as I read Nichols' account of their numerous syncing problems, my mood changed from mild bemusement to numb disbelief. My first thought was: These guys have too much spare time!

As I pondered Donald Fagen's quest for rhythmic perfection, it slowly dawned on me why it's been eight years since his last record. With these work methods, his current album could easily take another eight years to finish. Please don't think for half a millisecond that I'm knocking Fagen's artistic vision, but there are aspects of his recording methods that I find disturbing.

First, why couldn't the drum machine parts simply be recorded to tape? Nichols says it's because they didn't want the "pussilanimous sounding samples to deteriorate any further by going through the delays." Did I miss something or aren't these just scratch tracks for the live drummer to follow? Who cares what they sound like if they aren't going to be used in the final product? Or is the mere thought of recording something less than perfect just unbearable to Nichols?

And what a cruel ordeal the session must have been for Chris Parker, the "live drummer." I'm sure he was well paid for his "zillion passes," but couldn't they have saved a lot of time and effort by just hiring his roadie to set up his drums and strike one of each for their sampling pleasure? Hey, what a concept! Record the drum sounds digitally and then trigger them from a sequencer. The only problem is I think it's been done and I think it's called a drum machine.

I'm just a struggling small studio owner without access to all the high tech toys that Nichols and Fagen employ, but I have a feeling that with my SMPTE synced Mac, my R8M, and a couple of samplers, I could have done the same thing they did in a fraction (or should I say decimal) of the time. All the MIDI drum and percussion parts could be shifted track by track and still lock up to my multitrack. Of course I couldn't guarantee that the hi hat wouldn't be off by a nanosecond here or there, but I promise I wouldn't tell anyone except my cat Maxwell.

One last thought: Wouldn't it be great if Donald Fagen put out one album a year? Just imagine, there would be eight Donald Fagen CDs in existence instead of one. I guess when you're a perfectionist, recording takes time. I'm just glad the Beatles weren't perfectionists like Donald Fagen or we wouldn't have some of the most memorable songs ever written.

Devin Thomas, SouthWest Sound
Sierra Madre, CA

TECHNOCRAT?

I loved Roger Nichols' new comedy column, "Across the Board," in the April EQ. At least I hope it was intended as humorous, because I surely laughed all the way through it. I find it implausible that people with the talent and experience of Roger Nichols and Donald Fagen waste their time splitting nanoseconds over sync tones. With the awesome musicians they could assemble, Donald could just count off the songs and get great performances with plenty of "feel" and other desirable human qualities.

I guess it's just disappointing to have one of my favorite artists falling victim to the same scourge that plagues all the dance/techno/haircut/makeup/attitude/no-talent music that we seem to be complaining so much about — namely, an over reliance on technology. Sometimes it seems that if it didn't exist, we couldn't make music any more.

C'mon guys, let's get "real"!

Randy Porter, Milan, OH

Editors' reply: These are two samples (no pun intended) of the mail we got in response to Roger's latest column. Turn to page 90 for "The Flip Side." But wait. Isn't a producer or engineer's job to do what the artist wants? If he wants accuracy, you make it accurate.

REAL REVIEWS

I really enjoy the different approach that EQ takes regarding product reviews, [as well as] your informative technical articles. I am rather tired of product reviews...that read like an endorsement rather than a critique.

I enjoy "Room With a View" very much, as it helps me to keep abreast of modern project studios. Also, as I am in the process of building a new room, I found Tom Hidley's article

(Feb. '91) informative. The technical articles by Richie Moore are also straightforward and useful. Keep up the good work.

David L. Shew, Manager
Davenhill Studio, Snellville, GA

MORE WORAM

When I began reading John Woram's column in the April issue, I hadn't noticed that it was to be a regular feature. Up to the final paragraph, I had resolved to write, requesting that he be a regular contributor. His lively, readable writing style and realistic perspective have me looking forward very much to his future columns.

Mike Jennings
San Jose, CA

SOUND SURGERY SPECS

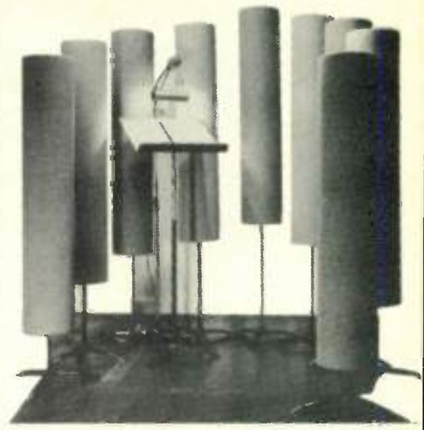
Your latest issue was filled with interesting articles that apply to my work. The article by Bernard and Birnbaum, "Sound Micro Surgery," was good — but I'd like a follow-up discussing which IC's are interchangeable, with and without external compensation, and basic differences (both audible and purely electrical) between various IC's that could be utilized. More details on how to safely revamp a circuit's capacitors would be helpful as well. You might look into a new product called a "Feedback Eliminator" made by Sabine Musical of Gainsville, Florida, and perhaps bench or field test it. If it works well, I'd say it's a very useful product. Thanks again, and keep up the great work!

Tom Young
Watertown, CT

CORRECTIONS

In the April issue of EQ, we neglected to credit the photograph of the Yamaha YPE 201 Optical Encoder/Gotham Audio Spot 90 System on page 22. On page 64, the SSM-2134 should have been identified as a high-performance replacement for the NE5534 *single* op amp instead of the NE5532 *dual* op amp. Two SSM-2134s can be used in place of a single NE5532.

At Last...



The Ultimate Mic Space

Time...
After Time...
After Time...

QSF™ Gobo . . . by Studio Traps™

ASC ACOUSTIC SCIENCES CORPORATION
1-800-ASC-TUBE
P.O. Box 1189
Eugene, OR 97440
MANUFACTURER OF TUBE TRAP™

CIRCLE 04 ON FREE INFO CARD

Peace Of Mind.

Designing a sound system puts your reputation and integrity on the line. People depend on you to create a complete solution to their audio needs; to put together just the right components to satisfy the job requirements and hold up over the long haul.

Specify products from ASHLY. You have our commitment of at least Five Years of trouble-free performance.* At ASHLY, every product we manufacture is hand constructed right here in the USA from the finest components available, and 200% quality controlled before ever leaving our plant. Designed to be the best possible investment in professional audio equipment.

Whatever your audio application, design in reliability and system integrity. Design in peace of mind. Design in ASHLY.

ELECTRONIC CROSSOVERS
COMPRESSOR/LIMITERS
NOISE GATES • PREAMPS
PARAMETRIC EQUALIZERS
GRAPHIC EQUALIZERS
POWER AMPLIFIERS • MIXERS

ASHLY
AUDIO TOOLS FOR THE TRUE PROFESSIONAL

*All ASHLY products are covered by our exclusive Five Year Worry Free Warranty. Contact ASHLY AUDIO directly for full details.

ASHLY AUDIO INC. • 100 FERNWOOD AVE • ROCHESTER, NY 14621 • TOLL FREE (800) 828-6308 • (716) 544-5191 • FAX: (716) 266-4589

CIRCLE 06 ON FREE INFO CARD

EQ&A

Q How much clipping is OK? When I set the limiters in my sound system, should I set them so that the clipping lights on the power amplifier never flash, or can they flash once in a while? What are the guidelines? Are the guidelines different for bass amps as opposed to mid and high amplifiers?

Barry Canada, Mesa, AZ

A Clipping is the point at which a sine wave begins to resemble a square wave, where the upper and/or lower most excursions of the waveform begin to be clipped off. This occurs when an audio signal applied to the input of an amplifier exceeds what the input is capable of accepting. The excess voltage drives the amp into "input clipping," resulting in distortion and possibly in a damaged amplifier input.

No amount of clipping is actually deemed acceptable. The clip LEDs located on the front panel of most power amplifiers illuminate between 1 dB and 5 dB below the actual point of clipping, hence giving the operator an early warning signal. You should check the manual or contact the manufacturer regarding your specific amplifier. The general rule of thumb is to operate the amplifier so only the extreme transient peak levels cause the clip LEDs to illuminate, while still maintaining good average operating levels. I would recommend setting up the limiter for approximately 2 dB below the actual maximum level specifications of the amplifier. This would certainly guarantee the amplifier would never be driven into clipping, while at the same time limiting your dynamic range. Setting up individual limiters for low- , mid- and high-frequency amplifiers is no different than using one limiter and amplifier to drive a passive full-range cabinet. From here this takes you into the Fletcher-Munson theory on perceived verses actual loudness, which we will have to save for another time.

Scott Heineman,
dbx Pro Product Manager

Q Can I use a shotgun microphone as a handheld vocal mic? It seems that it would help to improve the rejection of backline instruments, but would its use present any problems?

Travis Balcerak, Getzville, NY

A Shotgun or line + gradient type microphones have a very narrow acceptance angle for sound, especially at mid to high frequencies. This high directive (lobar) pickup pattern will result in some increase in isolation of sound arriving from the sides and potentially the rear of the microphone. Current technology allows "shotgun" microphones to handle high sound pressure levels and have very well controlled on-axis frequency response curves.

Two reasons not to use shotgun microphones for handheld vocal application are:

A) The polar pattern of a shotgun (sometimes called lobar) has lobes in the side pattern that vary with frequency and position. Thus, off-axis sounds aren't just attenuated, they are altered in terms of frequency response. Thus, a trumpet picked up off axis won't have the true sound of the actual instrument. This is off-axis coloration. It is very important in recording studio application, but may not be as much a concern in today's contemporary live music.

B) Generally, on stage in live application, the sound pressure level going into the front of the microphone is a combination of the vocalist, monitors and instruments on stage. Whatever is present at the entry of the microphone, the microphone will amplify.

By contrast, a well designed unidirectional (cardioid, hypercardioid, or supercardioid) will have a very linear off-axis response, and as you speak into the microphone and rotate it about the axis of the diaphragm, the output will decrease about the same at all frequencies. This provides a natural vocal pickup as well as a natural off-axis sound.

In summary: yes, you can use a shotgun (probably a short shotgun) for hand-held vocal, but also you will probably not get an overall improvement in vocal pickup or isolation over a good hypercardioid or supercardioid microphone.

Ken Reichel, Vice-President Marketing,
Audio-Technica U.S., Inc.

Q Is there any way to use an automatic microphone mixer for stage monitor use? Would it be usable for mains? How would the automatic mic mixer be added to the regular board? Could it go into the inserts somehow?

Chris Bell, Toronto, Canada

A Most automatic mixers are nothing more than sophisticated, multichannel gates. They are sophisticated in that they provide many features that are interactive between the various channels and the main output. These features, among others, include channel priority settings and output gain reduction dependent upon the number of active channels. However, the individual gate functions are fairly simplistic and don't always provide much in the way of adjustable parameters.

Now, it is not uncommon to find gates used in sound reinforcement applications. In fact, some of the most elaborate reinforcement consoles have them built right in. These gates, however, are designed specifically with the necessary features, controls and performance capabilities for this application. On the other hand, the gate functions of an automatic mixer may not live up to the demands of certain sound reinforcement applications.

Automatic mixers are designed primarily for the spoken word, and may not perform as would be desired for music. If, however, the device has a quick attack time and possibly an adjustable release time, it may indeed do the job. The next problem you may

continued on page 60

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

You've Heard The Applause For The PRM™ 308S . . .

The **PRM™ 308S** is being applauded by top professionals — like ***Don Was, Kenny Loggins, Joe Hardy, and Walter Afanasieff*** — for their uniform frequency response, minimum coloration of signal, a selectable switch for reference and EQ playback, and most of all just for "THE SOUND!"

PRM 308S Features

- **Mirror-image (left & right) pairs**
- **Switch-selectable response modes (equalized and reference)**
- **Acoustic foam blanket reduces baffle reflections**
- **Impedance 4 ohms**

Plus, there's more where that came from . . . Peavey. Namely the PRM 205A, 208S, 310S, and 312A phased reference monitors, all with many of the same great characteristics as the 308S to meet your particular needs and budget.

. . . Hear Here!

PRM™ 205A

- High-accuracy two-way system
- Impedance 4 ohms
- Small size fits atop most consoles

PRM™ 208S

- High-accuracy two-way system
- Switch-selectable response modes (equalized & reference)
- Impedance 8 ohms
- Acoustic foam blanket reduces baffle reflections

PRM™ 310A

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

PRM™ 312A

- Mirror-image (left & right) pairs
- High-accuracy three-way system
- Independent variable control of mid and high frequency levels
- Impedance 8 ohms
- Acoustic foam blanket reduces baffle reflections

PEAVEY®
AUDIO MEDIA RESEARCH

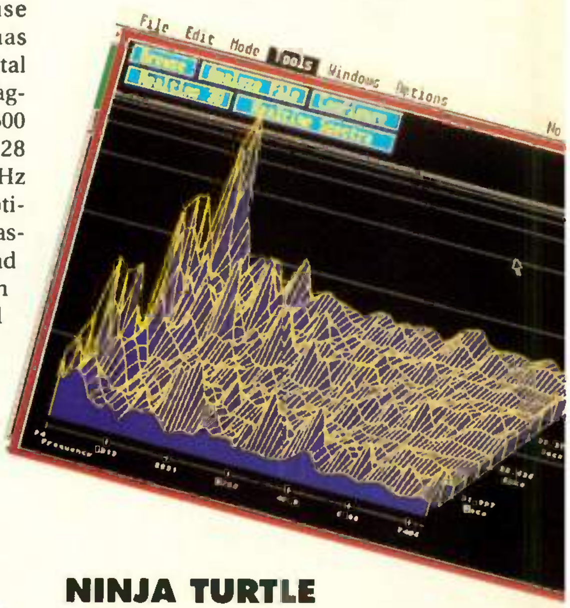
CIRCLE 23 ON FREE INFO CARD

Peavey Electronics Corporation • 711 A Street • Meridian, MS 39302-2898 • Telephone: (601) 483-5372 • Telex: 504115 • Fax: (601) 484-4278 • ©1991

EQ PRODUCT VIEWS

STORE GALORE

Never seem to have enough storage space? Never say never because Eltekon Technologies has introduced the MX-2D Digital Audio Series Removable Media Magneto Optical Disk Drive. The dual sided 600 megabyte removable cartridges hold up to 28 minutes of stereo audio at 48 kHz and 31 at 44.1 kHz per side. The MX-2D is equipped with a state-of-the-art optical Read/Write head mechanism that makes it immune from catastrophic "head crashes." It incorporates several proprietary hardware and software innovations designed to increase its speed and data transfer rate. In tests using "Disk Timer," average access time was 26 ms. The MX-2D can be used with hard disk based recording systems, like SoundTools by DigiDesign, and is priced at $5495. Contact Eltekon Technologies, 37493 Schoolcraft Road, Livonia, MI 48150; (313) 462-3155. Circle EQ Free Literature Number 170.



SAMSON HAIR RAISER

Getting too much wireless interference? Samson's UR-4 receiver operates on a higher plane, in the "blue sky" UHF frequency bandwidth of 944-952 MHz. With the low amount of traffic in this frequency, you will be able to perform and record with a minimal amount of interference and noise. When it's coupled with the UR-4's diversity circuitry and dbx noise reduction, you get an overall improved wireless mic performance. Also included in this model are di-electric filters, gas-fet and SMD cellular technologies. Contact Samson, 262 Duffy Avenue, Hicksville, NY 11801; (516) 932-3810. Circle EQ Free Literature Number 171.

NINJA TURTLE

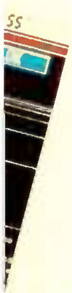
Turtle Beach's new release, SoundStage v1.2, is a two-track editing software system for their 56K Digital Recording System. This updated version includes new features such as sample rate conversion, time compression/expansion, a scrubbing window, separate mono Left/Right control, Pitch shifting, SMPTE Chase-Lock, Invert, Reverse, 3-D Realtime Frequency Analysis, and Record undersampling support. The version 1.2 will be free to all registered users of the 56K System, which includes the 56K-PC Digital Signal Processor Card, the 56K-D Digital Interface Box, and the SoundStage Two Track Editing Software. Contact Turtle Beach Systems, PO Box 5074, York, PA 17405; (717) 843-6916. Circle EQ Free Literature Number 172.

FROM MIX TO MAX

The CAD Maxcon Mixing System has introduced an expandable system with frame sizes that can support rackmounts from 16 to 144 inputs. The In-Line format, eight bus system offers two input types; the Quad mic and Octaline modules and two different master output group modules. Patchbays and CAD MegaMix Automation are also available for this system, which because of its unique Servo topography, offers near perfect transparency. The CAD Maxcon provides signal handling that is flat from 3 Hz to 200 KHz. Contact CTI Audio, Inc., PO Box 120, Conneaut, Ohio 44030; (216) 593-1111. Circle EQ Free Literature Number 173.

SYNTH-PLICITY

The Roland JD-800 Programmable Synthesizer lets your creative juices flow, combining the sound quality of a digital synthesizer with the creative ease of an analog synthesizer. The JD-800 is host to an impressive array of features including a new sound source that contains 108 preset waveforms sampled at 44.1 kHz. The large number of sliders and buttons on the front panel allow you to easily edit and access a variety of options, including a Palette function which lets you modify a single parameter for up to four tones at a time. The 61-key, weighted keyboard makes it ideal for both performing and recording. Suggested retail price for the multipurpose synthesizer is $2895. Contact RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040; (213) 685-5141.Circle EQ Free Literature Number 174.



A-T GETS CONDENSED

For all you movers and shakers, Audio-Technica is introducing the PRO 37R Remote Power Condenser Microphone. Suited for digital sampling, studio recording, and line recording, the PRO 37R features a very flat frequency response as well as high sensitivity. The PRO 37R also provides a distortion-free signal in sound fields as loud as 141 dB. The microphone can be powered with any phantom supply of 9 to 52V DC, and it offers a 30 to 15,000 Hz response. The mic comes complete with a foam windscreen and a protective carrying case. Contact Audio-Technica, Inc., 1221 Commerce Dr., Stow, OH 44224; (216) 686-2600. Burbank, CA 91501; (818) 841-1078. Circle EQ Free Literature number 175.

CLASS MONITOR

KRK Monitoring Systems has been supplying both close-field monitors as well as main-monitoring systems for many of the heaviest hitters in recording for many years. Their new line of monitors (KRK-703/1002/1303) incorporate the newest aerospace materials with the most advanced technology, design and construction that are available in speakers today. The KRK-703 is a compact two-way system for console-top close-field monitoring; the KRK-1002 is a medium-sized two-way system with extremely smooth frequency response, wide bandwidth and efficiency; the KRK-1303 is a high-performance 13-inch 3-way system capable of generating SPLs above 108 dB and frequencies from 38 Hz to 19 KHz. Contact KRK Monitoring Systems, 16462 Gothard St., Unit D, Huntington Beach, CA 92647; (714) 841-1600. Circle EQ Free Literature Number 176.

WRH

PRO PHONES

Take care of your ears with Sony's new headphones. At the top of the line is the MDR-7506, ruggedly designed with folding construction for storage ease. Gold connectors and an OFC cord make for a strong connection, while a stereo unimatch plug lets the unit interface with both 1/4- and 1/8-inch connectors. The MDR-7506 has a closed ear design for comfort and comes with a soft case for portable protective storage. It features a 40mm driver unit and a frequency response of 10 Hz to 20 kHz. The other two models, the MDR-7504 and MDR-7502, have 30mm drivers and offer frequency responses of 50 Hz to 18 kHz and 60 Hz to 16 kHz respectively. They list price for: $109 (MDR-7506), $89 (MDR-7504), and $49 (MDR-7502). Contact Sony Business and Professional Group, 3 Paragon Drive, Montvale, NJ 07645; (800) 635-SONY. Circle EQ Free Literature Number 177.

DUPE-IT-YOURSELF

KABA wants to back your band. With the purchase of a KABA duplicator, KABA throws in over $1100 of additional equipment and supplies free — setting you up with everything you need to produce and package your own cassettes. The pictured system can produce as many as 116 C-45s, completely shrink-wrapped and packaged, in under four hours. The system sells for $7596 and includes the master control deck, three dual transport slave decks, an L-Sealer, heat gun, 2000 ft. roll of shrink film, 1000 cassettes custom loaded with chrome tape imprinted to your specifications, and 1000 Norelco boxes. A similar four-position system with 500 cassettes is available for $5,358. Contact KABA, 24 Commercial Blvd., Novato, CA 94949; (800) 231-8273. Circle EQ Free Literature Number 178.

AFFORDABLE AKG

The famous AKG condensor sound can be yours with the C1000S microphone. It features high sensitivity, high output, and a stage-rugged construction. It can be used without compromise with consoles that don't have phantom power because the special, built-in battery clip accepts a standard 9V alkaline battery for full headroom and over 200 hours of operation. The C1000S is switchable between two polar patterns, cardioid and hypercardioid, for different applications. It comes with W1000 Foam Windscreen, SA 43 Stand adapter, and PPC 1000 Polar pattern converter. Suggested retail: $325. Contact AKG, 1525 Alvarado Street, San Leandro, CA 94577; (415) 351-3500. Circle EQ Free Literature Number 179.

CIRCLE 09 ON FREE INFO CARD

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?

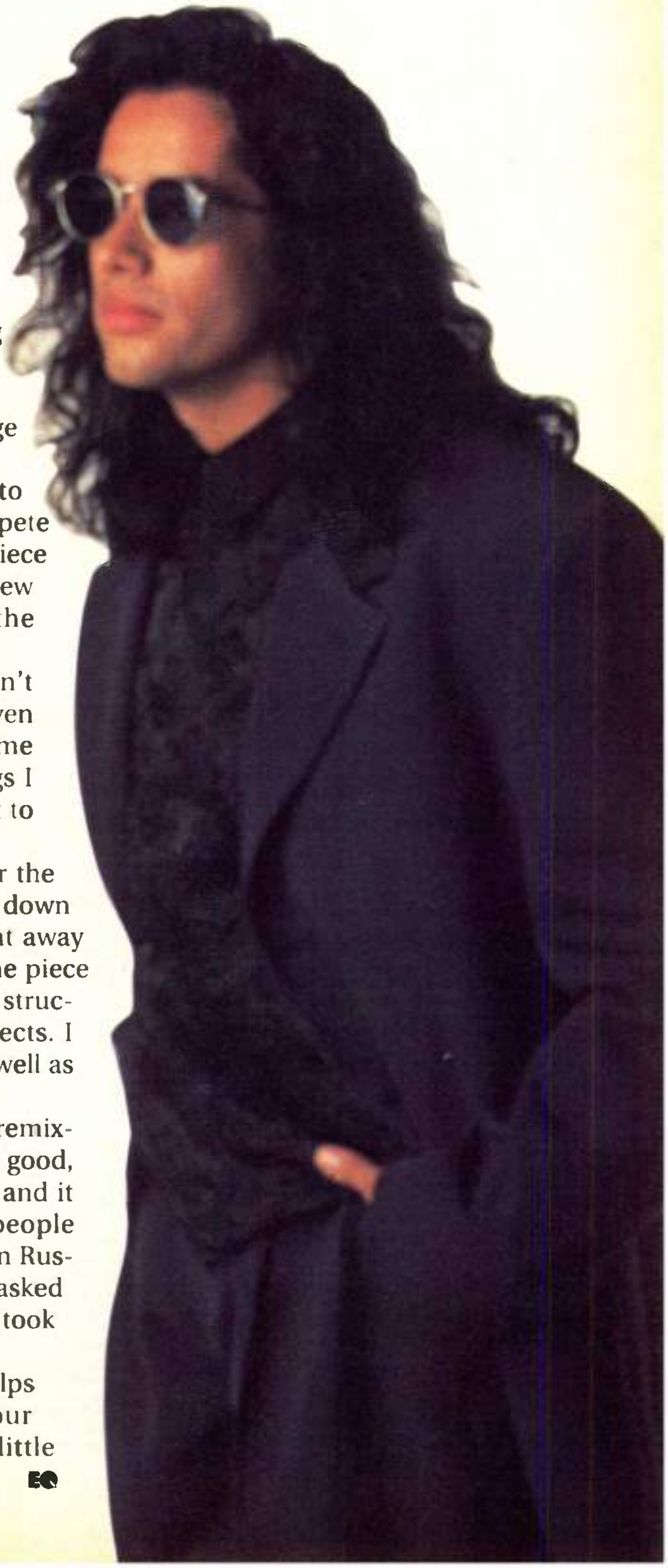
ROCKTRON
STUDIO SERIES

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

CIRCLE 24 ON FREE INFO CARD

GET 'EM MOVING

BY JELLYBEAN BENITEZ



How do you make music that gets people moving on the dance floor? You give it a groove. People only dance to the rhythm — to the strength and the pounding of the bass and drums. But they also need something new, something electric. That's why dance clubs are the forum where new music gets born, where old musical rules are broken and new ideas are made.

That's probably why more and more dee jays are proving their capabilities as leading producers, remixers and engineers. They are the talents that are leading the leading edge. The techniques are basically the same. You have to rearrange the music so that it moves people the most.

When you're remixing a pop tune for dance, you have to rearrange the rhythm of the song so that it will be able to compete on the club level, while keeping the original integrity of the piece intact. It's a matter of careful overdubbing. You might put in new drums and some driving synthesizers, but you still keep the melody and the vocal performance as is.

You can't be afraid to experiment with technology. I don't have any formula; I trust my instincts and my ears. I don't even really care if the end result is musically correct. Don't get me wrong — it has to sound good, and 99.9 percent of the songs I remix are correct. But I'm not doing it for science. I'm doing it to make people dance and build up a sweat.

Being tied into the club scene is an absolute necessity for the dee jay turned producer. I'll often hop out of the studio and go down to a club, pop in a tape and watch the reaction. I'll know right away whether I've hit it perfectly or if it needs work. Listening to the piece over a hot sound system is also a good way to determine the structural aspects of the cut — the levels, the mix, the sound effects. I always listen to a piece over a club system and a Walkman as well as the studio monitors before I finalize it.

Making the transition from dee jay to remixer — and from remixer to producer, for that matter — takes talent (you've got to be good, of course), a desire to make music on a more technical level, and it takes exposure. You've got to be spinning at clubs where people hang out. I was dee jay'ing at the Funhouse in New York, when Russell Simmons, who owns Def Jam, came in with a record and asked me if I wanted to mix it. I fell in love with remixing, and it all took off from there.

You really have to want to make it happen. It certainly helps to have a strong background as a dee jay — since it puts your musical ideas directly to the test. With work, talent, and a little bit of luck, you can get a whole floor moving. EQ

WRH

HOT PICKS

• **Electro-Voice** has unveiled a single-channel compressor/limiter, the **COL-1**, intended for sound reinforcement where transient protection and level control are required. The COL-1 utilizes a feed-forward design for stability at all compression ranges, and it employs an RMS detector for logarithmic gain changes. EQ Free Literature #180.

• **Shure** has introduced the **Model FP410**, a portable automatic microphone mixer, as part of their Field Production (FPP product line). The product incorporates the patented Shure IntelliMix technology. EQ Free Literature #181.

• **The Sapphyre, Soundcraft's** newest recording console, includes I/O modules that incorporate individual noise gates with a four-band EQ design. A Dual Line input option enables increased input capability for effects returns or virtual tracks. EQ Free Literature #182.

• **TAC's** newest sound reinforcement console, the **SR6000**, features a VCA output group and split auxiliary sends. It also includes parametric EQ, input metering, stereo effects returns, eight VCA/Mute groups and a 10 x 8 output matrix.EQ Free Literature # 183.

• **Echo Audio's EASE III** medium-sized, tri-amped speaker enclosures offer 13 ply/per inch birch ply shell construction, a 16 gauge steel grill, and recessed hardware. Utilizing JBL components, these enclosures are available in several flight hardware configurations. EQ Free Literature #184.

DANCES WITH WOLF & EPIC

Dance music buffs are familiar with the work of Richard Wolf and Bret "Epic" Mazur, who — with their special blend of rhythm and blues, pop, and rock — have created a singular style that helped create Bel Biv Devoe's premiere album, and Poison achieve triple platinum success.

The duo is currently turning their ears toward a mix for Prince's upcoming album. And they promise "to bring a certain funkiness to what he's doing." To achieve this funkiness, they don't plan on using any special equipment. "We're probably going to cut it all live," they say. "The special pieces of outboard equipment are actually our brains."

Most of their tracks are cut at their home studio in Sherman Oaks, California, but they usually finalize their mix elsewhere.

They are primo proponents of E-Mu's E-Max line and they often utilize the E-Mu Emulator 1 and 2 because of their preference for their filtering capabilities. "All samplers have their own sound, and E-Max let's achieve a sound that's not so electronic."

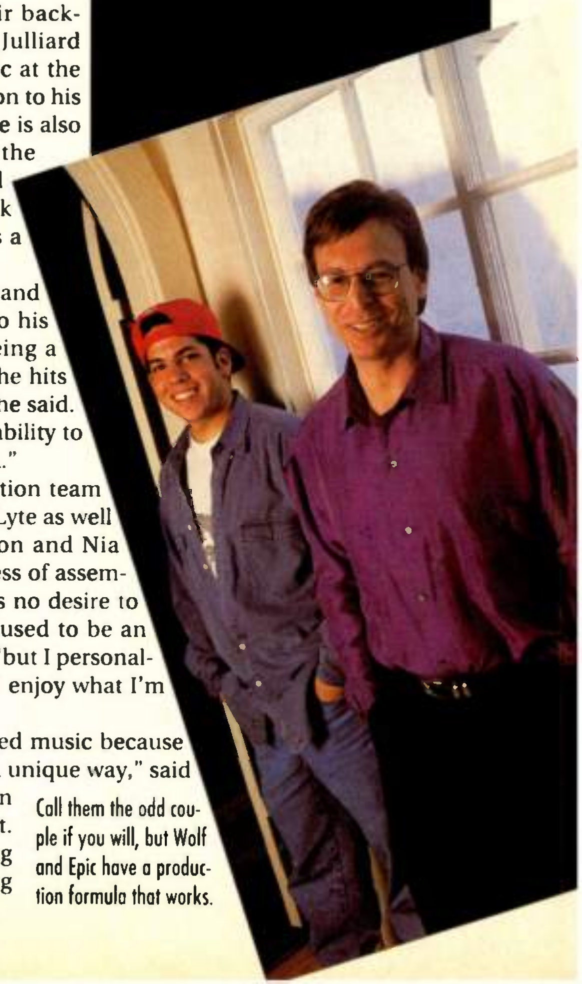
In addition to E-Max I and II Turbo, their rack includes the E-Mu SB1200 Drum Machine, as well as the Roland D50. They also use the Yamaha GX7 and FB01, the Korg M-1, and the OBXA. Despite the abundance of samplers and sequencers, the team says they only use sampling for added flavor rather than as the basis for a song.

Their styles are reflective of their backgrounds. Wolf studied piano at the Julliard School of Music and electronic music at the Manhattan School of Music. In addition to his work as a producer and songwriter, he is also a well-versed keyboard player. "All the musical experience that I've had being a part of jazz, funk and rock bands, I'm now finally able to use as a producer," said Wolf.

Epic's experiences as a drummer and club dee jay have also contributed to his production skills. He claims that being a club dee jay helped him recognize "the hits that the public could really get into," he said. "And being a drummer gives me the ability to know instinctively what rhythms work."

Other future plans for the production team include an upcoming album by M.C. Lyte as well as new mixes for both Sheena Easton and Nia Peeples. Epic is currently in the process of assembling a funk-rock group, but Wolf has no desire to step out from behind the scenes. "I used to be an artist on Capitol Records", Wolf says, "but I personally have no interest in doing it again. I enjoy what I'm doing now."

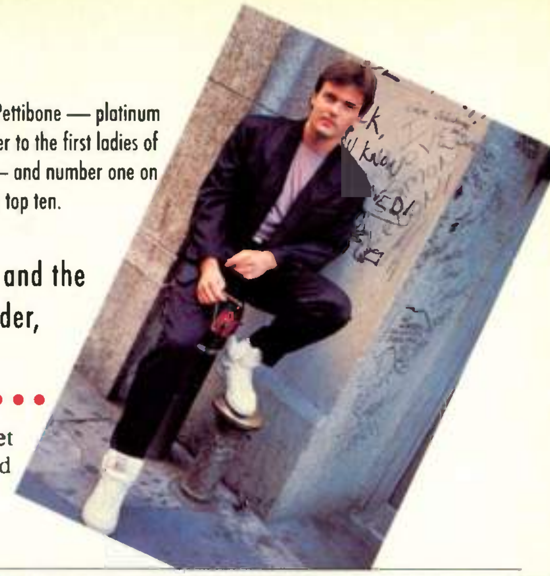
"Bel Biv Devoe, as a group, changed music because they fused hip hop, R&B, and pop in a unique way," said Epic. "We were part of that creation and we'll continue to expand on that. Hopefully, we will always be pushing the envelope of authentic, hard-hitting pop music." —*Anthony Savona*



Call them the odd couple if you will, but Wolf and Epic have a production formula that works.

EQ'S TOP 10 REMIXERS

Somebody had to do it — sooner or later. Somebody had to recognize the contributions of remixers to the state of today's music. So, the editors of EQ have compiled our list of the Top 10 Remixers in the industry based on the quality of their work and the popularity of the final product. The winners are listed in order, followed by some of their more popular remix credits.



Shep Pettibone — platinum remixer to the first ladies of pop — and number one on the EQ top ten.

1 SHEP PETTIBONE Madonna's "Immaculate Collection" LP; Janet Jackson's "Pleasure Principle," "Miss You Much," and "Rhythm Nation"; Paul Lekakis' "Tattoo It On Me"

2 FRANKIE KNUCKLES Alexander O'Neal's "All True Man"; Lalah Hathaway's "Heaven Knows"; Alison Limerick's "Where Love Lies"; Victoria Wilson-James' "Through"

3 DAVID MORALES Londonbeat's "I've Been Thinking About You"; The Adventures of Steve V's "Dirty Cash"; Safire's "Made Up My Mind"

4 STEVE ANDERSON Cybil's "Love So Special"; Nomad's "I Wanna Give You Devotion"

5 ROGER SANCHEZ Underground Solution's "Luv Dancin'"; Tribal House's "Mainline"

6 STEVE "SILK" HURLEY Black Box's "I Don't Know"; Kym Mazelle's "Don't Scandalize My Name," "Got To Get You Back," and "Never In A Million Years"; Unity 2's "Buckwheat's Revenge"

7 DAVID COLE & ROBERT CLIVILLES All work on C&C (Cole & Clivilles) Music Factory; Seduction's "It Takes Two"; New Kids On the Block "Games," "Never Gonna Fall In Love," "Step By Step," and "Valentine's Girl"

8 JUNIOR VASQUEZ Tevin Campbell's "Round & Round"; Janet Jackson's "State of the World"

9 DANNY TENAGLIA Double Dee featuring Danny "Found Love"; Escape Club's "Call It Poison"; Patrick O'Hearn's "Black Delilah"

10 TONY HUMPHRIES Dee-Lite's "Power of Love"; Cybil's "Love So Special"; Living Color's "Elvis Is Dead"; D-Mob's "That's The Way Of The World"

DISC MAKERS

Complete Package Prices for Demos

500 cassettes Studio Series for $800 (Includes color inserts, clear shell, and Dolby HX Pro)

500 cassettes B & W Express Series for $700 (Includes Black & White inserts, clear shell, and Dolby HX Pro) READY IN 7 DAYS!

500 CD's & 500 cassettes Combination Package for $2,990 (Includes color printing, clear shell, and Dolby HX Pro on cassettes, and blister pack for CD's)

500 CD's Studio Series for $2,490 (Includes color inserts, jewel box, and blister pack)

All prices include design, typesetting and color separations!

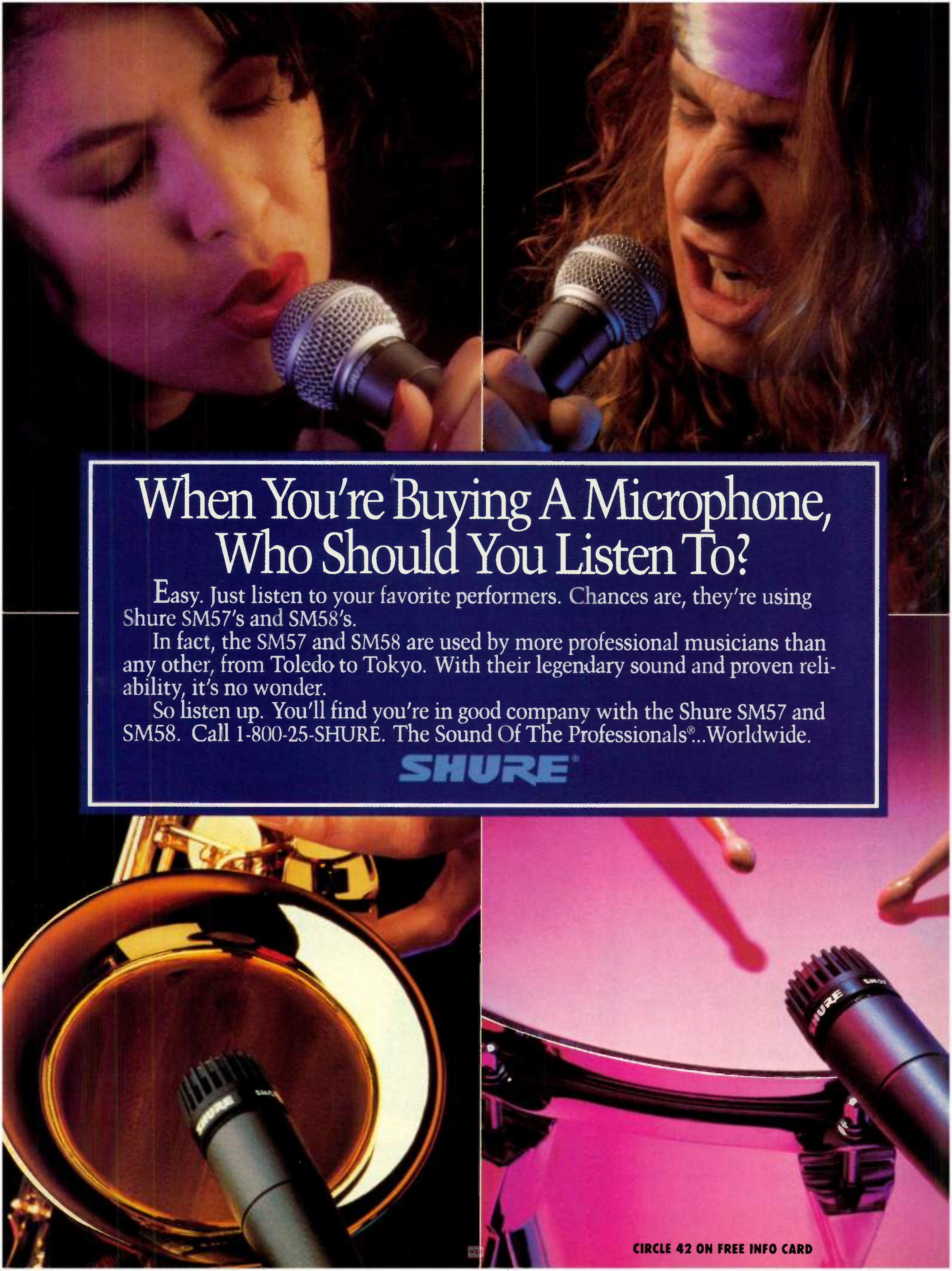
- Full color printing
- Major label quality
- Deal directly with the largest independent manufacturer in the industry

Add $25 to order for shipping cassettes via 2 day air service. Add $50 to order for shipping CD's via 2 day air service.

AVL1190-2

Call Today For Our Free, Full Color Catalog: 1-800-468-9353
In PA call: 215-232-4140

1328 N. 4th Street, Philadelphia, PA 19122
1-800-468-9353 Fax: 215-236-7763
CIRCLE 16 ON FREE INFO CARD


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
CIRCLE 42 ON FREE INFO CARD

Flying Machine

Taavi Mote's mountainous rack cooks sounds the way you like 'em.

BY JON RUZAN

TAAVI MOTE — veteran of high-flight projects like the Hollywood remix of U2's *Desire* and engineer on projects for big names like Madonna — has amassed an outboard box collection more like a mountain than a rack.

Filling one long, Jan-Al case and about 12 four-space and two-space cases, Mote's rack is the product of seven years' assembly. Built for remixing power, it boasts an arsenal of samplers including the T.C. Electronics 2290, Eventide's H-3000 SE/B/Vai with sampler card, and the Yamaha SPX900 and SPX1000. For unique special effects, Taavi uses Roland's DEP5 as well as a number of other Roland pieces including the SRV2000, the SDE3000S delay, and four Yamaha SPX90s that have been updated to II's for short percussion sounds.

To get true stereo from mono sources, he owns the Sound Technology Stereo Simulator, and the Dynacord CLS222 Leslie simulator because, as he says, "it is the closest thing to a real Leslie you can find." His choice of reverbs: the Eventide H3000 UltraHarmonizer configuration and Yamaha's Rev5, Rev7, SPX900, and SPX1000. The Alesis MidiVerb 2 is added to this array. "I like to use it for mixdowns; it has a sound that is just right for getting certain effects."

Though his rack contains a slew of effects, such as the T.C. Electronics 12D chorus/flanger/doubler, Mote uses them discerningly. "I use effects in the same way that an artist would use a brush stroke," he says.

Considering the enormity of his rack, compressors, limiters and EQs seem scarce. "Most of the studios (Larrabee Sound, Skip Saylor, and Elumba Studios) I work in have a lot of the models I like to use," he explains. "And, to add a lot of clarity to a piece without overusing EQ, I use BBE's 202R, 802, 822's."

Other notables: the dbx 120X-DS subharmonic synthesizer and the Dytronix FS-1. "I use the dbx for dance remixes because it helps me add more bottom, and the FS-1 lets me have a track pan to the tempo via trigger input and make sounds move from the background to the foreground."

His collection of vintage tube mics features an AKG C24 (which he used for Madonna's vocals), the Telefunken 251, and the Neuman U47, U67, KM254, and SM69FET. Added to his mic ensemble is a tube mic preamp by James Demeter, which he also uses as an instrument preamp.

If variety adds spice, Mote knows how to cook. "My rack is an assortment of high- and low-end equipment, because the sounds that I envision aren't always available from the latest and greatest gear," he says. "The object of putting together a rack is not to be able to brag about having the coolest gear. It's a matter of having equipment that will help me work out my ideas." EQ

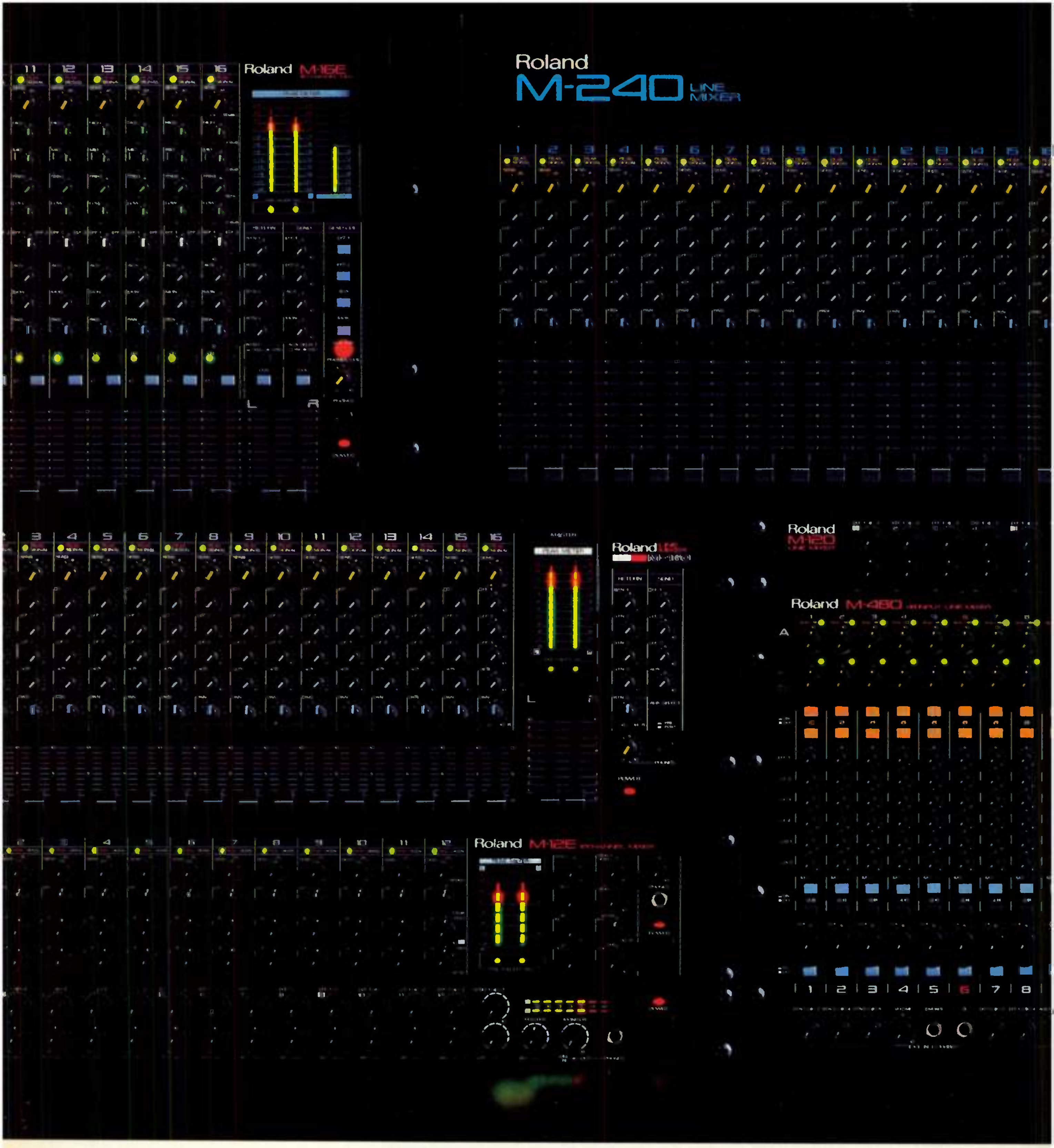
Photo by Ed Colver

Axel's Baby

STUDIO: European American Recording, New York, NY.
OWNER/OPERATOR: Axel Kroell.
PRODUCER CREDITS: Wet, Wet, Wet; Blow Monkeys; Grayson Hugh; Jimmy Helms; Arthur Baker; programming and writing for Quincy Jones.
CONSOLE: Tascam M-600 console.
RECORDER: Tascam ATR-80 24-track.
MONITORS AND AMPS: Westlake BBSM-8 monitors, Bryston 4B amplifier.
KEYBOARDS AND SAMPLERS: Akai S-1000 16-bit sampler (47-second sampling), two Akai S-900 12-bit samplers, two Roland D 550's, two Yamaha TX 802's, Yamaha DX7 and DX7FD, Korg M-1, Super Jupiter, Roland MKS 50, Oberheim Matrix 6R, Moog synth.
SEQUENCERS AND DRUM MACHINES: Linn-Akai MPC 60, Linn 9000, Atari ST (w/Creator sequencer), SMPTE-Track sequencer, Pro 24 Sequencer; E-Mu SP12, Alesis HR-16, Siel MP-100.
SIGNAL PROCESSORS: Lexicon PCM-70, TC Electronics TC 2290, TC Electronics SCF stereo flanger/chorus, Yamaha Rev5, Ibanez SDR-1000 reverb, three API 5502 EQ's, API 3124M preamps, dbx 166 compressors.
MICS: AKG 414, Shure SM 57 & SM 58.
TAPE: Ampex 456.
EQUIPMENT NOTES: "The main workhorses are the Akai MPC 60 and S-1000. The MPC 60 is the fastest gun in town and the S-1000 is very powerful and easy to use. The E-MU SP12 has a special sound that you just can't get with anything else. I love the Tascam board. It's extremely clean and straightforward, and I can have all the synths operating at the same time as the tape returns. The monitors impress everybody. I've had more than one person leave my studio and go buy his own Westlakes."
STUDIO TIPS: "You don't need a million-dollar console to get great sounds. We often record here, and spend a day in a top-dollar room replacing the drums. The rest of the overdubbing we do in-house, using the outboard API EQs. In the end, it works out as a completely discrete recording." EQ

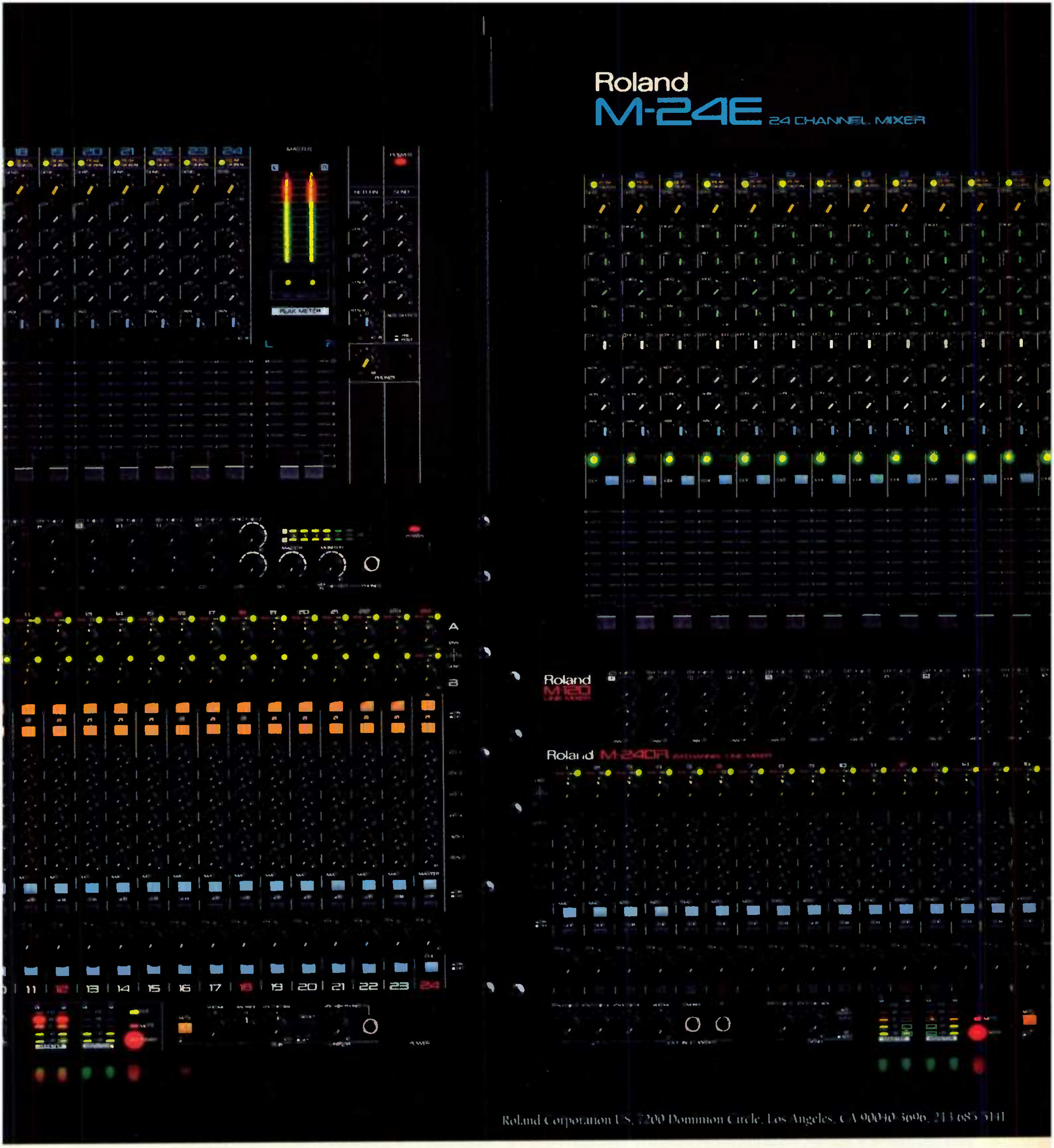
"You don't need a million-dollar console to get great sounds."

Photo by Claude Rolo



If you have trouble in candy stores,

Here's what you say to the little angel standing on your left shoulder: "No matter what I play or how I play it,



perhaps you should turn the page.

Roland has the mixer I need." And here's what you say to the little angel standing on your right: "Let's rock, dude."

Roland®

CIRCLE 37 ON FREE INFO CARD

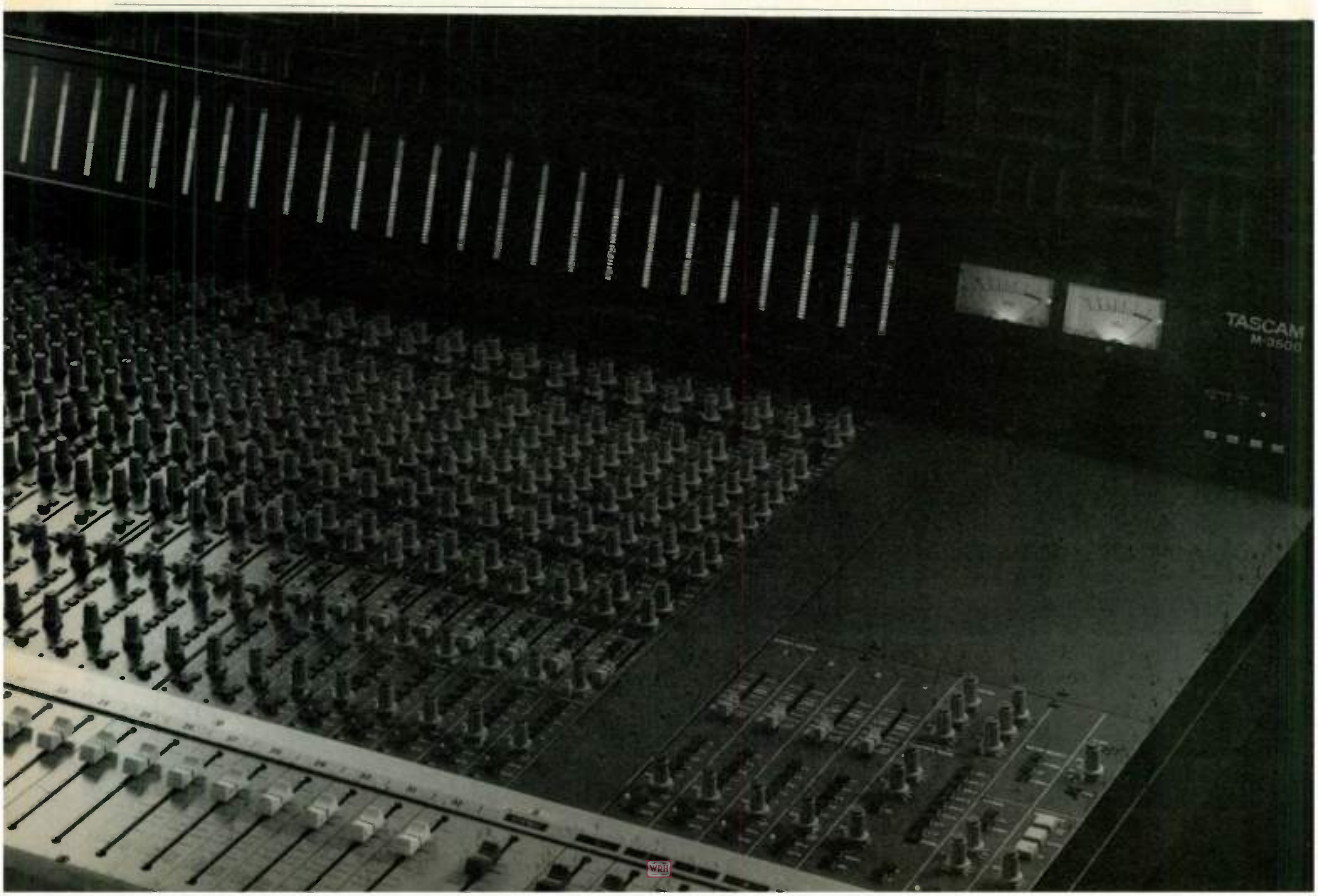
When the error rates increase to 300, clean the heads. One head, totally clogged, yields error rates of 4,999, while both clogged will read out the meter maximum of 9,999. Rates in the low teens to twenties are what our machines show normally. These error numbers, totaling 900 or so per minute, are typical of DAT's super-robust error correction schemes.

CODE CRACKING

Like any well-behaved DAT machine in the United States, the 3700 supports the Serial Copy Management System (SCMS). Being a professional machine, SCMS can be controlled when copying digitally through the AES/EBU ports. There are dip switches on the back of the machine for setting the status of the SCMS code. Switch 1 must be in the AES/EBU position for you to control what kind of protection coding goes (or does not go) onto your tape. When Switch 1 is in the AES/EBU position, the SCMS selection switches also set the SCMS status for a tape recorded from an analog source.

Unlike the new generation Professional DAT machines on the way, the 3700 does not support SMPTE or EBU timecode. It does support the "absolute time" subcode pack, which shows continuous hours, minutes and seconds. While this is handy when playing tapes with this subcode recorded on it, when playing back clients' tapes without absolute time written on them, I found it difficult to locate or time selections. However, the 3700 lets you put absolute time and automatically insert index numbers on a tape that has previously been recorded without them without affecting the audio.

Compared to the older 3500, the second best part of the SV-3700 is the input/output interface. The analog ins and outs offer both +4 dBu and -10 dBu operating levels (-10 dBu is available on the output only). The digital in and out is either the normal RCA pin-plug cable (SP-DIF to the hip, IEC 958/Type-2 to the even hipper) or the professional AES/EBU XLR connector interface. The new AES/EBU ports are a great help when several DAT machines are daisy-chained together. No longer does stopping the first machine take the others out of pause due to lack of a word clock. The AES/EBU offers a secure 5-volt interface connection, in



addition to the half-volt RCA pin-plugs that are often not as reliable.

ON THE LEVEL

There is one poor design problem with the analog stage. The unit has an input level control, but unfortunately there are no output level controls. You're stuck using Panasonic's 18 dB headroom reference level. The Neve Digital Transfer Console uses a 14 dB reference, and most studios I know use this or occasionally 16 dB. Panasonic's peculiar standard can create a very high analog output level (4 dB louder than if one is calibrated to a 14 dB headroom). This makes A-B comparisons impossible. A bad design oversight, as so many people had complained about this shortcoming with the older 3500.

To initially set up this machine by the book: Connect the output of your console to the DAT (pin 2 hot) and the DAT's output to the console monitor. Put in a new tape and put the machine in RECORD-PAUSE. Send a 1 kHz tone at 0 VU from the console. Raise the single RECORD LEVEL control until the meters just read "-18." This will mean a fair amount of fussing with the BALANCE control, as one channel always seems to get there first. Connect the output of the DAT to the console monitor. The output should read 0 VU on the console monitor meter.

The best news is that it sounds good, amazingly good for the money.

Do a trial mix while you observe the digital meter very carefully. The idea is to create a recording that doesn't ever go over on the meter, causing distortion, yet isn't so under-recorded that it forfeits the dynamic

continued on page 71

64 CHANNELS. CABLE READY.

Sometimes it seems like you can't get there from here. You've got a thousand great ideas, and just about as many plugs in your hand. What you don't have is enough input channels.

Well, allow us to give you some input about a new way to solve your dilemma. It's a Tascam M3500 in-line mixing console. Choose either the 24 or 32-track mixer and by simply flipping a switch, you can double it to 48 or 64 mix positions.

And, with a suggested retail price of $7,499 for 24 inputs or $8,499 for 32, it won't take up a lot of your budget, either.

If you're planning to build a 24-track development studio, here's another advantage: The M3500 is the perfect match for the MSR-24, Tascam's one-inch 24-track recorder. Together, they make the most cost effective studio available.

It just may be that you don't need a huge console to enlarge your capabilities. The M3500 offers you a new, more effective approach to traditional mixing that is both compact and low cost. And when you need more inputs, all you'll have to do is switch channels. From 24 to 48. Or from 32 to 64.

TASCAM®

© 1990 TEAC America, Inc., 7733 Telegraph Road, Montebello, CA 90640, 213/726-0303.

CIRCLE 45 ON FREE INFO CARD

Soundman's Tool Kit

IF THE SHOW must go on, then the sound system must work. And take it from me — there's nothing worse than staring a simple repair job in the face half an hour before curtain and having to hunt around for the tools to get it done with.

The show is 100 or 1000 miles from the shop. What do you want to bet that the inevitable equipment failures will extend well beyond the capability of your pocket-sized, do-anything tool? There is always a place for the Swiss Army knife type of sound tools, but there are a few more to consider while on the road. Here's what you need:

SOLDERING PENCIL
This can be a 100-watt, temperature-controlled soldering iron with a small tip or the not-so-temperature-controlled butane type if you think that finding AC power may hinder your emergency repair work.

RESIN CORE SOLDER
If possible, carry at least two sizes of solder — a small diameter one for microphone connectors and printed circuit boards, as well as a larger one for speaker connectors. Of course, this has to be the resin core solder for electronic work and not the acid core solder used in plumbing.

SOLDER WICK OR DE-SOLDERING TOOLS
Many of the repairs done in the field involve reconnecting terminals — and this usually means cleaning out old solder first. These two types of tools have their advantages; Solder Wick desoldering braid is good for smaller amounts of solder (like that found on circuit boards) and Soldapullt snap vacuum solder is made to remove larger solder globs, like those found on connector terminals.

MULTI-METER
This is a tool that can perform hundreds of useful tests in the field. Learn the operation of your multi-meter thoroughly and you will be able to trace common wiring and component faults with relative ease. It is also handy to assure yourself that the power hook-up is the voltage you were expecting on the terminals you're connecting to. When it comes to powering your sound system, trust no one until you have confirmed the polarity, ground and voltage that reaches your power connection. Checking the disconnect is essential before applying power to the system.

CLIP LEADS
A few different types of probes for your multi-meter will make awkward tracing easier, and spring clips can free up at least one hand from holding the leads. Extra leads that have small insulated alligator clips at both ends are very good for making temporary connections within equipment while troubleshooting.

OSCILLATOR
Although many mixing consoles include a simple oscillator that can help in tracing signals, a battery-operated tone generator (such as the Shure A 15TG) is ideal for tracing faults in multi-pair snakes, checking cross-patching and verifying continuity/levels through the signal chain.

SIDE CUTTERS
If you carry a lot of large gauge wire, two sizes of cutters may be handy.

NEEDLE-NOSE PLIERS
A set with long, thin jaws can dig stray bits out of awkward places and sink away less heat when holding wires on terminals during soldering.

LINESMAN PLIERS/VISEGRIPS
These tools can fill a lot of different functions. The linesman's pliers will usually have a cutting position for larger gauges of wire, so you may not need the large set of side cutters. A

Put this sound system first-aid kit together and don't leave home without it.

BY WADE MCGREGOR

good set of these can be very useful for both gripping and tightening loose friction-type connectors, such as the flag connector common to loudspeaker terminals. The vise-grips can even serve as a vise for small parts when you don't have a bench or suction-mount vise near at hand.

SCREWDRIVERS
Robertson, Phillips, Blade — three sizes of each. Get a small blade screwdriver or a set of jeweler's screwdrivers.

ADJUSTABLE WRENCH
A six- or eight-inch adjustable wrench will take care of many tightening jobs where lots of torque is not required.

NUT DRIVERS
Get 1/4" and 1/2" Nut Drivers. Include the driver sizes that are common to your sound equipment; rarely is an entire set required.

POCKET KNIFE
It's one of the most useful emergency tools there is; if you don't already carry a Leatherman tool, that is.

LEATHERMAN POCKET SURVIVAL TOOL
The Leatherman tool is a good example of the sound technician's Swiss Army knife. It has a folding 8-inch ruler, needle nose and regular pliers, wire cutters, a knife blade, can/bottle opener, three sizes of blade screwdrivers, #1 Phillips screwdriver, awl/punch, and a file/saw in a leather pouch that fits on your belt. Contact Leatherman, 10300 N.E. Marx Street, Portland, Oregon: 503-253-7826.

SUCTION VISE (SMALL PLASTIC RS TYPE)
Don't forget a small vise to hold things until the glue sets or the filing is complete. For soldering connector terminals, a box or plate with all the connectors used in your sound system mounted on it can be even more effective for holding the mating connector.

Pictured: The Leatherman Pocket Survival Tool

TAPE MEASURE
The tape measure can be used for everything from checking if doors are big enough to making speaker stacks symmetrical on stage.

POWER DRILL/SCREWDRIVER
Battery-powered screwdrivers can be a godsend. The best have an adjustable clutch to prevent over-tightening.

DRILL BITS
A complete set from very small to 1/2" drill bits as well as the appropriate screwdriver bits are absolute musts.

POP-RIVET TOOL AND POP-RIVETS
Like the powered screwdriver, the pop-rivet tool is an essential part of keeping things (like road cases) together over the long haul.

BATTERIES
Carry spares of size AA, D, C and 9-volt, and other types in your equipment — especially in the wireless microphones and cordless drills.

LUBES, GLUES & TAPE
Make sure to have lightweight oil, a can of spray-on lubricant and 5-minute epoxy with you. Also don't neglect to bring heatshrink or vulcanizing tape to mend slashed cable jackets and a lighter or soldering iron to heat it.

HEADPHONES
A higher impedance (greater than 600 ohms) set of headphones can be extremely useful for signal tracing, especially as it won't load the output of line level equipment as severely as an 8 ohm set.

FLASHLIGHT
A small, durable flashlight worn around your neck or on your belt will probably see as much work as all the rest of your tools combined.

PENS & PAPER
Bring a china marker; they can make temporary marks that can be wiped off most surfaces. Bring a felt-tip pen for making permanent marks, and always bring a note pad.

SMALL BOXES & ZIP-LOCK BAGS
These are useful for storing all the little screws, washers, etc. that like to disappear while you are dismantling a device.

HAMMER
It helps to persuade the otherwise stubborn to conform to your needs.

C-CLAMP, PONY CLIPS, ETC.
These are very useful for holding things in place until the show is over or until the glue dries.

KNEE BENCH
A knee bench is a platform made to sit snugly on your lap. It makes it possible to have a reasonably stable work surface in almost any situation.

MECHANICAL METRONOME
Handy for ringing feedback out of the sound system, for using as an acoustic reference and for torturing drummers.

MIDI DATA ANALYZER
This MIDI equivalent of a multi-meter is useful if you're relying on MIDI to communicate between crucial effects or sound generators.

AC POWER CHECKER
These small neon indicators are handy to check if an outlet is live, but can't replace the multi-meter for assurance that the voltage is appropriate and that there aren't dangerous voltages between the neutral and ground.

OSCILLOSCOPE
This is the electronic technician's version of the Swiss Army knife. It allows visual display of thousands of electronic conditions and can yield the same information as a room full of test equipment. There are now oscilloscopes available that fit in the palm of your hand, if you can afford them. EQ

The Joy of Lexicon

IN JANUARY of this year, I became the owner of two Lexicon 300 digital effects systems. A long-time user of Lexicon products (who isn't?), I was interested in upgrading the reverb fidelity in my project studio. The 300 turned out to be not only a great reverb, but a terrific source of a variety of effects.

Film scorer and musical engineer talks about life and times with the Lexicon 300 Digital Effects System.

BY GARY CHANG

Aside from the omnipresent Lexicon Reverb Algorithm, what impressed me the most was how quiet the 300 was. Demonstrating the unit for inquisitive ears amounts to lengthening the reverb time to maximum (about 64 seconds), playing a few piano notes into it, then soloing its output. It's so quiet, it sounds more like a sound generator than a sound processor.

One may say "As the 200 is to the 224, so is the 300 to the 480L." The 300 has fewer functions and editing parameters than the 480L, but has similar sonic specs, reverb algorithms, etc.

USING THE MACHINE

The front panel layout of the 300 consists of LED indicators (for MIDI and sampling rates), level masters, and an alpha-numerical display on the left, with various button arrays on the right, separated by the central value-change knob.

On the rear panel, you will find just about any kind of input/output configuration that you will require. Digital is well represented,here,with HLR-type AES/EBU, optical, and coaxial connectors. MIDI and SMPTE timecode connections accompany standard HLR analog inputs and outputs, making the 300 eminently interfaceable.

There are five modes that you can run on the 300. Modes are groups of associated variables. Changing from mode to mode is achieved through button selection to the right of the standard 10-key pad. Once in a mode, a particular parameter can be selected for modification (signified by an underlining of the parameter's name) by either the eight buttons above or below the alpha-numerical display, or, if the parameter is not on the page presently being displayed, pressing PAGE UP or PAGE DOWN will toggle through the various pages of that mode. Once selected, a parameter can be changed by twirling the change knob.

The five modes on the 300 are: RUN, where you load setups; CNTRL, where global changes are made (MIDI, Event change Lists, etc.); SETUP EDIT, where you modify I/O variables and load effects (it has 8 presets and 64 user memories); EFFECTS EDIT, where you modify effects variables (75 presets, 64 user memories); and MOD EDIT, where MIDI and Soft Knob patches are assigned. Of the five modes, most of the user's initial activities will be in the RUN, SETUP EDIT, and EFFECTS EDIT modes.

Each program loaded while in the 300's RUN mode actually contains a SETUP and an associated EFFECT. The SETUP is simply a listing of I/O variables (AES/EBU or SPDIF ins? 48KHz sampling rate? Emphasis?) plus a name of an associated EFFECT ALGORITHM — either REVERB, AMBIENCE, STEREO PITCH SHIFT or STEREO ADJUST.

Personally, I found the SETUP concept to be a bit overkill for my purpose. After hardwiring the units to my patchbay, I felt little need to change the SETUP, except for a few occasions where I used the digital input/output.

Accessing programs in one of the 64 user memories in RUN mode entails naming and saving the SETUP and the EFFECT - rather uninspiring tedium during creative sessions. To circumvent this situation, I avoid RUN mode after first selecting my personal SETUP file. From SETUP mode, I can select and load any of the

75 preset or 64 EFFECT user memories. From EFFECT mode, I can create or edit programs, and then simply save it in the EFFECT user memory. Doing this saves me the step of resaving the SETUP with the newly created or edited EFFECT's name associated with it and the returning to RUN mode, which is a big help. Since I use only one SETUP, I never return to RUN mode.

One bizarre character of the 300 occurs in the edit modes. When displaying variables, the 300 shows either an abbreviation of the name of the parameter (reverb time, predelay, etc.), or the value of the parameter. The VALUE button lets you toggle between the name and its value. However, when the 300 displays 8 different parameters at once (as it often does), the value button switches from a display full of abbreviations to a display full of numbers — this takes getting used to.

Shortcomings aside, the machine runs admirably. Its timecode reader and event list are very handy: each entry in the event list has an assigned timecode that matches an entry in the list, and the 300 will load the listed programs automatically. This can be useful in film/video sync situations. Furthermore, it acknowledges MIDI systems exclusive parameter, so dynamic changes to the 300's programs can be recorded with most MIDI sequencers on the market.

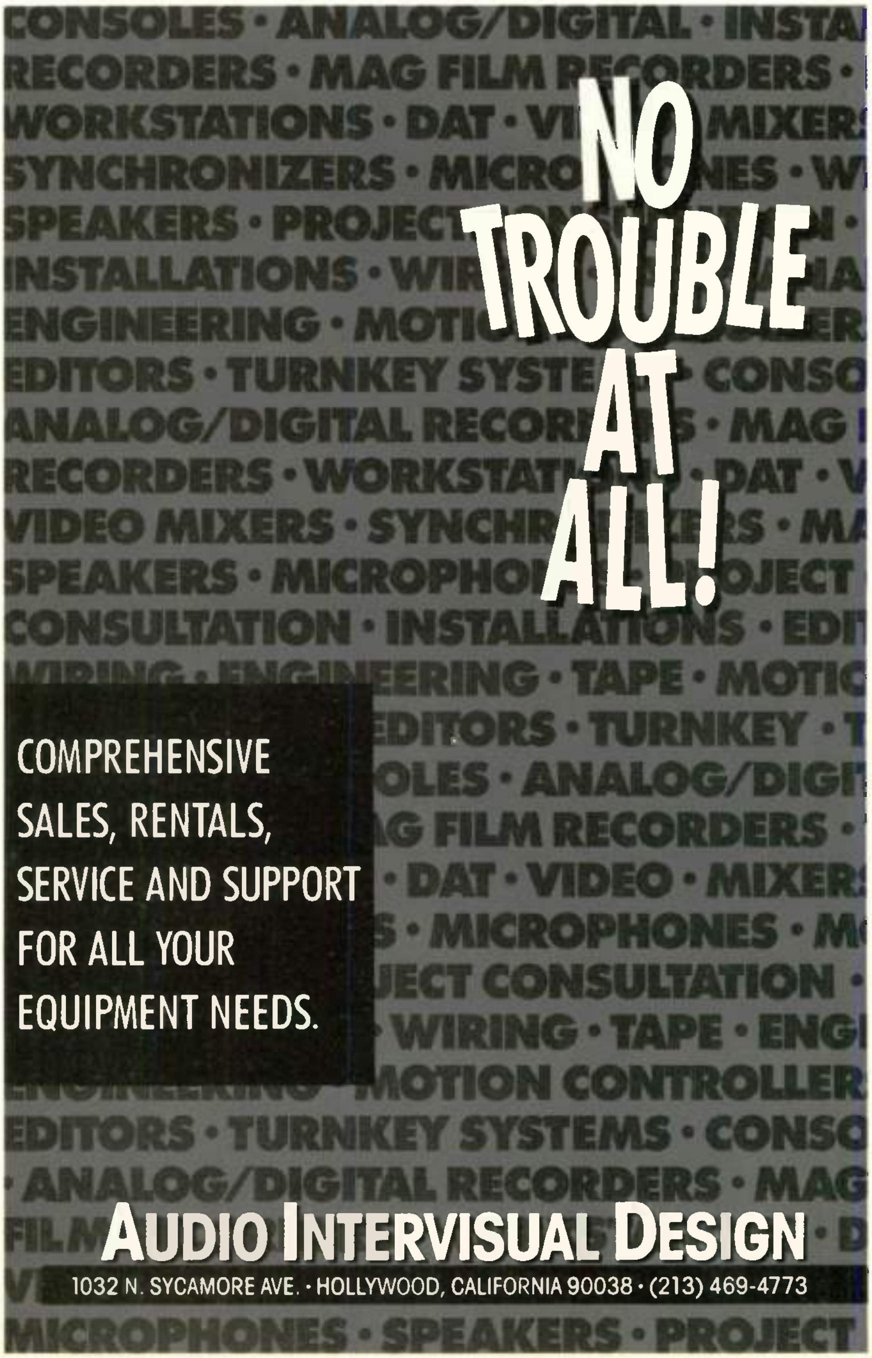
Sonically, the 300 is really a great performer. When my studio is configured for writing, I have thirtysomething inputs wide open on the board. Needless to say, noise is a big issue. I find it pleasing to note that the 300 delivers a very dense effect (some say denser than some of the other Lexicon reverbs) which is, in many cases too loud!

Great input and output headroom lets you forget about the 300 while you're in your creative throes. Both the REVERB and AMBIENCE algorithm produce great reverb, with the REVERB producing a more authentic long reverb, and AMBIENCE having more details early on. The STEREO PITCH SHIFT algorithm is very useful for your basic harmonizer effects, as well as the STEREO ADJUST algorithm, which seems ideal for the final processor in a digital mixdown. The one thing that I really found missing from the unit is that fantastic Lexicon Chorus, found in many of Lexicon's reverb products. Oh, well, I'll get over it.

Great input and output headroom lets you forget about the 300 while you're in your creative throes.

So, there you have it. Digital I/O on a stick. The Lexicon 300 is ideal for those who are looking for 480L-quality reverb without all the bells and whistles. For a couple of thousand dollars less than what a 480L costs, you can hook yourself with two totally accommodating 300's. If you're looking for an affordable, professional-quality reverb for your project studio, this is the box. EQ

Gary Chang ***is a Newhall, California based independent composer for motion pictures and television.***



CIRCLE 40 ON FREE INFO CARD

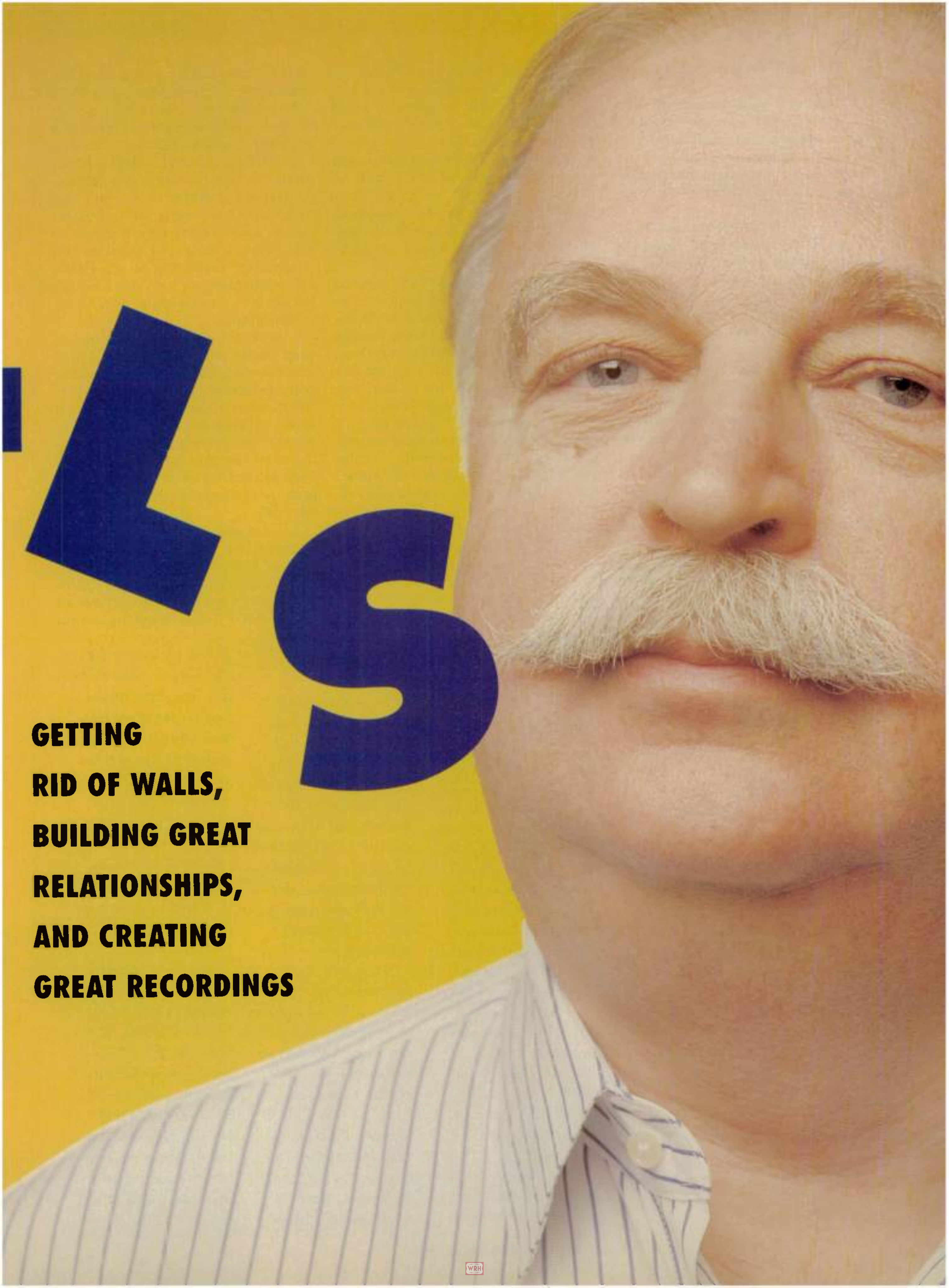
TEAR DOWN THE WAL

BY BRUCE SWEDIEN

I hate walls when I'm recording, either real or imagined. I don't like barriers or roadblocks of any kind. Anything that gets between me and the music. Between you and the feeling. It's a passion for openness and togetherness that has been a driving force in my career since I began recording music, as a professional, in my hometown of Minneapolis back in the 1950s.

I am fortunate enough to be in the process of expressing this facet of my character in physical form these days. I am building a very personal music studio at my ranch — and it's like no studio you've ever seen before. The main room in my new studio has no walls. There's nothing between what you would call the "control room" and the performers' area of the studio. Whoever's singing vocals can sing them right behind me, or beside

WRH



'LS

GETTING RID OF WALLS, BUILDING GREAT RELATIONSHIPS, AND CREATING GREAT RECORDINGS

me, or right in my ear, if that's what we wish.

I guess it boils down to the fact that I am bored with the standard clinical approach to music recording. Everyone being assigned to little "boxes" and the resultant feeling of us being separated during the performance of the music.

My love for togetherness is an attitude that influences all aspects of my work. It has been at the root of my long-term musical relationships and friendships with geniuses like Quincy Jones and Michael Jackson.

When you work hand-in-hand with someone on a creative musical project, you can't allow any obstacles to arise in your communication. You have to develop a working rapport that lets you accomplish your tasks in a way that allows for musical expression of both parties involved — in a way that perfectly translates the musical concepts onto the recording medium, so that the rest of the world can appreciate them, play them over and over again, and make them a part of their lives. Tearing down walls in your working relationships lets you get to the heart of the matter — the music — without any barriers getting in the way.

THE CLASSIC TOUCH

Working closely with the artist is something that always came naturally to me, but I have to give a lot of credit to the conductor of the Chicago Symphony Orchestra, Fritz Reiner, for bringing it out. I met Dr. Reiner in 1957 in Chicago. I had just come from Minneapolis, where I had been recording chamber music and choirs for local record companies and musical organizations. I came to the big city to work for RCA Victor, and I ended up recording a couple of albums with the Chicago Symphony Orchestra, with Fritz Reiner conducting.

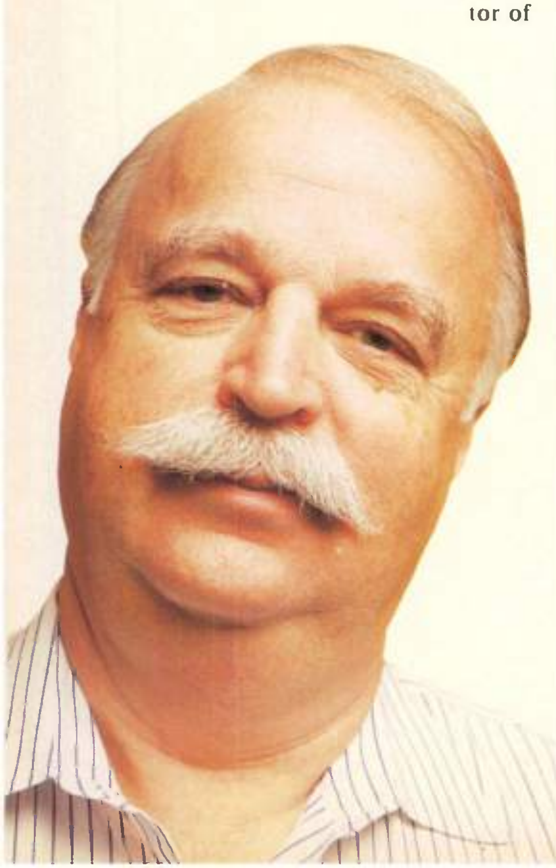
Dr. Reiner was not only a gifted conductor, he also dearly loved the recording process. I think I could say that he seemed to really enjoy making records. We would work together, hours into the night, editing the orchestra recordings bar by bar, and sometimes note by note. It was, in a way, an early model of the kind of relationship I've tried to build with all the artists I've worked with over the years. It's one thing to love the music, it's another to have the kind of intellectual curiosity and passion for quality that it takes to delve continually into the minute decisions of the recording process.

There's one thing I've never been able to forget about those formative years of working with Dr. Reiner. He immediately made me a part of his innovative, new incentive program — "one mistake and you're through." It worked; I paid attention. Every young engineer should be as lucky as I was.

Starting off with a strong background in classical music has certainly made a difference for me. An orchestral balance is an orchestral balance no matter whether you're recording acoustic or synthesized music. And learning to record classical music takes an incredible ear for detail. It has to be right; there's no middle ground.

In recording music — no matter what you're recording — there's only one criteria that transcends all other considerations, and that is quality. That's what appeals to me the most about working with Michael Jackson. All Michael cares about is quality in the music we do together. Quality of melody, quality of performance...in fact, incomparable popular music is our goal.

Classical music is also a tie between me and Quincy Jones. Of course, the projects we worked on weren't classical. One of the first projects Quincy and I did together was an album with Dinah Washington on



MIKING MICHAEL

Bruce Swedien talks about studio techniques for M.J.

The first thing that dictates my mic choice — with Michael Jackson or any other artist — is the music. For a love song, such as "She's Out of My Life," I'd use something very warm, like a Telefunken U-47 tube. For a somewhat harsher sound, like what you hear in "Workin' Day and Night," I'd put a Shure SM7 dynamic to work. For a lot of Michael's background tracks, especially for vocal block harmonies, I will even use one of my RCA 44-BX's.

I don't usually use a room mic. I prefer to move Michael around. When I do the background, I'll have him close to the mic, and then I'll have him step back. This forces me to raise the volume level of the mic, and it captures more of the early reflected sounds in relation to the direct sound, plus this technique gives a good mixture of the reflections off the floor.

I use a very visual approach, not only in miking but in all aspects of production. This probably stems from the fact that when I hear sounds, I see color. That's the way I EQ Michael. The high frequencies appear to me as silver in color, and the lows are dark colors, brown and black. When we were at the Grammys in New York this year, Bette Midler had on this absolutely bewitching purple velvet gown. When I saw her on stage, all I could think of was the bass sound in "Man in the Mirror." —Hmm...

I also like to experiment a lot; that's another reason for having my home studio. I love tubes, for example. No, I don't mean vacuum tubes. I mean paper shipping tubes. Pipes. That kind of thing. There's a line in "Billy Jean" where Michael's alter ego speaks to his consciousness. I had him say this line into the mic through a mailing tube. It gave it a real sonic personality. You can hear it if you listen.

Photos of Swedien by Ed Colver

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

Soundtracs distributed exclusively in the United States by:
Samson Technologies Corp., P.O. Box 9068, Hicksville, NY 11802-9068
TEL (516) 932-3810 FAX (516) 932-3815

© 1991 SAMSON

CIRCLE 44 ON FREE INFO CARD

Mercury Records. We also did projects together on Roulette Records with Sarah Vaughn. Quincy later moved to Paris and became a star pupil of Nadia Boulanger, who taught Stravinsky. Our mutual love of the classics is a big tie between me and Quincy; I think it's a love that's evidenced in *Back On The Block*.

Besides, it was also classical music that got me into recording in the first place. Both my parents were classical musicians, and they poured on the piano lessons. It was soon apparent to me (and my piano teachers!) that due to a severe lack of keyboard ability, I should do something else for a living. Recording was a natural fit.

THE MUSICAL REVOLUTION

And recording I did. I was incredibly lucky to have had the privilege of working with artists like Count Basie, Woody Herman and Duke Ellington, Stan Kenton, Jack Teagarden — and of course, Quincy Jones, who I met very early in my career. It was this incredible diversity of experience that has served me well to this day.

Right now, I'm working on a book that goes over all this history in detail. After all, I've seen the inside of the control room through the musical revolution. This book is going to be about my experience and the people I've worked and dealt with before, during and after that revolution.

What do I mean by musical revolution? Well, in the early days, what we tried to create was a concert-like image of the music. But in the late 1950s and early '60s, we began to think that reality in recording was not necessary, or even desirable. That's when I really started getting excited about recording.

In my opinion, one record changed pop music forever: Les Paul and Mary Ford, 1951, *How High The Moon*. Only one instrument plays all the parts. There wasn't a shred of reality in it — and it was wonderful. The attempt to present it in a concert-like atmosphere was totally absent, and people dug it. Music making has never been the same since.

It all comes back to what I was saying before about walls. You don't need them. Trying to recreate a concert on tape is like being a rat running through a maze — you start with a

continued on page 75

WESTVIKING RECORDERS

For 35 years, I've been dreaming of building the perfect recording studio. Finally, I'm getting the chance to make the dream come true.

I have a ranch outside of Los Angeles. We have horses, chickens, geese, ducks, dogs and cats. I've torn out the tennis court and put a chapel-like studio where it used to be. The studio — which I'm calling Westviking Recorders (in homage to my Scandinavian heritage) — represents a total departure from clinical studio design.

As I said, I've always hated being separated from the musicians, looking through the glass from a separate control room. So the structure contains one main room: 35 feet long x 21 feet wide with a 15 foot ceiling. It's large enough to produce a 30 Hz full wave, so it's going to be just about the sweetest sounding room you've ever heard. For recording rhythm tracks and the like, there's a large garage — fully equipped and linked back to the main room by video.

I designed the studio acoustically with double wall construction and no two parallel surfaces. The east wall is canted and the ceiling is coffered, so standing waves are kept to an absolute minimum.

The desk is a gorgeous Neve 8032 modified by my pal Stewart Taylor in Canada. It's a 32-bus board capable of running two 32-tracks or any combination smaller than that. I'm a real fan of Rupert Neve's earlier efforts — especially this particular one. I love the sonic quality of pure class-A electronics.

My dear friend Allen Sides, of Oceanway Studios, is doing all my speakers. The monitors themselves are on a perfect 10-foot, 8-inch triangle to the sweet spot, built into special soffits constructed for acoustic isolation from the rest of the building.

I've also built this room for scoring films. There's every format in there from 8mm to one-inch video, and I can link any audio format to picture. We'll use a 12 x 9-foot Cinemascope screen for projection.

Of course, we've got enough outboard gear in there to blow your mind. Some of the highlights are Neve limiters, Yamaha Rev5's, Rev7's, SPX-1000 and SPX-90, the Eventide 949, UREI 1176 limiters, dbx 165a's, a Dynatronic CX-5, an EMT-250 and two echoplates custom-built by Jim Cunningham in Chicago.

Tracks can be laid down on my 16-track two-inch MCI. I have a Mitsubishi X-850 digital master machine with Apogee filters. I also have a Mitsubishi X-86HS complemented by several other analog and digital tape machines.

I have 105 microphones; it takes 15 Anvil cases to hold them. I still have the first mics I bought after high school: old Telefunken tube U-47's. I still use them in every one of my projects. I've also got M49's and M50's, along with three Telefunken ELAM 251's. But you'll also find units like the AKG 414, 451 and 452 in there, along with a ton of Neumann U64's, KM56's and M84's. There are ribbon mics too — RCA 44-BXS's, 77-DX's and BK-5's for percussion. And how could I forget my B&K's and Milabs? I love them.

Gear aside, the point is that this is my dream studio. I'm going to be co-producing a couple of songs with Sergio Mendes for Elektra, and co-producing with Rene Moore (of Rene and Angela) — all at Westviking Ranch. I can't wait to get started.

A few important words about the new A-T 40 Series:

AT4049
Omnidirectional
Capacitor

AT4051
Cardioid
Capacitor

AT4053
Hypercardioid
Capacitor

Tony Bongiovi
Power Station

"The 4051 is a great mike, especially for rock. It sounds fat and you can bang away at it with a lot of level without a pad…for a rock studio like the Power Station that's important. When you put it on horns it has a nice clean sound and it holds the dynamics well…it's just an excellent sounding mike."

David Cook
Dreamland Studios

"…real nice top end and a warm bottom end…very versatile. I didn't have to pile on a lot of EQ to capture the air in the studio…very present, very natural sounding mikes."

Milan Bogdon
Masterfonics

"The S/N ratio is superior to some of the other mikes we used. They're bright and clean so we don't have to push the EQ. Superb mike…great for vocals, overheads, snare, toms, electric and acoustic guitar...it seems to work well wherever we put it."

Jeff Baxter
Producer/Artist

"If I'm not getting what I want from another microphone…I've been putting up the 4051 and it nearly always does the job."

Mack Emerman
Criteria Studio

"The response is very flat…it holds the natural tonal qualities even at high sound pressure levels."

Now it's your turn!

Compare the new Audio-Technica 40 Series against the very best in your studio. Contact your A-T pro sound dealer today.

CIRCLE 35 ON FREE INFO CARD

audio-technica
1221 Commerce Drive, Stow, OH 44224
(216) 686-2600 • FAX (216) 688-3752

THE COMPLETE SAMPLER

SAMPLING SMARTS

A BRIEF BUT UNIVERSAL TOUR OF SAMPLING ETHICS, AESTHETICS AND EDITING TECHNIQUES.

EVERY ONCE in a while I see the statement that there is no "new" music. That every note or combination of notes has already been written. Every new song is just a re-statement of previous works.

Well, I guess there's a ring of truth in it. I heard a song on the radio the other day, and until the vocal came in I couldn't tell who's song it was going to be. When Bowie started singing, my kids looked up with astonishment, proclaiming that he must have stolen the song from Vanilla Ice. Right! And my wife got her good looks from our kids.

I believe that guns don't kill people; people kill people. The newest weapon available to the musician is the sampler. The samplers aren't responsible for ripping off material from other artists, the people who use them are responsible for the inappropriate use of these tools.

THE ETHICAL SAMPLER

I built one of the first samplers in 1978. We used it to play drums and percussion on Steely Dan's "Gaucho" album. Roger Linn built the first commercially available drum machine that used samples of real drums in 1979. In 1991, drum machines and samplers are a dime a dozen, and some of them sound great. Sampler sound quality is now getting right up there where it should be. Some new sequencing programs do a pretty good job of adding a human feel to the sequenced parts. Sample editing systems like Sound-Tools from DigiDesign and the Akai DD1000 make creating new samples very easy. They also make it easy to sample pieces of someone else's record and use it for your own project.

I guess that the bottom line here is sampling *responsibility*. If you record a song that someone else wrote, then they get paid for their creative endeavor. By the same token, if you use a piece of someone else's recording in your song, they should get paid for their creativity. During the "Gaucho" album, we had musicians come into the studio and we recorded their drum sounds for time displacement (for use after they were gone). The musicians knew that they were playing sounds for the sampler. They not only got paid for the sampling session, they also got paid when we used their samples on other tunes. Everything worked out. We were happy, the musicians were happy and the union was happy. I like this method. You can't take clips out of movies and use them in your own film without permission from the copyright holder, so why should you be able to do it with records? If you can't afford to pay for the bass line you're stealing, then come up with one of your own.

BY ROGER NICHOLS

THE QUALITY SAMPLER

Now that I got that off my chest, let's talk about quality sampling. If you are going to sample something, you should at least try to do a good job. When you listen to records, you can usually tell immediately when a record has a drum machine instead of a drummer. It is almost as easy to tell whether the instruments are samples or the real thing. The two qualities that give it away most easily are the fact that most samples are mono, and there is an absence of ambience or leakage from one instrument to

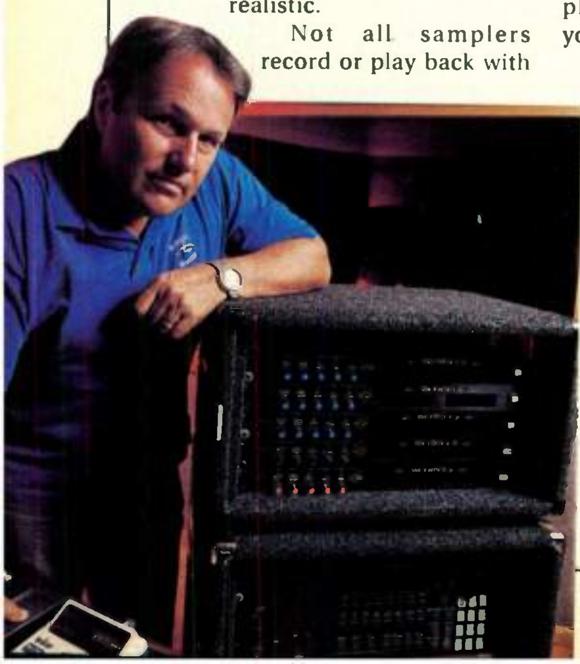
Nichols hefts Wendel, his homemade sampler. Photo by Ed Colver.

If you can't pay for the bass line you're stealing, then come up with one of your own.

another during the recording process. A little pre-planning can go a long way toward eliminating most of these problems.

First, stereo samples. When recording drums, you usually have from seven to twelve microphones that are mixed together into the stereo image. When the snare drum is hit by the drummer, the sound of that drum gets into all of the other microphones. Some of them may be gated, but usually multiple microphones contribute to the snare drum sound. Part of the image that is heard in the control room is due to the fact that all of these microphones are not the same distance from the snare drum. The time delay of the acoustic energy from the snare drum reaching these different microphones dictates the snare drum's perceived position in the stereo field. If when you sample a snare drum, you use the sound of all of the contributing microphones, in stereo, then the sample stands the most chance of sounding realistic.

Not all samplers record or play back with equal fidelity. Check out the sampler that you plan to use and make sure that it is compatible with your needs. Some samplers, like the Akai S1000 and S1100, sound better if the sample was recorded on a DAT machine or some other A/D converters and then transferred digitally into the sampler for editing. Make the sample sound as good as possible. A bunch of crappy samples added together end up sounding worse, not better.



SAMPLE EDITING

After you record the sample, it needs to be edited. Remember, the stereo position is dependent on the timing of the right and left channel information. If you can edit the sounds in stereo, you stand a better chance of maintaining that relationship. If your sampler can play back stereo samples and can accept stereo samples dumped over from the editor, you're in pretty good shape.

You are in big trouble if you have to edit each side of the stereo image independently. I did some testing with two overhead mics on cymbals. I recorded the samples in stereo and then loaded them into SoundTools for editing. When you zoom in to get a closer look at the wave form, you can see the relationship between the cymbal attack as seen by each of the mics. If the drummer hit a cymbal on the left, then the attack wave form on the right will be as much as 100 to 200 samples later. This is due to the speed of sound, approximately 1100 feet per second. If the microphone is

IN REVIEW

AKAI S1100 & S1000

Starr Parodi
Keyboard Player, Arsenio Hall Show

I've been using the Akai S1000 on The Arsenio Hall Show and its upgraded version, the S1100, for my own record. One of the reasons for getting the S1000 is that Arsenio wanted a new opening that required us to be able to lock to the picture. So I use it with sampled sounds and lock it to the picture. I've even sampled Arsenio yelling, "Crank It Up", and the first two bars of the intro.

Both models give me a lot of freedom — I've sampled drum loops and five or six different grooves to be ready for any of Arsenio's impromptu requests.

The S1100 is a great upgrade to the S1000. The S1100 seems to have revitalized sounds that I didn't like on the S1000. I like to use the S1000 for darker, more organic sounds. The S1100 has improved stereo imaging and its high end is more transparent with a nice amount of air to it.

"A great upgrade to the S1000..."

The thing I love about the S1100 is the Mix Page in Select, which is really easy to use. Both models, in general, are very user-friendly. They're fast to get around on, which is an advantage both live and in the studio.

I like that the S1100 can be expanded to 32 megabytes of memory. I find its 8 independent outputs plus stereo inputs very useful since I like to sample in stereo quite often. But I would like to see it expanded to even more outputs.

On a live show things happen unpredictably every night. I like these samplers because they can be ready at a moment's notice, and also enhance my creativity in the studio.

Circle EQ Free Literature #101.

IN REVIEW

E-MU EMULATOR-3

Frank Serafine
Composer and Sound Designer

I use the E-3 on nearly everything I do. All of the sounds I created for "The Hunt for Red October" were done on the E-3 connected to a Sony Pinnacle 650 megabyte computer with an optical read/write drive.

Using it for a music score is where it really shows what it can do; it has the most realistic orchestra sounds from the factory that I've ever heard. I've managed to pull off some orchestral sounds with it that producers have refused to believe were done without a full orchestra.

It only has a few minor drawbacks. It does not have polyphonic outputs, and I often experience aliasing in low registers. But with just a little work, you can overcome these problems quite easily.

It has six Macintosh interfaces, but I only use two for library functions. Believe me, no other model has as good a Macintosh front end.

The E-3's 56-disk Optical Media Jukebox allows for a gigantic library with auto-loading computer access, and it makes editing unbelievably fast. Click twice and the sound loads itself. 30,000 effects would be more than just a hassle to load otherwise. And the fact that it operates on optical disc is great; 11 CD-ROM disks are much more manageable than 5,000 floppies.

Another E-3 feature I really like is the scrub function. It allows you to pinpoint a sound with its pitch wheel. This feature has never really been available for digital but it's very valuable to me when I'm editing.

"The most realistic orchestra sounds I've heard..."

It seems that E-Mu has set the standard for sampling. The E-3 will be the sampler

ten feet away from the cymbal it takes nine milliseconds for the sound to get there. That's 450 samples later. As long as the channels are locked together in stereo, there's no problem. You can edit on the attack of the earlier channel. If the left and right channels have to be edited separately and then locked in stereo, there's no way to tell how much later one channel should be in relation to the other. If you edit the attack on each channel, the spatial representation will be off during playback.

MONO RAILING

If mono is unavoidable, keep a common timing reference between the two channels — SMPTE timecode, perhaps, or a standard start point. Usually the time from the start is the same for both channels. If the earliest channel starts at 2.164 seconds, make the edit on the other at 2.164 seconds (not at 2.198 seconds, where the attack shows on the display). Another option: put a click on both channels just prior to the cymbal hit. When you edit, measure the distance from the click to each of the attacks and edit them both at the time reference of the earlier one.

Also, let samples decay. If the sound ends prematurely and you listen to it too many times there is a possibility that your brain will be

continued on page 72

SAMPLER BUYING TIPS

EVERYTHING YOU NEED TO KNOW WHEN YOU GO TO YOUR LOCAL DEALER

A SAMPLER IS a sonic chameleon that can reproduce just about any sound you want, from a set of timpani to a thunderstorm. While contemporary samplers have more similarities than differences, there are still several important considerations to take into account when you're out at the dealer's for some sampler-shopping.

■ **Sound library.** A sampler without sounds is like a razor without razor blades. You can always do your own sampling, but this may not be possible for sound sources like Bosendorfer pianos and string quartets. Several manufacturers (such as E-Mu, Ensoniq and Roland) have been very aggressive in expanding the sound libraries for their instruments.

An additional source of sounds is "shareware," non-commercial samples that musicians trade with each other; music stores will sometimes let you access their public domain samples as an extra perk when buying a sampler.

■ **Visual editor software support.** For serious sample editing, a visual editing program is essential. Such software packages are available for all major computer families (Sound Designer, Sound Apprentice and Alchemy for the Mac, Sample Wrench for the Amiga, GenWave and Avalon for the Atari ST, and Sample-Vision for the PC and compatibles). Visual editors show samples onscreen, and allow digital processing options such as cut, paste, truncate, resample, EQ, mix, and translate format (e.g., you could bring an Akai S1000 sample into the editor, then send it to an Emax II).

Not all sample editing packages support all samplers, so check carefully. However, virtually all visual editors support the MIDI Sample Dump Standard (SDS), so if your sampler can transfer samples in this manner, you're covered.

Also note that some samplers (e.g., Roland S-50 and S-770) will also include a video out jack for hooking up to a monitor. This may actually eliminate the need to tie up your main computer with a sample editing package — a plus.

continued on page 72

Sample your favorite synth sounds, then add some aftertouch, layering, splits...whatever.

FUN WITH SAMPLERS

IDEAS AND APPLICATIONS FROM CRAIG ANDERTON'S LAB

SURE, you can load in disks and play back sounds — but consider these other sampling applications.

■ **Sampling vintage synths.** Remember that Casio CZ-101 sitting in your closet? It made some great sounds in its time, but without any aftertouch, velocity, filters and other contemporary features, it may have outlived its usefulness. Or has it? Sample your favorite synth sounds, then use the sampler's processing capabilities to add aftertouch, layering, splits, etc.

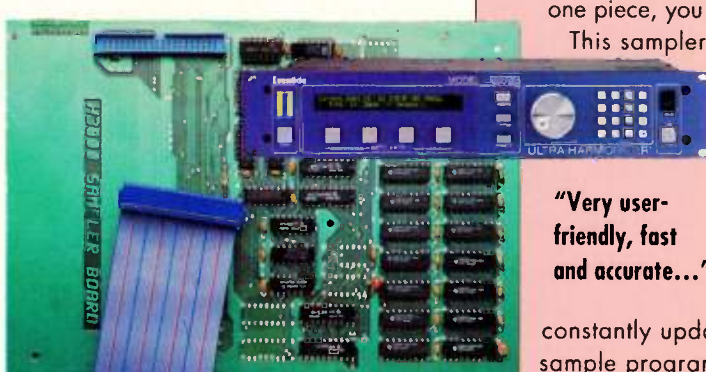
■ **Fixing in the mix.** With a sampler and (for best results) a visual editing system, you can pull sounds from a tape, process them, then record them back on tape again. For example, if a musician's instrument hits the mic, sample that part of the tune, cut out the "thunk," then crossfade the audio on either side of the thunk to smooth over the place where the glitch occurred. Transfer the sampled version to tape, and the problem is solved.

■ **Test equipment.** Record test tones, tuning tones, white noise, spoken left/right channel identification, and other audio test signals (and/or MIDI test notes if there's a built-in sequencer) into the sampler. Boot the disk up at the beginning of your session to check if everything is aligned properly.

■ **Massive vocals.** Sample the vocal track to be "massed." Copy the sample several times to various layers, each with slightly different detuning, start times, vibrato rates, etc. Send the composite sound back to tape.

■ **Audio for video.** Load sound effects into a sampler whose MIDI out feeds a sequencer synced to video (this generally involves syncing to SMPTE via a SMPTE-to-MIDI or SMPTE-to-MTC converter). Press keys associated with particular effects at the appropriate times, and the sound effects will sync up with the picture as long as you don't change the tempo. To get around that problem, check into MIDI Time Code and Events Lists.

—Craig Anderton



of the future because it's a powerful, flexible tool with great sound manipulation capabilities. *Circle EQ Free Literature #102.*

IN REVIEW

EVENTIDE H3000S

Richard Landers
Engineer

I've used nearly every H3000S configuration possible. What I like best about it is that Eventide is constantly experimenting with new things and updating all of its modules — especially the HS322 sampler module.

As a sampler, it is an excellent piece of equipment that allows me to move vocal passages around. It's about as easy to use as a tape machine, yet is extremely powerful.

Its 23 seconds in mono is more than long enough to sample in a piece of a vocal track. Using the scrub wheel you can change the starting point to make drop-in easy..

One of the nicest things about it is that you can change the pitch of a sample without affecting its speed. Conversely, it's capable of changing the length of a sample without affecting its pitch. In many cases, a sampled vocal track sounds better than the originally recorded vocal.

It's very user-friendly. And it's faster and more accurate than other samplers. The sampling programs include expert modes that bring in some parameters that allow you to perform certain things you might not ever need, but if you ever wanted to achieve six different delays in one piece, you could.

This sampler is one of the best available pieces on the market, and it will never be outdated since it is a module that is being constantly updated with new sample programs. *Circle EQ Free Literature #103.*

"Very user-friendly, fast and accurate..."

IN REVIEW

ROLAND S770

Gary Barlough
Primary Engineer for Jon Anderson (Yes); Creative Director for Opio Productions (engineer/co-producer)

While I was working on the new Yes album, *Union*, there were a number of other samplers available to me, but I started playing around with the Roland S770. It had some interesting sounds in its large internal library, and I wanted to hear more. I loved it, and ended up using it on the album.

I was hooked on its excellent vocal sounds, and I liked its strings samples as well. As for its orchestral sounds — particularly the percussion effects — I found them to be very applicable when I was working on a musical with Jon Anderson and Don Black called *Forever and Ever Amen*.

I used it without a computer monitor on *Union*, which made it more difficult to access samples, but I managed to work it out using its readout. Accessing sounds is still excellent.

The sound structure of its library is one of its main strong points. It makes constructing patches a little

easier and better.

"Accessing sounds is really excellent..."

Its large RAM capacity lets you access a wide variety of patches instantly. I know that John Anderson is using it on his current tour. It is a bit more complicated, but that's because its internal architecture is very thorough. It has a lot of room for memory expansion and for the hard disk to be accessed.

For a 16-bit sampler it's also pretty quick. There was obviously a lot of thought put into this model. *Circle EQ Free Literature #104.*

SAMPLING AND THE LAW

DESPITE THE fact that everybody is sampling these days — and that legal issues are hotter than ever in the musical world — there are still no firm legal guidelines to follow. Why not? Because the sampling suits that have been filed are being settled before trial. Therefore, no court has decided whether sampling without permission constitutes copyright infringement.

In the meantime, here's a look at the legal issue sampling raises, as well as some of the licensing procedures now in use:

Using a sample involves the copyright in both the recording and the musical composition. The sampled recording is usually copyrighted by the record company, though only sound recordings made after 1971 are protected by law. A musical publisher will likely own the rights to the musical composition.

Anyone, including an engineer, involved in production and sale could be technically liable for infringement. But the producer, the artists, the record company, the distributor and anyone who earns income from the recording will usually be named as defendant in a sampling suit. A plaintiff can ask for an injunction against further manufacture and sales of the defendant's recording, plus profits or statutory damages (of up to $10,000 for each infringement).

Recording agreements typically include a provision where an artist warrants that product delivered is "original and uninfringed." The artist also agrees to indemnify the record company for settlements, judgments, attorney fees and court costs that may arise from violations of the third party rights. Indemnifications may also cover record producers and production companies. An artist will at least want shared responsibility.

Two primary issues in a copyright infringement suit are whether the defendant has access to the plaintiff's work and whether, to the average listener, the defendant's work sounds "substantially similar" to the plaintiff's work. Substantial similarity is tough to prove if a sample has been altered to bear no resemblance to the original. If the sample is essentially unaltered, access will be easy to prove, but the question still remains as to how much of the preexisting work must be used before the defendant could be liable for copyright infringement.

BY STAN SOOCHER

continued on page 89

THE SAMPLER UNIVERSE

BRAND	MODEL	TYPE	STEREO	PROCESSOR (BITS)	SAMPLING RATE	SAMPLING TIME	STORAGE	EXPANDABLE	HARD DISK	PLAY/LOADING	SEQUENCER	SIGNAL PROC	INPUTS	OUTPUTS	POLY OUTS	SCSI	SDS SUPPORT	LOOPING TYPES	FREE LIT #
Akai	S1000HD	m	y	16	44.1-22.05	23.76	y	y	y	n	n	n	2	11	y	y	y	2	140
	S1000KB	k	y	16	44.1-22.05	23.76	y	y	y	n	n	n	2	11	y	o	y	2	141
	S1100	m	y	16	44.1-22.05	23.76	n	y	o	n	n	y	2	14	y	y	y	2	142
	S950	m	n	12	48-7.5	63.35	n	y	n	y	n	n	2	11	n	n	n	2	143
Casio	FZ-1	k	n	16	36-9	14.5	n	y	n	n	n	n	2	9	y	n	n	8	144
	FZ-10M	m	n	16	36-9	29	n	n	n	n	n	n	2	10	y	n	n	8	145
	FZ-20M	m	n	16	36-9	29	n	n	n	n	n	n	2	10	y	y	n	8	146
DigiDesign (w/MacII)	AudioMedia/ Sample Cell	c	y	16	44.1-22.05	46	y	y	y	n	n	y	2	8	y	y	y	i	147
	SoundTools/ Sample Cell	c	y	16	48-22.05	46	y	y	y	n	n	y	2	8	y	y	y	i	148
E-Mu	EIII	k/m	y	16	44.1-33	94	y	y	y	n	y	y	2	16	n	y	y	2	149
	Emax II	k/m	y	16	39-20	107.4	y	y	y	n	y	y	2	8	n	y	y	2	150
Ensoniq	EPS-16 Plus	k	n	16	44.6-11.1	11	o	y	o	y	y	y	1	2	y	o	n	6	151
		m	n	16	44.6-11.1	23	o	y	o	y	y	y	1	8	y	o	n	6	152
Eventide	HS322	k	y	16	44.1	24m/12s	n	n	n	n	n	y	2	2	y	n	n	n/a	153
Korg	M1*	k	n	16	variable	n/a	y	y	n	y	y	y	0	4	y	n	n	n/a	154
	T1*	k	n	16	48-32	variable	y	n	n	n	y	y	0	4	y	n	y	n/a	155
	WS-AD*	m	n	16	variable	n/a	y	n	n	y	n	y	2	4	y	n	n	n/a	156
	WaveStation*	k	n	16	variable	n/a	y	y	n	y	n	y	0	4	y	n	n	n/a	157
Peavey	SX/SP	m	n	16	48-16	380	y	y	n	n	n	n	2	4	y	y	y	2	105
Roland	S550	m	y	12	30-15	28.8	n	n	n	n	n	n	1	9	y	o	n	4	159
	S750	m	y	16	48-22.05	22.5	n	y	n	n	n	n	2	8	y	y	y	7	160
	S770	m	y	16	48-22.05	22.5	y	y	y	n	n	n	2	8	y	y	y	7	161

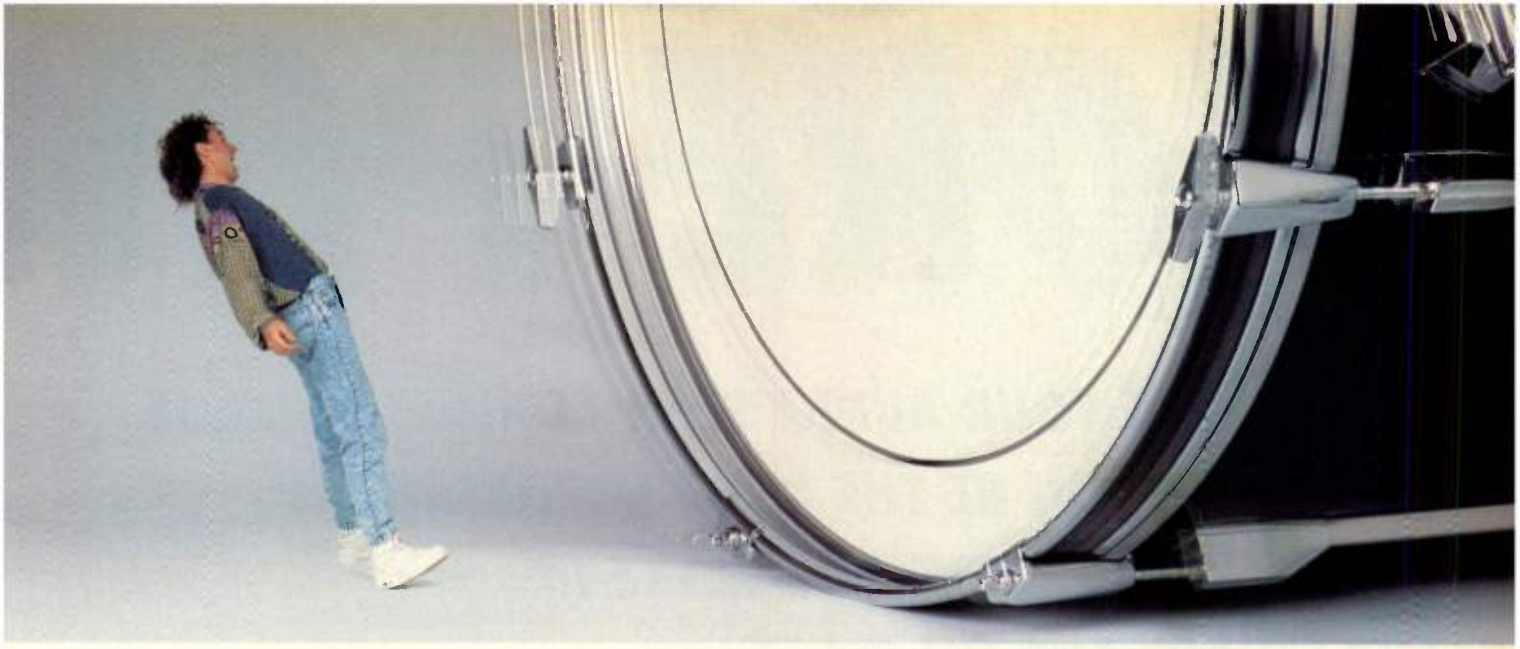
LEGEND

Type: c=card, k=keyboard, m=module. Stereo: refers stereo sampling capability. Rate: in kHz, highest and lowest. Time: in seconds, at highest rate. Play/Loading: unit plays while loading. Other symbols: o=optional, i=infinite. All information provided by manufacturers.

*Playback only

NEW PEAVEY EXPANDER

Peavey has just released the DPM-SX Sampling Xpander module, but it wasn't available for review at press time. The DPM-SX expansion card lets you turn your DPM 3SE keyboard into a sampler. The SX digitally records your own 16-bit samples and then loads them into the DPM SE Sample RAM area (256K on-board RAM). Meanwhile, trimming, looping, mapping and storing is done in the DPM3 with SE (version 2.0) software. Once you've created a sample, it becomes a DPM3 sample wave that you can use in voices and sequences, and even store to floppy — letting you build your own library. The single space rackmount SX offers phantom powered XLR and standard 1/4" inputs. It supports SDS, is expandable to 16 megabytes with industry-standard SIMM's, has selectable sample rates up to 48 kHz, offers armed and manual sampling and can optionally be fitted with an SCSI interface. *For more information, circle EQ free literature #105.*



MAXIMUM IMPACT.

E-mu's new Procussion™ Maximum Percussion Module doesn't do anything in a small way.

Over *1000* drum and percussion sounds based on 16-bit samples from the Emulator™ III library flawlessly reproduced by our celebrated G-chip. Each one too big, too hot, too real to be described by words.

So stop reading. And beat a path to your E-mu dealer to hear Procussion. Since you'll probably have to wait in line, you'll have time to read the rest.

Whether it's rock, hip hop, heavy metal, or Latin percussion, Procussion delivers maximum selection. Organized into 128 drum kits, there are literally hundreds of killer kicks, snares, and toms. And with our multi-channel MIDI capability, you can play up to 16 different kits at the same time.

Of course, great drum parts require more than great samples. You also need the subtle nuances and dynamics that define virtuoso drumming. Exactly where Procussion shines. Extensive real-time modulation and expressive controls, including our remarkable Super-switch™ software, allow you to control virtually every element of your sound by inputs such as velocity, trigger rate, tempo, and more. And to reproduce all your intricate rhythms perfectly, Procussion has a lightning-fast MIDI response time.

With unprecedented editing and layering capabilities, you have maximum flexibility to create startling new sounds. Our 32-voice polyphony ensures that even truly huge sounds won't reduce you to a 3-piece drum kit.

Since Procussion's from E-mu, we don't have to tell you how easy it is to use. Or that it's made in the U.S.A.

So put yourself in front of Procussion. Crank it up. Then brace yourself for maximum impact.

E-MU SYSTEMS INC

E-mu Systems, Inc.
applied magic for the arts

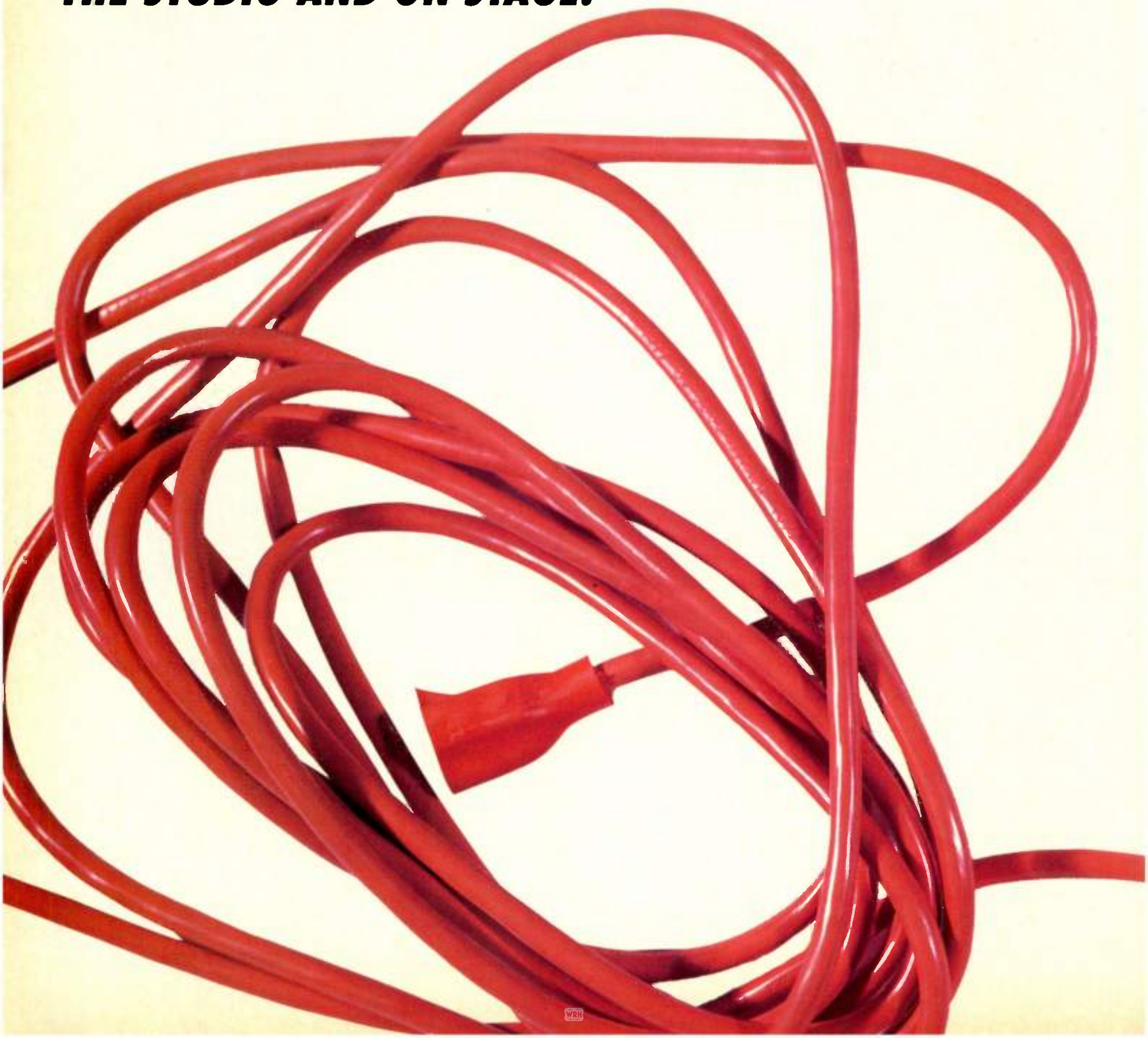
Contact: E-Mu Systems, Ltd., P.O. Box 1, Prestonpans, East Lothian, Scotland EH32 0DG
© 1991 E-mu Systems, Inc. 1600 Green Hills Road, Scotts Valley, CA 95066
All trademarks are property of their respective companies.

CIRCLE 17 ON FREE INFO CARD

WRH

MORE POW

PROTECTING YOUR AUDIO GEAR AGAINST VOLTAGE GONE WILD CAN BE THE MOST IMPORTANT MAINTENANCE MOVE YOU MAKE IN THE STUDIO AND ON STAGE.



ER TO YOU

ALL THINGS ELECTRIC, so the ancient and dusty joke goes, run on smoke. Let out the smoke and the machine stops working. Of course, the joke isn't so funny if it's your gear smoking. From ultra-high voltage lightning strikes that can blast your gear into the afterlife to noisy electromagnetic interference, electricity run amok can be a powerful studio destroyer. Electrical deviations cause problems as serious as smoke-charred power supplies to unexpected and otherwise inexplicable lost or corrupted data.

Overvoltage and overcurrent failure fit into one of three classifications: hard, upset, and latent. With hard damage, the unit fails almost immediately. An upset is a temporary malfunction that is sometimes self-restoring. And latent failures occur when the components are weakened by poor power, leading to premature breakdown.

Since most of your equipment was designed to run off of a 60 Hz sine wave at 110-120 volts, any deviation can lead to failure of one class or another.

OVERVOLTAGE

So how do you know if what's coming out of your wall socket is the right stuff, and how do you fix it if it isn't? There are several types of problems and solutions when it comes to AC power line disturbances.

The most common problem is electrical overstress (EOS): too much electricity. EOS is often identified as surges and transients. Most definitions will hold that a transient is an overvoltage lasting less than 8.3 ms or 1/2 cycle of the 60 Hz waveform, while a surge lasts longer. Any deviation in the transient range of less than ten volts can also be classified as noise. Distortion can be either pulse (one sharp deviation) or oscillatory (several successive disturbances, either consistent or falling off in time).

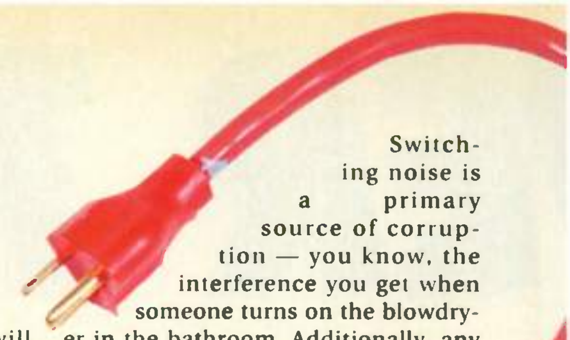
While there are a number of possible sources of overvoltage, the most likely hazard is a spike (possibly up to 6,000 volts at a wall outlet) from a sudden atmospheric electrostatic discharge. Since the theological ramifications of a lightning strike in or around your studio are beyond the scope of this article, we'll look at possible ways to protect your gear from surges and transients.

SURGE STOPPERS

The basic idea is to use a protection device that will pass the standard voltage but clamp or shut any excess before it gets through. Originally, surge protection was designed to protect telephone operators from lightning strikes and consisted of carbon blocks with an air spacing. These blocks would arc or short out at high voltages (air gaps at about 30,000 volts

BY CALIX LEWIS RENEAU

IT MAKES NO SENSE TO TRUST ALL OF YOUR GEAR — AND YOUR ART — TO THE WHIMS OF THE WALL SOCKET.



per inch.) Today there are three basic components available for Transient Voltage Surge Suppressors (TVSS): MOV's (metal oxide varistors), gas tubes and silicone suppressors or avalanche diodes (also referred to by the trade name Transorbs). MOV's are generally used on AC lines for high voltage protection, while gas tubes are the most common protection for telecommunications lines and high current/energy lightning protection. Silicone suppressors are primarily used for protection on the microchip level. Most surge and transient protection systems use at least one of these three basic components, while the better units use a combination to balance out the advantages and disadvantages of each component.

There are also three IEEE categories used for TVSS, designated not by component but by where in the system the protection is used and the physical distance from the utility entrance. Category C covers the service (from the utility pole to the meter base) and will need components which can handle upwards of 30,000 volts to 20,000 amps. Category B includes the main and sub distribution panels (the breaker and fuse boxes) and will usually see no more than 6,000 volts and 3,000 amps of surge. Category A, meanwhile, covers all long branch circuits and points of use, such as the wall outlet to the instrument to be protected (although since the standard specifies that Category A locations are at least 30 feet from the B location distribution panel, it is quite possible in most residential and small studio locations that your wall outlets actually fall into Category B. This is important, as the maximum current and voltage you will need to protect against is to a great extent determined by the inherent impedance of your wiring system.) In general, transient voltage and current decreases as it travels from Category C to A.

CATEGORY, PLEASE

So what does this mean when it comes to protecting your gear? Well, for the most applications Category C protection will be unnecessary, both because not everything in the building will need more protection than the natural wire impedance provides and because the power handling requirements of protection at this point make it very expensive. For a small studio application, a combination of protection at B and A will suffice. Your TVSS should consist of a combination of MOV's, gas tubes, and silicone suppressors installed both in your distribution panel for general protection and at the point of use for more sensitive equipment.

TVSS can be installed either in series or parallel with your AC power lines. A parallel arrangement has certain advantages; it's smaller and can be easier to install, but the main difference is economic. At 200 amps (Category B) a basic series TVSS will cost several thousand dollars, while a high-quality, top-of-the-line parallel system can be found for under a thousand.

At point of use, however, a series arrangement becomes more desirable. A series TVSS will usually consist of a parallel suppressor (a gas tube or MOV) for primary protection, followed immediately by a series reactor (either a resistor or choke) and then a secondary parallel suppressor (this time usually an MOV or silicone diode). The reactor provides forward impedance to block transient rise time, giving the primary suppressor time to react, while the secondary suppressor takes care of whatever passes.

A second advantage of the series TVSS is that, while DC and communications systems will use a resistor, most AC series TVSS use chokes for the reactor — which means half of a filter circuit is already built. A well-designed series TVSS includes capacitors to complete the filter.

LINE NOISE

Since so much of the musical and studio equipment in use today is microprocessor based, line noise is an increasing problem.

Switching noise is a primary source of corruption — you know, the interference you get when someone turns on the blowdryer in the bathroom. Additionally, any device that uses a transformer or coil will induce unwanted noise when turned off (as the magnetic field collapses.)

Electromagnetic and Radio Frequency Interference (EMI and RFI) can also corrupt your power, so filtration can often be essential to avoid trouble. It's also a good idea to keep sensitive gear on its own home run or individual branch circuit to minimize power-induced noise from other components, and further to keep all of your studio as electrically isolated as possible. One quick check for potential trouble is to measure ground-to-neutral voltages with a reliable meter. Most computer companies specify a limit of .5 volts; anything higher can cause problems for your gear. If necessary, you can install an isolation transformer at the point of use to eliminate common mode noise (or noise measured in reference to ground.) A transformer of this type can be found in the 250-500 VA range (which will provide around 4 amps) for under $200.

A brief note about grounding and ground loops: a solid operation path to earth or ground is essential for the safe operation of any electrical equipment. That 60 Hz hum in your speakers is an audio problem, not a power problem, and quite often indicates either trouble in the way you've connected your equipment or a breakdown in one piece of gear or another. Removing the ground from a three pronged plug is a temporary solution at best and removes the designed "safety valve" grounding path should something go wrong — which could

For years you've been using our consumer headphones on the job. We decided it was time you had some professional help.

Here's the help you've been waiting for. The new Sony 7500 Series Professional Headphones.

They give you the reliability you expect from Sony headphones. They're rugged, lightweight, comfortable and they give you terrific sound quality. So stop by your professional audio dealer and try the new Sony MDR-7506, MDR-7504 and MDR-7502 headphones. And give your ears the professional treatment they deserve. To hear more, call the Professional Audio Group at 1-800-635-SONY, ext. 911.

SONY

BUSINESS AND PROFESSIONAL GROUP

Sony Business and Professional Group, 3 Paragon Drive, Montvale NJ 07645-1735. © 1991 Sony Corporation of America. Sony is a trademark of Sony

CIRCLE 43 ON FREE INFO CARD

In the right hands they're incredible tools.

We'd like to introduce you to the Crown PZM®-30F and 30R,* two extremely versatile and indispensable microphones for the recording studio.

Like other professional tools, it takes some experience and skill to discover everything they're capable of doing. But, when used properly, they do what no other microphones *can* do.

Unlike conventional mics, the pressure-zone design of the 30 series uses a miniature condenser mic capsule to receive direct and reflected sound simultaneously. Direct sound from the source and reflected sound from a wall, floor or other boundary combine in-phase to produce a wide, smooth frequency response free from phase interference. The result is increased sensitivity, superior clarity and "reach" with little or no off-axis coloration. All you receive is clear, natural sound.

Because of their unique PZM design, the 30 series excels when used to record sound near hard reflective surfaces such as pianos or instruments surrounded by reflective baffles. They're also an excellent choice for recording room ambience or, when used in pairs, for recording near-coincident or spaced-pair stereo.

With some experimentation, you'll find the 30R and 30F open a whole range of possibilities and solutions not available with any other microphone. We think you'll find them an important part of your recording "toolkit."

Like all Crown professional mics, the 30 series carries a full three year unconditional warranty against malfunction with a lifetime warranty on the acoustic system.**

For more information on our complete line of PZMs or a free copy of the Boundary Microphone Application Guide see your Crown dealer or call toll-free: 1-800-535-6289.

Call for your free copy of the Boundary Microphone Application Guide

*The PZM-30F features a flat frequency response, the 30R features a rising response. Also available with a smaller profile as PZM-6F and 6R

** See your Crown dealer for complete warranty details.

crown®

P.O. Box 1000 • Elkhart, IN • 46515-1000

© 1990 Crown International, Inc. PZM and Crown are registered trademarks of Crown International, Inc.

CIRCLE 12 ON FREE INFO CARD

WRH

prove highly dangerous or even fatal. There's no excuse for lifting a ground in the studio situation where you can take the time to solve the ground loop problem the right way.

LOW VOLTS

Once you've installed a TVSS to take care of surge and transients and arrange to filter out unwanted noise, the third problem to consider is low voltage. Low voltage is most often caused by situations out of your control, such as brownouts and blackouts, but there are several options. A voltage regulator can be installed in the system to automatically adjust for variations in the line voltage, but usually this will only be effective in dealing with the smooth fluctuations your gear can handle without problems anyway.

If your area is prone to more serious voltage drops, install a battery backup system. There are limitations here as well. First, battery backups are expensive, especially if they will be expected to support a large amount of equipment for some period of time. Second, battery backup units require some period of time to make the switch from utility to backup. This time is measured in milliseconds and usually is faster than one cycle (about 16 ms), but some equipment can't handle even a brief delay, a situation which will become more prevalent as your gear grows more sophisticated.

This ugly scenario brings us to the Uninterruptible Power Supply, or UPS. A UPS is the most common form of power synthesis equipment, and uses utility power to generate a new, clean waveform for the user. Since this is a new 60 Hz sine wave, the UPS should output clean power in addition to providing instantaneous access to battery power. This level of protection doesn't come cheap. A small UPS (say, the size to run a home computer) will cost from five hundred to a thousand dollars. A system sized large enough to run an entire studio will cost significantly more.

Additionally, even the biggest UPS systems are seldom installed to carry a site through an extended power outage. Most UPS's are installed to give the user (usually in a computer application) time to shut down his system without losing data or damaging equipment or to allow time for a generator system to switch on. In most cases, this type of power protection supply is hardly justifiable for a small studio in terms of expense, but for a larger digital recording studio it could be an important advantage, both in terms of protecting your equipment and avoiding costly down time.

If you 're shopping for a UPS system, the primary thing to look for is that the UPS provide a sine wave at the output with less than 5% THD; some less expensive units output a squarewave (which makes good reed sounds but lousy power). Also keep in mind that the run time of the UPS

continued on page 73

THERE'S NO EXCUSE FOR LIFTING A GROUND WHEN YOU CAN TAKE THE TIME TO SOLVE IT RIGHT.

CIRCLE 51 ON FREE INFO CARD

The Patchbay Perfect

A little attention can keep patch problems at bay. Here's the fact, fiction and folklore about your plug port.

BY RICHIE MOORE, PH.D.

MUCH HAS BEEN written in the technical world about patchbays. The patchbay is that venerable part of the recording studio that is always on the tongue of mixers and technicians alike. Few moments of musical magic are possible without the patchbay. In fact, many rental companies fly them all over the country based on client demands.

If you believe the above lines, there's a bridge I really want to sell you. When EQ's editors asked me to write a piece on patchbays, I did a little research in my library to see what others had written. To my surprise (grin), not a whole lot has been penned on the subject. The patchbay has always been linked to audio voodoo. Knowledge and mastery of the patchbay have always been part of the rite of passage for recording engineers. Understanding of the patchbay was taught to me, and it is now time for me to pass that knowledge on to others.

SWITCHBOARD REVISITED

In my experience, the patchbay is the most important piece of gear in the control room outside of the console and tape machine. The patchbay is the system by which the hardwire design function of equipment can be easily altered. The bay makes the ins and outs of components of the system available at jacks for insertion or substitution of equipment, testing and maintenance.

The major components of the bay are the jacks and the patch cords. Pretty straightforward. There are, however, different ways to set up a patchbay. The original type of bay used a 1/4-inch jackfield. This was because it was inherited from the telephone company. In fact, back in the 1950s, most broadcast, motion picture and recording studios used a double plug type of patch cord and jack assembly. These were big and bulky, but allowed the user to change the polarity of the cable by reversing the plug position. To cut down on the size of the bay, Western Electric (remember Ma Bell?) developed an alternative. This was the single circuit 1/4-inch tip-ring-sleeve, or TRS convention. Even though the 1/4-inch TRS bays were a substantial improvement in space saving, they still took up a lot of room. In a standard 19-inch rack space, you could fit 24 jacks across or 48 in a double row.

Somewhere in the '70s there arrived a new convention that saved a lot of space. This was the TT patchbay and cords. "TT" is an abbreviation meaning "tiny telephone." The patch bays have also been called "bantam" because they are miniature compared to the original. In 19 inches of space, with TT you could put 48 across or 96 points in a double row.

Although jacks and cords come in several sizes, they are constructed the same. The contacts of a tip-ring-sleeve jack are shown in Figure 1. A TRS jack has two springs, the upper making contact with the ring of the plug and the lower with the tip. The frame of the jack grounds the sleeve of the plug.

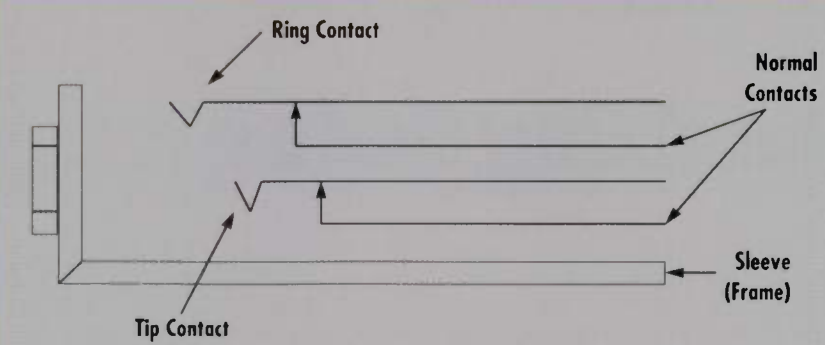


Figure 1: Patchbay Jack

HAVE A NORMAL, HALF A NORMAL

In addition to the two main contacts, there are usually two additional contacts if the bay is wired to have a normal. A "normalled" bay is one in which it is normal for the output of one jack to be tied to the input of another, and vice versa. A "normal" situation is a tape machine output to console line input. Another convention is a "half-normalled" jack, where the signal goes to two places. A "half-normal" situation is where the tape machine output is wired to both line input and the monitor section of a split console. In both cases, when a plug is inserted into the input, the normal is broken and the signal flow interrupted.

Wiring a patchbay should also follow a convention. In a balanced system, the hot (+) side is connected to the tip and the low (-), or neutral side, is connected to the ring. The sleeve is tied to technical earth at the bay. To eliminate possible ground loops and enhance shielding in a star ground wiring installation (most preferable), the shield is attached to the bay at the send and lifted at the device input. The shield is then attached to the device output and lifted at the patchbay. This procedure is known as telescoping the shields. The device ground is attached to technical earth at a central location.

In an unbalanced system using a TRS jack, you need to attach the (+) to the tip and the (-) to the ring. At the device you need to connect the (-) and the shield together for proper signal flow. At the output of the device, connect (-) and the shield together and lift the shield at the patchbay. If you are using an RCA-type (phono) patchbay, the same convention applies.

I always recommend using a metal type jack in the bay. There are some console and other manufacturers that use a plastic jack that attaches to a printed circuit card. These are just not reliable enough although cost efficient. The plastic case does not have enough tensile strength, nor is the contact metal material strong enough to give repeatedly good contact. RCA jacks, though relatively inexpensive, are not good for heavy duty use. The insulating material becomes deformed and makes for loose and intermittent connections.

The hardwire method to a metal jack is the best approach for wiring. This is done with a direct solder connection to the metal jack either free-standing or attached to a PC card. The interface can be either a multi-pin connector (ELCO or Cannon DL), or a ribbon connector to the jack field. The most unreliable connections I have seen are with molex-type connectors attached to printed circuit cards. Take it from me: try to avoid these if you possibly can.

The most important thing that you can do to maintain your patchbay is to keep it clean.

PATCH HYGIENE

The most important thing that you can do to maintain your patchbay is to keep it clean. This starts with keeping the patch cords cleaned. Although a thankless job in itself, it helps keep the jacks clean and minimizes poor connections. My two personal favorites for cleaning patch cords are Polysand sheets from Audio Accessories in New Hampshire (603-466-3335) or dealers around the country, and Doe's Plug Polish from Walter S. Doe Company in Cleveland, Ohio (216-845-7716). I don't know where else to get Doe's except from the manufacturer. Polysand does a beautiful job, but is a little on the expensive side. Doe's is cheap and lasts a long time. It has a strong odor and is a bit messy on the hands, but it really cleans the plugs. I even use it on my wedding ring. I recommend cleaning the patch cords once a month no matter how much they are used.

Cleaning the jacks is a whole other story. If you keep the cords clean, the jacks stay fairly clean if they are dusted and squirted with compressed air on a regular basis. Do not empty coffee or ashtrays into the patchbay. In the old days, a wire brush was used to clean the interior of the jacks. But with every insertion, small amounts of the jack casing would wear away, causing a loose fit. The devices that I use are the Vertigo burnishing and injector tools from Vertigo Recording Services in North Hollywood, CA (818-907-5161). For burnishing, you insert either the 1/4-inch TRS or TT tool and twist a few times. The injector allows you to attach some cleaning solvent to clean the normal contacts. Both of these should be used sparingly. The bad side of the burnishing tool is that it can wear down the contact if used exceedingly, and small amounts of metal can fall into the jack, possibly creating a short. These tools are wonderful when used in moderation.

All of the above has to do with jacks and plugs made with brass. Mogami makes a wonderful patch cord that uses a Neutrik nickel-type alloy that does not tarnish easily. I have been using them for about two years and think they are wonderful. Polysand is great for cleaning these plugs when the patch cord has been sitting in a cup of coffee overnight.

As you may now be aware, there is not a lot to the patchbay. It is just real important to purchase quality jacks and plugs and to keep it clean. Not much else to do.

Until next time, stay connected. EQ

Richie Moore, *Ph.D., of Novato, CA, covers studio maintenance techniques for EQ.*

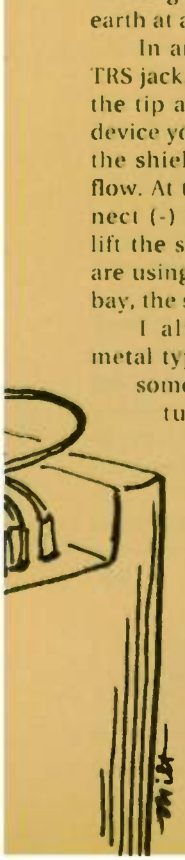


Illustration by Milton Reyman

Remixing Redux

IN THESE DAYS of multi-format marketing in the music business, remixing is taking over as the major record company strategy in trying to guarantee exposure and success for a new single release.

Here's a list of some of the most important pointers in the process of mixing a song. They'll help you save time, budget money and have your remix come out hassle free, and hopefully, sounding slammin'.

Simple strategies to help you get the most out of a remix.

BY TUTA AQUINO

■ A few basic items should be on everyone's checklist: Healthy Time Code/SMPTE, click tracks (whenever possible), tones (1 kHZ, 10 kHz and 100 HZ), and track sheets. Believe it or not, most of the time, one out of four of these essentials is forgotten.

■ It is always good to organize your slave reel so you have sets of composite mixes of drums, keys, bass and vocals. It is also good to make a second slave pass of the song on your slave reel. At the last minute, you might want to whip up an alternate set of tracks for production.

■ When choosing an overdub studio, make sure you know how the room sounds in relation to the mix room you are using for the project. You don't want any surprises when you get down to mixing. Try to rely on the same set of nearfield monitors in both studio environments. Don't rely so much on big speakers.

■ Try to hire a studio, programmer and an engineer for your overdub date who are used to working with the kind of music you're producing. This will ensure that most of the equipment you will need will be available and that the personnel will be most effective both in time and usage of equipment. Create a "team" consisting of programmer, engineers (both for overdub and for mixing) and recording studios. This will help you in creating your "sound," which should be the basis for getting hired for work.

■ When trying to add to the original production of a song, you have to worry about locking to the tempo of the song. Two scenarios could come up:

A) You get lucky and get a click track on the original master. Piece of cake! All you have to do in this case is create a tempo map using the existing click and a sync box like the Roland SBX-80.

B) You have to sweat it and try to create your own click track. Big hassle! This puzzle could take a while to sort out. But there are various ways to deal with the situation. 1) You can use a couple of pieces of equipment that will put out a click from "seeing" all of your tracks or a combination of a few of them, namely: Garfield's Human Clock, Aphex's Studio Clock and The Russian Dragon. 2) You can try to manually play a click. Very time consuming! 3) You can use a combination of delayed click and snare tracks, and a couple of very fast fingers on mute buttons. 4) Or you can use a very elaborate but effective rig consisting of a combination of Trigger to MIDI boxes and computer with Performer software.

■ When printing your new tracks on the slave reel, be sure to organize them so a minimal amount of cross patching has to be done at the mixing stage. Try not to make composites of tracks, especially drums and/or percussion, to avoid mixing headaches. This way, you will also have more flexibility.

■ Try to split your schedule in three 12 hour days. One for your overdub date, one for setup of your mix, and the last day to fine-tune and print your mixes and various passes. Recording studios, in this day and age, are more than willing to deal, so press the issue and split that 24 hour lockout in two 12 hour periods.

■ A personal preference is to make use of tape compression when printing kick drums and bass tracks, especially the fat analog sounds of old synthesizers. Also, I am still an advocate of analog two-track 1/2-inch at 30 ips. Usually, I print the mixes on DAT and transfer the final or edited versions to tape. I find +6/185 alignments give the necessary room and best results.

■ Finally, it is always hard to keep whoever hired you for a job off your back, but with the use of some diplomacy, you can gain their confidence and buy yourself some peace of mind. Nothing worse than having the A&R person, artist or manager breathing down your neck while you're trying to receive those inspirational vibes from the music gods.

Be creative and trust your instincts. It always works. Good luck and good sounds! EQ

Tuta Aquino *is the owner and operator of Prime Cuts in Manhattan, a 24-track/MIDI recording studio.*

FROM THE MASTERS OF MASS STORAGE

INDUSTRIAL STRENGTH STORAGE SOLUTIONS !

•REMOVABLE •FIXED •OPTICAL •CD ROM

RX-2

44 Megabyte Removable Media Hard Disk Drive 20 ms seek time

AVAILABLE FOR: ▪AKAI S1000 ▪EMAX II,III ▪ENSONIQ EPS16+ ▪MACINTOSH ▪ROLAND S-770

CD-2RX

CD ROM Drive / 44 Megabyte Removable MediaHard Disk

AVAILABLE FOR: ▪AKAI S1000 ▪EMAX II,III ▪ENSONIQ EPS16+ ▪MACINTOSH ▪ROLAND S-770

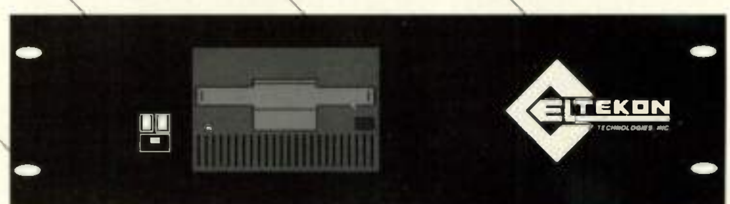
EX-2

Internally Mounted Fixed Media Hard Disk Drive 20Mb-1 Gigabyte

AVAILABLE FOR: ▪AKAI S1000 ▪EMAX II,III ▪ENSONIQ EPS16+ ▪MACINTOSH ▪ROLAND S-770

MX-2D

650 Mb. Removable cartridge Read / Write Magneto Optical Drives



AVAILABLE FOR: ▪AKAI S1000 ▪EMAX III ▪MACINTOSH ▪ROLAND S-770 ▪SOUNDTOOLS

NOTE: OUR OPTICAL IS FAST ENOUGH FOR DIRECT TO DISK AUDIO USING SOUNDTOOLS™.

INTRODUCING THE SAMPLERS EDGE SOUND LIBRARIES FOR THE FOLLOWING SAMPLERS:

S1000,EPS 16PLUS AND SAMPLE CELL

AVAILABLE ON FLOPPY DISK, REMOVABLE MEDIA AND CD ROM.

THESE ARE COMPLETE LOOPED SAMPLES NOT RAW DATA.

ELTEKON TECHNOLOGIES INC

"THE MASTERS OF MASS STORAGE"

ELTEKON TECHNOLOGIES, INC. 37491 SCHOOLCRAFT RD.
LIVONIA, MICH. 48150 313-462-3155

CIRCLE 41 ON FREE INFO CARD

EQ&A

continued from page 12

encounter is addressing the individual gates.

An automatic mixer usually provides its own mix output, and may only accept microphone inputs. If you are also able to extract gated signals from each channel, the automatic mixer could be placed between the mics and the console inputs. If, in addition, the automatic mixer will accept line-level input signal, then the automatic mixer could be connected to the console insert points.

Incidentally, the gate functions of Biamp's automatic mixer are provided by a unit which is separate from the mixer itself. The Auto Gate, as it is called, not only accepts line-level inputs and provides separate gate outputs, but does so on a three-conductor jack, which interfaces to a standard three-conductor Patch/Insert jack using (1) three-conductor 1/4" Phone cable.

The major determining factor on whether or not you use an automatic mixer/gates on your main or monitor console is: does it eliminate unwanted noise without cutting off the desired signal or being audible in its function?

Jay Bradshaw, Marketing Support
Biamp Systems

Q My question concerns signal-to-noise ratio with outboard effects processors. While my dry mixes are good, as soon as I turn on my effects modules, Voila!, a cornucopia of noise. How do you determine the optimum settings for the (a) Aux Send (individual and Aux Outs); (b) input and output levels of the effect; (c) internal output setting (e.g., the master effects mix level on the Alesis Quadraverb); and (d) the console Aux Return?

Jeff Jacob, Corinna, ME

A Your letter states that your "dry mix is usually very acceptable" indicating that the gain structure of your effects device, not grounding, is the most likely problem. At least, you won't have to tear your whole studio apart and rewire it to fix a gain-structure problem. (Troubleshooting rule number one — always check the easy stuff first).

A signal processor has a dynamic range that is dictated in part by the analog-to-digital (A/D) encoding process. Most processors currently available use 16-bit encoding. The dynamic range of an A/D is 6 dB per bit, which results in a theoretical dynamic range of 96 dB (16 bits x 6 dB = 96 dB). Theory and practice rarely yield the same results, so actually, the digital signal processor will have dynamic range performance of 75 dB to 90 dB. Of course, these processors are probably going to be connected to an analog mixing console with dynamic range greater than 110 dB.

So how do we best fit the 75 dB dynamic range of the processor into the 110 dB dynamic range of the console? If we send too much signal to the input of the processor, the result will be clipping. Over-driving a digital processor results in a particularly nasty sound, so most audio system operators tend to go too far in the other direction to avoid it. And what lies in the other direction? The noisefloor of the processor.

Noise is less objectionable than clipping and is frequently dismissed as an inevitable by-product of a "noisy" processor. The fact is that many owners of digital processors never actually use more than 12 of the 16 bits they paid for. When too little signal is sent to the input of a processor, the result will be a low-level output signal. In order to get enough effect into the mix the engineer will then increase the gain of the effect return. This brings the effect to the desired level, but also brings excessive (and unnecessary) noise to the mix. Here's a method of steering between the Scylla of clipping and the Charybdis of noise [check your mythology books, readers]:

In the following example we are going to add reverb to a synthesizer track. The digital reverb is being fed from the console's AUX 1 bus. The stereo outputs of the digital reverb are being returned to the console's AUX RETURN 1.

✓ Match the input and output levels of the console with the input and output levels of the digital reverb. Many processors have inputs and outputs that can be set (via rear-panel switches or front-panel level controls) to match the input and output of either professional audio equipment (+4 dBu) or instruments and home recording equipment (-20 or -10 dBu).

✓ Check the owner's manual for your console and digital processor. Match (as closely as possible) the input level of the processor to the nominal output level of the console. Let's use the Yamaha MR1642 console and Yamaha SPX90 as examples. The rear-panel markings of the mixing console indicate that both the AUX SEND and the AUX RETURNS operate at a normal level of +4 dBm. This tells me that the SPX90 input and output level switches should both be set to +4 dB.

✓ AUX RETURN fully counter-clockwise. We'll get back to it in a moment.

✓ Channel AUX SEND to nominal (about 2 o'clock or two thirds of the way clockwise).

✓ Play back the synthesizer track and bring the AUX SEND MASTER to nominal (about 2 o'clock) or bring up the AUX SEND MASTER until the PEAK LED in the AUX 1 Meter flickers during the loudest passage.

✓ Continue to play back the synthesizer track and adjust the reverb processor's INPUT LEVEL control while watching its input meter. When the synth track hits a dynamic peak, the top segment of the processor's LED meter should flicker briefly. To get the best signal-to-noise performance, try to operate the processor as close to clipping as possible without actually getting into clipping.

✓ Slowly increase the level of the AUX RETURN until you hear the desired amount of reverb in the mix. You will probably find that the AUX RETURN is being run at a much lower level than previously used. Listen for clipping distortion. If you hear it, slightly reduce either the console AUX SEND or the processor INPUT LEVEL controls just enough to eliminate the clipping.

✓ If more than one input channel is

being processed, balance the amount of reverb on the various channels using the channel AUX SEND controls — but all effected channel sends should be near the nominal level.

✓ The amount of reverb in the mix is controlled by the AUX RETURNS. Never decrease overall reverb by reducing the AUX SEND level. This allows all the residual noise from the reverb to remain in the mix. The AUX SEND level should only be reduced if the reverb processor is being overdriven.

✓ Repeat this procedure for each AUX SEND/RETURN and processor in your system and you should experience considerable reduction of noise.

If not, it's time for more complex trouble-shooting. Start by inspecting your cables and connectors. Are you using good-quality shielded cable for all audio signals? Check all solder connections. A good solder connection looks smooth. Be sure to look for tarnished or corroded contact surfaces on the connectors — a major trouble spot. If you are suspicious of a cable, substitute a cable you know to be good and see if you observe any change in the symptoms.

It is also possible that some of the low-frequency noise is being induced by the proximity of audio cables to AC power cables. Separate them! Audio cables should cross AC power at right angles. Speaking of induced hum, it may be that the magnetic field from some high-current piece of gear (like the power-amp) is generating hum in a signal processor. If processors are near your amplifier, try separating them.

Some of the signal processors you mentioned use external wall-mounted power supplies. Check to be sure that each processor is connected to the appropriate power supply. Connecting the wrong voltage to a processor can result in hum or distortion (or worse). If you are using an XLR to phoneplug adapter to connect an unbalanced signal processor to balanced console input, be sure that console phantom power on the input channel is switched off.

Gerald W. Tschetter
Pro Audio Division,
Yamaha Corporation of America

CONSERVATORY OF Recording Arts & Sciences

At Last! An Affordable Master Recording Program Taught by Award-Winning Professionals

If your goal is to succeed in the music and recording industries, our professional training program will give you the hands-on experience and technical know-how you need.

CALL FOR INFORMATION AND CATALOG

1-800-562-6383

1110 E. Missouri, #400
Phoenix, AZ 85014

Licensed By The State of Arizona.
Approved For Foreign Students.
NATTS ACCREDITED

CIRCLE 11 ON FREE INFO CARD

BE A RECORDING ENGINEER

★ TRAIN AT HOME ★

Learn to become a professional recording engineer at home . . . at a fraction of the cost of most resident schools.
Easy Home-Study practical training in Multitrack Recording including the latest in Digital and MIDI.
Join the Audio Institute's successful working graduates or learn how to build your own studio. Career guidance. Diploma. Job placement.

Audio Institute of America
2174 Union Street Suite 22 AN
San Francisco, CA 94123
AIA
For A Sound Education™

CIRCLE 07 ON FREE INFO CARD

THE STUDIO SOURCE.

INNOVATIVE SOLUTIONS FOR PROFESSIONALS

Prompt, Courteous Service • Always the Lowest Price

• SOUNDTRACS IL48/ERIC/CMX/MRX STUDIO CONSOLES • N.Y.' EXCLUSIVE AMEK DEALERSHIP • C.A.D. CONSOLES • EVENTIDE DIGITAL SIGNAL PROCESSING • LEXICON 300 DIGITAL PROCESSOR • ROLAND DIGITAL SIGNAL PROCESSING SYSTEMS • YAMAHA DSP SYSTEMS • YAMAHA DMP7D DIGITAL MIXING CONSOLE, CONDENSER MICS • CONDENSER MICS BY JOSEPHSON, AKG, CROWN, NEUMANN, SONY & SENNHEISER • APPLE MACINTOSH COMPUTERS (NY's ONLY APPLE MUSIC DEALER) • SOUNDTOOLS DIGITAL AUDIO RECORDING & EDITINGSYSTEM • TASCAM ATR 80, 24 TRACK 2", MSR 24 TRACK 1" • MONITORS BY TANNOY, EV,JBL & UREI • PROCESSORS BY DBX, SYMETRIX VALLEY PEOPLE, ALESIS, T.C. ELECTRONICS & AUDIO LOGIC • D.A.T. FEATURING THE TASCAM DA-30, JVC DSDT 900, TIME CODE CAPABLE • SAMPLERS FEATURING THE AKAI S-1000 & S1000HD, THE ROLAND S-770, S-550 & S-330, THE EMU EMAX II, EIII & PROTEUS • INTRODUCING D.A.R. DIGITAL WORKSTATION - POST PRODUCTION POWER TOOL • DRAWMER • CALZONE CASES

Tape • Cables • Software • Parts

• DESIGN • SALES • SERVICE • PARTS • INNOVATIVE FINANCING PLANS • RUSH DELIVERY

SamAsh PROFESSIONAL

ASK FOR ANY OF OUR SALES REPRESENTATIVES:
BOB QUINONES
TIM FINNEGAN
ROBERT MILLER
DAVID CHARLES
BRUCE Mc PHERSON
BOB CAPUTO

723 SEVENTH AVENUE (THIRD FLOOR)
NEW YORK CITY • 10036 • (212) 719-2640

Spin the Groove

Putting together the ultimate dee jay system.

BY J.D. SHARP

WE'RE OUT TO create a premium dee jay system, one that will pack the punch that gets dancers moving. The elements of these systems are quite straightforward: a mixer, several sound sources, some signal processing to spice things up, and a playback system that won't quit when the action gets hot.

But as simple as it sounds, there are quite a few choices to be made, and the harder you look at it, the more difficult it is to say that a certain piece of gear is the absolute top in its class. The manufacturers have turned out an incredible array of high-quality stuff.

MIXER & PLATTER SPINNER

Nonetheless, the time has come to name our favorites, so here goes: The heart of any dance system has to be the mixer; here we're partial to Numark's DM 1975, a top-of-the-line unit with four stereo inputs, assignable to EQ, and (here's the kicker) four seconds of sampling, ideal for injecting that creative sparkle into an evening's work. As an alternative, consider one of Biamp Systems' fine mixers; there's no sampling, but the audio quality is great and they're available in several price ranges. If you're looking for a cost-effective alternative, the Yamaha MJ100 should be very attractive, combining a full complement of features with a very reasonable price tag.

Sound sources are next; the leading edge definitely leans toward CDs for their scratch resistance and digital-quality sound. Here once again, Numark fills the bill with their CD 5020. This is a dual-CD rack mount unit with optional remote control, and it features a ±8% pitch dee jay adjustment, making beat matching and time fitting a breeze. Of course, there's many a dee jay who's a magician with a turntable, so we'd better specify at least one and perhaps a pair of Technics SL 1200 MkII units; even after years on the market these are still the standard, with plenty of torque to bring cuts up to speed post haste. Alternatively, try units from companies like Gemini. At some point, we'll probably end up with a CD player that you can 'scratch' on, but it hasn't come to pass yet.

SYSTEM SPEAKERS

Moving down the signal line, it's time to consider a playback system. If you have the wherewithal to move them, it's hard to top JBL's SR Series for their combination of high-fidelity sound, incredibly rugged cabinet construction, immense power handling capacity and reasonable prices. For thunderous lows, we'd recommend a pair of JBL SR4718's featuring a 600 watt, eighteen-inch woofer in a vented cabinet with response down to 30 Hz. The mids and highs can be handled very nicely by a pair of JBL SR4722's, each with a 12-inch speaker and a Bi-Radial horn mated to a titanium-diaphragm compression element. If the budget is really unlimited, substitute JBL's three-way SR4732's instead, with two 12-inch drivers, a two-inch mid/high compression driver, and an ultra-high-frequency 2404H Bi-Radial transducer.

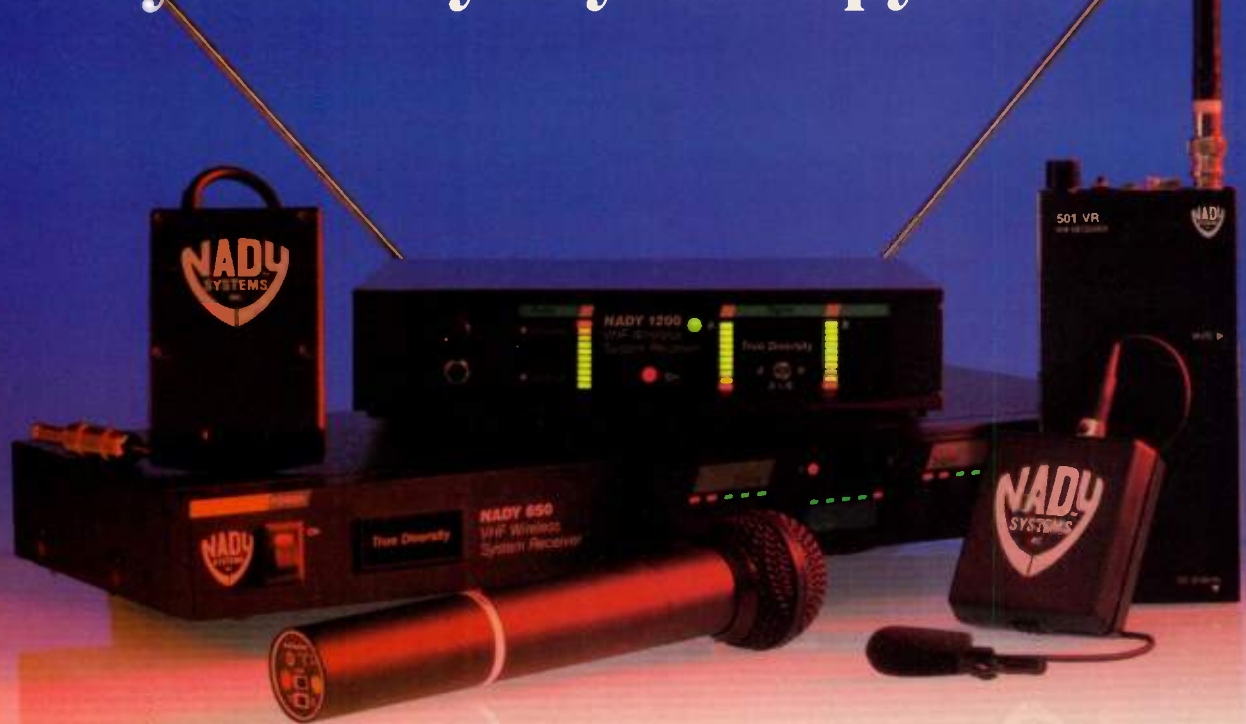
For a money-saving alternative that can still handle tons of power and pour on the sound, check out Peavey's revised SP-4ti's. Power handling has been upped to 350 watts continuous, and the one-inch compression driver now sports a titanium diaphragm.

Dual 15-inch woofers put out pounding bass. A surprising number of people buy these speakers without taking advantage of Peavey's Dynamic System Controller, and it's a shame; this unit takes care of all electronic crossover needs, and has intelligent circuitry that monitors how much power is going to each component of the system, keeping everything clean yet allowing you to run close to the max constantly. Active EQ also extends both the high and low end. Get a pair (for stereo) if you go with SP-4ti's.

For our JBL system, we'd employ a Rane AC22 stereo two-way crossover for biamplification, although you could get fancy, bypass the internal passive crossovers in the SR speakers, plug in a Rane AC23 two-channel 3-way crossover, and tri-amp — but it's probably not necessary. If you're looking for a system that one person can move alone, the speakers of choice would be Ramsa; four WS-A200 two-way 12-inch systems, combined with four of their WS-A240 compact subwoofers put out a surprisingly solid wall of high-fidelity sound when stacked in an array, yet the whole system fits in a (big) station wagon. Crossover functions, as well as active

The Gemini Quartz Turntable

If every wireless company claims theirs is the best, why do they try to copy ours?



First, let's talk technical.

A lot of newcomers to the wireless field are making pretty bold claims these days, but they all have one thing in common. They all say theirs are the best. Well, we've got news for them: all wireless systems are not created equal.

At Nady Systems, when we say we are the leader, we can back up our claims. Everyone knows modern wireless systems use companding for noise reduction, yet few realize that we invented and patented audio companding circuitry for wireless over 12 years ago.

Sure, other companies have attempted similiar designs. Unfortunately, most of them simply fail to utilize the existing technology properly, so at 120 dB dynamic range, we still enjoy up to a 20 dB edge over many of our competitors.

In fact, what we find interesting is that even our early prototypes are still significantly quieter than these so called "new and improved" wireless systems. You may not always need the kind of performance our wireless systems can deliver, but isn't it good to know that the technology is there if it's required?

Let's talk product.

Our Nady 650 VHF Wireless System packs a surprizing number of professional features into a mid price system. It's rack mountable and can be used with up to 10 channels operating on 10 different frequencies at the same time. And, of course, the 650 VHF has a highly sensitive True Diversity front end for drop out free performance in any environment.

The top of the line Nady 1200 VHF Wireless System defines state of the art wireless technology. Features include 20 channel simultaneous operating capacity on custom frequencies. The 650 and 1200 are ideal for any wireless application, such as the theatre, churches, meeting rooms or live entertainment. Musical instrument and lavalier mic body pack transmitters are available. Our 1200 HT handheld is a sleek, metal wireless mic with no protruding external antenna—another much copied Nady innovation.

Our 501 VR Wireless System delivers the highest performance available and features a portable, rugged metal case receiver perfect for video and film production.

So whether you need a wireless system for a modest facility with a small budget or a multi channel wireless for a large location, you can count on Nady to work flawlessly, give you complete coverage and sound superb doing it.

Finally, experience.

Nady has been in the wireless business for over 12 years. And we now sell more wireless than the rest of the industry combined. Nady almost singlehandedly pioneered the use of wireless in the live entertainment industry, where road ruggedness and dependability in a wireless system is essential. This means that when you install or use Nady, it will not only work the first time—but keep on out performing other wireless units for a long time to come.

Being on the leading edge of wireless technology practically since its inception, we have given the competition a lot of things to try to copy.

So why settle for imitations? In Wireless there's really only one leader—Nady.

NADY™ SYSTEMS INC.

CIRCLE 27 ON FREE INFO CARD

Contact your local dealer or Nady Systems, Inc. 6701 Bay Street, Emeryville, CA 94608. (415) 652-2411 for more information.

© 1990 Nady Systems, Inc.

WRH

EQ, are optimized by Ramsa's WS-SP2A processor.

POWER IT UP

Of course, power amps make the whole thing go. This is an area where it's particularly hard to make choices, because everybody seems to have a different opinion. In fact, just about any high-powered unit from any of the major manufacturers could do the job very well, so you're free to substitute your preference at this point.

> We're out to create a dee jay system, one that gets dancers moving.

With that disclaimer aside, here are our recommendations. Crown amplifiers have a remarkable record of reliability and a fantastic warranty, along with the ability to drive all sorts of troublesome loads and impedances without shutting down. A pair of Microtech 1200's (or Macrotechs if you must have front-panel volume controls) should take care of most needs. Power-mad freaks should substitute the Macrotech 2400 instead for low end.

If keeping weight to a minimum is of the highest priority, take a look at the new power amps recently introduced by Stewart Electronics; their PA-1200 and PA-1500 units are reasonably priced, very lightweight, and yet are capable of pedal-to-the-metal performance just about indefinitely. Then again, you'll find many sound jocks who swear by Crest amps.

We'd make the crucial connection between amp and speakers with Monster Cable to deliver the maximum punch. A dual 5-band EQ provides a means for overall system tone shaping: the Rane ME-15 combines superb specifications with a moderate price tag, so we'd include it.

We also need a drum machine — the Alesis SR16 should fit in nicely. A digital effect patched into the mixer can add a new dimension to a dee jay's rap, so we'd insert a Boss (Roland) SE-50; it offers a great combination of pitch shift, delay, reverb and even vocoding effects, all with very high quality.

This system provides all the firepower needed to put over a compelling performance. Just add record-spinning expertise and you're in business. EQ

FOR MORE INFORMATION

COMPANY	LIT #	PHONE
Alesis	107	(213) 467-8000
Biamp	108	(800) 826 1457
Crown	109	(219) 294-8000
Crest	110	(201) 423 1300
Gemini	111	(718) 851-6000
JBL	112	(818) 893 8411
Monster Cable	113	(415) 871-6000
Numark	114	(201) 225 3222
Peavey	115	(601)483-5365
Ramsa	116	(201) 348 7000
Rane	117	(206) 355-6000
Roland	118	(213) 685 5141
Stewart Elect.	119	(916) 635-3011
Technics	120	(201) 392 6140

J.D. Sharp, EQ's Systems columnist, can be found masterminding systems like this one at San Francisco's pro audio dealership Bananas-at-Large.

Why pay through the nose to please your ears?

Yorkville YSM-1 Near Field Monitors

Dimensions	16.25" x 9.75" x 8.75"
Frequency Resp.	40 Hz - 20 kHz
Efficiency	90 dB
Impedance	8 Ohms
Woofer	6.5"

"Under $300 ? ! ? ! ?"

That was Chris Meyer's response when he found out the price of our new YSM-1 near field monitors, while reviewing them for Home & Studio Recording. Here's a sampling of his findings...

"A couple of things hit me right away. The first was their smoothness...the second thing was their imaging...

Their sense of air and stereo positioning was very nice, even when I got quite close to them.

...ideal in a cramped bedroom studio."

"The bass response extends further down than you would expect.

Transients were far tighter than I expected...

Vocals sounded especially good...

These little fellas are simply great value for the money."

After 28 years of designing and building speaker cabinets, we know how to get it right...the sound and the price. Why not hear them yourself at your Yorkville Dealer?

And by the way...the price is $280 – PER PAIR.

Yorkville

IN CANADA
Yorkville Sound Ltd.
80 Midwest Road
Scarborough, Ontario M1P 4R2

You'll be hearing from us. . .

IN U.S.A.
Yorkville Sound Inc.
4600 Witmer Industrial Estate, Unit #1
Niagara Falls, New York 14305

CIRCLE 50 ON FREE INFO CARD

A Sound Investment

Substantial profits can be earned by duplicating cassette tapes in-house. But before committing capital, make sure the system you invest in offers the following:

- Audiophile sound quality. Do not settle for less. Such quality reflects your company image and translates into repeat and referral business.
- Flexible modular design. Increase the capacity of your system economically as you build your business.
- Rugged industrial duty construction. Trouble free operation.
- Maximum flexibility. Interface with any master format on the market.

KABA is the recognized leader in audiophile quality tape duplication equipment and production. We offer and support the whole package. We built our business by understanding your business.

The KABA "Profit Package" includes:

- A four-position KABA four-track real time and 2X duplication system.
- Shrinkwrap L-Sealer, heat gun and film.
- 500 top quality cassettes, custom loaded with chrome tape and imprinted to your specifications.
- 500 Norelco boxes.
- Promotional support material to help you build your business.
- A supply of pre-recorded cassettes to demonstrate the quality your KABA system can provide to potential customers.

Save Over $1,100.00

Only $5,385

(limited time offer)

KABA REAL TIME

KABA
Research & Development

For more information on this or larger packages call:

24 Commercial Blvd., Novato, CA 94949 (415) 883-5041
1-800 231-TAPE FAX (415) 883-5222

Stick It To 'Em

Ten off-the-wall suggestions for getting great sound with PZMs.

BY BRUCE BARTLETT

ANYWHERE THERE'S a hard surface, someone will stick a PZM microphone on it and try to record something!

You've seen those funny-looking plate mics called Pressure Zone Microphones, made by Crown. A PZM is meant to be used on a boundary or surface such as a wall, floor, table, or panel. You can place it almost anywhere, yet it manages to sound airy and lifelike.

Standard PZM techniques have evolved over the years: you can mike a grand piano with two PZMs taped to the underside of the lid, pick up room ambience with PZMs on the walls, or record cymbals with two PZMs over a drum set.But thanks to ingenious experimenters, many bizarre applications have surfaced. Like these...

1 On a drummer's chest. Innovator Chris Altizer once made a drummer's special PZM chest mount. However you do it, the PZM picks up the set as the drummer hears it — all the toms, snare, bass drum, and cymbals. Altizer says, "In small clubs, or on large outdoor stages, one PZM beats ten regular mics for convenience any day in my book."

2 On your head. Binaural recordists such as Frank Zappa have taped a PZM over each ear, then played back the recording over headphones. The result? An amazingly realistic sense of space, for less than the cost of a dummy head.

3 On a kick drum head. J. Paul Hancock of AUDIO I/O taped a PZM to a Maxi-Pad and attached the pad to the kick drum beater head. According to Hancock, "this gave a great kick and snare balance with an open, punchy snare sound."

4 In a garbage can. For a trashy snare sample, nothing beats a PZM in the bottom of a waste container.

5 On the floor by a guitar amp. To fatten the sound of an electric guitar in a demo, Vince Motel laid a PZM on the floor 3 feet from the speaker cabinet. Then he placed a hard panel 6 feet from the cabinet, behind the PZM. Acoustic reflections from the panel thickened the sound. By changing the distance between speaker, PZM, and panel, Motel got an ambient sound without adding a second room mic.

From top to bottom: SASS-P stereo PZM; PZM-6R; PZM-180. All names are trademarked by Crown International, which provided these shots.

6 On drum baffles. Mark Wright, who plays in a Top-40 band, used two PZMs inside a large plexiglass barrier surrounding his drums. He placed the mics low and in the corners. He also used one conventional cardioid mic centered overhead and one on top of the snare drum. With this setup, "I got incredible low end from the toms and bass drums, great highs from the cymbals and Rototoms, and used only four channels on the main board. The barrier also cut down my stage volume."

Alfred Grunwell of Calf Audio devised this unique method of drum miking: "We have a flying V plexiglass unit that we use for stereo overheads with PZMs. We usually use a PZM in the kick and on the high-hat. Our newest breakthrough is to tape the PZMs onto the top of the kick, directly under the mounted toms. And voila — more great sound, bright but deep, lots of attack and plenty of air."

7 In a plastic dish. Producer Gary Reber recorded Buddy Rich's drum set overhead with two PZMs, each mounted inside a plastic dish (actually a dome window from a van). A baffled PZM picked up the bass drum. PZM pioneer Ken Wahrenbrock invented these boundaries, plus a slew of other intriguing baffle shapes.

8 On a percussionist's chest. Trying to cover a large array of congas, bells, gongs, and wood blocks? Strap a PZM to the chest of the percussionist. The PZM follows the musician as he or she moves from one instrument to another, getting every sound. Since only one mic is used, there is little ambient pickup.

9 In a stereo folded baffle. Gary Pillon, a soundmixer for General Television Network, devised a PZM wedge that he used to record the Fort Street Chorale and Orchestra — a recording that won him an Emmy. He also invented a PZM stereo shotgun. Each PZM is mounted in the apex of a plexiglass pyramid and spaced apart like two ears. Using the shotgun, Pillon won Emmys for his soundtrack recordings of the Michigan Muzzleloaders Festival and the Stearman Fly-In air show. He often mounts the stereo shotgun on a Steadicam platform.

10 In a stereo microphone. The Crown SASS-P is a stereo PZM microphone. Engineer/Producer Tom Edmonds used it to record Zoro, the exciting drummer with the Lenny Kravitz band. Edmonds placed the mic under the cymbals, between the rack tom and floor tom, aiming at the snare about three feet away. Then he added two more SASS mics back in the room for ambience. Within five minutes, he came up with a sound that, he said, "would have taken hours in the studio to create."

PZMs have non-musical applications too. In the movie *Days of Thunder*, a SASS-P inside the car picked up ambience in stereo. One PZM was mounted inside the engine compartment, and another outside the car to pick up the slipstream and passing cars. To create watery effects for the submarine movie *Hunt for Red October*, sound-effects wizard Frank Serafine put a PZM in an oil-filled film can and sunk it in his swimming pool. PZMs have even been laid on a cat to pick up purring.

> For a trashy snare sample, nothing beats a PZM in the bottom of a waste container.

PZMs are great fun to experiment with because conventional mic techniques go out the window. In fact, I think I'll go tape some to my house to sample a thunderclap.... EQ

Bruce Bartlett *is the author of Recording Demo Tapes at Home, published by Howard W. Sams & Company.*

SPEAKER CABINETS

Great sounding, easily transportable, 13 ply/inch Baltic Birch Cabinets.

·JBL Drivers
·Design Direct Horn
·16 Guage Perf. Steel Grill
·Truck Packable Size
·Financing Available

FEATURING JBL SPEAKERS

EXPERIENCE THE ECONOMIC ADVANTAGE

ECHO AUDIO

908 968 2871 FAX 908 424 0507
328 THIRD STREET
DUNELLEN, NJ 08812

CIRCLE 18 ON FREE INFO CARD

"I protect my $150,000 Synclavier with a Furman Line Voltage Regulator"

Starship veteran Mark Morgan doesn't gamble with his investments. That's why he depends on the Furman AR-117 Line Voltage Regulator to protect his precious electronics and recording equipment. An easy choice, since no other power conditioner offers more features for the cost, or provides better protection for sensitive audio equipment than the AR-117.

In a single rack space, the AR-117 automatically converts a wide range of high or low AC voltages to the North American standard 117 volts. It handles a hefty 15 amps, yet it radiates only a minimal stray magnetic field. The AR-117 is frequency-independent, so it can be used successfully with generators. It filters out spikes, surges, and RF interference — and can even protect you from a disastrous accidental connection to 220V.

Find out why professionals like Mark Morgan depend on the Furman AR-117. See your local dealer or call or write for the complete details.

FURMAN Power Conditioning

Furman Sound, Inc., 30 Rich St., Greenbrae, CA 94904 (415)927-1225 or Fax: (415)927-4548

CIRCLE 20 ON FREE INFO CARD

Build Your Own D.I.

This do-it-yourself direct interface can revolutionize your project studio.

BY CRAIG ANDERTON

Photo by Ed Aiona

IF YOU RECORD guitarists in your studio, or play guitar yourself, here's the direct box for you. Optimized for guitar, bass and Stick, the do-it-yourself (DIY) DI:

✓ Matches these instruments (as well as guitar-level signal processors) to pro-style balanced gear...

✓ Buffers them from the loading effects of long cable runs or signal processors/amps with low impedance inputs, thus improving clarity and reducing muddiness...

✓ Mixes stereo instruments to mono...

✓ Delivers 6 - 24 dB of gain...
...and it's not hard to build, either.

Input J1 is almost always the best input to use, but if you experience problems with some active electronics or signal processor outputs, try J3. J6 is for stereo instruments (e.g., Chapman Stick); it mixes the two signals into a mono output. J2 carries the same signal present at J1, J3 or J6. Use this to split the instrument signal to a tuner, amplifier, etc.

J4 provides an unbalanced 1/4" phone output, J5 a balanced 1/4" phone output, and J7 a balanced XLR connector output.

There are two controls. R10 (Gain) sets gain from X1 to X20. S1 (Polarity) chooses whether XLR pin 2 and the 1/4" phone tip, or XLR pin 3 and the 1/4" phone ring, carry the "hot" signals.

HOW IT WORKS

IC1A provides the amplified, non-inverting ("hot") signal and also feeds unity gain inverting amp IC1B, which generates the inverting ("cold") signal. S1 routes the two op amp outputs to the output connectors differently, depending on which polarity you select.

Regarding the inputs, J3 provides traditional capacitive coupling and a discharge path to ground (R11). R11's high resistance avoids loading down an instrument's sensitive pickups, but unfortunately, higher resistance leads to more noise.

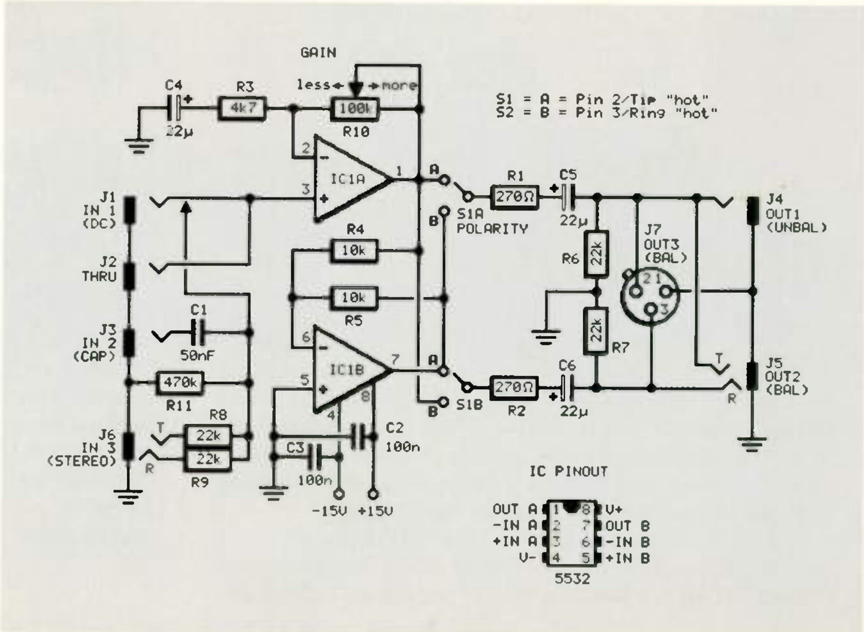
Plugging a guitar into J1 improves the noise performance, especially if the instrument's volume control is all the way up. In this case, the op amp "sees" a low resistance path to ground through the pickup wiring, which reduces noise; meanwhile, the pickup "sees" the op amp's high input impedance. The result: very little loading and very low noise. The NE5532 (IC1) is a dual op amp that features high slew rate, quiet operation, and the ability to drive 600Ω lines.

Note that the schematic parts values are in international notation but the parts list gives values in both American and international notation. The main conversion to remember is that 1 nanofarad (nF) = 1000 pF = 0.001 µF.

FINDING PARTS

Except for the NE5532, all parts are relatively common and available at Radio Shack. However, I'm also happy to report that Paia Electronics, who have supported several of my projects in the past, is providing a complete parts kit for the DIY DI (see the parts list), as well as a repair service in case of trouble.

The power supply can be any well-filtered, bipolar, DC power supply between ±9 and ±18 volts (more volts gives more headroom). You can



even use two pairs of 9V batteries (wired in series) if hum is a problem, or to help avoid shock hazard. However, batteries are the least ecological power option.

Route the input and output wires away from each other. The leads going to J1-J3 and J6, as well as connections to IC pins 2 and 6, should be as short as possible. I'd recommend mounting the DIY DI in a rack mount enclosure, but if you use a plastic case, R10's case must connect to ground.

Speaking of ground, the connections from IC1B pin 5, the junctions of R6/R7 and C2/C3 and pin 1 of XLR connector J7 should connect directly to J1's ground connection. J7's shield pin (pin 4, not shown) can connect to any chassis ground point.

If you experience RFI, connect a low value (10 pF or so) capacitor from IC1A pin 3 to ground. Also, note that pin 3 has no protection other than what's inside the chip. If you're the cautious type, insert a 1 K resistor between IC1A pin 3 and the line going to the various input jacks.

To double the available gain (but also the potential for more noise and clipping distortion), change R3 to 2k2 (2.2k).

USING IT

The DIY DI is well-suited to driving any kind of balanced input (e.g., pro signal processor or console) with an unbalanced output, particularly if it's low level. Choose the right kind of input and output connector, adjust the polarity switch as needed, and set the gain for the desired kick.

Important: Don't overlook using the unbalanced output (J4) to drive studio-oriented signal processors that, even though they have a 1/4" phone jack input, load down pickups due to a low input impedance (even some guitar boxes do this). Using the DIY DI to provide proper matching between the guitar and processor can greatly increase the guitar's "sparkle."

And that's about all there is to it. Happy recording, and may you never hear a muddy guitar, buzzing bass or sick Stick sound again. EQ

Craig Anderton *is the author of several books on musical electronics, including Electronic Projects for Musicians.*

Parts List

Radio Shack part numbers are indicated with #.

CAPACITORS

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

C1	50n or 0.05µF (#272-134)
C2, C3	100n or 0.1µF (#272-135)
C4-C6	22µ tantalum or electrolytic (#272-1026)

RESISTORS

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

R1, R2	270Ω (#271-1314)
R3	4k7 or 4.7k (#271-1330)
R4, R5	10k (#271-1335)
R6-R9	22k (#271-1339)
R10	100k linear taper pot (#271-092)
R11	470k (#271-1354)

JACKS

J1	¼" mono phone, switching jack (#274-255)
J2-J4	1/4" mono phone jack (#274-252)
J5-J6	1/4" stereo phone jack (#274-312)
J7	Female XLR connector (#274-013)

OTHER PARTS

IC1	NE5532 or equivalent (see text)
S1	2-pole, 2-position toggle switch (#275-614)
Misc.	IC socket, perfboard, socket, knob, power supply, etc.

SPECIFICATIONS

Frequency response: ±0.2 dB, 20 Hz-80 kHz

Signal-to-noise ratio (unweighted, 6 dB gain, output 1): -102 dB

Signal-to-noise ratio (unweighted, 6 dB gain, output 2): -96 dB

Signal-to-noise ratio (unweighted, 24 dB gain): better than -88 dB

Input impedance, J3: >400kΩ

Input impedance, J1: >1 MegΩ

Output impedance: 600Ω

Max headroom (±15V power supply): >26V peak to peak

Gain range: 6 dB to 24 dB

A parts kit including circuit board, all components and connectors, #9110/K, $39.95. Punched, anodized and lengended rack panel, #9110/FP, $15.95. Include $2.50 postage and handling for each item ordered. Power supplies and full enclosures are also available. Write Paia Electronics, 3200 Teakwood Lane, Edmond, OK 73013. Tel. (405) 340-6300.

LEARN THE ART OF RECORDING

THE RECORDING WORKSHOP IS THE WORLD'S LEADING SCHOOL FOR "HANDS-ON" TRAINING IN MUSIC RECORDING TECHNOLOGY. IN OUR INTENSIVE, 5-WEEK PROGRAM, YOU'LL LEARN THE CREATIVE OPERATION OF PROFESSIONAL RECORDING EQUIPMENT. WE'LL TEACH YOU THE JOB SKILLS NEEDED TO SUCCESSFULLY START YOUR CAREER AS A RECORDING ENGINEER, PRODUCER OR STUDIO MUSICIAN.

- 6 STUDIOS FILLED WITH THE LATEST EQUIPMENT: DIGITAL RECORDING TO HARD DISK, DAT MASTERING, AUTOMATED MIXING, MIDI AND MUCH MORE
- SMALL CLASSES, PERSONAL ATTENTION
- NO PREVIOUS EXPERIENCE REQUIRED
- INTENSIVE, 5-WEEK PROGRAM GIVES YOU MORE QUALITY, IN-STUDIO EXPERIENCE THAN MOST LONG-TERM SCHOOLS
- JOB/INTERNSHIP PLACEMENT ASSISTANCE
- FINANCIAL AID AVAILABLE
- LOW-COST, ON CAMPUS HOUSING

FOR FREE BROCHURE, PLEASE CALL OR WRITE TODAY

THE RECORDING WORKSHOP

1-800-848-9900 1-614-663-2510

445-J MASSIEVILLE ROAD, CHILLICOTHE, OHIO 45601

OHIO STATE BOARD OF PROPRIETARY SCHOOL REGISTRATION #80-07-0696T

THE RECORDING WORKSHOP ®

©1989 THE RECORDING WORKSHOP

CIRCLE 47 ON FREE INFO CARD

It's Time You Experienced PMX Syndrome.

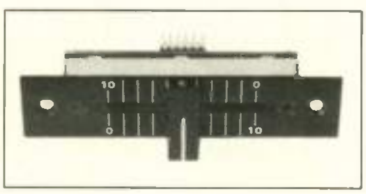
PMX-10
(10" rack mixer)

Introducing Gemini's new PMX series mixers—designed to get you hot, priced not to get you bothered.

PMX-70
(15" rack mixer)

PMX-350
(19" rack mixer)

Removable fader control can be replaced in minutes



PMX syndrome—it starts the minute you see our new line of mixers. A warm rush comes over you as you realize these mixers have exactly the features you want, loads of inputs (phono, mic. & line), removable faders, voice changer so you can go from munchkin to monster at the push of a button, sound effects with speed control to add a hot new dimension to your gig, rack mountability and more.

Then the warm rush fades when you realize that mixers with these features are usually priced way out of your range. Fortunately, these mixers are by Gemini—a name that means you're getting a whole lot more mixer than you're paying for.

Experience PMX syndrome today, at your authorized Gemini dealer... you'll be glad you did.

Corporate Offices: 1221 38th Street, Brooklyn, NY 11218
Phone: 718-851-6000, FAX: 718-972-8178
Gemini South: 2848 J Stirling Rd., Hollywood, FL 33020
Phone: 305-920-1400, FAX: 305-920-4105

gemini

SOUND PRODUCTS WITH SOUND IDEAS

CIRCLE 21 ON FREE INFO CARD

LUDWIG

continued from page 31

range advantages of digital. You should be peaking from -2 dB to 0. If you do encounter an over, play it back to see if the distortion is audible. If your style is to "peg the VU meters a lot," this "-18" level may in fact be OK. For most people, the "book" setup will create an under-modulated recording.

To remedy this, feed the 1 kHz tone from the console again and carefully raise the RECORD LEVEL until your test recording is hot enough. Again, the output level will now be too loud compared to the input level from the console. As Panasonic left out the output level controls, either live with the loud output or pad-down the output level yourself after you have established your reference (alternatively, switch the output to the "-10 dBu" position). If you have a relationship with a mastering facility, I'm sure they will be happy to make you a -14 dB calibration tone that is digitally generated to facilitate the setup procedure.

It has enough gain to drive most headphones into the deafening range cleanly (be careful!)

USER INTERFACE

The machine cannot be instantly slammed into record like an analog machine. To record on the machine, either select an analog or digital input, then press RECORD. The tape will advance four seconds past the zero time mark. If the input is digital, the digital word clock will lock on (if everything is hooked up correctly), then the machine backs up to the two second mark and PAUSES itself in RECORD, waiting for the user to push PLAY to commence the actual recording. By pressing END SEARCH, the last absolute time recorded can be found automatically.

In the RECORD-PAUSE mode, the input levels appear on the digital meter with accurate ballistics. Strangely, when one inputs a digital signal 14 dB below the zero dB mark, it appears on this meter at only minus 12, but unlike some famous professional DAT machines, when this machine says you are OVER level, believe it!

The meter also shows which index number (like CD "track" number) one is on. The machine can automatically insert an index number at the start of each tune if there is a small amount of silence before the beginning of the music. This feature has to be selected with the PNO/START ID AUTO switch. I always use this mode, and it is annoying to me that it doesn't default to this mode. If you're doing a live album with no silence between songs, or if the songs are sequenced with almost no pause, the auto-ID feature won't work. For those index numbers that it misses, the 3700 has a lot of facility for manually inserting them. Unlike some decks that allow one to audition the start index point before printing it, the 3700 demands that you stare at the absolute time numbers to see when the song begins, and then back the tape up with the nifty large knob (that lets you select 1,2,3 or 4 times normal speed). Take the unit out of AUTO mode, turn on the START ID button and then when the correct absolute time number comes, press the ID WRITE button. To erase a wrong start index, do the same, but press the ID ERASE after the "wrong" START ID light comes on.

The headphone output sounds good, much better than the poor circuits used on some portable machines. It has enough gain to drive most headphones into the deafening range cleanly (be careful!)

A cursory look at some of the many other features: The SV-3700 records at either 48 kHz or 44.1, but it will play back tapes made at the four hour 32 kHz rate. There is a great remote control with lots of programming features. A CLEANING indicator lights when the heads need cleaning and the 5000 Hour Meter shows record and playback head use. The analog input has a 2.5 second auto fade-up and a 5 second auto fade-out feature.

The operating instructions manual is clear and well written, and reveals even more little features that help make this my personal DAT machine of choice at the moment. EQ

You'll find digital guru **Bob Ludwig** *at Masterdisk in NYC.*

HEAD RECONDITIONING

Restore your worn heads to original (new) performance specifications at a fraction of the replacement cost. Our laboratory services include:

- Digital/Optical & Electrical inspection
- Precision recontouring of heads
- Complete digital/optical alignment of assembly
- Exclusive "Audio Magnetic Head Test Report™" & Data Sheets

We also carry a full line of replacement heads and parts. Our 25 years of experience and reputation are unmatched in the industry.

JRF MAGNETIC SCIENCES
249 Kennedy Road
P.O. Box 121 • Greendell, NJ 07839
Tel (201) 579-5773
Fax (201) 579-6021

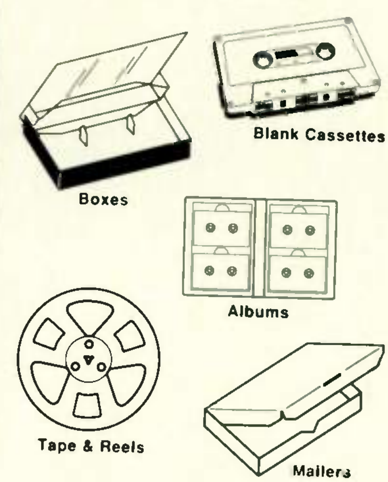
CIRCLE 28 ON FREE INFO CARD

Q101-6

AUDIO PRODUCTION & PACKAGING SUPPLIES

Immediate Shipment · Competitive Pricing

Ask for our Free Catalog



PolyQuick
1243 Rand Road Des Plaines IL 60016
Phone: (708) 390-7744 Fax: (708) 390-9886

CIRCLE 34 ON FREE INFO CARD

SAMPLE SMARTS

continued from page 45

sucked out through your nose due to the implosion at the end of the sample. An otherwise dog-meat sample could be OK if it decayed naturally. Fade it gradually, starting about halfway along the sample length.

One more trick: When sampling, record additional samples with the mics far away in the room. Put these mics on separate tracks, or on a second DAT machine. Play them from a second sampler or two other outputs from the same sampler. Mix them in with the close-miked sounds, and change the perspective by making the ambient sounds happen earlier or later.

So, if you are going to steal, at least do a good job of it. If you are going to rob my house, please don't track mud all over my new carpets.

Now that we are at the end of my ranting I have to admit that I haven't written a word of this. I scanned it all in from other articles. I am going right now to turn myself in to the Sample Police (headed up by Joe Sample, no doubt). Hopefully I'll be out of the slammer in time for my next piece. EQ

CIRCLE 49 ON FREE INFO CARD

BUYING TIPS

continued from page 45

■ **SCSI port (or SCSI option).** SCSI (Small Computer System Interface) lets the sampler access PC-type peripherals such as hard disks and CD-ROMs. These hold more data than floppy disks, and transfer it to the sampler faster. A CD-ROM is a read-only device; you cannot save modified samples to it, as you can with a hard disk. However, the CD-ROM can hold the equivalent of hundreds of floppies. Some visual editing programs (e.g. Alchemy, Avalon) support SCSI as well as MIDI because SCSI allows a transfer in seconds instead of minutes. Consumer Alert: just because a sampler has a SCSI port does not mean that it will work with all SCSI devices, or software that supports SCSI.

■ **Specs.** The 16 bit/ 44.1 kHz sampling is the de facto standard for decent sound quality. (Nonetheless, some second-hand 12-bit units still sound very good and sometimes come cheap, complete with disk library. Another important spec is the number of voices. The more voices, the greater the sampler's usefulness in multi-timbral applications.

■ **"Synthesizer" options.** A sample can only be changed via synthesizer-type signal processing options such as filters, amplitude envelopes, LFOs, keyboard pressure response (monophonic or, even better, polyphonic), and so on. Look for flexibility and lots of modulation options.

■ **The operating system.** The easier it is to use, the better. Some important convenience features include a choice of looping options (standard, bidirectional, crossfade), multiple outputs, operating system "macros" to cut down on button-pressing, an informative display, the ability to load several sounds into memory at once (and preferably still be able to keep playing), and response to external MIDI controllers.

Samplers often include features such as on-board sequencers and signal processing. If you have a computer-based sequencer and a bunch of signal processors, such features may seem redundant. Then again, sometimes you'll need to expand your number of MIDI channels by syncing sequencers together, and having another reverb unit kicking around never seems to hurt.

—Craig Anderton

POWER

continued from page 55

(how long it will support your equipment) is dependent on the amount of battery storage and the size of the load connected to it; the more you need, the more it's gonna cost you. Make sure that your UPS is sized to give you enough time to save your data before it runs out of power. A standard rule of thumb is to oversize the UPS by three times the load expected.

Obviously, installing power protection and conditioning to cover every possible situation is beyond all but the most extravagant budget. The first step is to determine how much protection and conditioning you really and truly need. In most situations, careful attention to grounding and the provision of lightning with some filtering will be more than enough for your home studio. If problems still persist, a power line study can be done, but in most cases the study will cost more than just installing the appropriate TVSS, isolation transformer, or even a UPS. A number of companies offer power conditioning and regulating equipment designed for the studio and for the musician, including Furman and Pacific Coast Technologies, which might handle your particular needs.

A UPS uses utility power to generate new, clean waveforms for the user.

The main consideration in power conditioning is the importance of the gear to be protected. It doesn't make sense to spend $10,000 to protect a keyboard that costs less than a grand, but it makes less sense to trust all of your gear — and your art — to the whims of the wall socket. Careful consideration of the options available and your own power conditioning needs can be a good start to making sure that only the music ends up smokin'. EQ

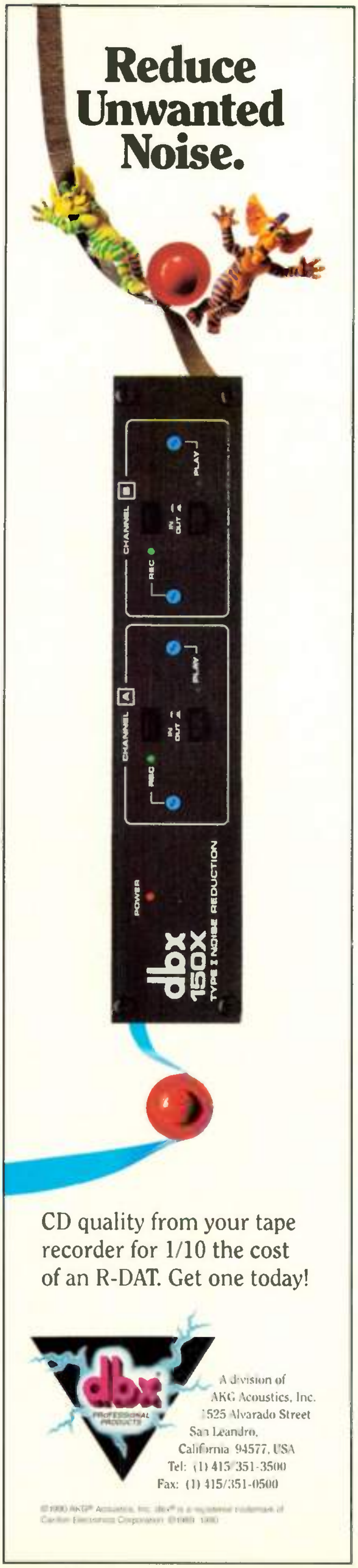
POWERFUL PRODUCTS

Several innovative power products protect your gear from nasty spikes, sags or surges. Furman Sound offers the industry's most complete line of voltage regulators and power conditioners for project studio and live sound reinforcement needs. The Furman AR-117 A.C. Line Regulator accepts input voltages from 99 to 129 volts A.C. and converts them to the North American standard 117 volts. Voltages beyond that range may also be converted to usable levels, depending on how far out of range they are. The unit is rated at 15 amperes.

The AR-117's technology differs from other voltage regulators since it uses a toroidal multiple-tap autoformer; the proper tap is switched electronically at the voltage zero-crossing to absolutely minimize line disturbances. The unit does not use a ferro-resonant transformer, which is generally heavy and bulky and radiates a large stray magnetic field.

Housed in a compact, single space rack-mount chassis, the AR-117 features eight regulated convenience outlets on the rear panel. A bargraph LED meter indicates input voltage and another LED indicates proper function. A special feature senses dangerously high overvoltages (such as inadvertent connection to 220 volts) and it shuts down the output before damage is done.

The AR-117 sells for $549. For more information contact: Furman Sound, Inc. 30 Rich Street, Greenbrae, California 94904; Tel: 415-927-1225; EQ Free Info # 106.



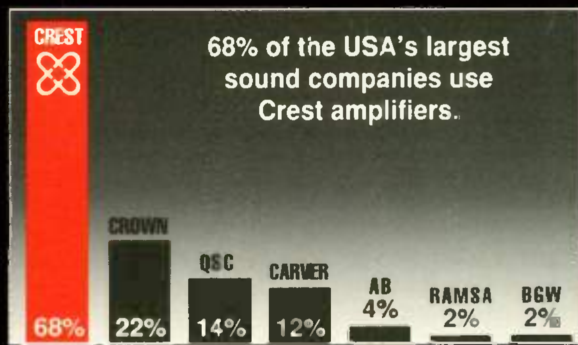
CIRCLE 13 ON FREE INFO CARD

© 1990 Crest Audio I

Power is serious business...

Just ask the owners of the fifty largest and most prominent touring sound companies in the USA. Over the last ten years, Crest Audio has established itself as the dominant amplifier manufacturer in the touring sound industry. The results of an independent survey, conducted in July 1990, attest to that. When asked, "What is the main brand of power amplifier your company uses?", these serious businessmen cited Crest Audio by an overwhelming margin.

Until recently, the quality demanded in this market was unaccessible for retail sales and most installations, because...quality has its price.



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

...never before has Crest quality been so affordable.

FA Crest Audio has taken its proven technology, road-tested reliability, & sonic excellence and incorporated them into the new FA Series. The FA Series offers the features, construction, and audio quality that have made the Professional Series industry standard. Totally modular construction, IGM™ impedance sensing protection, a Clip Limiter, variable speed DC fan and full load protection for thermal and DC voltage conditions are all standard. Complete LED status indicators, front panel detented gain pots, and a full complement of input and output connectors mean that the FA Series offers the discerning user more features and value than any of the competition.

FA SERIES Power Specifications	FA601	FA901	FA1201	FA2401
8 ohms/channel	125 watts	280 watts	300 watts	350 watts
4 ohms/channel	250	350	475	600
2 ohms/channel	N/A	440	680	770

EIA at 0.05% THD

Find out more about the FA Series. Write for a free color brochure and information on our full line of power products.

CREST AUDIO®
power is serious business.

CREST AUDIO INC. 150 FLORENCE AVENUE HAWTHORNE, NEW JERSEY 07506 USA tel: (201) 423-1300 fax: (201) 423-2977

CIRCLE 10 ON FREE INFO CARD

BRUCE SWEDIEN

continued from page 40

very constricted endpoint in mind and go through endless shenanigans to get there. It doesn't exactly inspire artistry or spontaneity. To me, it's not even very satisfying.

When I work with Michael Jackson, or any other pop act for that matter, the images I create are in my imagination, not in reality. Together, we create an all-new set of rules, a construct of axioms that are all our own and no one else's. That's beautiful, when you stop and think about it. Every album you hear — not just the ones I produce, obviously — is an entirely different reality. It wasn't built to conform to anyone's preconceived notions of good and bad. Thanks, Les and Mary.

PASSION OF THE GROOVE

Freedom, of course, has its limits. Unfortunately, the lack of structure in pop music seems to have created a least common denominator toward which too many poorly-engineered projects gravitate. A lot of what I hear nowadays in pop music sounds a lot alike; I hear far too much "knobbing around" in the control room — throwing on limiters and gated reverbs with little thought about how it effects the emotional statement.

A lack of adequate technique destroys the passion of the music. Obviously, modern studio technology — effects and the like — have opened up a tremendous new palette to musical artists. But it's important to keep them under control. My personal rule of thumb is that anything I try to do has to be involved with the passion of the groove. Any effect, any trick, has to augment the musical statement. How do you make those determinations? Gut reaction. I always trust my instincts. There's really no other way.

Another example: digital recording. On Michael's upcoming album, there are two songs I mixed to analog. And not analog 1/2-inch, mind you. Quarter-inch. Why? Because the songs sound better on 1/4-inch analog tape. Now, I'm not saying that digital is bad. What it does well, it does so dramatically well that there isn't even anything you can say. But there are times when analog just sounds better. Again, you can't let technology lead you; you have to lead technology.

If I could give any piece of advice to people getting into the business it would be this: get some education. Engineering is a lot more than deciding which "cool effect" to throw on. Having a solid background in music and in technical matters — like electrical engineering, for example — gives you an incredibly solid ground floor from which to make constructive decisions in the studio.

As in all arts, you've got to know what the limits are in order to push things to the limit. You've got to know WHY people built walls in the first place before you can tear them down. EQ

COMING NEXT ISSUE EQ

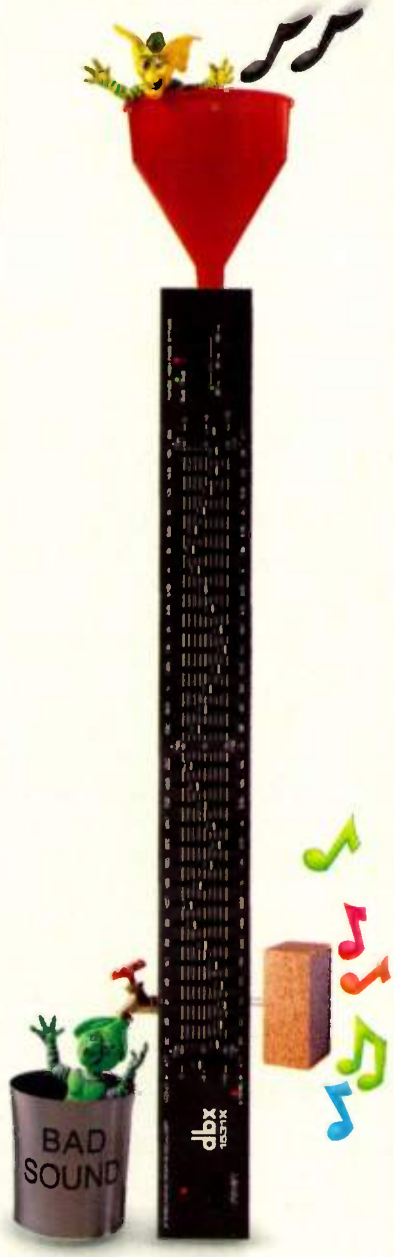
✔ **The Complete DAT Primer**
Machines, editors, tape, mastering, consumer units, copyright coding and more.

✔ **Low-Cost Digital Workstations**
Workhorse workstations that won't bust your wallet. Units from all the majors from Akai and Alpha to Studer and Turtle Beach.

✔ **Total Techniques**
A complete primer on recording keys...Using subwoofers in live sound reinforcement...Hot products reviewed by the pro's.

✔ **Workshops**
Upgrading FSK to SMPTE...Miking saxophone...Guide to gunk (chemicals for maintenance)...Woram, Nichols, Anderton and more.

Filter Out Bad Sound.

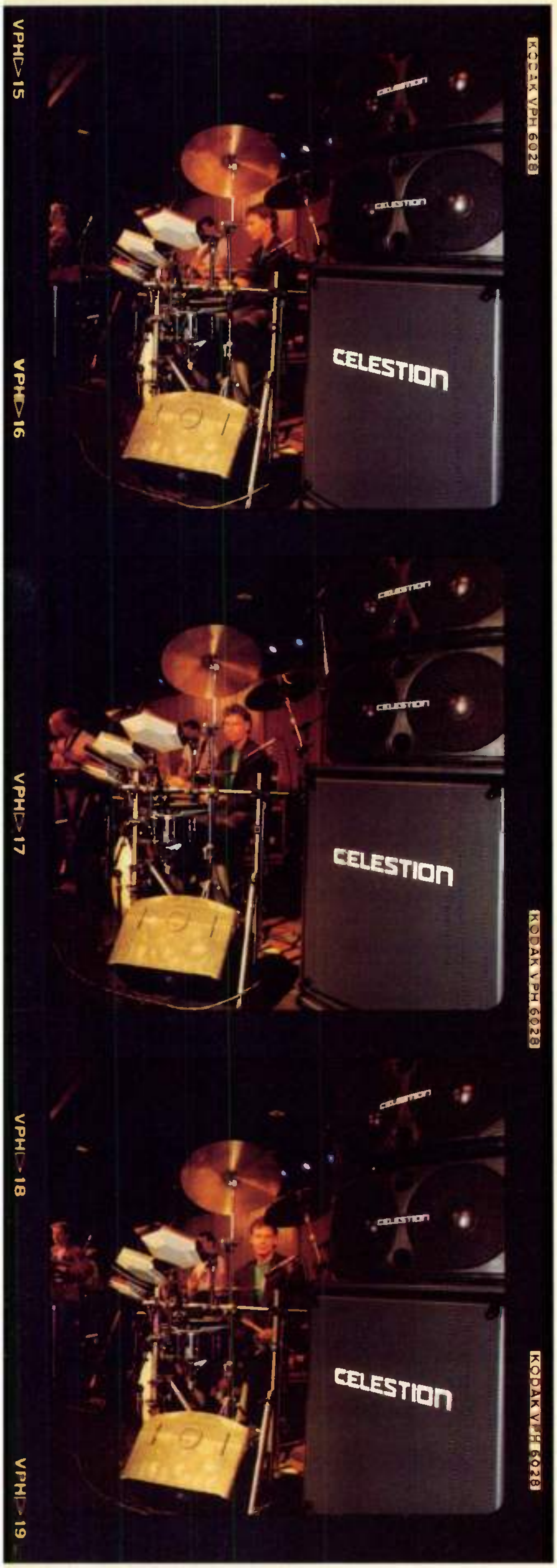


Get music to your ears with the world's only EQ which transforms from a dual 15-band to a mono 31-band at the touch of a button!

dbx PROFESSIONAL PRODUCTS

A division of
AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro,
California 94577, USA
Tel: (1) 415/351-3500
Fax: (1) 415/351-0500

CIRCLE 14 ON FREE INFO CARD



EARTH MOVING

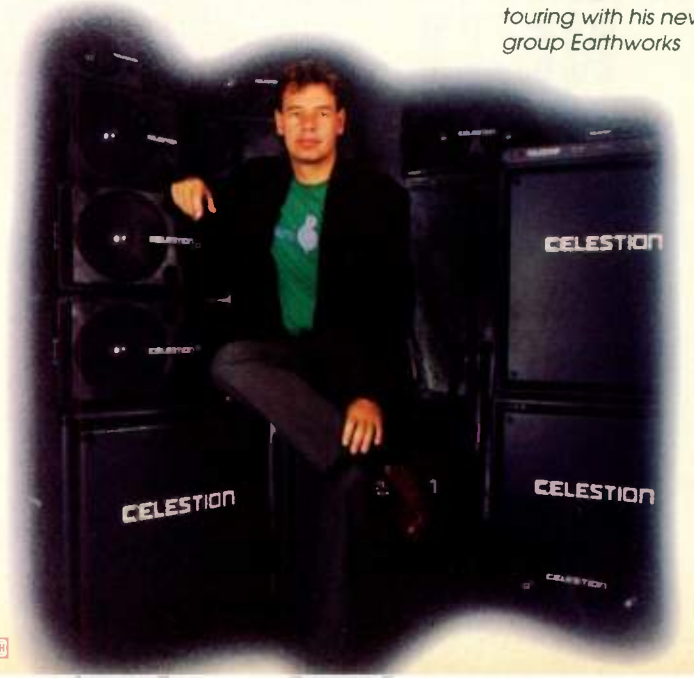
Celestion SR and Bill Bruford

Celestion's SR System is more than sound reinforcement, it's high fidelity music reinforcement. Bill Bruford is known for his melodic and complex rhythmic style. His music requires extremely accurate audio reinforcement. From Bill's recent work with Anderson, Bruford, Wakeman & Howe to his most current ensemble Earthworks, he demands the most from himself, his players and his equipment.

Celestion's new SR speaker technology offers pure and accurate audio reinforcement for both touring and permanent installation applications, and does so in a compact, easy to move, easy to hide package. The hemispheric dome radiator is what makes the SR Driver so unique. The hemispheric dome and cone are integrated as one. In essence, a Single Source Radiator. This completely eliminates the need of traditional high end devices and the harsh, unnatural sound that goes with them. Apply this truly innovative speaker technology to Celestion's SRC intelligent controllers and the SR subwoofers and you have a fully protected sound reinforcement system capable of producing extraordinary sound pressure levels with a frequency range of 40 Hz to 20 kHz.

Audition the full range of Celestion's SR System components and accessories. We guarantee it will be an earth moving experience.

Bill Bruford is currently touring with his new group Earthworks



CELESTION

89 DOUG BROWN WAY • HOLLISTON, MA 01746
(508) 429-6706 • FAX (508) 429-2426

CIRCLE 38 ON FREE INFO CARD

Advertiser Index

PAGE	BRAND	INFO	PHONE
35	A.I.D	40	(213) 469-4773
9	A.R.T.	03	(716) 436-2720
Cover 2	AKG	05	(415) 351-3500
11	Acoustic Science Corp.	04	(503) 343-9727
3	Ampex	02	(415) 367-3889
11	Ashly	06	(716) 544-5191
61	Audio Institute of America	07	(415) 931-4160
41	Audio-Technica	35	(216) 686-2600
55	Audix	09	(415) 463-1112
86	C-T Audio Audio Systems	08	(407) 738-0622
7	C.A.D.	09	(216) 593-1111
76	Celestion	38	(508) 429-6706
61	Conservatory of Recording Art	11	(602) 493-9898
74	Crest	10	(201) 423-1300
54	Crown	12	(219) 294-8000
73	dbx R-DAT	13	(415) 351-3500
75	dbx equalizer	14	(415) 351-3500
77	dbx 263X	15	(415) 351-3500
21	Disc Makers	16	(215) 232-4140
49	E-Mu	17	(408) 438-1921
67	Echo Audio	18	(908) 968-2871
59	Eltekon	41	(313) 462-3155
87	Ensoniq	19	(215) 647-3930
67	Furman	20	(415) 927-1225
70	Gemini	21	(718) 851-6000
81	A R	22	(212) 677-7580
Cover 4	JBL	26	(818) 893-4351
79	JLCooper	25	(213) 306-4131
71	JRF Magnetic	28	(201) 579-5773
65	KABA	29	(800) 231-8273
5	Lexicon	30	(617) 891-6790
89	Midiman	31	(818) 449-8838
83	Music Makers	32	(818) 407-0744
63	Nady Systems	27	(415) 652-2411
89	Paia Electronics	33	(405) 340-6300
13	Peavey	23	(601) 483-5365
71	Polyline	34	(708) 390-7738
69	Recording Workshop	47	(614) 663-2544
18	Rocktron	24	(313) 853-3055
26-27	Roland	37	(213) 685-5141
67	Roland "Studio M"	36	(213) 685-5141
61	Sam Ash Professional	39	(212) 719-2640
22	Shure Bros	42	(708) 866-2527
53	Sony	43	(201) 358-4196
39	Soundtracs	44	(516) 932-3810
30-31	Tascam M3500	45	(213) 726-0303
84-85	Tascam MSR 24R	46	(213) 726-0303
88	Turtle Beach	48	(717) 843-6916
72	Ultimate Support Systems	49	(303) 493-4488
64	Yorkville	50	(716) 297-2920

Clean Up The "S" Mess.

Sufferin' from sibilance? Give those excessive "ssss's" in your vocals a chance and they'll ruin your sound. Now you can afford the same dbx® circuitry that has sssaved the pros for years!

dbx PROFESSIONAL PRODUCTS

A division of AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro,
California 94577, USA
Tel: (1) 415/351-3500
Fax: (1) 415/351-0500

CIRCLE 15 ON FREE INFO CARD

Phase and Polarity

Reversing a pair of speaker wires doesn't always do the trick when it's time to fix your phase.

BY JOHN M. WORAM

Photo by Peter Monroe

My system didn't sound quite right, so I decided to do a little experimenting. I reversed the leads to one of the speakers and the improvement was so great that I did it to the other one too.

HO-HO-HO. Everyone(?) knows what's wrong with that little scheme: the speakers need to be "in phase" and if they're not, reversing the leads to either speaker fixes the problem; reversing the leads to both unfixes it again. That's about all you need to know about phase.

Again, ho-ho-ho. There's much more to the subject than the simple wire-flip, which would be more accurately described as a reversal of polarity, not of phase. In fact, unless you get off on listening to pure sine waves, the phase relationship between any two dissimilar signals — say, the left and right channels of that stereo system — is almost meaningless.

PHASE IN

So, what is this thing called phase? Phase: at any point in a pure sinusoidal waveform, a measure of that point's distance from the most recent positive-going zero crossing of the waveform. OK, in English: To find out, we need to take a closer look at how a simple audio frequency can be generated. Figure 1 shows the general idea: a single conductor is seen rotating in a magnetic field. As the conductor moves from point p1 to p12, its direction with respect to the magnetic (dashed) lines of force continuously alternates.

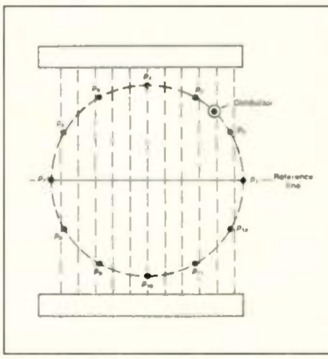


Figure 1: A single conductor rotating in a magnetic field.

At any instant, the current induced in the conductor is directly proportional to the number of lines through which the conductor is passing. At certain points (p1 and p7) it cuts through no lines of force at all, since at these instants the conductor moves parallel to the direction of the lines. At other points, (p4 and p10) it cuts through the maximum number of lines, moving first in one direction (at p4) and later moving in the opposite direction (at p10). We might call the two directions left and right, but that would make things difficult when we try to describe what happens at other points in the rotation. "Half left" or "one-quarter right" sound more like ship commands than electrical jargon. So, we take the easy way out and refer to one direction as positive and the other as negative.

Just "watching" the conductor spin round and round, we can see that the induced current varies from zero to some positive maximum, then back to zero, then to negative maximum, and then back to zero to begin the next cycle. On the printed page, we might also spot the direct relationship between the amplitude and direction of the current, and the vertical position of the conductor with respect to the horizontal reference line in the illustration.

To plot the ever-changing amplitude of the induced current, we could find a piece of graph paper and mark off a horizontal line in increments of say, 10 degrees or so. As the conductor rotates through 360 degrees, we could measure its height above or below the reference line and transfer that information to our graph. Or we could take the easy way out and reach for the calculator. At any angle, the amplitude of the current is simply the sine (abbreviated sin) of the angle. Thus, at zero degrees (p1), the amplitude is sin 0 = 0; at 30 degrees (p2), sin 30 = 0.5; at 90 degrees (p4), sin 90 = 1.0, and so on around the circle. The decimal value (0 through 1) indicates the relative current amplitude, which varies from zero (0) to maximum (1). At angles between 180 and 360 degrees, the sine comes out as a negative number, indicating a current flow in the opposite direction.

The speed at which the conductor rotates through the magnetic field is — or rather, was — referred to as cycles-per-second. However, when it was discovered that too many people understood cycles-per-second, the preferred unit of measurement was changed to hertz (abbreviated Hz).

Figure 2 shows three 500 Hz sine waves, as might be seen (on paper at least) by measuring (a) the direct output of a signal generator, and that same output delayed by (b) 0.5 ms and (c) 1 ms (ms = millisecond, or one-thousandth of one second). Note that wave b begins at the point at which wave a reaches its positive maximum; that is, wave b has a phase shift of 90 degrees with respect to wave a. Likewise, wave c is shifted 90 degrees with respect to wave b, and is therefore shifted 180 degrees with respect to wave a. Also, wave c is the mirror image of wave a; that is, its current flow is always equal in amplitude but opposite in direction to wave a.

POLARITY PROBLEMS

This direction of current flow is commonly referred to as polarity, which is either positive or negative. Since the polarities of waves a and c are always opposite, the combined current (a + c) is no current. Since c is always equal to -a, we have at all times, a - a, or if you like, -c + c. Either way, the sum is zero. That is, if the waves are combined electrically. But if one wave goes to the left speaker and the other goes to the right speaker, they are converted into separate acoustic waveforms, both of which eventually reach the listener. The result is an acoustically blurred impression of where the sound is actually coming from. Assuming this happens in error, the quick fix is reversing the leads to one set of speaker cables, which inverts the polarity of that signal. A simple solution, right?

Well, maybe not. Take a look at waveform b. It begins 90 degrees after waveform a, and 90 degrees before waveform c. Now, "correct" waveform c by flipping its polarity. What does this do for waveform b? Nothing much — it's still 90 degrees removed from waveform c, but now the shift is in the opposite direction. If the wires to one of the speakers in the listening room were indeed inverted, then all's well. But what if the wires were crossed earlier in the mix? Some signal was accidentally routed left and right, with a polarity reversal to the right track. Then what? The "quick fix" at the speaker terminals takes care of that particular signal, but at the cost of reversing a lot of other stuff that doesn't need to be reversed.

OK, so it's unlikely — maybe even impossible — for such a goof to happen during a mix. We'll assume that you (or someone) put your hardware together the right way. There is absolutely no possibility of a wiring inversion anywhere. Which is right up there with "We'll fix it in the mix" and "The check is in the mail."

Well, accidents do happen, but this time you're lucky. It turns out that the only wiring error in the entire system is indeed at the speaker terminals. At one speaker the red wire goes to the "+" terminal, at the other it goes to the "-". No problem, just flip one pair of wires. Uh, which pair? EQ

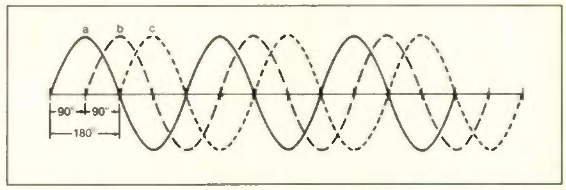
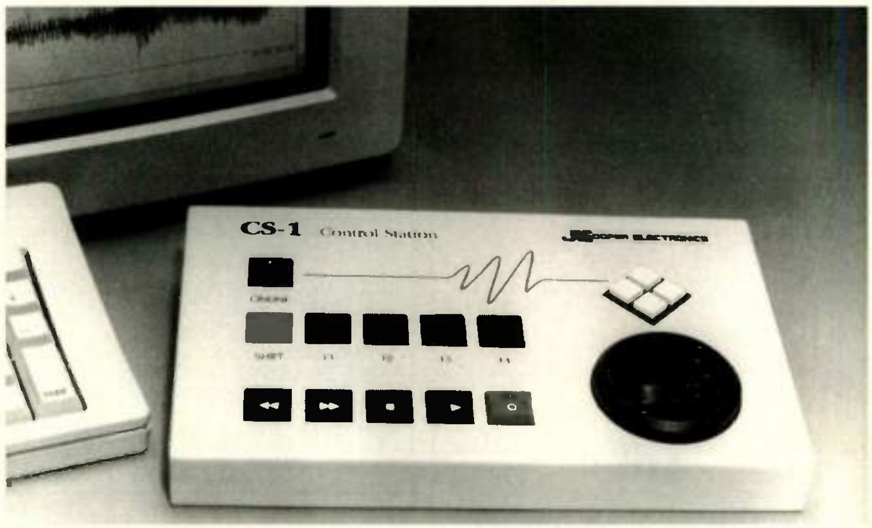


Figure 2: Three 500 Hz sine waves. Curve (b) lags (a) by 90 degrees; curve (c) lags (a) by 180 degrees.



QUIT MOUSING AROUND.

Mice are handy for word processing and general use. But for precision audio, video, and MIDI editing you need more than just a mouse.

Introducing the CS-1 Control Station™ — the easy-to-use mouse replacement for hard disk-based recording systems, MIDI sequencers, and multimedia software, from JLCooper.

The CS-1 features conventional controls that look and feel like a tape recorder transport, with large buttons for Record, Play, Stop, Fast Forward, and Rewind. A footswitch input permits hands-free operation, and the smooth, optically encoded jog/shuttle wheel provides precise positioning and effortless "scrub" editing.

Optional CS-1 software allows you to quickly create custom presets, that can be saved to disk or non-volatile memory. The CS-1 can be programmed to trigger any keyboard or mouse command. Used in conjunction with macro utilities like QuicKeys™, Tempo II™, or MacroMaker™, the CS-1 can initiate any click, drag, or key sequence. It lets you to create a personalized control station for virtually any Macintosh program.

The CS-1 (Macintosh version) comes factory configured to control Sound Tools™, Vision™, and Performer™. The CS-1m (MIDI version) works with the Atari, and the CS-1rs (RS-232 version) works with IBM compatibles.

Get the input device that lets you concentrate on your project, not your computer. The CS-1 is so fast and easy to use you'll wonder how you ever got along without it.

Contact your authorized JLCooper dealer for more information. Suggested retail price $599.95. Optional CS-1 Editor/Librarian $49.95.

JLCOOPER ELECTRONICS

13478 Beach Ave • Marina del Rey, CA 90292
Phone: (213) 306-4131 • Fax: (213) 822-2252

CS-1, JLCooper Electronics, and JLCooper logo are registered trademarks of JLCooper Electronics.
Other product names and trademarks are property of their respective companies. © 1990 JLCooper Electronics

CIRCLE 25 ON FREE INFO CARD

ATTENTION MUSIC DEALERS

SELL EQ MAGAZINE

THE MONEY-MAKING READ FOR YOUR RECORDING & SOUND CUSTOMERS

(312) 427-6652

CARL FISCHER
312 SOUTH WABASH AVENUE
CHICAGO, ILLINOIS 60604

NFI SUPERCOUSTIC ACOUSTIC FOAM

GUARANTEED LOWEST PRICES IN THE COUNTRY.
WE'LL BEAT ANY DEAL.

48" x 48"

IMMEDIATE SHIPPING

2"-$15.50
3"-$23.50
4"-$27.50

Medium Blue or Custom Charcoal Colors.

Stop Sound Now! With the Highest Density Polyurethane Foam available anywhere. Excellent sound proofing for all applications. Custom colors and sizes, also available. *Warning* - Don't buy low density bed size mattress foam that has no sound absorbing properties or specs. Buy from the only company that specializes in foam and Guarantees satisfaction! Call now for FREE Catalog, Specs. & Samples

NATIONAL FOAM, INC.

1-800-247-497
IN CA (213) 650-8906
6311 De Soto Ave., Suite A
Woodland Hills, CA 91367

MasterCard VISA Check C.O.D

YES! Small ads are read!
To make this space yours,
call Matt Charles
212-213-3444

EFXS Unlimited

Offering a complete line of Studio Effects, Guitar Processors, Pedals, Cases, Cables, and Much More !!!

Call, FAX, or Write For Free Catalog
EFXS Unlimited
1216 Pembroke Street
Uniondale, New York 11553
24 Hrs (516) 538 - EFXS MC/VISA

3/4" JVC & SONY VTR's for Audio Lockup Rebuilt w/90 Day warranty (some with address track time code)

JVC CP-5500U, CP-550U, 8200, 8280, 850U w/JOG Boxes

SONY VO-5000, VP-5600, 5800, 5850, BVU-8000

AMPEX & SONY 1" VTRs also available

Call 212-580-9551 or 212-206-1475
or Fax (212) 929-9082

CALL TEMMER VIDEO, Ask for Michael

YOUR MUSIC ON CD FOR $195

UP TO 1 HOUR OF AUDIO TRANSFERRED TO CD FROM DIGITAL OR ANALOG SOURCES

A Limited Offer, Surcharges for some tape formats

INQUIRE ABOUT LONGER PROGRAMS & MULTIPLE COPY DISCOUNTS

NY DIGITAL RECORDING
212-874-4660

SINGERS! REMOVE VOCALS!
Unlimited Backgrounds™

From Standard Records & CD's with the Thompson Vocal Eliminator™ Call for Free Catalog & Demo Record.
Phone: (404)482-4189 Ext 28
Singer's Supply, Dept EQ-1
7982 Hightower Trail
Lithonia, GA 30058
24 Hour Demo/Info Request Line (404)482-2485 Ext 28

Singer's Supply - We Have Anything & Everything For Singers

Recording Studio Management Software

- Recording Session Logs
- Mix & Track Information
- Also included:
  - Mailing Lists
  - Client Profiles
  - Schedules
  - Receipts
  - Inventory

Entire Pkg. Only $39.95

Call or Write for a FREE Brochure

SRB Software
P.O. Box 641 • Lombard, IL 60148-0641
(708) 627-4710

Low Prices You Can Afford!
— Real Time and High Speed —
CASSETTE DUPLICATION

- Complete packaging services
- *Every* Tape Checked for Quality Control
- Blank Cassettes - *Custom & Stock lengths of normal & high bias*
- Serving Duplicators, Studios, Musicians, etc . . .
- Hard & Soft shell Cassette Boxes in 5 colors
- Labels in 12 colors, variety of formats
- Complete Audio/Video Cassette Labeling Software Package only $29.95!!
- Free Catalog - 100% Guaranteed quality

EASTCO PRO
AUDIO/VIDEO CORPORATION
91 HALLER AVENUE, BUFFALO, NY 14221
1 (800) 365-TAPE (8273)

MICS, MIXERS, MONITORS, AMPS
DATs & Processors
AKG, E-V, JBL, K-T, Neumann, Tannoy, Yamaha, etc.
CALL FOR FREE CATALOG
AMERICAN AUDIO SYSTEMS
(800)222-6460 in WI, (414)354-2440

CLASSIFIEDS

CATEGORIES: Equipment for sale • Parts/Accessories • Sounds & Software • Tapes/CDs/LPs • Instruction • Employment opportunities • Business Oppertunities • Recording Services • Repair/Maintenance • Studios for Sale

AD SPECS: Col. Width 2¼ • Lines per col. inch: 7 • Characters per line:30

RATES: $80. per column inch. • Color: $20 (red, yellow, blue, or green)
- Frequency discount: Prepay 5 issues, get the 6th free.
- Blind boxes: $30. Replies are forewarded upon reciept for up to 6 months.
- All classified ads must be prepaid by check, money order, MC Visa or Amex.

NEED TO KNOW MORE? Write: EQ Classifieds • 2 Park Avenue • Suite 1820 • NY, NY 10016
Phone: 212-213-3444 • Fax: 212-213-3484 • Ask for Matt Charles

FALCO AUDIO SYSTEMS

PROFESSIONAL SPEAKER SYSTEMS

33-54 165th Street • Flushing, N.Y. 11358 • 718-939-5096

EUROPADISK, LTD.

MASTERING / MANUFACTURING

- Compact Disc
- Cassettes
- Vinyl Records
- Printed Graphics
- Digital Mastering Studios

The U.S. most complete facility with a 20-year reputation for the finest quality. Call for our information-packed brochure.

EUROPADISK, LTD.
75 Varick Street, New York, NY 10013
(212) 226-4401 FAX (212) 966-0456

CASSETTE ALBUMS FAST

no minimum

BASF Standard & Chrome Tape
48 Hr. Color Inserts •On Cassette Printing
Blank Cassettes – C-10's - C-92's
Ampex Reel-to-Reel Tape & Supplies
Call 904-399-0424 •Fax 904-398-9683

THE WAREHOUSE STUDIO SALES
2071-20ps Emerson St. · Jacksonville, FL 32207

SALE! PLUS FREE ADHESIVE!

MARKERFOAM™ ACOUSTIC FOAM

GIANT 54"x54"

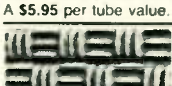
Immediate Shipping

2" Reg. $29.95 Now $19.99 • 3" Reg $39.95 Now $29.99!
KILL NOISE QUICK! High performance, full-size sheets of high density Markerfoam. EZ mount. Blue or gray. Super-effective sound absorption for studios.
WARNING—Don't buy low density bed size mattress foam that has no sound absorbing properties or specs. Markerfoam offers best value, looks professional & is proven in studios worldwide. Request Foam-Buyers Guide Catalog, specs & free samples today. VISA, MC, AMEX, COD, PURCHASE ORDERS, QTY. DISCOUNTS

MARKERTEK BLADE TILES™

HIGH PERFORMANCE LOW, LOW COST!
$3.49 per tile, 16x16x2" America's best acoustic tile value, only from Markertek! Blue or gray. Also available 16x16x3" as shown $4.49 ea.

MARKERSTIK™ foam adhesive FREE with any Foam purchase in this ad! Limited offer. A $5.95 per tube value.



SONEX
All the colors and sizes plus great prices!

Get our FREE 150-page catalog of over 5,000 exclusive and hard-to-find supplies for audio & video.

800-522-2025 *America's most unique catalog for audio & video!*

MARKERTEK™ VIDEO SUPPLY
145 Ulster Ave., Saugerties New York 12477 U.S.A.

IT'S HERE...

$14.95

THE ULTIMATE TOOLS FOR YOUR COMPUTER

A collection of the greatest Shareware, Public Domain and Demo music software available for the Macintosh.

This is not just one program. EQ's Mac Music Pak is loaded with applications (it's really 3 disks worth — we *Compact* it). Everything from a full featured Assitant Engineer demo to a whole slew of Midi-helpers. From the latest cutting edge technology to a Midi Tutorial. This is a collection that only EQ could put together. Requires a Mac with a hard drive or two floppy drives.

Remember to support Shareware - if you use a program, pay for it.

Name ______
Address ______
City ______ State ______ Zip ______

Clip & Send with check or money order for $14.95 to: *PSN, 2 Park Avenue, Suite 1820, New York, NY 10016, Att: EQ MMP*

***Please make all checks payable to* MPA.**
NY State & Nassau County residents add sales tax.
Allow 4-6 weeks for delivery.

It's not enough to study the "art" of recording. You simply must learn the *technology* of recording. That's the only way to insure your career for today *and* tomorrow!

"IAR grads are the most highly motivated and best trained of anyone entering the engineering field. I give IAR my highest endorsement."
Dan Healy, Chief Engineer
The Grateful Dead

800-544-2501
NY, NJ, CT 212-777-8550
Lic. by NYS Ed Dept / HS Diploma or GED Required
App for Veterans Training / **Financial Aid if Eligible**

Institute of Audio Research
64 University Place, Greenwich Village, New York, NewYork 10003

CIRCLE 22 ON FREE INFO CARD

IN REVIEW

Tascam M-2516/2524 Mixer

MANUFACTURER: Tascam, 7733 Telegraph Road, Montebello, CA 90640. Tel: 213-726-0303.

APPLICATION: Series of in-line, eight-bus mixing consoles suitable for studio recording or live sound applications.

SUMMARY: Extraordinarily flexible, good-sounding and affordable boards with advanced features like MIDI muting, and pluses including eight subgroup/buses, four aux sends, balanced and unbalanced line and mic inputs.

PRICE: $2,995 (16-input M-2516); $3,995 (24-input M-2524).

FREE LIT NUMBER: 121

UNTIL FAIRLY RECENTLY, you'd be hard-pressed to find a decent eight-bus mixing console for under five grand. Well, here's some good news: Tascam's newest series of in-line mixing consoles, the M2516 and M-2524 (identical except that the M-2516 — list price $2,995 — has 16 channel inputs while the M-2524 —list price $3,995 —has 24), break new ground in both price and performance. Both models offer a number of features not usually found in this price range, including the provision of eight subgroups/buses, four aux sends, balanced and unbalanced line and mic inputs, and even MIDI-controlled muting.

While there is, of course, no such thing as the ideal mixing console for all situations, a good console should be flexible enough to work well in most environments. While designed primarily for eight-track recording applications, these new offerings from Tascam would be a good choice for pretty much any project studio or even for live applications, particularly as a keyboard submixer. For example, the M-2516 used for this review was integrated into a large synthesizer-based system utilizing digital hard disk recording as opposed to analog multitrack tape. The amplification and monitoring in this system are extremely accurate, and the M-2516 temporarily replaced a console with a much higher list price, making it easy to judge its sonic performance. As a test, I recorded a MIDI sequence (driving four synths and a drum machine) to DAT, once through the M-2516 and once through the more expensive console. While I can't honestly say there was no difference in quality whatsoever (the noise floor of the M-2516 was somewhat higher, though by no means unacceptable), it would be equally unfair to state that there was any huge difference in quality between the two tests. In short, the M-2516 passed this listening test with flying colors.

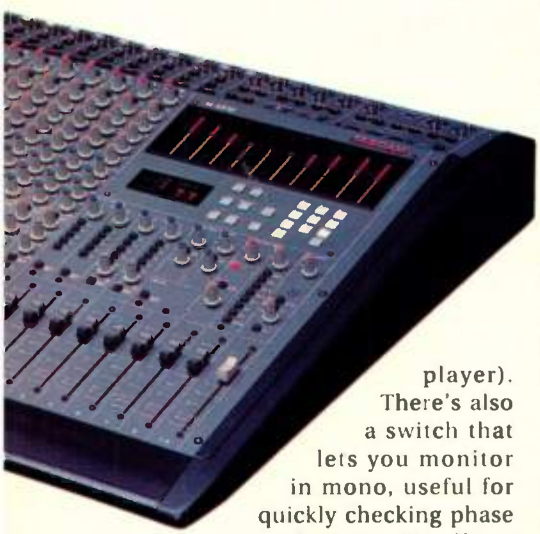
BOARD BASICS

The layout of these consoles is simple and intuitive. They feature 16 (or, in the case of M-2524, 24) identical input channels, each with a 30 dB pad, a trim pot, and a "flip" switch that brings the in-line monitoring (tape returns) down to the faders. Each channel can route signals to any or all of eight subgroups and/or to the L-R stereo bus. The last eight channels (eight through sixteen in the M-2516 and sixteen through twenty-four in the M-2524) also provide "group" switches that automatically route that group output to the monitor section. In addition, each channel offers a panpot, a mute/solo switch (when soloed, a channel is monitored post-fader and in its stereo position), a PFL (Pre Fader Listen) switch, three-band equalization (15 dB of cut or boost, with the low and mid center frequencies adjustable), and four aux sends (sends 1 and 2 can be pre- or post-faders, while sends 3 and 4 can be either post-fader or can receive signal from the monitor section). One unusual feature in the aux section is the use of two-way knobs (instead of pre/post or monitor/post switches), with the "off" position at 12 o'clock. To increase the amount of post-fader aux signal, you turn the knob clockwise, as usual. However, to increase the amount of pre-fader or monitor aux signal, you turn the knob counterclockwise. This can take a little getting used to; presumably, this odd implementation helps make the console more cost effective.

Another economic factor worth mentioning is that the M-2516/2524 console design is integrated into one front panel as opposed to using a frame with drop-in modules, as are typically found in more expensive consoles. What this means to the user is that if there is a maintenance problem, the entire console has to be disassembled. However, Tascam's internal design is modular, and therefore does allow for a single channel circuit to be removed and separately repaired.

At the right-hand side of the console is the "master" section. Here you'll find a master mute for the monitor section and faders for the eight subgroups, as well as a stereo master fader. In addition, this section contains the aux send masters and returns (each of the aux returns —two are stereo and the other two mono — can be mutes, soloed, or routed to any or all of the eight subgroups and/or to the stereo bus). There's also a control room module, where you can set the main, PFL, and headphones level. Switches in this module allow you to monitor the stereo bus, any of the aux sends, a connected two-track machine, or an external signal (such as from a connected CD or record

WRH



player). There's also a switch that lets you monitor in mono, useful for quickly checking phase correlations. Finally, a talkback module contains the internal talkback mic (there's no provision for plugging in an external one) as well as a level control and slate/talkback momentary switches. From this module you can also switch on attached studio monitors.

Above the master section and below the console meters (ten LED bar graphs showing the VU levels of the stereo group and each of the eight subgroups) lie the MIDI mute automation controls. Here's where things get really hip: The M-2516/2524 lets you create up to 99 scenes, with each scene containing a series of mutes. You can mute individual channels, as well as each aux return or even the entire monitor section. The machine stores these scenes in non-volatile memory and can instantly recall them via MIDI program change commands. In addition to simply changing from scene to scene, you can use MIDI control change or even note messages to instruct the console to carry out mutes in real time!

PLAY THAT BOARD

Yes, folks, we're actually talking about being able to "play" your console from your MIDI controller — play middle C hard (with a velocity above 64) and channel 13 is muted, play it softly (with a velocity below 64) and it is unmuted. The main purpose of this feature is not to automate your mix — that would require MIDI control of at least fader position if not also pan position, eq and aux settings, etc. — but, very simply, to be able to turn off unused channels automatically. If you've ever listened closely to the not-so-sweet sound of a digital synthesizer idling (clock noise abounds!), you'll know how valuable this feature can be, in both recording and in live performance. Without all this extraneous noise, your tapes will sound much cleaner and your PA will sound much quieter.

All input and output jacks to the M-2516/2524 are mounted on a flat horizontal behind the sloped front panel. What this kind of design loses in aesthetics (plugs and wires are clearly visible), it more than makes up in convenience, since it's eminently simple to plug and unplug cables, thereby obviating the need for a patch bay, an element virtually demanded by consoles that use a vertical rear panel for input/output. A variety of jacks are used; for example, mic inputs (which can double as balanced line inputs) use XLR; line inputs (balanced or unbalanced) use 1/4" mono; channel inserts use 1/4"stereo; tape inputs and direct outputs use RCA phono.

Yes, folks, we're actually talking about being able to "play" your console from your MIDI controller.

Overall, the M-2516/2524 stacks up well against consoles costing lots more. In fact, it's difficult to see how you could ask much more of a mixing console in this price range. Sure, there are a few minor complaints — there's no mic/line switch (in fact, both are always on and

WE HAVE THE KEY(BOARD) TO YOUR HAPPINESS!

and the audio console, and the multitrack, and the digital effects, and the...

- To get the most and the best--Don't buy from **ANYONE** until you talk to **US!**
- Small or large, we have **FINANCING/LEASING**, No one has a better program. **Call Now!**

Nominated "Most Effective Recording Products Dealer" in the 1991 Music & Sound Awards

512-684-8400

MUSICMAKERS

4100 SAN PEDRO • SAN ANTONIO, TEXAS 78212

It can be sloppy...we don't grade it! Jot down your want list and FAX it to **512-737-0691**

CIRCLE 32 ON FREE INFO CARD

summed at the fader so gain differences are controllable at the trim) and the mute switches can impart an audible click but then again, solenoid switches are designed not to click, so something unusual may have caused this. These are definitely outweighed by the positives: Extraordinary flexibility, good sound quality, and advanced features like MIDI muting. If you're in the market for either a project studio mixer or for an onstage keyboard submixer, the M-2516/2524 should fit the bill admirably.

—Howard Massey

Howard Massey's ***books include The Complete DX7 and A Synthesist's Guide To Acoustic Instruments. Currently, he heads up Workday World Productions, a music production studio, and On The Right Wavelength, a MIDI consulting company.***

E-Mu's Proteus (es)

EQ LAB REPORT

MANUFACTURER: E-Mu Systems, 1600 Green Hills Road, Scots Valley, CA 95066. (408) 438-1921.

APPLICATION: A multi-timbral ROM sample player in a lightweight single rack-space box. Plays 32 voices on 16 MIDI channels from 6 audio outputs. Used with a MIDI keyboard and sequencer, can be a standalone "studio in a box".

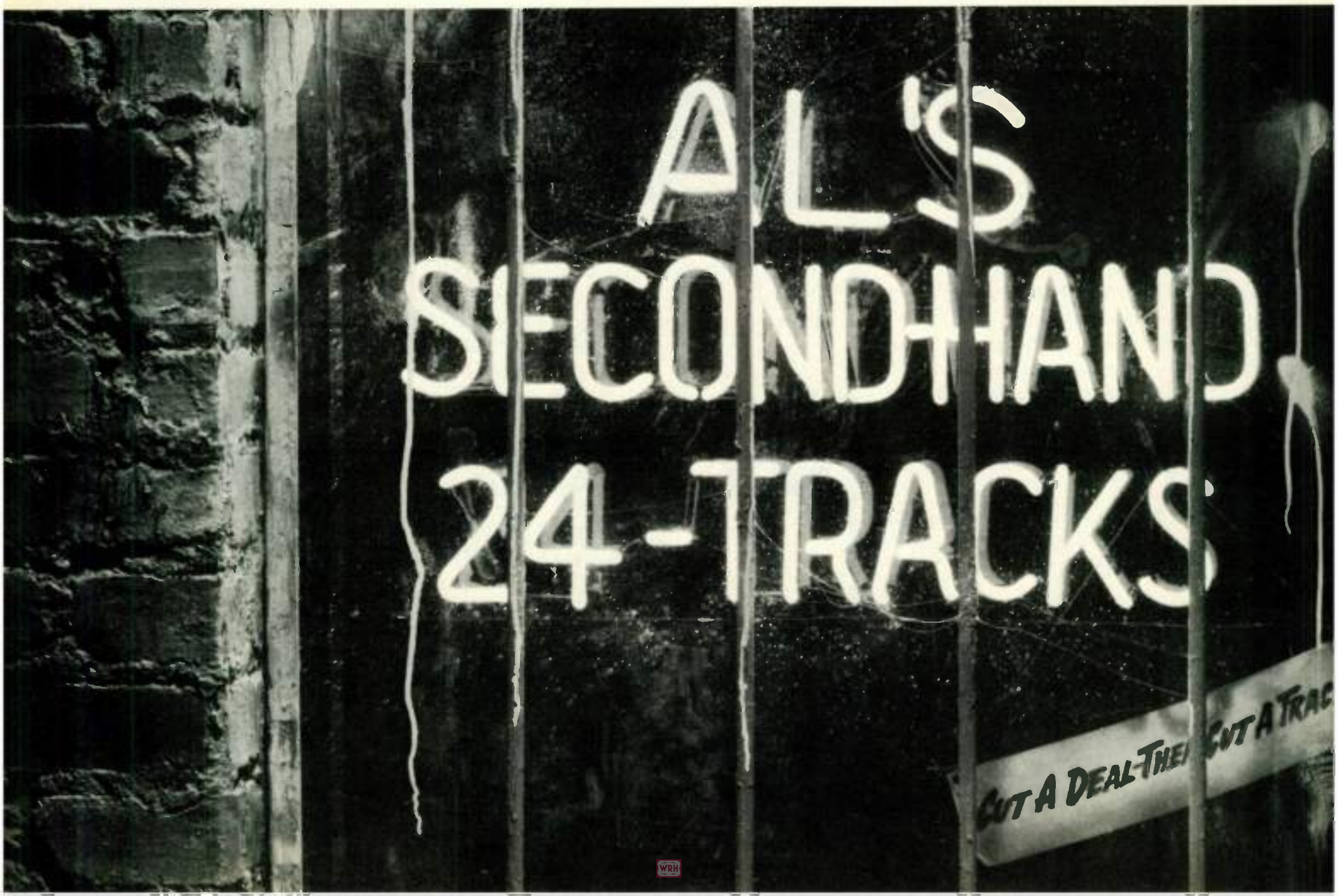
SUMMARY: Excellent-sounding, highly flexible devices with a wide range of sounds right out of the box, and good user programmability as well. Proteus/1 best for rock-oriented music; Proteus/2 lends itself more to symphonic and and chamber music.

FREE LIT NUMBER: 122

TWO YEARS AGO, E-Mu Systems, makers of state-of-the-art samplers, came out with a $1000 playback-only sampler called Proteus. It proved to be a phenomenal success, and has since spawned the Proteus/2, as well as several other variations.

The Proteuses (or is it Protei?) are MIDI-controlled ROM sample players. Like samplers, they play sampled sounds, but you can't record your own.

The Proteus fits into a single rack space and is only about eight inches deep. Depending on the model, it contains 4 or 8 megabytes of ROM. In that memory are 125 or more waveforms, known as "instruments," the majority of which are multi-samples: the same instrument sampled at differ-



ent pitches, and spread across the keyboard range. In the unpitched percussion instruments, as many as 31 different sounds are mapped across the keyboard. All the samples are taken from E-Mu's library for their flagship sampler, the EIII, and are all 16 bit.

The other ROM instruments are sine waves with various combinations of overtones, single- and multi-cycle segments of sampled sounds and other digitally-created waves. These are used alone or in conjunction with the longer samples.

SAMPLE SOUNDS

The samples are all of extremely high quality. A full-blown sound in a Proteus is called a Preset. A preset corresponds to one MIDI channel. It can contain either one or two instruments, and each instrument can have its own key range, tuning, relative volume, stereo pan position, chorusing, delay, sample starting point (for changing attack transients) and solo mode setting (for simulating single-line instruments). You can impose a different volume envelope on an instrument than the one it comes with, and you can crossfade or switch between two instruments using key ranges, velocity or a MIDI footswitch.

In addition, each preset has two programmable LFOs and an auxiliary envelope. Pitchbend, mono and poly aftertouch, four selectable MIDI controllers and the aux envelope can be configured to affect pitch, volume, crossfade position, the relative length of one or more envelope segments, and/or each other. MIDI footswitches can be assigned to hold onto different envelope segments. This gives the unit a tremendous amount of real-time expressive capability — and makes it sound like a real instrument.

Furthermore, any preset can be "linked" with up to three other presets to create more complex sounds. Each linked preset can have its own key range and can be assigned its own scale. Besides the standard equal-tempered scale, there are four alternative scales in ROM and one user-definable scale, in which every single key — not, as in some instruments,

OR, FOR JUST $12,499
YOU COULD BUY A NEW MSR-24.

Maybe at one time it was worth risking second-hand sound, iffy reliability and outmoded technology to save money on a used 24-track.

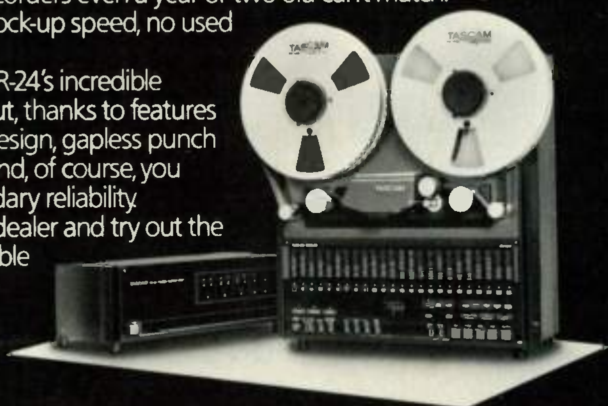
Not any more.

At $12,499,* the MSR-24 actually costs thousands less than most used machines. And its cost-effective one inch format saves you even more.

But despite its low price, the MSR-24 has micro-electronically controlled functions that recorders even a year or two old can't match. And when it comes to lock-up speed, no used machine can compete.

Best of all, the MSR-24's incredible sound will knock you out, thanks to features like our superior head design, gapless punch in/out and spot erase. And, of course, you also get Tascam's legendary reliability.

See your Tascam dealer and try out the brand-new, very affordable MSR-24. And let Al keep the hand-me-downs.



*Manufacturer's suggested retail price

TASCAM®

© 1989 TEAC America, Inc., 7733 Telegraph Road, Montebello, CA 90640, 213/726-0303.

CIRCLE 46 ON FREE INFO CARD

each step in the octave — can be individually tuned with a resolution of 1/100th of a semitone.

VOICES AND OUTPUTS

The Proteus is multitimbral; different presets can be used on each of the 16 MIDI channels. It can produce 32 voices at one time, and has dynamic voice allocation, so you don't have to worry about pre-assigning a specific number of voices to each preset or channel. However, the polyphony is reduced significantly when more complex sounds are used. In the worst case, when a preset contains two instruments and is linked to three other presets with two instruments, only four MIDI notes can be received — but chances of this happening in the real world are slim. A "MIDI Overflow" sends excess notes to the MIDI Out jack.

There are three pairs of audio outputs on the back of each Proteus: Main, Sub1, and Sub2. A preset can be assigned to a particular stereo pair, or a MIDI channel can be assigned to a pair, overriding the preset setting. Besides letting you treat the Proteus as three separate stereo (or six mono) sound sources, the Sub outputs also act as effects sends and returns.

MIDI MAPS AND DUMPS

The presets are stored in both ROM (128 slots) and user-programmable RAM. The number of RAM slots varies between models. Since there are more total presets than the 128 programs accessible by MIDI, the Proteus contains a program change map, so that presets with numbers higher than 127 can be called up with a sequencer.

Only one program map can be stored on board, but different maps can be off-loaded using MIDI system-exclusive dumps. Many sequencers can now record and play back system-exclusive information, so you can dump your program map at the beginning of a sequence, and load it back automatically later.

You can also dump the Master settings (overall tuning, MIDI channel enables and output assignments and MIDI controller assignments), the tuning table, a single preset and/or all of the RAM or ROM presets. This means that an entire Proteus configuration can be stored within each sequence and recalled in a few seconds. If your sequencer doesn't have system-exclusive capabilities, there are plenty of editing and librarian programs available.

THE MODELS

Since the introduction of the original Proteus, six more versions have appeared. Here's the rundown.

The original Proteus, now called the Proteus/1, has 69 sets of samples and 56 smaller waveforms contained in 4 megabytes of ROM. The samples are mostly "pop band" sounds, and include grand piano, string ensemble, choir, flute, sax, brass, electric and acoustic guitar, lots of electric basses, drum sets, Latin percussion and more. It has 64 user slots for new presets (using the same instruments) in RAM. The Proteus/1 XR has exactly the same sounds, but has room for 256 user presets. If you only want the factory sounds, or you're happy with the idea of frequently downloading your custom presets to a computer, you probably don't need the XR version.

The Proteus/2 is more orchestrally oriented. It is designed in many ways to complement the /1 — there isn't a lot of overlap. The /2 has 8 megabytes of ROM. Its string library includes solo, ensemble and pizzicato versions of each member of the orchestra, and its woodwinds include oboe, English horn, bassoon and variations. There are more varieties of brass (including a Harmon-muted trumpet), and the percussion leans more towards timpani, tom tom, clash (not crash) cymbals, etc. The Proteus/2 XR, again, adds more user preset slots but has the same instruments.

HEARD BY MILLIONS BUT SELDOM SEEN

C-ducer

Studio Quality Contact Condenser Microphones

- SUPERB CLARITY
- TRUE ACOUSTIC SOUND
- COMPLETE ISOLATION
- EASY TO USE, NO HOLES TO DRILL, CAN BE MOVED TO OTHER INSTRUMENTS
- MORE THAN 10,000 USED WORLDWIDE
- LOWER COST AND BETTER PERFORMANCE THAN CONVENTIONAL MICROPHONES

New CP Series just introduced.
- Super quiet
- computer matched
- miniatured circuitry
- mics acoustic instruments

NEW FROM OMNICRAFT

OMNICRAFT

GTS Stereo Noisegate

Omnicraft's unique optical system has enabled us to design a noisegate with the lowest levels of noise, coloration of any noisegate on the market!! The GTS also features:

- 24dB/Octave hi & lo-pass filters
- Balanced inputs & outputs (on XLR-type connectors)
- Duckling, Keying and 2 modes of triggering
- 2 seperate channels can be ganged for stereo
- Low Cost

...and many more features.

Call 1-800-CT-AUDIO

In Canada call 604-688-2444

CT-AUDIO SYSTEMS, INC.

3050 SW 14th Place, Suite 3, Boynton Beach, FL 33426

Tel.407-738-0622 • Fax 738-0319

CIRCLE 08 ON FREE INFO CARD

Recording great samples is only part of making great music.

Let's face it, most samplers on the market today are essentially sophisticated digital recorders—they use advanced technology to enable you to record and play back sounds. But as a musician, you probably want to do more than trigger recordings. Wouldn't you prefer to have an instrument that takes those recordings and allows you to shape them into expressive musical sounds?

The EPS-16 PLUS keyboard

That's the basic idea behind the EPS-16 PLUS Digital Sampling Workstation from ENSONIQ. Use sampling technology to make truly expressive sounds, then give you the tools to perform and compose effortlessly with them.

Available either as a keyboard or rack-mount unit, the EPS-16 PLUS gives you true 16-bit resolution with fidelity that surpasses samplers costing twice as much! A variety of sample editing options and an extensive voice architecture allow you to turn samples into sounds that combine the clarity of sampling with the imaginative possibilities of synthesis.

The EPS-16 PLUS rack-mount unit

And our onboard 24-bit dynamic effects (a first for a sampling instrument) allow you to add reverb, chorusing, delay, and even distortion effects to your music.

The extensive library (over 1000 sounds) for the EPS-16 PLUS includes innovative "Signature Series" volumes from top artists—and new collections are being released every month.

For performance, you'll find our Poly-Key™ Pressure keyboard, Patch Select buttons, and ability to load new sounds while you're playing—all unique to ENSONIQ instruments—open up all kinds of expressive possibilities. Want more? Add our optional FLASHBANK™ to keep your favorite sounds loaded in backed-up memory.*

Have an idea for a new tune? The EPS-16 PLUS will be right there for you—20 voices and an onboard 16-track sequencer with the easy-to-use recording and editing features that have won ENSONIQ rave reviews around the world. Add a SCSI interface and a hard drive, and you've got a versatile MIDI recording system that's small enough to set up at home or take on stage.

It takes more than great sampling specs to make an instrument truly musical, and the ENSONIQ EPS-16 PLUS is the first sampler to do just that. Check it out at an Authorized ENSONIQ dealer today. Call 1-800-553-5151 for the one near you.

**Actually, you can make the EPS-16 PLUS extremely friendly by getting it factory-loaded with the FLASHBANK™, SCSI, and expanded memory options—and save up to $500 in the process! Ask your dealer for details.*

***The SD-1 Music Production Synthesizer**—Advanced synthesis (including new 16-bit piano waves!), 16-bit output circuitry for unsurpassed fidelity, 24-bit dynamic effects, and a 24-track sequencer with extensive mixdown capabilities. Truly, the next generation in integrated music production synthesizers.*

***The SQ-R Synthesizer Module**—The great sound of the SQ-1 in a single-space rackmount module.*

***The SQ-1 PLUS Personal Music Studio**—Advanced synthesis (including new 16-bit piano waves!), 24-bit dynamic effects, and a 16-track sequencer with mix-down capabilities. The low-cost MIDI studio with the high-quality sound.*

***The SQ-2 Personal Music Studio**—A 76-key synthesizer with great sounds (including new 16-bit piano waves!), 24-bit dynamic effects, 16-track sequencing, and performance features. The perfect choice when 61 keys just aren't enough.*

Please send me information on:

☐ EPS-16 PLUS ☐ SD-1 ☐ SQ-2 ☐ SQ-1 PLUS/SQ-R

☐ **I would also like the FREE ENSONIQ Guide to choosing a synthesizer versus a sampler.**

Name

Address

City

State Zip Phone

Mail to ENSONIQ, Department Q-24
155 Great Valley Parkway, Malvern, PA 19355

ensoniq®

THE TECHNOLOGY THAT PERFORMS

CIRCLE 19 ON FREE INFO CARD

IN REVIEW

There are two expansion boards available for the Proteus/1 or /1 XR. The Protologic, from InVision, adds 4 megabytes of ROM samples to the unit, expanding on the pop band concept, including Leslie organ, distortion guitar, fretless bass, new drums, ethnic tuned percussion and winds, electric piano and various types of human voices. It must be installed by InVision or an authorized dealer. E-Mu's expansion board puts 4 MB of orchestral sounds into the Proteus/1 (or XR). It also has to be installed by a qualified technician, and you can't put both boards into one unit.

The MacProteus, made by Digidesign, is a Proteus/1 on a NuBus card that fits inside a Macintosh II. It has only one set of stereo outs, and its ROM and RAM cannot be expanded. It can't run without the computer, and needs a MIDI Manager-compatible sequencer to play it.

IS IT FOR YOU?

The Proteus's popularity is well deserved. As a group, they are fantastically useful instruments, with great sounds out of the box and a high degree of user programmability to boot.

They serve the high-end and low-end user alike very well. For the beginning MIDI studio, they represent high quality and versatility for relatively little money, while for the experienced composer or producer, they provide a way to free up samplers for more esoteric uses, without sacrificing the quality of the "bread and butter" sounds the samplers would otherwise be handling.

The strength of these modules is in the realism of their sounds, and although you can certainly create "synth" sounds with them, you will probably find them most immediately useful at emulating real instruments. The sounds tend to be more "in your face" than other sample players, but this means they have the power to slice right through a mix. If need be, you can usually calm them down with a little upper-midrange cut.

If your music tends towards rock, jazz, R&B, or funk, the Proteus/1 is your best bet. I didn't get a chance to hear the InVision board, but it seems it would be useful if you're looking for a greater palette of more modern, synthetic textures. On the other hand, if you write symphonically, go for the Proteus/2. The woodwind sounds of the /2 are among the best I've ever heard, and coming from a former bassoon player, that's a major compliment. If you cover a lot of different ground, a good compromise would be the /1 with the E-Mu expansion board — that's what I'm getting. Whatever you end up with, you'll have a lot for your money.

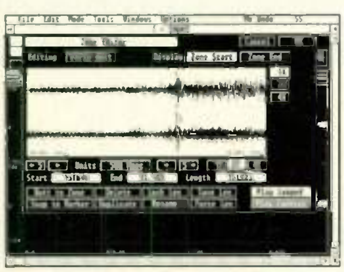
—Paul D. Lehrman

56K digital recording system

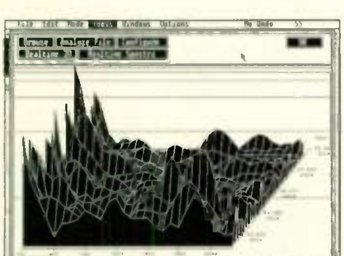
The Best Gets Better!

Turtle Beach Systems is now unveiling version 1.2 of our SoundStage software for the **56K Digital Recording System**. If you want to edit CD-quality sound on an IBM computer, there is no better system, no better price, no better choice than the 56K!

New Features



The New Zone Editor



The 3-D Frequency Analysis

- Sample Rate Conversion
- Time Compression/Expansion
- Scrub editing
- 3-D Realtime Frequency Analysis
- Reverse/Invert
- Mono recording (left, right, or sum of both channels)
- SMPTE Chase Lock
- Record undersampling support (2X, 4X)
- Numerous improvements to existing software
- Still $2689 (No improvement necessary!)
- Free demo disk!

turtle beach systems

PO Box 5074, York, PA 17405 717 843 6916 Fax 717 854 8319

CIRCLE 48 ON FREE INFO CARD

SAMPLING LAW

continued from page 47

Plaintiffs can argue use of even a small sample is a copyright infringement if the portion sampled constitutes the "essence" of the original work. But then again, a defendant can argue the sample was a "fair use" allowed by federal copyright law. The effect of the defendant's recording on sales potential of the original work is a primary fair-use issue. Courts generally give greater leeway to educational, rather than commercial, fair uses.

A plaintiff in a sampling suit might also proceed under federal and state trademark principles, unfair competition statutes and right of publicity laws, among other things. These are particularly useful to an artist who doesn't own the copyright in the sampled recording, but who can claim that he or she is known to the public by a unique style. A sampling suit (that has since been settled) filed by former Turtles lead singers and copyright owners Flo and Eddie, over De La Soul's use of a portion of the 60s hit "You Showed Me," even claimed violation of a California anti-privacy statute.

Some artists, producers and remixers don't seek permission to sample a portion of a pre-existing work unless the new recording sells well. They argue the cost of seeking a license before that would make the cost of recording too prohibitive. On the other hand, once a record becomes a hit, the copyright owner of the original work will likely ask for a higher licensing fee — money is money, after all.

License fees can range from one-time payments as low as $100 per use to several thousand dollars or even higher, depending on the length of the sample and the popularity of the sound recording that is being sampled. Sometimes the copyright owners of a pre-existing work will ask for additional payment if product reaches a specified sales plateau. If a high fee is paid, the party doing the sampling could ask for full ownership of the copyright in the new recording.

Some record companies and music publishers share from the new work. This is sometimes resisted on the ground that it further complicates the accounting process.

The owners of a pre-existing work might also ask for written credit on the new recording. EQ

MIDI TO CV

MV-8 From PAiA

PAIA's new MV-8 is the simplest way possible for the contemporary MIDI studio to access the great sound of classic analog equipment. This easy to use rack mount processor converts MIDI input to control voltages or control voltages to MIDI outputs. RCA phono jack connectors provide front panel connections to 8 control voltage inputs, 8 control voltage outputs, 8 gate inputs and 8 gate outputs.

There's a lot of flexibility here. The MV-8 has user selectable operating modes that provide complete control of a single synth, pitch control of eight synths or 8 general purpose CV ins/outs using MIDI control change data.

Available as an easy to assemble kit or fully assembled and tested.

MV-8 Kit **$299.00**
MV-8 Assembled **$399.00**

Add $5 for shipping for your order for either version.

For more information or a catalog of other PAIA products, call or write:

PAIA Electronics, Inc. **(405) 340-6300**
3200 Teakwood Ln., **Edmond, OK 73013**

CIRCLE 33 ON FREE INFO CARD

Industry Standards

Syncman Pro - State of the Art. Supports every MIDI synchronization format plus our own "Spot Lock" Video Sync. Other features include JAM Sync, SMPTE Regenerator, SMPTE/MTC/DTL Display, 786 Foley/MIDI Hit Recorder/Editor, Auto Punch In/Out. Create Tempo Maps automatically, manually or by "Tap Tempo" entry. Over 30 other features. No other sync box even comes close.

Syncman - Worldwide Best Seller. Sync *any* sequencer to tape. Reads and writes *all* SMPTE formats including MTC, Smart FSK and DTL. Supports auto downloading of tempo maps, and up to 384 tempo changes. Includes JAM Sync, MIDI merge and exclusive sync stripe duplication feature.

ALL NEW LOOK!

SmartSync - Unsurpassed performance, quality and reliability in the lowest priced MIDI sync box available today. Supports Smart FSK Sync, JAM Sync, unlimited tempo changes, MIDI merge.

midiman™
Guaranteed for life.

30 North Raymond Avenue, Suite 505, Pasadena, California 91103 Tel.: (818) 449-8838 Fax.: (818) 449-9480 Toll Free: 1-800-969-6434

CIRCLE 31 ON FREE INFO CARD

The Flip Side of The Coin

The ultimate, high tech guitar processor comes to life

BY ROGER NICHOLS

Photo by Ed Colver

BESIDES THE Donald Fagen project, I have been working with Walter Becker on a few other albums during the last year. In the same amount of time that it took to record drums and piano on two tunes for Donald, we recorded ten hours of music on 11 albums — and half of them are already in the stores. The two experiences are really the flip sides of a coin.

The musician list on the Walter Becker project has been a veritable who's who: John Patatuci, Peter Erskin, John Beasley, Bob Sheppard, Dean Parks, Dave Weckle, Jackie McLean, and many, many more. What a pleasure it has been.

These were jazz albums for Windham Hill and Triloka records. It is a lot of fun to go into the studio, record a whole album in two days, mix on the third day and then call it done. Most of the albums contained about 60 minutes worth of music. Mixing was at the rate of ten tunes per day as opposed to ten days per tune.

IT'S ALIVE

The best part of all was the fact that there were warm bodies in the studio. Remember them? Live musicians? People with names and families? If you wanted one of the instruments to play a different part, all you had to do was push the talk-back button and ask. What a concept!

So many thoughts were going through my head during the tracking session: Wow, listen to how the sound of all of these instruments blends together so nicely. Every instrument is playing a part that fits right in with all of the others without any conflict, all automatically. What a great source of samples I've got (sorry, just kidding).

I realized that there is additional quality that you should be aware of when using samplers to replace musicians. The first two aspects of this are the tuning of the instrument and the "feel" of the sound. The third, and most overlooked, is intent. Yup. There is a subtlety in the sound of an instrument that is being played by a real musician that is determined by his technique in playing the instrument as well as his approach to the material being played. These parameters are not readily available in samplers that are being played by sequencers. Sure, you can change the volume of a note or a chord, but a real instrument played delicately sounds completely different than the same instrument played ferociously and made softer with a gain control. It is these inflections (which I think are curable with a dose of penicillin) that make it hard to duplicate the performance of a live musician by automated means.

How about spontaneity? There's another quality I have yet to see automated. Intentionally, I mean. I have had runaway sequencers come up with some pretty esoteric constructions, but I think that the legal ramifications of someone breaking a leg while trying to dance to it more than offsets any potential commercial value.

"HAL" FOR THE STUDIO

What if we designed our own machine to play more like real musicians? The perfect machine for this task would be one with artificial intelligence. It would have to pack a separate processor for each instrument and process all of the necessary information needed to play each instrument so that it would blend well with all of the other pieces in the band.

Let's take a guitar part, for instance. The musical arrangement should be fed into the machine easily — ideally via a musical chart that you could scan in directly. That would be better than having to type it in with a keyboard or play it in through a MIDI keyboard. The processor would look through a library of all of the possible rhythms and usable notes that might fit this particular piece of music. The guitar library must also contain an almost endless array of samples that offer all the permutations of note value, volume, attack, sustain, decay, aftertouch, pitch bend, duplicate notes played on different strings, hammer-ons, single note bends within a chord, French arm simulation, slide simulation and effects such as chorusing, repeats and volume pedal swells that can change dynamically as necessary during the performance.

Next, the guitar processor would interactively communicate with all of the other processors assigned to this particular tune. Through a recursive process, the entire tune would be scanned by all of the processors until they came up with parts that would fit together in a way that was pleasing to the producer of the piece of music. The entire performance would be modified by a routine in each processor that simulates "taste" (in the musical sense). After a few passes through the tune, the tape machine would be placed in the record mode and the final results captured for all future generations to hear.

Finally we have to give this guitar processor a name. I think it would have to be Dean Parks. EQ

NEW MQ™ SERIES. MASTER QUALITY DIC///DAT™ FOR THE ULTIMATE QUALITY MASTER.

DIC///DAT introduces its new MQ Series DAT tape. Master Quality media crafted with second-generation technical improvements based on extensive R&D and feedback from audio pros in the field. The one DIC///DAT perfected for professional use.

- **NEW MQ "Formulation Finish"** reduces friction and provides more efficient tape-to-head contact for superb performance even in the face of high search and rewind speeds.
- **NEW MQ proprietary DIC MicroFinity metal particle technology** creates a consistently high quality magnetic medium for uncompromising professional DAT recording.
- **NEW MQ exclusive DIC///DAT tape lengths** include an extra two minutes for tone recording, test signals and pre-roll identification—as well as the industry's only 15–30 minute DAT cassettes for small programs, demos and test tapes.
- **NEW MQ dust-busting cassette design** virtually eliminates recording errors and tape jamming due to environmental contamination.
- **NEW MQ unique window calibrations** for easier reading and more accurate time judgment.
- **NEW MQ extra-strong DIC///DAT cassette shell** made in the U.S., with new window ribs for increased stability, 100% anti-static plastic, and the industry's only fiber-filled slider for structural integrity, stands up to the most rigorous studio use.

DIC///DAT™ MQ SERIES

TAKING MUSIC TO THE *Nth* DEGREE.

222 Bridge Plaza South, Fort Lee, NJ 07024 · 1/800-DAT-1-DIC (in NJ, 201/224-9344)

DIC Digital

CIRCLE 52 ON FREE INFO CARD

© DIC DIGITAL SUPPLY CORPORATION 1991

Even Our Smaller Systems Can Fill A Big Hall.

SR4700 Series

You've seen and heard our big systems, installed in major venues around the world and on tour with some of the biggest names in the music business. The SR4700 Series Loudspeaker Systems are equally at home on the road or in the hall.

Their legacy comes from Cabaret® Series, a prime player in small to medium club sound reinforcement since the seventies. Their influence is Concert Series,™ the new standard in packaged JBL systems. With Cabaret as the starting point and Concert Series as our current direction, we listened to your comments and suggestions, turning them into the sound system of the nineties and beyond.

The SR4700 Series is pure pro audio. Pure JBL components, the same caliber as those in our larger systems, including the Vented Gap Cooling™ low frequency drivers and the 2445J compression driver. Pure JBL enclosures, five times stronger than traditional designs, and protected by plastic-coated steel grills and super-durable fabric-laminated plywood cabinets. Pure JBL sensibility, interlocking enclosures with heavy-duty steel handles and recessed input connectors, so they travel tighter, set-up easier and last longer.

SR4700 Series, pure JBL. Be the first in line to hear them today at your local JBL dealer.

JBL Professional
8500 Balboa Boulevard, Northridge, CA 91329 USA
A Harman International Company

CIRCLE 26 ON FREE INFO CARD