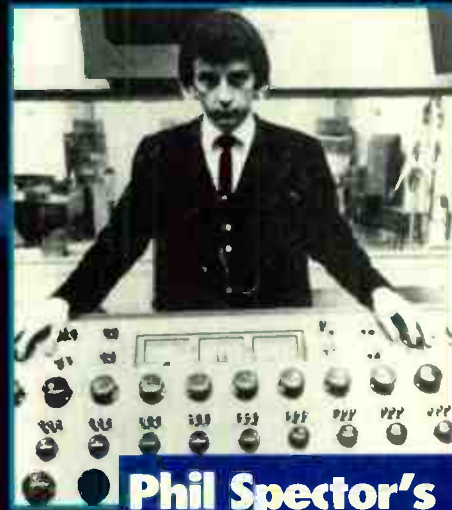
PROJECT RECORDING & SOUND

# EQ

JAN/FEB 1992 · PSN PUBLICATIONS

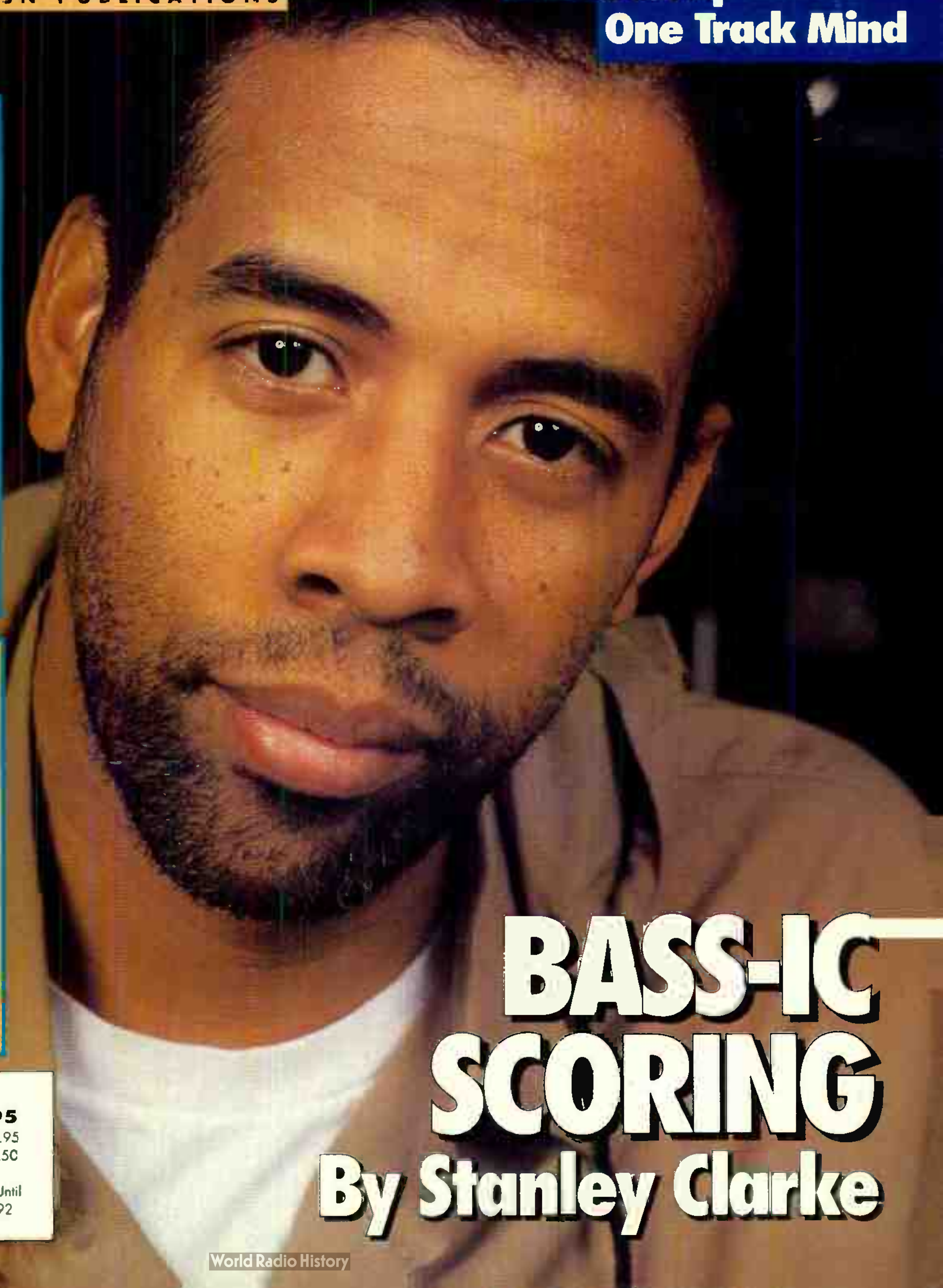


Phil Spector's One Track Mind

EXCLUSIVE PRODUCT REVIEWS

- PEAVEY SP
- QSC EX 1600
- DIGITECH VOCALIST

- CRAIG ANDERTON'S NAMM PREVIEW
- THE COMPLETE COMP/LIMITER
- SOUND REINFORCEMENT SOUNDCHECK LIST
- STAGE MONITORING
- DIGITAL SHOOTOUT AT ELECTRIC LADY



# BASS-IC SCORING

## By Stanley Clarke

02

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Obviously, our "smart" digital processors don't look like the old, traditional processors you're accustomed to. What may surprise you is that they don't process that way either!


Roland's custom VLSI chips give our Processors amazing power at a price that will astound you!

For example, here are some comments from Roger Nichols about his own Roland R-880 Digital Reverb. "The detail with which you can construct a process is amazing! You can select the size and shape of the room, the number of early reflections, and the time and amplitude of each individual reflection. Or you can have two different choruses going at the same time, each with its own depth and rate. So you think the left input should go through a chorus and then to the reverb section, but you want the right input to bypass the chorus and go into the reverb, maybe with a little compression along the way? No sweat."

# ROLAND'S "SMART" PROCESSORS. HOW DID YOU EVER GET ALONG WITHOUT THEM?


With 20Hz to 20kHz frequency response, greater then 90dB dynamic range, and less than 0.015% THD, the R-880's transparent performance provides natural ambience for acoustic instruments.


The R-880 Reverb gives you an LCD "workbench" so you can lay out and interconnect functions until you have exactly the effect you need. And there's no need for you to dedicate separate DSP functions like equalizers or compressors to effects sends or returns, because these are built-in and user-configurable right at the controller.

Actually, those two words best describe the idea behind buying and using all our digital processors. "No sweat!"

Consider our E-660 Digital Parametric Equalizer. It's a new breed of studio tool operating entirely in the digital domain. Designed for the ultimate in sound quality and ease of use, it offers eight bands of equalization in a variety of configurations. It even gives you 99 storage memories for individual curves,


It's a whole new way to look at reverbs. The R-880's effects exist totally in software, so you can create unique effects unattainable with other devices. Optional memory cards let you use Roland's pre-programmed effects, or you can use the same card and write your own. And after you've custom-programmed exactly what you need, these programs are stored into memory for future use!

each recallable at the touch of a button. And the "660" is MIDI compatible!

And while we're on the subject of doing things right, you should check out our SN-550 Noise Eliminator. First, it's affordable. But it's also an extremely sophisticated, yet easy to operate, single pass unit that works in real-time.

Laboratory-grade construction, components, and grounding techniques deliver a flat frequency response from 20Hz to 20 kHz, a dynamic range greater than 94dB, and 0.015 THD. (The screen shown lets you reverse the polarity of either or both channels from the front panel.)


Not only can you control analog and digital levels separately (screen 1), but the E-660 Parametric EQ lets you store and recall a precise EQ history, from microphone to master, either as an exact numerical setting, or as a representative curve (screens 2 & 3).

And because all signal processing occurs in the digital domain, the integrity of the original signal is preserved at every stage. We even put a hum canceler in the "550" that *really* works. As one of *R-E-P* magazine's testers put it, "Its hum cancellation capabilities are nothing short of amazing."

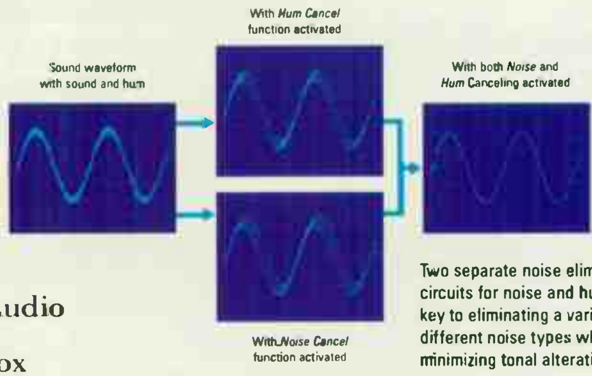
16-bit linear A/Ds and 18-bit D/As with 48kHz sampling frequency gives the SN-550 Noise Eliminator exceptional dynamic range and sound quality.

So before you buy *any* audio effects processor, you owe yourself a look at Roland's line of "smart" processors.

You really *will* wonder how you got along without them!

And don't forget to send for illustrated brochures on these, and other fine Roland Pro products, like the new DM-80 Digital Audio Workstation, the SBX-1000 MIDI Cueing Box and the Roland *Sound Space*™ Processor.

Call Roland at (213) 685-5141, ext. 337.




Two separate noise eliminating circuits for noise and hum is the key to eliminating a variety of different noise types while minimizing tonal alteration.

Roland ProAudioVideo

CIRCLE 36 ON FREE INFO CARD

Mr. Nichol's comments courtesy *EQ Magazine*  Roland Corporation US, 7200 Dominion Circle, Los Angeles, CA 90040-3696

# EQ

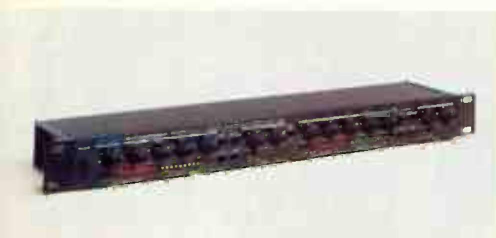
PROJECT RECORDING
& SOUND TECHNIQUES
**VOLUME 2, ISSUE 6**
**FEBRUARY 1992**

## FEATURES



## TECHNIQUES





## WORKSHOPS



## COLUMNS/DEPARTMENTS



Stanley Clarke by Ed Colver; Phil Spector: Michael Ochs Archives

EQ (USPS 002-952) (ISSN 1050-7868) is published bi-monthly by P.S.N. Publications, 2 Park Avenue, Suite 1820, New York, NY 10016. Second class postage paid at New York, NY and additional mailing offices. POSTMASTER: Send address changes to EQ, P.O. Box 57294, Boulder, CO 80332-7294.

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## SPECIFICATIONS

| Model | 4 ohms | 8 ohms | bridged mono |
|---|---|---|---|
| Power-Tech 1 | 300 watts | 220 watts | 575 watts |
| Power-Tech 2 | 440 watts | 320 watts | 965 watts |

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EQ

A PSN Publication
Vol. 2, No. 6
February 1992



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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

EQ (USPS 002 952) (ISSN 1050 7868) is published bi monthly by P.S.N. Publications, 2 Park Avenue, Suite 1820 New York NY 10016 ABC audit applied for. Second class postage paid at New York, NY and additional mailing offices POSTMASTER: Send address changes to EQ, P.O. Box 57294, Boulder, CO 80332-7294 All product information is subject to change; publisher assumes no responsibility for such changes. All listed model numbers and product names are manufacturers' registered trademarks.



## TAPE THAT!

Upon reading the article "Snap, Crackle and Pop Music" by Roger Nichols in your October issue, we were surprised to learn that he did not approve of Mobile Fidelity Sound Lab's ULTRADISC version of Steely Dan's *Gaucho*, which we released in March 1991. Roger's assertion that our source tape was incorrect was a true surprise in that he himself recommended the 3M digital tape which we used for mastering.

*Michael Grantham*
*Vice President*
*A&R, Product Development*
*Mobile Fidelity Sound Lab*

*Roger Nichols responds: Thanks for your letter. Yes, I recommended the 3M tapes, but since I did not receive a DAT or CD ref of the project, I had no way of knowing whether or not you got the proper tape. It seems that since the MCA version of Gaucho and the ULTRADISC version suffer from the same problems, MCA probably sent you the same incorrect tape they used for their own CD prep.*

*Just to set the record straight, I have been a Mobile Fidelity fan ever since the Drag Strip recordings of 1963. It was also my pushing and shoving that got Mobile Fidelity the rights to the Aja album for their F-1 format tapes. I believe that Mobile Fidelity may have been just another innocent bystander affected by the record companies' "don't-give-a-damn" (expletive deleted) attitude towards proper handling of artists' masters.*

## HEY! DON'T WAIT ON-LINE

Every issue of *EQ* answers more of my real working questions than any other magazine. EQ&A is an especially valuable column that I wish you'd expand. Unfortunately, sometimes I need my questions answered right away. What can I do?

*James Rivlin*
*Ann Arbor, MI*

*If you own a modem you'll soon be in luck. EQ will be introducing an EQ&A FORUM on America On-Line this year. This will allow our editors or other EQ subscribers to answer your questions right away. It will also allow us to stay in closer touch with our readers so that we can address your most pressing editorial requirements. We'll keep you informed about the details. America On-Line already has several active forums that address the information needs of computer musicians.*

*—Editors*

## SPARRING WITH ROGER

Like many your other readers, I'm sure, I was amused by Roger Nichols' column in the December *EQ* entitled "Big D." His tongue-in-cheek labeling of an "AD-Ad-ddDaD-Ad-D-D" recording was not only perceptive but what I would call "post-prophetic."

I use this fabricated term because, on one hand, he sensed the need for the demise of what has been known as the "SPARS code." What Mr. Nichols had no way of knowing though was that after considering changes to the code for several years, on October 6th of last year, the SPARS (Society of Professional Audio Recording Services) Board of Directors came to the conclusion that the code had outlived its usefulness and should be retired. A news release was issued to the press late in November, too late to be noted in Mr. Nichols' column.

The SPARS code was devised in the mid-Eighties to help identify which portions of the recording process were digital and which were analog. Times were simpler then! The changes that have taken place in the recording process since make the simplicity of the three-letter code unrealistic in its ability to describe or judge a final product. Not only are there now "super-analog" systems that compete head-to-head with top-end digital systems, but there's such a proliferation and wide range of "digital" products that just stamping an *A* or a *D* on a release services no immediate

## 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.

indication as to the end quality of a recording.

As I hear my 3-year-old son "reading" his book with one-bit digitized dogs barking from its pages at him, I wonder if we should come up with a code for books, too. No? Well, let the SPARS code rest in peace and let's press on to other matters.

*Richard Trump*
*President*
*SPARS*

## BOOGIE ON DOWN

Your recent section entitled "Boogie Boards" was right on the money — literally. I've been shopping for an inexpensive mixing console for several months and your comparison chart did most of my shopping for me. Thanks. I'm also looking for ways to buy a console that also works in my home studio, any ideas? And, by the way, how about more articles for bands like mine that make their living in small clubs. Every other magazine covers sound reinforcement for the stadium circuit. Keep up the great work. I'm a reader for life.

*Harvey Nelkin*
*Tamarac, FL*


*Check out the articles on doing a soundcheck and designing a stage monitoring system in this issue. And, yes, we'll be devoting more space to articles like you suggest in future issues. Just as we provide how-to information for the project recording studio, we also specialize in sound reinforcement for the average working band.*

*—Editors*

## STILL MORE ABOUT LES

After reading your summer cover article by Les Paul I have seen numerous other articles about the maestro in other magazines. I also was lucky enough to see him perform and talk at AES. I was most impressed by his quick wit. Any suggestions on how I can get more of Les?

*Ellen Wilkins*
*Sacramento, CA*

*Be sure to check out the box set entitled "Les Paul. The Legend and the Legacy" just released by Capitol. It contains more than thirty unreleased tracks and contains track-by-track comments by Les himself. Don't forget that you can hear him live when you come to New York. He still gigs every Monday night at Fat Tuesdays on Third Avenue.*

*—Editors*

# EQ &A

## DRUM ROLAND

**Q** *I have a Roland TR 505 drum machine. I would like to create eight total outputs (the unit has two) by accessing the unit's internal circuitry. How would I do this without ruining the device? Does Roland offer a retrofit kit that would accomplish this task?*

*T. J. Smith*
*Hurst, TX*

**A** The Roland TR-505 does internally produce eight separate sounds so it is possible to custom modify the unit to provide eight outputs. However, no such instructions exist at Roland for this nor do we endorse such changes. Changes such as these are sometimes done by third-party service centers, but we do not presently know of any offering this change to the TR-505. We can, however, provide a service manual that includes schematics, printed circuit board layouts, etc. Call Roland at 213-685-5141 (ext. 233) to order. Achieving eight outputs goes beyond making simple changes in connections and requires rather extensive rework. It might be more feasible to consider upgrading to a Roland R-8 drum machine with the eight outputs already built in.

Edward J. Coker
Technical Services
Roland Corp., US

## PLANT FOOD FOR THOUGHT

**Q** *I saw the ART Power Plant review in EQ some time ago [Feb. 1991] and purchased a unit. Although I like it a lot, I heard another Power Plant that a friend was using and it seemed to have more sustain. Am I just hearing things, or is there a possibility that something is wrong with my unit?*

*Frank Lindsey, Amarillo, TX*

**A** According to Buck Brundage, the main designer of the Power Plant's circuitry, the original production run of Power Plants used a 20k resistor for R10. This was later changed to 10k to increase the sustain. To find out which version you have, with the line cord disconnected, remove the four screws that hold the top lid in place. On the part of the circuit board near the front panel, locate the words APPLIED RESEARCH & TECH. R10 is right next to RESEARCH. If the value is 20k, tack solder another 20k resistor in parallel with it to reduce the resistance to 10k. Put the lid back on, and the modification is complete.

Craig Anderton, West Coast Editor

## ONE GOOD APPLE

**Q** *I'm putting together a major studio facility and want to have a couple of small, Mac-based preproduction MIDI rooms. I'd like to keep costs down by using a Mac Plus or SE, but some people say you really need a Mac II for musical applications. Apple has so many CPUs these days it's hard to keep up! What's the best choice for my situation?*

*Keith Gaudette,*
*City of Light Recording and Sound*
*Lakeport, CA*

**A** The Plus, Classic and SE are not good choices for musical applications. New software makes ever-increasing demands on the computer, and the older models aren't fast enough to guarantee stable timing with the current generation of sequencers.

Probably the most cost-effective option is the Classic II, which retains the small size and convenience of the Classic/Plus/SE family but uses a 68030 to run approximately twice as fast. The basic configuration is 2 Megs of RAM and 40 Meg hard disk for about $1300. 4 Megs and an 80 Meg hard disk costs $1800. Even with the 68030, however, the Classic II is not quite as fast as the now discontinued SE/30. If you can find a good deal on a used SE/30, you'll have excellent performance at a reasonable price.

The next step up would be a Mac LC. It offers similar performance to the Classic II but is larger and requires a separate monitor (usually color; the Classic II is gray-scaled). The LC is also more expensive. A typical system with a street price of $3100 consists of a Mac LC, 14-inch color monitor, 4 Megs of RAM, 40 Meg hard drive, Math co-processor, and extended keyboard. One caution: if you plan to work with digital audio, then a Mac II family machine with a big hard drive is pretty much essential.

Daniel Rose, Mark of the Unicorn
and Craig Anderton

## ON THE CIRCUIT

**Q** *What does the term "balanced" mean? What's the difference between "transformer" and "transformerless" balancing?*

*Phil Hill, Easton, PA*

**A** A balanced circuit has an inverting (negative) and a non-inverting (positive) voltage path, as well as a separate path which acts as a "shield" and (hopefully) directs stray electrical signals to ground. Usually, an unbalanced circuit uses the minus side to carry the ground path and, therefore, was once considered more susceptible to induced hum. In practice, a high-quality modern circuit will not usually require balancing when cable runs are under 20 feet long.

A "transformer balanced" circuit uses a passive (non-powered) coil (a transformer) to power the signal. An "active," or transformerless, circuit uses active amplifiers to accomplish this task. With today's advanced technology, neither approach can be called right or better than the other in every situation, but rather must be judged by the specific application and

This is where your questions get answered. Send your query with your name and address to:
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overall quality of the design and components in a particular product.

Jimmy Yamagishi and Bill Stevens
TEAC/Tascam

## JUMPING TO CONCLUSIONS

**Q** *While most of us are using compression to make vocals sound "bigger," often the result is just the opposite — the vocals sound thin, like someone has his hands wrapped around the vocalist's neck. I record and produce commercials and I've recently heard a few commercials that seem to "jump out" of the speakers. These spots are evidently highly compressed, yet have great brightness and warmth. How do the rest of us, who when we use high ratios and quick attack and release times end up with dull overcompressed recordings, go about getting a loud, full sound? Compression usually means making the loud parts soft, then raising the overall volume level. However, how do I make the soft parts loud!?*

*Dan Popp*
*Akron, OH*

**A** Making a commercial jump out of a speaker is possible (and also a good trick). The secret is a careful blending of vocal delivery, mic choice, mic technique, compression, limiting, and EQ. In a nutshell, you must preprocess the signal that you deliver to the station so that it is more or less immune to the station's compressor(s).

The station's management wants their signals to leap out of the radio, too, of course. The problem is that if the station overmodulates (exceeding 100 percent modulation), the FCC can fine them. The end result is that most stations use several compressors, each covering a separate frequency band, then feed it all into an overall limiter. The key is to get your audio signal through this audio gauntlet unscathed.

Peaks are responsible for setting off the station's compressors. These peaks may be in voiceovers, but more likely they're in the music track accompanying them. Once the compressor reacts, the loudest thing in the mic controls the compressor, and everything else around it gets beaten down. Try sending your overall mix through a compressor before going to tape. Use a moderate attack time, a release time long enough to prevent pumping, and a moderate (4:1) ratio.

For your vocals, you may be trying too hard. Lighten up on the ratio. Try 4:1 — 8:1, and use a peak limiter to gain some absolute peak protection. A soft-knee compressor will sound subjectively better without being really obvious. Six to 9 dB of gain reduction are all that you'll need. If the compressor has an auto release function, try it. In many units, the auto function does things that you can't do with the knobs. Put your EQ after the compressor so the compressor won't undo your EQ.

Last, there's no substitute for a great sounding mic (start with good ingredients, right?), voice talent that knows how to deliver the words, and, most importantly, mic technique.

Rick Chinn
Applications Engineer
Symetrix

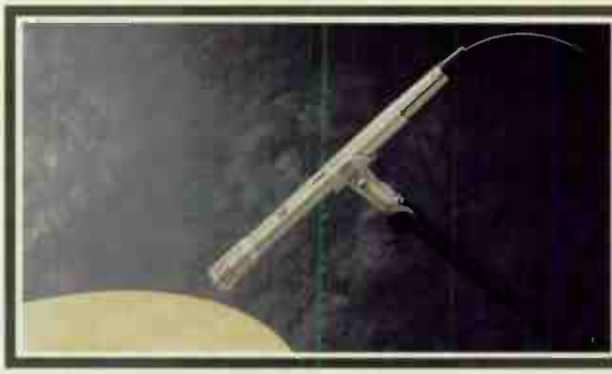
# SHURE

# DRUM KIT MICROPHONES

## The Standard of Versatility and Performance

SNARE: BETA 57

OVERHEAD: SM81



KICK: SM91

TOMS: SM98

HI-HAT: SM94

OVERALL: SM57

There are many reasons why Shure microphones are the first choice for percussion sound reinforcement and recording, but it all comes down to performance. You can always rely on Shure microphones to give your drum sound the extra drive you need to get to the top. The Beta 57 will deliver your snare drum tone with maximum punch and impact while isolating the "bleed" from adjacent drums and cymbals, and its steel grille will survive the worst abuses of the road. The SM81's ruler-flat frequency response will capture every nuance of your cymbals and the natural ambience of your entire kit, with a reliability and durability found in few condensers. The SM98 is a natural for toms--small, unobtrusive and easy to set up. Its polar pattern can be modified to supercardioid with the optional A98SPM. The SM91 will provide power, definition, crispness and isolation in the kick drum position, and it's easily positioned without the use of a stand. The SM94 will help the natural sparkle and personality of your hi-hat and individual cymbals cut through the mix. And the world standard SM57 may be used in any position, as it has been in defining live and studio drum sounds for over 20 years. For more information on Shure drum kit microphones, call 1-800-25 SHURE.

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# With Strings Attached

Producer, engineer, mixer Glenn Rosenstein plays (guitars) hard, too

**Top gun:** Glenn Rosenstein.
**Studio of choice:** Sixteenth Avenue Sound Studio, Nashville.
**To his credits:** Ziggy Marley & The Melody Makers, *Jahmekya* and *One Bright Day* (Virgin), produced, engineered, mixed and played guitar; Brent Bourgeois (Charisma), produced and engineered; Michelle Shocked, *Arkansas Traveler* (Mercury), co-produced and engineered. Also worked with: Madonna, Lisa Lisa & Cult Jam, Talking Heads, U2, The Ramones, Tears For Fears, and mixed (with David Byrne) soundtrack of *The Last Emperor* (Virgin) and *Married to the Mob* (Reprise).
**Main rack (top to bottom):** Aris 615 rack illuminator, Eltekon RX2 removable hard-disk drive, Roland S770 Sampler (with 16 MB expanded memory), Drawmer DS201 dual audio gates, Lexicon 300 digital effects system, Lexicon 200 digital reverb.
**Left of rack:** Princeton Graphic Systems HX-12+ RGB monitor (for use with S770), Gibson Chet Atkins SST acoustic guitar.
**Right of rack:** KEF C55 studio monitors, Korg DT-1 Pro Digital Tuner, Panasonic SV-3700 DAT recorder, Gibson Les Paul Custom electric guitar. (Not shown: Roland Juno 60 analog synthesizer, Roland D-70 digital synthesizer, two beyerdynamic M-88 mics, six or seven vintage Fender amps.)
**Equipment notes:** The Roland S770 is one of the warmer, more natural-sounding samplers. I like that I can stack backgrounds in it with a minimum of intermodulation distortion and a maximum of clarity. The Lexicon 300 sounds incredibly clean and is very, very quiet. Its cue list lets me program EQ and effects changes in real time. I find the Panasonic SV-3700's A/D and D/A converters to be a bit cleaner than the SV-3500's. The KEF C55s are really happening, with minimal phase shift between tweeter and woofer. Roland's Juno 60 (pre MIDI) has that warm, fat bass sound; it sounds funkier than the newer stuff. I use basically Gibson guitars. The Chet Atkins has a very hi-fi pickup and sounds beautiful direct. The beyerdynamic M-88s were a gift. They're the only mics I own right now.
**What works for him:** As a producer or engineer, understanding music helps you communicate more effectively with the artist. I don't try to sell myself as a musician, but being able to play a part here or there can only help in the studio. As for lugging stuff around, I only buy and carry the stuff I can't live without. The rest is easy to find; if the studio or engineer doesn't have it, I rent it.
**Pro tips:** To eliminate vocal pops without a windscreen, try using a "dummy" mic directly below your actual mic. Have the singer "aim" at the inactive one. With a little practice, the vocal will still be "on-axis" but free of explosive breaths. EQ

Photo by Beth Gwinn

# Das Zimmer

**The Man:** Hans Zimmer (Zimmer is "room" in German)
**Credits:** Scored the films: *Rain Man, Driving Miss Daisy, Days of Thunder, Pacific Heights, Backdraft, Green Card, Thelma & Louise, Regarding Henry.*
**Consoles:** Euphonix CSII 8064 and 5 Yamaha DMP7s
**DATs:** Otari DTR-900II, Panasonic SV-3900 DAT, Sony VO 9800 U-Matic
**Monitors:** Three Yamaha NS40Ss, two Yamaha NS10s with a Yamaha DSP-A100 Surround Sound amp.
**Outboard Stuff:** Lexicon 300, Yamaha Rev 1, Yamaha SPX 90, Eventide H3000, Roland Phazer, Roland Flanger.
**Keyboards:** Three Roland Super Jupiters, two Yamaha TX 816, a Roland MKS 20 and a MKS 70, Roland D550, Roland System 700, Midi-Moog, Korg M1R, Fairlight series III, two Korg Wavestations A/D, twelve Akai S1000/S1100 samplers. (I must like these!)
**Mics:** Bruel & Kjaer.
**MIDI:** Dash 6800 computer tower, Cubase sequencer, and The Mark of the Unicorn Midi timepieces.
**Equipment notes:** I really like the Euphonix because it's totally automated and it doesn't take up too much real estate. And it has a helluva lot of inputs (64 x 6). If I've got to do a film in Australia, I can just pack it up and be on the go or I can take a floppy disk with me if I know there will be one on the site. It's solid and can do everything it claims to be able to do. I always work in Surround, so I love using the Yamaha NS40s, because they allow me to get a good surround sound without bouncing sound all over the place.
**Studio tips:** Get the wiring right. Figure out exactly how you like to work and then set up your patchbays. A lot of my stuff is even. And no one has invented a box that can do everything. So don't try to cram everything you need into one piece. I don't like equipment that I have to spend three days reading the instruction manual in order to use. You should be able to sit down in front of it and play with little or no hassle. EQ

*I REALLY LIKE THE EUPHONIX BECAUSE IT'S TOTALLY AUTOMATED AND IT DOESN'T TAKE UP TOO MUCH REAL ESTATE.*

Photo by Ed Colver

*My feet just, like, took over.*
*It was eerie. But you know, good eerie.*

The new Fostex X-28 four-track. Rehearse with your feet. Play with your hands. Listen with whatever's left over.

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CIRCLE 17 ON FREE INFO CARD

# APOCALYPSE NAMM

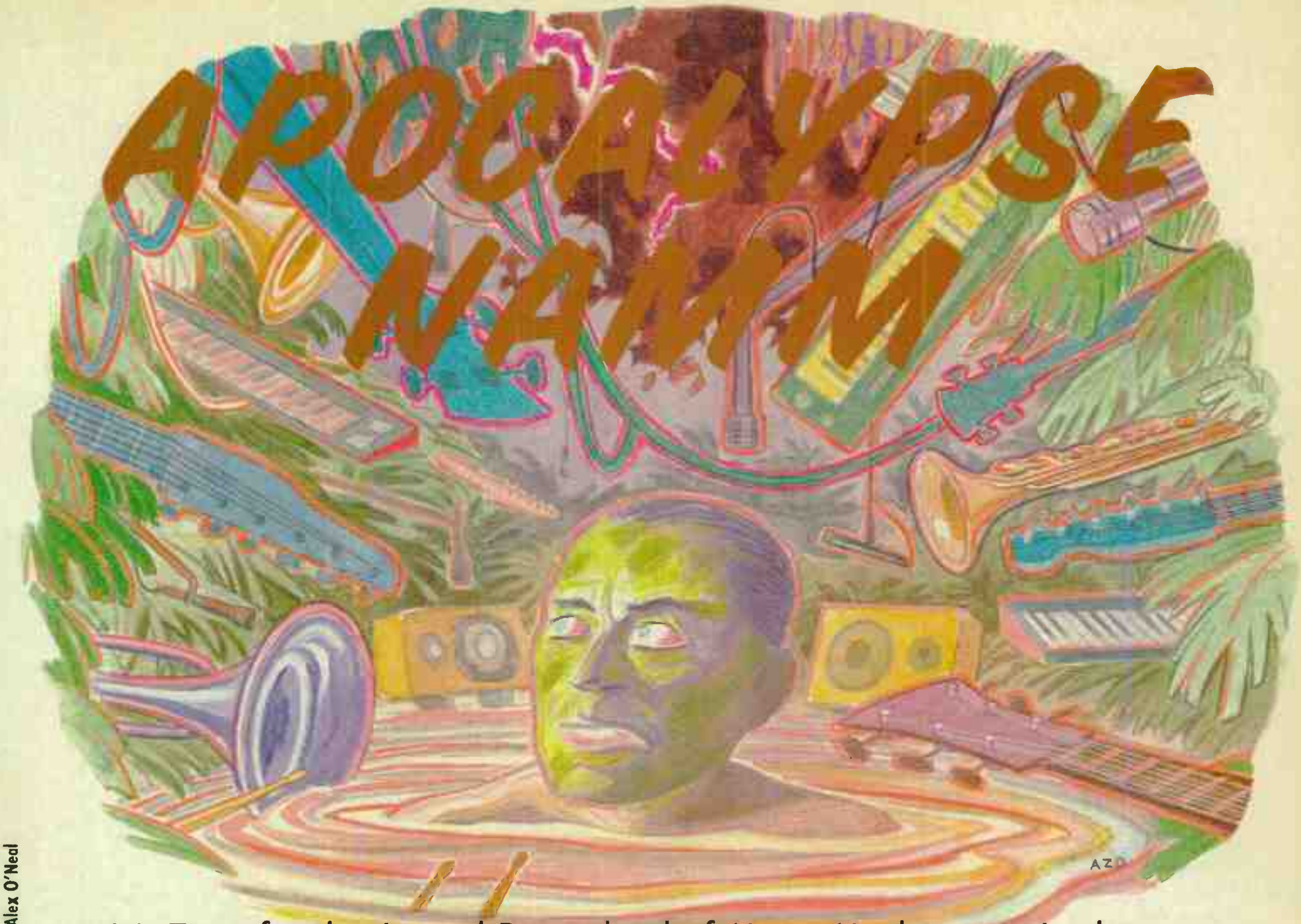


Illustration by Alex O'Neal

## It's Time for the Annual Disneyland of Music Machines in Anaheim. Perk Up Your Mouse Ears, Put on Your Commando Gear and Just Jump in — Brain First

**BY CRAIG ANDERTON**

Where's the music industry going in the years ahead? The place to calibrate your crystal ball is the NAMM (National Association of Music Merchants) convention, held each January in Anaheim, California (this year, from the 17th through the 19th). The show's premise is simple: manufacturers have three days to show off their gear, with a special emphasis on new stuff, to dealers in the hopes that thousands of orders will result, to journalists in the hopes that thousands of words will result, and to assorted rock stars in the hopes that thousands of endorsements will result.

# NAMM VIEWS

NAMM has seen its share of history. In 1978, the Prophet-5 made its debut and launched the age of the polyphonic, programmable synthesizer. In 1982, about ten companies that were curious about something Sequential was working on called the Universal Synthesizer interface attended a meeting. One year later, a MIDI cable connected a Sequential and Roland synth together, and the MIDI revolution was born. (I'd been following the progress of MIDI at the time and will never forget when Sequential's Dave Smith collared me on the show floor and said, with a look of relief and surprise, "We plugged it together and it worked!") In 1983, Yamaha showed a funny-looking keyboard with a bunch of membrane switches and claimed it was the future of synthesis; 250,000 or so units later, the DX7 had lived up to the hype. The modern era of quality, affordable sampling started in 1984 when E-mu did its now-famous Emulator demo, featuring sampled James Brown sounds.

And that's just keyboards; whether your bag is guitars, drums, signal processors, computer software, or whatever, NAMM is the place to be. It's like an extension of Disneyland, rolling Tomorrowland and Fantasyland into one package, along with a bit of Mr. Toad's Wild Ride. And a wild ride it is.

I'm writing this article in mid-November for an issue with a February cover date that comes out in January. (No wonder journalists never know what time it is.) So, I don't know yet what will be at NAMM, but here are some of the reality checks I'll be monitoring this year:

**THE COMPUTER DU JOUR.** Which booths have PCs and which have Macs? Do they dare run their software on anything less than a 386 PC or IIci/fx? How far will Atari's comeback go? You get extra points for spotting Amigas, an endangered species at recent NAMM shows.

**RETROVATION/INNOVATION.** A lot of companies have rediscovered secrets of the ancients such as tubes, tremolos and vocoders. I'm curious to see who makes the transition from retrovation to innovation. Retrovation could be a legitimate recognition that new is not always better, or it could mean that a company has run out of steam.

**GUITARS.** There's magic in a slab of wood covered with skinny little pieces of metal. If guitars are hot, then all's well with the world. When people lose interest in guitars, the music industry loses direction. *Really.*

**MULTIMEDIA.** Apple needs something to follow up desktop publishing. IBM needs new applications to give its technically ho-hum machines some pizazz. Commodore's last best shot for the Amiga is multimedia. Will the music industry believe the hype and jump into an immature market, or wait until there's an installed base of two million CDTV, CD-I, or MPC machines?

**REALITY CHECKS.** I ask everyone I see what they like at the show. People love to turn you on to the strange inventor in the corner booth with some absolutely whacked-out thingie. You may never see it again, but it's a consciousness-expanding experience to witness the fertility of the human mind. Remember the Gittler guitar? When I asked the inventor about its most important attribute, he answered "you can play it underwater." I haven't seen any Gittler guitars lately, but that phrase will stick in my mind until senility sets in, and possibly long after.

**SOFTWARE.** Software ain't what it used to be; a smaller number of players are dividing up the pie, so the illusion is that the market is good because the remaining companies are doing okay. But the folks who fell by the wayside may not feel the same way. What was the Wave of the Future has crested and turned into just another tool. Frankly, I like it better that way. I've always preferred a good tool over religion. Still, the ultimate piece of music software has yet to be invented. We're still seeing databases and word processors in musical clothes, just like MIDI studio mixers are still based on the PA model. Will this be the NAMM show where music software becomes a distinctive art form? One can only hope. Programs like Max and Alchemy, though, are pointing the way.

So you're probably saying, "Sign me up! I want to go!" But you have to be in the industry to enter NAMM's hallowed halls — the general public is not welcome. All is not lost, though. Here, in an *EQ* exclusive, is the world's first Virtual NAMM Show:

First, it's strategy time. A NAMM show is like one of those contests where you have three minutes to stuff a shopping cart full of goodies. The only difference is you have three days to stuff your head full of future shock.

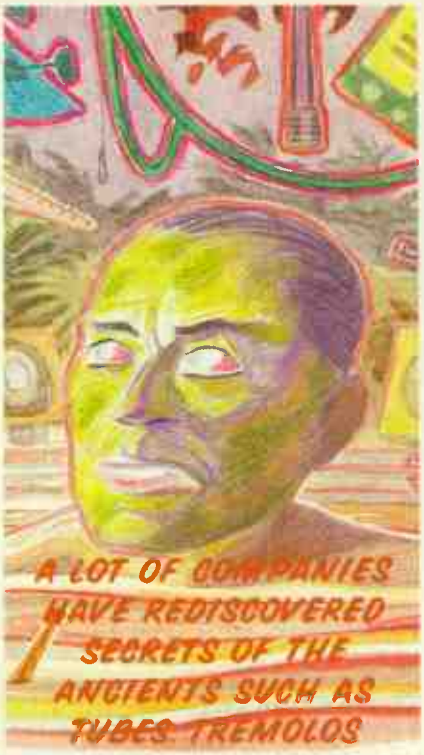
So, you arrive a day early to get a jump on the rest of the crowd. If you're extremely lucky, you're clutching an invite to the Peavey or Roland dealer conference, where you can shake off your jet lag and get a head start. Fortunately, you've been here before; you know which hotels have the swimming pools and hot tubs, and you've been taking walks a couple of weeks beforehand to build up your stamina.

Cruising the floor is nice, but you realize there are a few situations where you need quality time — to get a demo of something that really interests you, to smooth the feathers of someone who has ruffled feathers, or because the person you want to see always seems to be with some dealer who does $400,000 worth of business and you're not quite in the same league. This is where appointments come in. You've learned one of the top-secret NAMM tricks, which is to schedule the appointment for somewhere other than the person's booth. If they hang at their booth, they'll be interrupted; so you meet people at demos they want to see. That way you both get to see the demo, then socialize for a while.

Part of the NAMM experience is food, or actually, the lack thereof. At a three-day show, lunch is a luxury, so you go for a large, high-protein breakfast — the Japanese breakfast is ideal, and you've gotten used to handling seaweed and fish in the morning. Or maybe you go for steak and eggs. You learned the hard way last year that doughnuts and continental breakfasts are out; the quick rush of energy was okay at first but blood sugar rollercoasters are not a pretty sight. You decide it's always worth the couple extra bucks for room service so you can get an extra half-hour's sleep.

Where did the time go? You haven't seen anywhere near the number of booths you wanted to but the show has already closed for the day and it's time for dinner. You're scheduled to go to some parties where the food consists of little tiny things you eat with your fingers, some identifiable and some not. You make it back to the hotel just in time for a quick soak at the hot tub. Your muscles soothed, you mentally prepare yourself for the next day.

Actually, NAMM shows can be close to brutal. I've been going to them for over a decade, and they only get bigger, louder and more chaotic. And more and more people want to show their latest goodie, which often isn't really that different from last year's goodie, and each "just give us two minutes" demo often turns into a marathon.



A LOT OF COMPANIES HAVE REDISCOVERED SECRETS OF THE ANCIENTS SUCH AS TUBES, TREMOLOS AND VOCODERS.

So why do I still love going to NAMM shows? Easy. Because only crazy, dedicated people are in the music business. The opportunists got out a long time ago, once they realized that the only people who make money are Michael Jackson, Prince and record company presidents. The greedists have moved on to junk bonds, pyramid schemes or politics. All that's left at NAMM shows — especially in these recessionary times — are people who genuinely love what they do.

Yeah, NAMM shows are cool. Don't forget to give yourself a day or two to debrief — everyone will understand. And if you see a guy with glasses and a big smile walking around the floor in a state of amusement and fascination, do say hi. I'm sure we'll have lots to talk about in the one minute and seventeen seconds we'll have before one of us has to move on to the next appointment. EQ


## POWER AMP PARADE

QSC Audio is introducing its new line of MXa power amps, consisting of the 250 watt MX 1000a, the 350 watt MX 1500a, and the 450 watt MX 2000a. All units offer full short circuit, open circuit, ultrasonic, thermal and RF protection. New features include automatic fan speed control, an input slot for additional connectors, indicators and both active and passive input accessories. For more information, circle EQ free lit. # 142.


## MULTITRACK STAR

Fostex recently unveiled the X-18, a compact, 4-input multitracker that can be operated with either batteries or AC operation. Instruments or mics can be directly connected and each input is assignable to any of the tracks. Suggested list price is $399. For more information, circle EQ free lit. # 143.

## ROCKIN' & ROLLIN'

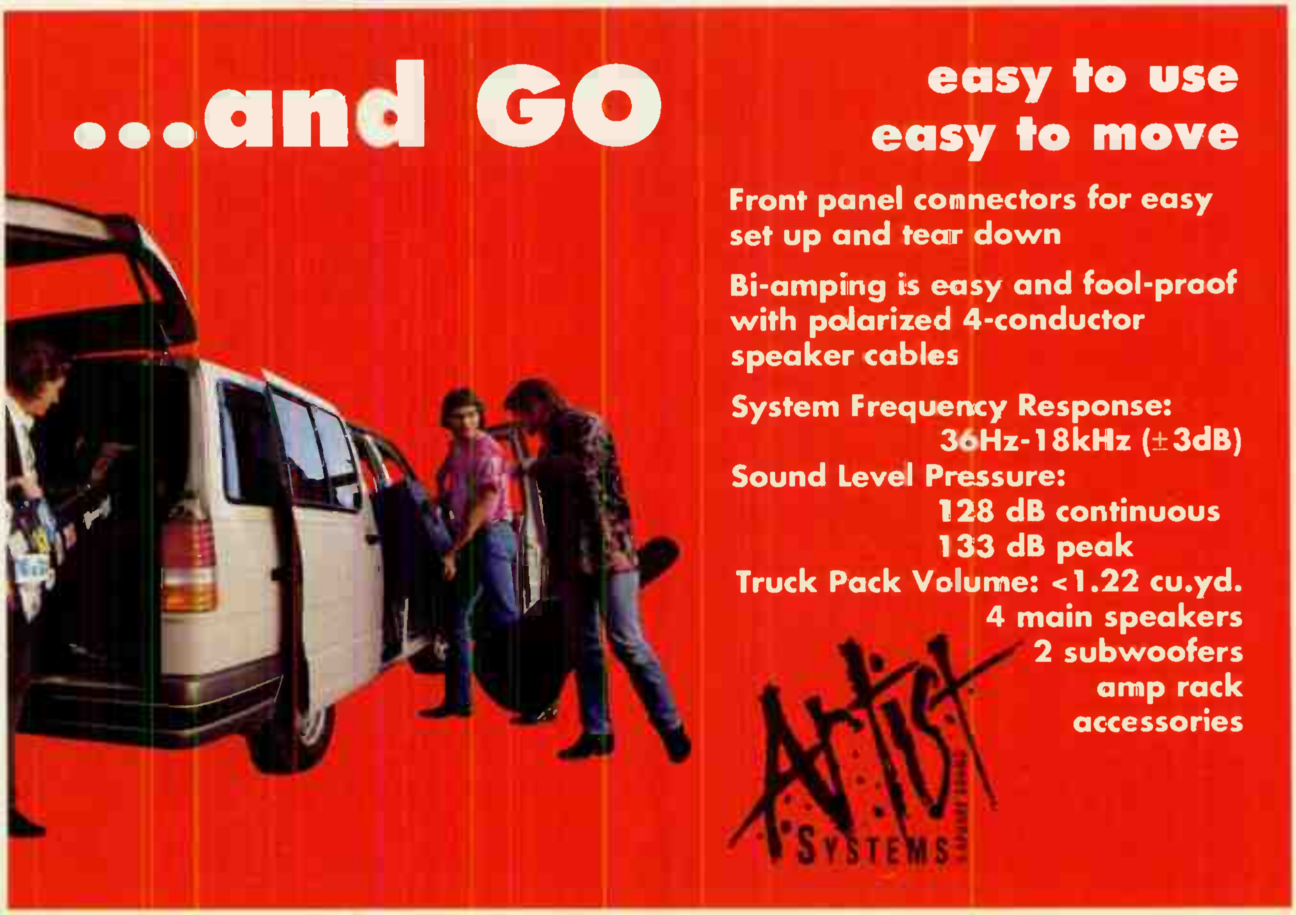
The long-awaited arrival of the Roland DM-80 digital audio workstation is upon us. This highly touted hard-disk workhorse can be run with a Macintosh front end or a remote controller/autolocator, and utilizes standard SCSI devices. The unit uses VLSI technology similar to that found on the company's acclaimed S-770 sampler. The four-track version lists for about $6995, while the eight-track version checks in at $9995. For more information, circle EQ free lit. #133.

## JUST FOR DAT

Designed specifically for DAT and high quality cassette recording, Audio-Technica's AT822 OnePoint X/Y stereo condenser microphone is suited for a range of audio/video field applications. The model is mono-compatible and is equipped with a pair of wide-range, closely matched cardioid condenser elements. Its user net price is $299. For more information, circle EQ free lit. # 134.

## GETTING BAFFLED

Acoustic Science Corporation is showing the Studio Trap, a free-standing, self-contained acoustic baffle that is tripod mounted with a floating suspension. The unit lets you set up an acoustic field in any room. Its suggested retail price is $255. For more information, circle EQ free lit. # 144.




CIRCLE 05 ON FREE INFO CARD

## MR BIG

JBL has introduced its line of MR Series sound reinforcement speakers. The line includes 10 models all incorporating new Selective harmonic Geometry, Thermoset Composite unique voice coil transducer technology and an improved titanium diaphragm compression driver.

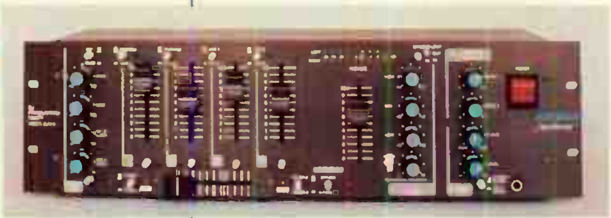
## MISSING LYNX

TimeLine introduced its Micro Lynx low-cost machine control system for synchronizing audio and video tape transports and MIDI. The basic unit supports two transports plus MIDI features including SMPTE and MIDI timecode generators, two transport synchronizer/resolvers, MIDI-to-SMPTE synchronizer, and direct to Macintosh interface. The system consists of a compact rack unit and a remote keyboard that sells for $2495. For more information, circle EQ free lit. #145.

## ACTIVE VERB

The new Alesis Quadraverb comes equipped with a a comprehensive real-time MIDI control of parameters that's unequalled by other programmable signal processors. It also includes digital reverb and delay, chorus/flange/pitch detune and Leslie effect, 11 bands of graphic EQ, up to 5 bands of Parametric EQ, 1500 msecs of delay and programmable panning. For more info, circle EQ free lit # 138.

## SOMEBODY PLEASE SAY BOOM



The new DJM-8 deejay production mixer from Furman includes the company's unique PUNCH Subharmonic Processor for extra bass. The unit has eight stereo inputs (2 phono, 6 line) that can be routed to faders, and then to the mixer's crossfader. This mixer lists for $849. For more information, circle EQ free lit. # 139.

## FROM AXE TO ZOOM

Zoom Corporation of America debuts their new ZOOM 9000 16-bit digital multi-effects guitar processor, and is also offering an optional total-access foot controller (model FC01) to complement the new product. The unit features 21 programmable effects and up to five can be used at once. The price for the processor and the foot-controller is $399. For more information, circle EQ free lit. # 140.

## EXQUISITE TAC

Amek/TAC has introduced the TAC SR6000 sound reinforcement and live teleproduction mixing console. The board incorporates TAC's traditional hard bussing system and rigid steel chassis, boasting a weight-to-strength ratio that allows for a forty input chassis that weighs only 330 pounds. It's available in three chassis sizes, and ranges in price from about $34,000 to $49,500. For more information, circle EQ free lit. # 141.

# NAMM VIEWS

## REFLEX MUSCLE

The Klipsch KP-480-SW is an 18-inch pro bass-reflex speaker, utilizing a 15-inch passive radiator, that is designed for applications in which high output, low distortion, rugged construction and linear frequency response are on call. The speaker has an 8-ohm impedance and a frequency response of 44 Hz to 200 Hz +/-4 dB. Output is -10 dB at 34 Hz and there is usable response to 2.5 kHz. It handles 300 watts continuous pink noise from 40 Hz to 2 kHz for 8 hours, with peaks to 3 kW. Measurements: 27 3/4-inch high x 24 1/2 inch wide, and 23 3/4-inch deep. Weight: 85 lbs. For more information circle free literature #146.

## RANE DROPS

Rane has begun delivering their new ME 60 Stereo 1/3-octave graphic equalizer, featuring two independent channels of Constant-Q filters from 25 Hz to 20 kHz. The two-space rackmount chassis offers low-noise/low distortion performance and a full complement of connector hardware. For more information, circle EQ free lit. # 147.

## TO YOUR MIKING

Neumann has debuted the KMS 150, a "hand-held" hypercardioid vocalist condenser mic. Due to its special acoustic filter and transformerless, high load amp, even "loud explosive sounds" won't overload it. For more information, circle EQ free lit. # 148.

# WELCOME TO ANAHEIM

Photo courtesy of Disneyland

Our snoops have been working overtime to get the latest scoop on the hot stuff they'll be showing in Anaheim. Here are some of the things you'll be hearing about from other magazines in the months to come ...There will be a major announcement from **Fostex** regarding interface capabilities and developments between its tape machines and the products of most leading music software companies; this could be the major announcement of the show...Several **Mark IV** audio companies are now offering the Interface series of modular mixing consoles. As part of this "multi-brand concept," **Altec Lansing, DDA, Dynacord and Electro-Voice** will all market these new consoles, which are available in 8-, 16-, 24-, 32- and 40-channel mainframes...**Ramsa** is debuting the WP-1000 Series power amps, featuring class H circuit design that continuously evaluates the amp's input signal so that it runs cool and efficiently; also from Ramsa, the new WR-S4400 Series of 12, 16 and 24 channel 4-Bus mixers...New from **DOD** is the R 844 Quad Noise Gate and the R 866 Stereo Compressor + Limiter + Gate; in addition, the company will be showing the new 1642 16-channel rack-mount mixer and the new **FX 54** Attacker effects pedal, which features both compression and distortion in one simple box ...**Yorkville Sound** will be introducing the first compact 1200 watt stereo 12 and 16 channel powered mixers, the Audio-Pro 1212/1216, which feature on-board Alesis 16-bit digital signal processing...Speaking of **Alesis**, make sure to show up at their demo of their **ADAT** digital recorder; it was the hot talk of AES last fall and should generate equal enthusiasm at this mega-music event...**Stewart Electronics** is showing the PA-800 follow-up to its successful PA-1200...**Ross Systems** will be introducing the RCS Series mixing consoles...**Kurzweil** has begun shipping their K2000 synthesizer...**Amek/TAC** will world-premiere the Einstein console...**AKG Acoustics** will show off the new Tri-Power line of mics...and **Audio-Technica** will unveil eight new mics.

# Skins Game

People are quite surprised when I walk into a gig carrying my own mics. But I've gotten to the point where I think it's essential, and on many levels. My sound is now what I want to hear, and it's consistent. The instrument's appearance is enhanced by the omission of numerous mic stands, and my setup time is faster. But most important, the audience now hears the same sound I hear from my throne. This is no small feat. And it gives me more confidence that the gig will go well, allowing me to concentrate on making music. And that's what it's all about, right?

I've been fortunate to have worked with some of the finest engineers in the business, in both recording studios and live venues. I watch (and hear the results of) what they do. One of the most important things I've learned is that the microphone represents the crucial link in the chain of events that takes place during a concert or a recording.

I've always been intrigued by the tech side of things, anyway, and I've obviously discovered that some particular mics work better on certain drums than others. A customized drum sound can be attained, but only through the microphone which most accurately reflects your drumming approach and speaks your sound philosophy.

What is my sound philosophy? There are some drummers who want their sound to be "bigger than God." In comparison, I like the drums to sound out front like they do where I'm hearing them and where I'm playing them. Not any bigger or smaller than life, and not lost in the music. You're playing to make the music fit, right? So that means the mics have to pick up the sound without adding characteristics all their own. I'm not looking for a mic to enhance my sound; I just want the mic to convey it.

Over the years, I've learned a lot about miking drums from recording sessions. When I'm working on the sound with the engineer I usually try to be as cooperative as possible. I'll let him or her know that I can make adjustments to some of the drum tuning or set-up if necessary. At the same time, if the engineer asks, "Can you put a little tape on the drum-head?" I'd probably counter with, "Let me try moving the mic a little closer to the center of the drum." It's surprising how repositioning a mic will usually take care of a problem such as too much "ring." (Or, I'll explain that the sound they're hearing is, indeed, the sound of the drum, and invite them to stand by the kit as I play it to hear it from my perspective.)

Anyway, I've noticed in recording studios with really good engineers that their miking position is as even and symmetrical as possible from drum to drum (toms in particular). That's when I realized I could probably accomplish the same kind of consistency from concert to concert that I heard in the studio.

Shure's SM98 hard at work

## If You Don't Want to Sound Like a Different Drummer Every Night, Consider Traveling With (and Setting Up) Your Own Mics

BY PETER ERSKINE

# The Top 500

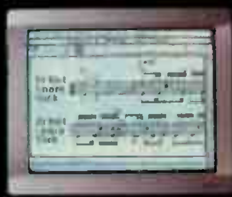
*Play the D4 with its onboard trigger inputs.*

Alesis drum machines are famous for their sounds. The HR-16's natural acoustic drums are still the standard for transparent rhythm tracks. The punchy aggressive samples of the HR16:B redefine how to make rhythm tracks burn. The SR-16 is an instant hit with its sampled reverb and ambience techniques.

Now you can have all this and more with the new **Alesis D4 Drum Sound Module.** There's an incredible 500 sounds in all. Right at your fingertips.

The D4's sounds are unparalleled for their realism. For example, when you hit a D4 sound harder, the tone *and* pitch change just like a real drum, thanks to the D4's Enhanced Dynamic Articulation.™ Plus, stereo reverb and ambience are built into many of the samples so you can keep your mind on the beat.

Using the D4 is a breeze with its large data entry knob and dedicated buttons for all major functions. There's even a touch-sensitive preview button and headphone output for instant gratification... and latenight drumset programming.



*Play the D4 with MIDI software or hardware.*

The D4's 21 user definable drumsets are accessible via MIDI or through the 12 onboard audio trigger inputs. You can even replace a wimpy drum sound on tape. Which you'll want to do if it didn't come from a D4. No rocket science here. Just pure honest incredible sound. The only reason to buy a drum sound module.

Everybody wants a hit. The D4 has 500 of them right now. At your Alesis dealer.

*12 audio trigger-to-MIDI inputs are built in for drum triggers, pads, or tape.*


## ALESIS THE TOP 500 DRUM SOUNDS

| THIS WEEK | LAST WEEK | WKS. ON CHART | TITLE / ARTIST | DRUM / PRODUCTION |
|---|---|---|---|---|
| 1 | 2 | 208 | **HONEST SNARE** HR-16 | ALL WOOD-BRASS RIM |
| 2 | NEW ▶ | | **STUDIO TOM** BRAND NEW D4 | 16" MAPLE TOM w/VERB |
| 3 | NEW ▶ | | **BIG "O"** BRAND NEW D4 | DOUBLE HEAD KICK w/VERB |
| 4 | 5 | 52 | **RIM SHOT ROOM** SR-16 | BRASS PICCOLO w/VERB |
| 5 | 10 | 156 | **BIG FOOT** HR-16:B | SINGLE HEAD 26" MAPLE |
| 6 | NEW ▶ | | **SLAM** BRAND NEW D4 | POWER TOM w/VERB |
| 7 | 23 | 156 | **COMBO SNARE** HR-16:B | PICCOLO PLUS WOOD |
| 8 | NEW ▶ | | **BIG BALLAD** BRAND NEW D4 | WOOD SNARE w/BIG VERB |
| 9 | NEW ▶ | | **FAT CITY** BRAND NEW D4 | SUPER FAT SNARE |
| | | | | ARTICULATED HI HAT |

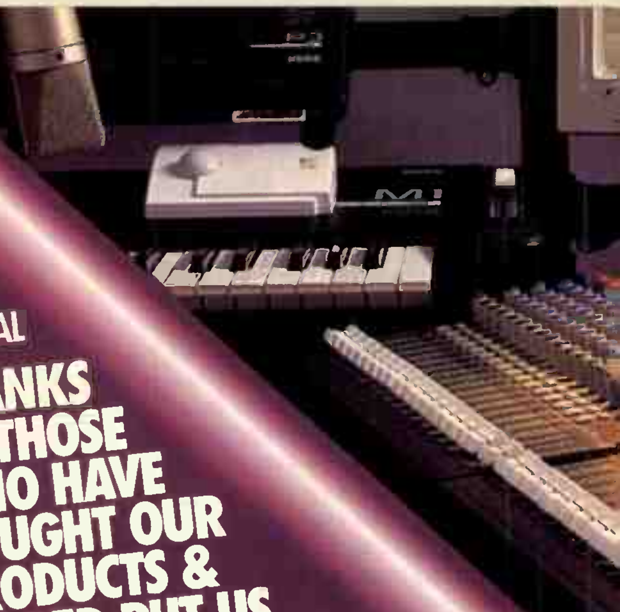
| THIS WEEK | LAST WEEK | WKS. ON CHART | TITLE / ARTIST | |
|---|---|---|---|---|
| 302 | NEW ▶ | | **RAW HIDE** BRAND NEW D4 | CUSTOM SN |
| 303 | 450 | 52 | **HI ROOM TOM** SR-16 | 10" MAPLE |
| 304 | NEW ▶ | | **WET HALF** BRAND NEW D4 | HALF OPEN HAT |
| 305 | 327 | 52 | **RIM 2 CENTER** SR-16 | ARTICU |
| 306 | 123 | 208 | **DOUBLE HEAD** | DOUBLE HEAD KIC |
| 307 | 223 | 15 | | |
| 308 | 401 | 5 | | |
| 309 | NEW | | | |
| 310 | 175 | 1 | | |
| 311 | NEW | | | |
| 312 | 171 | | | |

CIRCLE 03 ON FREE INFO CARD

**ALESIS** STUDIO ELECTRONICS

Alesis Corporation 3630 Holdrege Avenue
Los Angeles CA 90016

Over the past year, as the CR-1604 16-Channel Mic/Line Mixer's reputation has grown, we've been noticing more and more warranty cards returned by notable studio and performing musicians, composers, producers and engineers. This month we thought we'd diverge from our usual mondo-tech ad and take a minute to thank some of these individuals — as well as everyone else who has helped put us on the map (albeit up in the Northwest corner of the map, a few miles north of Seattle). Being more interested in spending money on improving our products than on pricey photographers, we just sent some of our notable users each a cheap Polaroid* camera and told them to fire away at their Mackie CR-1604 installation. The results were mixed. But then, we're a mixer company.



1ST ANNUAL

# THANKS TO THOSE WHO HAVE BOUGHT OUR PRODUCTS & HELPED PUT US ON THE MAP.

**FRANK HELLER** • Engineer, Producer, Programmer & Keyboardist • **RECENT PROJECTS:** Used his Mackie CR-1604 to record & mix Brian Slawson's "*Bach Beat*" for Sony Classical Records; mixed "*Mama Said Knock You Out*" LL Cool J for Def Jam; mixed "*All I Need is You*" for BMG/Zoo; producing Movement EX, an LA-based rap group for Columbia Records • **PAST ACCOMPLISHMENTS:** zillions of 7", 12", EP & album mixes & production for groups including Madonna, New Kids O/T Block, OMD, The Spinners, , Earth Wind & Fire, Jimmy Cliff, , Force MD's, Was Not Was, Pointer Sisters, New Order, Fat Boys, Heaven 17, Steve Winwood, Quincy Jones (filmtrack), Debbie Harry, 808 State, New Edition & many more.

**PAT MASTELOTTO** • Drummer, Percussionist, Programmer • **RECENT PROJECTS:** Rembrandts, Jude Cole, Peter Kingsbery, Richard Page, Keedy, percussion on upcoming Sugercubes and work on Voice of the Beehive and Big Car singles • **PAST ACCOMPLISHMENTS:** Drummer for Mr. Mister; session discography includes XTC ("*Oranges & Lemons*"), Too Much Joy ("*Cereal Killers*"), The Truth ("*Weapons of Love*"), Vanity ("*Skin on Skin*"), Danny Wilde ("*The Boyfriend*"), Jack Wagner ("All I Need"), Patti Labelle ("*Winner in You*"), Pointer Sisters ("*Contact*"), Hall & Oates ("*Change of Season*"), Eddie Money ("*Take Me Home*"), Shandi ("*Shandi*"), Al Jarreau ("High Crime"), Cock Robbin, Knight & Des Barres, Danny O'Keefe, Robyn Hitchcock & the Egyptians, Martika and Nick Gilder.

**MICHAEL WATTS** • Synthesist • **ON-GOING PROJECTS:** *The Young Riders* for MGM/ABC TV, *Harry & the Hendersons for Fox Television,* • **RECENT PROJECTS:** Soundtrack for upcoming Disney Productions television movie "*Still Not Quite Human*,", re-recording of all music for Tomorrowland section of Disneyland, "*The Seduction of Travis County*" (CBS TV Movie of the Week), *The Vidiots* (NBC pilot), "*The Usual Suspects*" (an audiophile album for Sheffield Labs), "*Karaoki Music*" for Pioneer LaserDisc/Weberworks • **PAST ACCOMPLISHMENTS:** Session work on "*The Rookie*" for Clint Eastwood/Warner Bros., "*The Two Jakes*" for Paramount Pictures, "*The Jetson's Movie*" for Hanna Barbera/MCA/Universal, *The New Lassie* for Fox TV, "*Major Dad*" for MCA/Universal, "*The Fisherman's Wife*" album with Jodie Foster for Windham Hill, Van Dyke Park's new Warner Bros. Records album "*Tokyo Rose*," *The Tonite Show* with k.d. Lang.

A photo provided by Mark after we ran out of cameras.

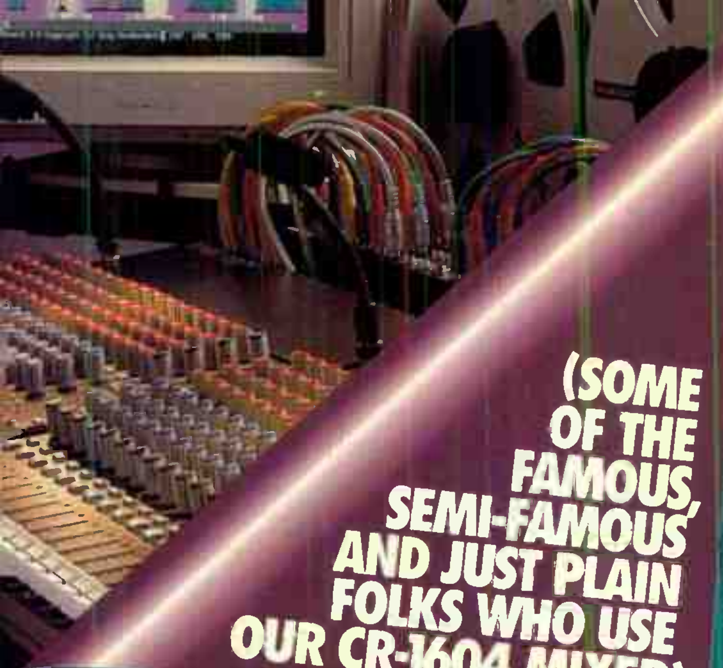
**MARK PORTMANN** • Composer, Keyboardist, Producer • **RECENT PROJECTS:** Releases from GRP acts including The Rippingtons' "*Curves Ahead*" album & tour, Nelson Rangell ("*In Every Moment*"), Carl Anderson ("*Fantasy Hotel*") and co-producer for new dance/pop artist Miralles; currently running 24-track recording studio as producer and songwriter • **PAST ACCOMPLISHMENTS:** Performing with the Rippington's "*Killimanjaro,*" "*Tour in Paradise,*" and "*Welcome to the St. James Club*" tours, GRP Christmas Collection II, Coors TV commercial, recorded and performed with David Benoit, Phil Perry, Syreeta Wright, Gary Herbig, Paquito D'Riviera; traveled and performed in over 20 countries.

**STEVE McELYEA** • Keyboardist, Producer, Arranger • **ON-GOING:** 24-tk. recording studio in Atlanta, IL, uses CR-1604 on various album and demo projects as well as audio for video productions. Runs two CR-1604's on the road as his house desk • **PAST ACCOMPLISHMENTS:** Toured with Ronnie Milsap (5 yrs), the Imperials (5 yrs), performed with Amy Grant, Reba McIntire, Merle Haggard, Ricky Skaggs, The Judds, the late Stevie Ray Vaughn and on *Austin City Limits, Phil Donahue,* The Nashville Network, *Mike Douglas* and the 1986 Grammy Award Show.

*Polaroid® is a registered trademark of the Polaroid Corporation.
Cheap is an unregistered trademark of people who buy brands of mixers other than Mackie.

Definitely *not* a Polaroid: Bob provided this photo of himself, wife Lola and son Kenny.

**BOB BLANK** • Studio Owner, Engineer, Producer • **RECENT PROJECTS:** SBK recording artist Phoebe Snow, produced by Phil Ramone, national TV campaigns for Yugo, Tyco Toys, ESPN, produced soundtrack for Miramax Pictures' "*Eversmile*" with Daniel Day Lewis • **PAST ACCOMPLISHMENTS:** Twenty Platinum and Gold Records for engineering, three Grammy nominations for engineering, three #1 chart records as producer, engineered over 500 chart records & over 100 Top-40 records including "*Push, Push in the Bush*" & "*I Got My Mind Made Up.*" Formerly owned Blank Tapes, NYC for 14 years; opened Blank Tapes, CT in 1987.



# (SOME OF THE FAMOUS, SEMI-FAMOUS AND JUST PLAIN FOLKS WHO USE OUR CR-1604 MIXER)

## CHARLIE BISHARAT

• Electric, MIDI and acoustic Violinist • **RECENT PROJECTS:** Albums: "*The Best of Shadowfax*" for Windham Hill, Chuck Greenberg's "*From a Blue Plant*" for Gold Castle Records. Soundtracks: "*Midnight Rider*" for Universal Studios, "*Hi Honey, I'm Dead*" for Fox Television. Commercial work for Intel, Hertz, Buick, Universal Studios. Yanni 1991 U.S. Tour **•PAST ACCOMPLISHMENTS:** Violinist for Shadowfax from 1985 to 1990. Kitaro 1990 World Tour. Awarded Grammy for Best New Age Performance for "*Folksongs For A Nuclear Village.*" Contributions to albums by Tracy Chapman, Jane's Addiction, Stunz & Farah, Kitaro, Scott Cossu. Soundtrack work includes "*Godfather III*", "*Kindergarten Cop*", "*Air America*", "*Russia House*", "*Joe Vs. The Volcano*", "*War of the Roses*", "*Ghostbusters*", "*Beaches*", *Murder She Wrote* for CBS Television, *B.L. Stryker* for ABC Television. Commerical work for Mercedes, LaBatt's, Oldsmobile, Anheuser Busch/Budweiser, AT&T.

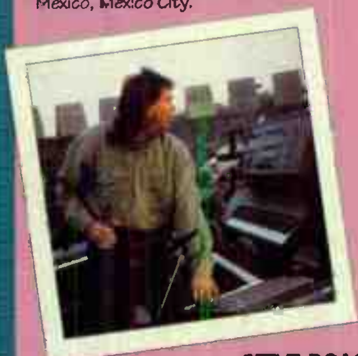
## TOM MGRDICHIAN

• Keyboardist/Programmer, Composer, Producer • **RECENT PROJECTS:** *Into the Night with Rick Dees* (Par 3 Productions), *Into the Night* (ABC TV Productions), *Studio 59* (ABC TV Productions) • **PAST ACCOMPLISHMENTS:** Used Mackie CR-1604 while playing with Air Supply, Andy Summers, 5 Star, Seals & Crofts, Richie Havens, Debbie Gibson, Ronnie Milsap, Englebert Humperdink, Maxi Priest, Marilyn McCoo, Riff, Mickey Gilley, The Osmond Boys, Thelma Houston and many others. Composed, produced and arranged musical cues as well as served as session keyboardist/programmer for: "*Die Hard 2/Die Harder*" (20th Century Fox), "*Millenium*" (20th Century Fox), *Totally Hidden Videos* (Fox Television), *Night Court* (Warner Bros./NBC), *Life Goes On* (Warner Bros.), "*Friday the 13th Part 8*" (Paramount), "*The Cellar*" (Cannon), "*The Fisher King*" (Tri Star), *Rick Dees Weekly Top 40 Show* (KIIS FM) plus commercial work for Nissan, Standard Brands, Dupont and others.

## CORY LERIOS & JOHN D'ANDREA

• Composers for movie and television scores • **RECENT PROJECTS:** Musical scores for Universal Picture's "*Child's Play III*", ABC Movie of the Week "*The Entertainer*" with Bob Newhart and Linda Gray, twenty two new episodes of *Baywatch* • **PAST ACCOMPLISHMENTS: John D'Andrea**: Scored and written songs for eight features films and numerous television projects including composition of four songs for "*Dirty Dancing*" and arrangement of its Academy Award-winning "*I've Had The Time Of My Life.*" Has been associated as composer/producer with twenty-five gold and platinum albums by artists ranging from Belinda Carlisle to Air Supply. **Cory Lerios**: As founding member and principal songwriter for Pablo Cruise, wrote numerous hit songs including "*Love Will Find A Way*" and "*Whatcha Gonna Do.*" More recently has worked on albums with Whitney Houston and Kenny G., and written songs for artists such as The Neville Bros., George Benson, Melba Moore and Santana. Scored and wrote the themes for ABC Television's *Max Headroom* and *O'hara.*

Steve & CR-1604 in a volcanic crater on the campus of the University of Mexico, Mexico City.

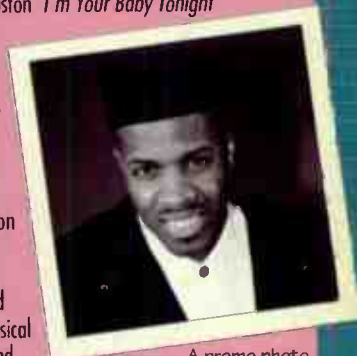


## STEVE ROACH

• Composer, Synthesist, Producer • **RECENT PROJECTS:** Double compact disc "*World's Edge*" just released on Fortuna Records; recording collaboration with artists in Germany, Spain, Mexico and U.S. in progress; upcoming European concert tour in 1992 • **PAST ACCOMPLISHMENTS:** Ten solo releases since 1981, including the groundbreaking "*Dreamtime Return*"; numerous collaborations including "*The Leaving Time*" with Michael Shrieve, and "*Strata*" with synthesist Robert Rich which hit the Top 10 on *Billboard*'s Adult Alternative chart in 1991, one of the few electronic artists performing live consistently for over ten years, Roach's engagements in the past year alone have taken him from concert halls in the U.S., Canada and Europe to lava caves in the Canary Islands and volcanic craters in Mexico.

## MICHAEL BEARDEN

• Keyboardist, Producer, Composer • **RECENT PROJECTS:** Keyboards for Whitney Houston "*I'm Your Baby Tonight*" World Tour; writing and producing for After 7, Vertical Hold and Jocelyn Brown **•PAST ACCOMPLISHMENTS:** Performed and/or recorded with Whitney Houston, Chaka Kahn, Patti Austin, Jonathan Butler, James Ingraham, Yoko Ono, Will Downing, Phylicia Rashad. Member of Grammy-nominated jazz-fusion group Special EFX for 4 years. Scored several PBS programs, produced acts for GRP Records including Nelson Rangell and Omar Hakim (Grammy nomination). Musical director for *Jazz Explosion* tour house band which included Freddie Hubbard, Noel Pointer, Stanley Turrentine and Angela Bofill.



A promo photo provided by Michael. (We temporarily ran out of Polaroid cameras.)

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# The Soundcheck List

## Avoid getting soundcheck-mated when you're working on the road

**BY GRAHAM THORNTON**

As the sound system engineer on a wide variety of musical acts, ranging from Kiss to The Church to a recently-revamped Donny Osmond, I've been lucky enough to get involved in many different segments of the musical spectrum. However, whether it's the volatile funk-metal of Jane's Addiction or the bouncy synth-pop of Howard Jones, there's always one binding technical neccessity that unites all live-acts — the soundcheck. Every band, regardless of genre, needs to go through this long, sometimes arduous rite of passage in order to take to the stage and present their compositions to a multitude of adoring fans (and a handful of sound-hungry critics).

Ideally, what I listen for in a concert is exaggerated studio sound. The solos are more dynamic. The drums hit harder. And the vocalist sounds more natural, uninflected. But if the system is not operating to it's maximum potential, we're going to hear it, and so is the audience. That's why I've set up this seven-step method to a better soundcheck: to catch the glitches before the band takes the stage, and the audience takes their seats.

✓ **Step 1: EQing the System.** Before the actual soundcheck process begins, there's a great deal of prep work to be done. While the crew on stage are setting up mics, monitors and the band's equipment, I'm at the controls, EQing or "tuning" the system.

The house system, on a typical night, supplied by Delicate Productions, is a powerhouse of Yamaha PM 3000 and Midas XL3 sound consoles, Crest and Carver amplifiers and Martin F2 speakers. In order to start at the same place every day, I like to run pink noise through the system. It sounds like a 747 coming in for a landing, but it's really just a full-spectrum noise in the system that gets picked up from a mic and sent through a real time analyzer (RTA). An RTA is a visual display of thirty frequencies (in one-third octave increments), usually on a bank of LEDs, which enables you to see which frequencies are lacking or sticking out. By comparing the flat signal that goes into the system with the distortion caused by the room, you can use a third-octave graphic to "flatten" the display on the RTA.

Next, I play a CD in a dbx player, preferably something of high quality such as Peter Gabriel or Sting. If you play the same track over and over throughout a tour, you'll get familiar with it and be able to make changes accordingly. While the song is playing, I'll walk around the venue with a level meter and check the coverage at various points. Sometimes, especially at outdoor concerts, I'll use an SPL (sound pressure level) meter to make sure we contain the sound within the perimeter and in front of the PA to see if we have to re-aim cabinets or shift individual speakers. We try to provide smooth coverage to every seat in the house, rather than blast out spectators up close.

✓ **Step 2: Prepare the Console.** While the band's equipment is being set-up, this is a good time to set up the board. I plan out the routing, subgroups, VCAs, channels and AMS reverb effects. This information can be saved by marking out the board and putting tape on the settings you've made. Later, before I completely finish the soundcheck, I'll take a portable recorder and describe every channel or chart the board on paper.

✓ **Step 3: The Line Check.** One of the most important steps in a soundcheck involves checking the continuity of each line. Nobody likes a burst of droning feedback disrupting the melody of a song. Once all the instruments are set up and the mics are in place, each line is tested individually by someone talking into or scratching each mic. Keep an ear open for any crackles and buzzes that could potentially sabotage an otherwise perfect production. If everything's working fine, you can concentrate on the sound, but if there's a problem, you're thinking about that repair.

✓ **Step 4: Drum Check With Technician.** Since the actual drummer's soundcheck could end up being a very time-consuming process (depending on the drum kit), I have one of the crew go through the drums just so I can mark initial settings. It's hard to EQ drums without the actual drummer, but if you're able to get an idea of the input levels and gate operation, it'll save you time during the band's soundcheck.

✓ **Step 5: The Soundcheck.** The band's drummer comes on-stage first. It would be pointless to have the entire band come out if all they're going to do is stand there for the next twenty minutes. The more time the drummer has to tune his kit, and the better they sound acoustically, the easier it will be to amplify them. If you're left with a ring or overtone it's very difficult to remove it all with EQ, so time spent here is well spent.

When the rest of the band comes out, I begin by checking the bass.

Photo bt Eric Landsberg

## HOW'S IT HANGING?

Even if your sound is flying high, wait until the band is playing a song to solo up the singer or back-up vocalist.

Basically (or bass-ically), most people use two channels for the bass, one a mic, the other a DI, and these are mixed together to get the desired sound. Since the bass and the drums have an important relationship on-stage, it's a good idea to check the two together at this point. When the guitarist arrives I have him play a variety of rhythm and lead. It's important that each musician play different types of music relative to what he'll play in the show. I usually have them all go through a loud song, a ballad and anything else they believe needs special attention.

After the drums, the next instrument to check is the keyboard.

Keyboards are perhaps the easiest instrument to deal with, although level changes between patches can be a problem. A slight difference in level when they're programmed in the studio or at home can translate into big changes in a powerful PA system, so we use compressors to keep them under control.

When you're checking the vocals, it's very hard to have the band members sing by themselves in the abscence of music. Wait until the band is playing a song to solo up the singer or back-up vocalist. Through the monitors, the vocalist can hear the band and himself, while through the PA, you can work on the vocal by itself.

✓ **Step 6: Charting the Board** I've made up board charts that I find helpful in saving information. The charts consist of a drawing of ten channels per page on which I mark the settings of every pot and switch on each channel of the board that I've used. This allows me to reset the board accurately if it's been used by another band between my soundcheck and the show.

I used this technique to good effect during last summer's Lollapalooza Tour, which had seven acts sharing two Yamaha PM3000s. Each band's engineer was able to start every show with the notes he'd made at the soundcheck from the first venue.

✓ **Step 7: Zero Out the Board** After you've made all your settings and notes, the board is sometimes completely reset for the support band. In that case, if there's no time for a second soundcheck, I try to assign the opening-act channels to the headline-act channels. You just trust that the levels and rough EQ are basically the same for both acts, then hope it all works. EQ

**Graham Thornton** *is a system engineer for Delicate Productions in Camarillo, CA. Recently, he applied his talents to the summer's most successful tour, The Lollapalooza Festival, featuring Jane's Addiction, Living Colour, Siouxsie and the Banshees, Ice-T, Nine Inch Nails and The Rollins Band.*



# Keeping Afloat With Sync

## The Time Is Right for Multimedia Computer Systems, But the Timing Among All Those Signals Is Still Off

BY PAUL D. LEHRMAN

Syncing MIDI to tape is easy. Just record a sync tone on your audio reel or videocassette and tell your sequencer, "Follow that stripe!" If the tape is a little slow or fast, no big deal, the sequencer will be a little slow or fast too. No matter how far off the speed is, the sequencer's pitch always remains the same, so all the MIDI stuff stays in tune.

Our ears don't find slight changes in tempo nearly as offensive as slight changes in pitch, so the process is pretty forgiving.

But now we have systems that let us combine MIDI tracks with digital audio recorded on a hard disk on the same desktop computer, and keeping these in sync is giving a lot of us some brand-new headaches.

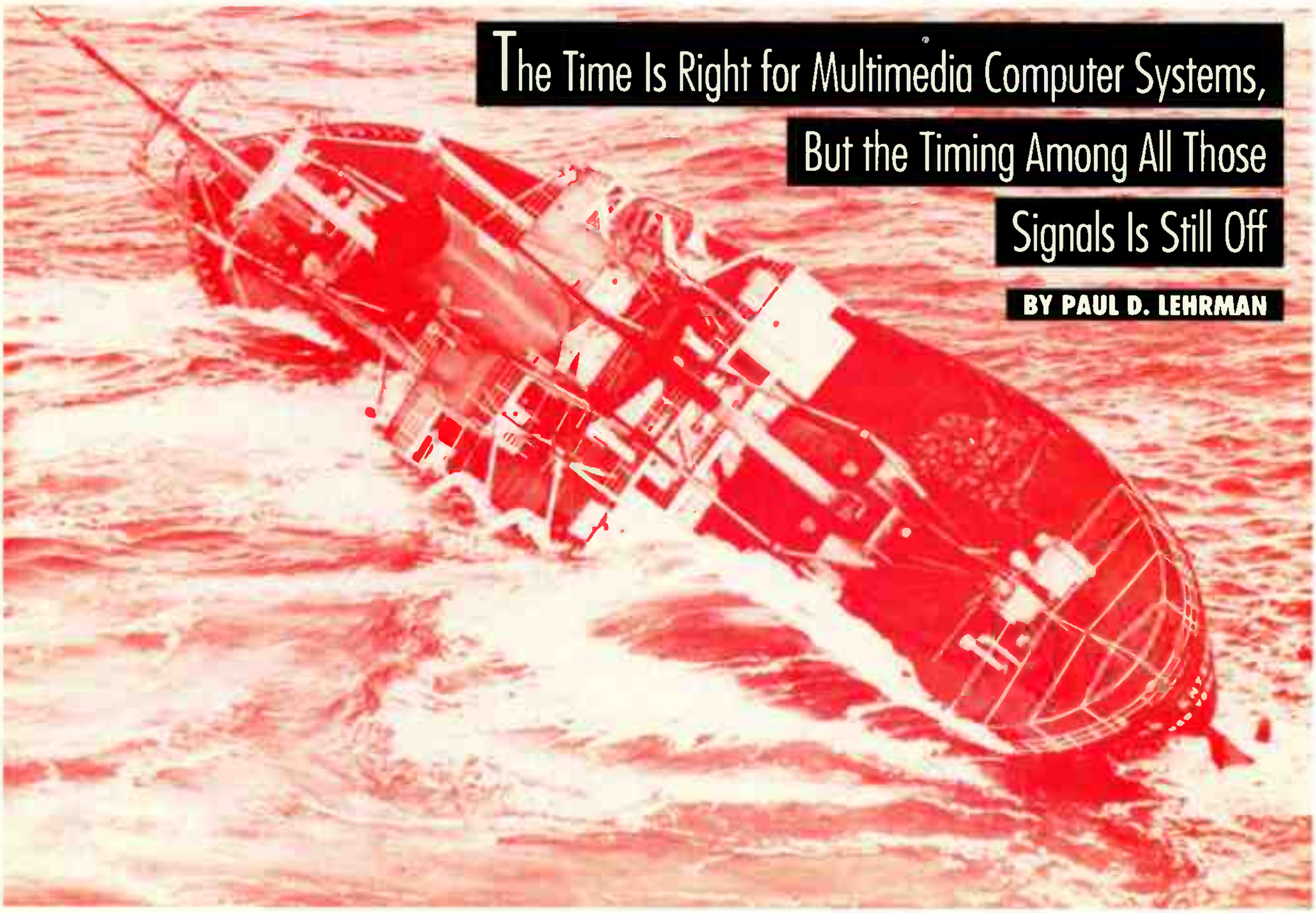
Think of a sampler that's used to store an entire musical phrase, which is triggered from a sequencer. Regardless of how fast the sequencer is going, the sampler is going to play that phrase at the same speed and pitch every time. If the sequencer happens to be playing back at a slightly different speed than it was when the sample was recorded, the sampled phrase will be out of time, ending either too soon or too late.

With hard-disk recording and MIDI sequencing this problem is potentially much worse. In many systems, a hard-disk track, which can be many seconds or even minutes long, gets a single trigger from its associated MIDI sequencer that tells it when to start. After that it's on its own, timing-wise. If the sequence plays back at the wrong speed, that rockin' sax solo in the second verse could end up barging in several beats early.

### OUT OF SEQUENCE

Why would a sequence not play back at the right speed? For one thing, different sequencers offer different tempo resolutions. One sequencer may let you specify tempos to the nearest 1/100th of a beat per minute (bpm), while another only lets you specify whole number tempos. If you record a sequence in the first at 118.51 bpm and then play it back in the second, the tempo will be 119. You may not hear any difference, but if there's a hard-disk track playing along with it, two minutes after the track starts it will be a beat behind.

A more common cause is that sequencers' and computers' internal



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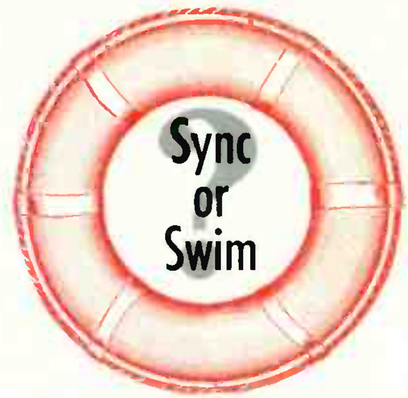


clocks are far from perfect. If you record an audio-plus-MIDI track based on an internal clock, and then play it back locked to an external clock like timecode, the MIDI may play slightly faster or slower, and the audio will drift.

There's also the fact that many sequencers assume that SMPTE timecode is coming in at exactly 30 frames per second (fps), but real SMPTE, especially if it's locked to a video source, counts only 29.97 frames in each second. Sequencer (and hardware) designers are just now beginning to get the message about this discrepancy, but there are plenty of disasters still waiting to happen. If you have a sequencer that doesn't know it's supposed to reference itself to a 29.97 fps clock, and you record an audio-plus-MIDI sequence using its internal 30 fps-based clock, and then play it back locked to videotape, the MIDI portion will be 0.1 percent slow. After two minutes, the audio will be ahead by 120 milliseconds.

### HARD DISK DECISIONS

So what to do? A simple solution is to keep hard-disk tracks short, breaking them up into individual events if necessary, so that timing problems don't have a chance to occur. A more flexible fix is to have a hard-disk system that, instead of relying on a single trigger to start playing, syncs *continuously* to the internal clock or incoming timecode, the way that analog tape decks in video post-production houses operate. But this can give rise to a new problem: If the clock speed changes, as it might, for example, when switching from internal to external sync, the *pitch* of the audio will change, but the MIDI pitch won't, and so, while the tracks will be in sync, they'll now be out of *tune*.

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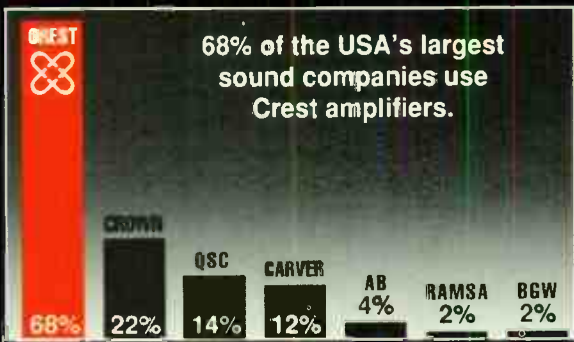

Totals are over 100%; some respondants mentioned more than one brand.

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**A more flexible fix is to have a hard-disk system that syncs *continuously* to the internal clock or incoming timecode**



Another potential danger of continuous sync is if the hard-disk system is designed to follow changes in tape speed too closely. Both SMPTE and MIDI Time Code (MTC) have some unavoidable "jitter" — slight, momentary speed changes — and if the audio follows this religiously, it can result in a "flutter" effect. While MIDI, as I said earlier, is immune to this, audio is very susceptible to it.

There is a best solution, and that is to lock *everything* in the studio — video and audio tape decks, computers, sequencers, hard-disk audio, even samplers — to a common, extremely accurate, timing source. Video houses routinely do this because they have to. You can't edit video unless every component in the room is locked to "house sync," a common sync source. House sync is commonly distributed in the form of a video signal, with color bars or with no picture at all (in which case it's called "black burst").

### DOING IT THE HARDWARE WAY

Hardware for generating black burst is becoming relatively inexpensive, and while hard-disk systems are available that lock to house sync, there are no sequencers yet that can follow it directly. Many sequencers can follow MIDI Time Code, of course, so if you have a box that generates MTC using house sync as a timebase, and you can lock your audio to the house sync and your MIDI to the MTC, you're in business. (This system has one drawback: There's no way to use it unless tape is running.)

What would solve all these problems would be a jack on the back of the computer that accepts a video signal and lets the computer base all of its musical calculations on the timing of that signal. Everything would always lock, and hardware kludges won't be necessary. (Of course, everything else in the studio, including audio tape recorders, would have to lock to the video signal as well.)

Smart computer makers should realize that it's not just MIDI and hard-disk audio that require this capability. As computers become more multimedia-oriented, the ability to sync their internal operation to real-world video signals will become more than a matter of convenience. It will be sync or swim. **EQ**

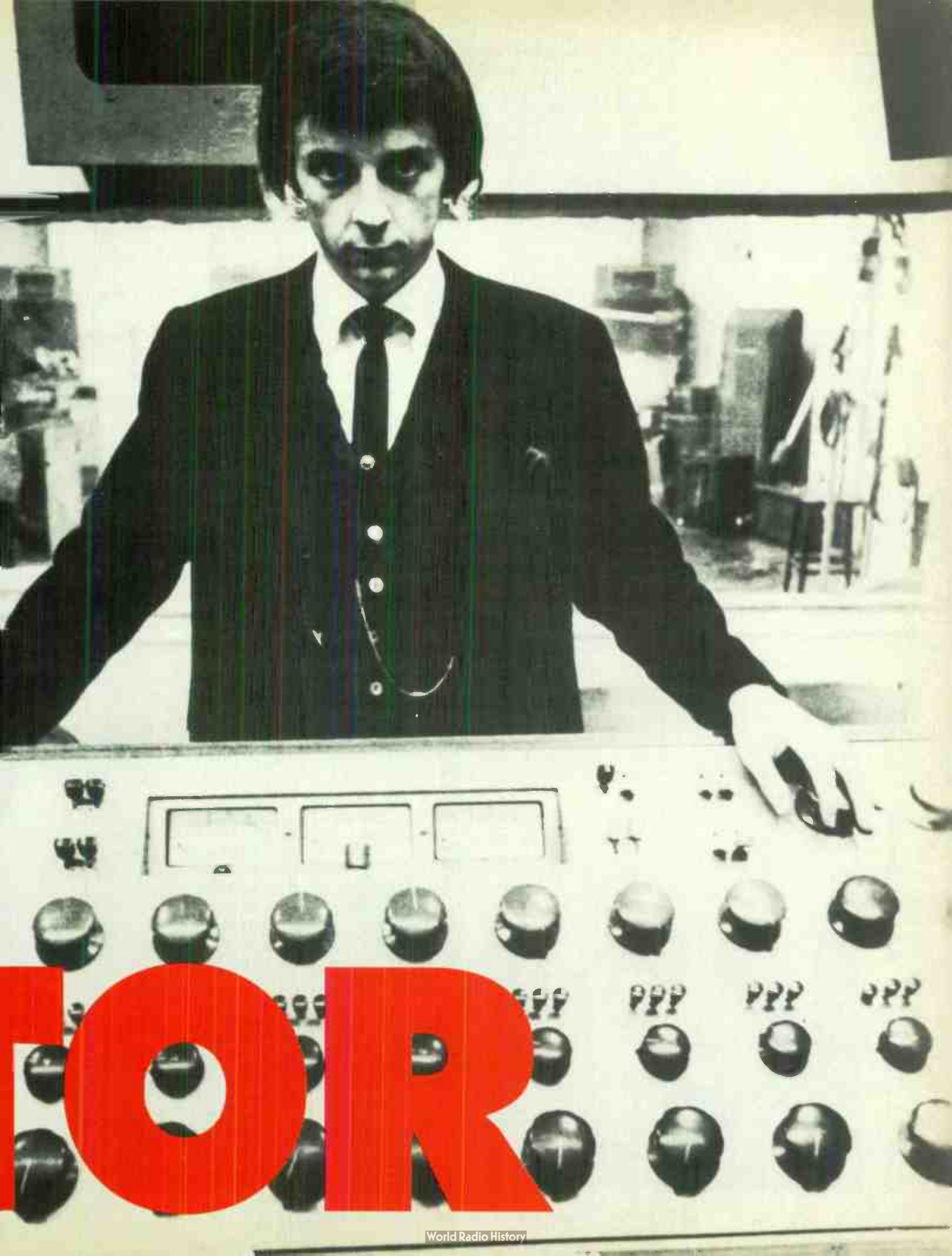
**An EQ Exclusive Memoir by The Man Who Engineered The Legendary Wall of Sound**

BY LARRY LEVINE

Phil Spector first came to Gold Star Studios in L.A. in the late Fifties to work with Stan Ross, the studio's owner and my cousin, on several projects, including the Teddy Bears and the Paris Sisters. Then, in 1962, he flew in from New York to record "He's a Rebel" in a big hurry because he was racing the release of Vicki Carr's version at Liberty Records. Stan was vacationing in Hawaii, though, and Phil was forced to use me instead. Later, with "Zip-a-Dee-Doo-Dah," I was to become his regular engineer.

Photo: Michael Ochs Archinves

# PHIL SPECT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I had met him earlier when he was working with Stan and I didn't like him at first. (Most people didn't.) I thought he had this aura about him that said "Spoiled Brat." But by this time I knew of his reputation and was in awe. I remember the "Zip" session well: We were setting up, trying to record for three or four hours and all of my pots were real hot — all my meters were pinning at the top and I knew I wasn't going to be able to record like that. Phil was really working us hard and I didn't know what to do to make him back off.

I finally decided that there was nothing else I could do, so I just turned all the pots fully off, all twelve of them. Phil looked at me like I was crazy and started screaming at me that he had just about had it and how could I do that. I told him that there was no way that I was going to get that on tape. I started bringing the pots back up one at a time, first the basses, then the drums, guitars and so on. I had eleven of the pots turned up and had mixed them pretty gently to where they were when Phil yells, "That's it! That's the sound. Let's record." I told him that I didn't have Billy Strange's mic open yet. (Billy was playing lead guitar.) Phil said, "Don't turn it on!" So we recorded it that way and the guitar sound on the tape is what bled through the other microphones. The room was small so you could hear it, but it didn't have the presence that a microphone can give.

It was my single greatest experience with a record and from there on in, I was Phil's engineer. I was so excited about "Zip-A-Dee-Doo-Dah" that I would take people into the control room and tell them, "If you tell me there's a chance, just a chance, that it won't make the Top 10 — I'll eat the tape right in front of you." They'd look at me like I was crazy because nobody could predict with that kind of certainty what would happen to a record after it was released, but I never did have to open that bottle of ketchup.

**LETTING THE GOOD TAPES ROLL**

The good times were the great times. If things were going good, Phil would do stand-up comedy in the control room. Sometimes he would do as much as twenty-five minutes' worth. He used to love to use me as the butt of his jokes, but I didn't mind because I knew that was the role I was playing.

The sessions were a lot of work, though, and you could never relax.

# RETRO-SPECTORED

*Phil Spector's Band, Studio Crew and Stars Recall the Heyday of the Wall*

*Compiled by Tony Savona*

**Phil Ramone, Producer/Engineer**

"Phil had his own way of doing things. The record industry was dominated by rules and he broke almost every one of them. At that time arrangers drove the business, but not with Phil. He would take the song from the arrangement and carry it to where it would work.

I hear his influence today, particularly in rap music with its layering, sampling and big orchestral drops. People aren't even conscious of where it came from. If Phil got back into recording today, he could walk all over the dance market."

**Hal Blaine, Drummer**

"We would always play Friday nights at Gold Star Studios. It was always a tremendous party. The biggest joke was that there was a sign on the door that said "Closed Session" — meaning that no one should enter — but Phil grabbed everybody that poked their head in and made them play a percussive instrument. Everybody from musicians like Brian Wilson of

Photo courtesy of Hal Blaine

1. Joy Migliori
2. Lou Blockburn
3. Steve Douglas
4. Al Delory
5. Don Randi
6. Roy Caston
7. Phil Spector
8. Irving Rubins
9. Hol Bloine
10. Jock Nitzsche
11. Ray Pohlmon
12. Lyle Ritz
13. Sonny Bono

When you played things back you had to play them back as a record. It was very tiring and there were times when we would argue over it. The Christmas Album was particularly exhausting. After we finished, I had to tell Phil I couldn't do it anymore. Our separation lasted about three months. Phil was working back in New York and Sonny Bono came to me and said that Phil really wanted to work with me again. By that time I missed making great records, so I was happy to have him come back to Gold Star. Of course, Phil and I worked well together, but there was something special about Gold Star itself.

When I first arrived there in 1952, it wasn't anything like the place that Phil Spector would help make famous. I started hanging around there in the evenings because there wasn't much else happening. At first I just did gopher stuff, but when business increased and they needed someone else to work the board, Stan and Dave Gold gave me my big chance.

It was great fun in those days (even at $25 a week). We were doing basically demos at that point. No one was doing sessions except larger studios. We were working with songwriters or publishers who would come into the studio, sit at the piano, and have their song recorded onto disc. You would count to three, point at the guy in the studio and drop the stylus onto the disk. And that was it.

We had one tape recorder then, a brush and a wire recorder, and a console that Dave had built. The console was very basic — six or seven inputs and an amplifier. It wasn't state-of-the-art because at that time there was no art. And although Gold Star Studios would soon become famous for its incredible echo, at this time we had a long way to go.

In fact, to get our first echo we used the hallway that ran from the main reception room to the main control room. The singer would stand in the doorway of the studio — half in and half out — and sing. We'd put a microphone at the other end of the hall and try to capture some semblance of echo. The technique didn't work too well and it was a pain trying to keep everybody from walking down the hallway. For our next attempt, we built this little attic above the hallway. It was a 3 feet by 30 feet by 3-feet-deep box. We painted it with slate to harden up the surface to make it reflect better. It's amazing we didn't kill ourselves with the fumes. Unfortunately, despite our efforts, it sounded like shit and were back to the hallway.

**THE BOOM-BOOM ROOM**

In 1960, shortly before Phil arrived, we really started living! Dave redesigned the studio and we had gotten an Ampex 350 mono deck and a 351 two-track. We also had a new console built by Dave and Bill Putnam that had twelve inputs and three outputs. This console had a divider, so as you increased the echo send, you decreased the send to the fader, which was the way we thought echo should be. Echo was to make things sound farther away, not to create the effect.

Dave had designed two chambers that were angled, so that they were two geometric shapes that would fit side by side. These were the chambers that Phil Spector would make famous. Inside the chambers, we put a little eight-inch speaker and a RCA flat ribbon microphone to capture the echo and bring it into the board.

Ray Avery/Michael Ochs Archives

**Jack Nitzsche, Darlene Love, and Phil Spector.**

the Beach Boys to other producers who came down to see what Phil was doing. Everybody would play. There was magic when we worked with Phil.

He would pack us all in the studio for the Wall of Sound effect. General recording sessions have four rhythm tracks, but Phil used two or three basses alone. He had four or five percussionists, although I was the only one who played drums, and four or five guitarists as well as horn and string sections. It was always huge and always a party.

I worked with him years later on a Leonard Cohen album. It was still a big party and Phil still made any onlookers, which in this case included Cher, Bob Dylan and poet Alan Ginsberg, perform."

**Sonny Bono, Percussion West Coast Promotion for Philles Records**

"Phil was the Wagner of pop music — larger than life, grandiose. He brought a whole new way of thinking to recording. Rather than thinking about an entire album, Phil focused 100 percent of his energy into a single song. He wrote it and produced it. Then he crammed the studio full of musicians and pushed all the sound up into the distortion range.

The first time I saw it, I thought, What the hell is going on here. He was breaking every rule. And to my knowledge, Phil invented the technique called bouncing. He got two two-track recording machines and bounced the various instruments back and forth, which enabled him to squeeze even more sound onto the tape. Nobody had ever done that before. Call it clever, call it genius, it was unique."

**Darlene Love, Singer**

"Although I had worked with other producers, Phil was the first one who had produced a number one hit. He was different in that most producers wouldn't teach you the song that you were performing —


Ray Avery/Michael Ochs Archives

The Beach Boys, Phil Spector and The Righteous Brothers' Bobby Hatfield (right).

that I had originally put on.

I didn't admit it was an accident, though Phil knew how it had happened. But he liked what he heard. He listened and understood exactly how it had happened. He had that kind of knack.

Everything that was created during those sessions was done by Phil. Nobody else should get any credit for the "wall of sound." (The phrase was coined by some disc jockey or reviewer in response to Phil's revolutionary rhythm section.)

**SCALING THE WALL**

Here's how we built the wall: We'd fill up the studio with twenty to twenty-five people. The room was very small and there would hardly be room enough for the musicians to move around one another. The room was filled with musicians playing their

most of the songs were sung by the songwriter — but Phil would sit down with you and tell you exactly what he wanted. He would sit with me at the piano at Gold Star Studios and go through the entire song with me.

The big Wall of Sound sessions were great — you could actually see Phil's genius. He only had three tracks to work with, whereas today you have 64. He put an amazing amount on those three tracks. Plus he had an incredible ear. Today you can push a button and hear only the instrument that you want to hear. Phil couldn't do that, but he didn't have to. He could hear anything — everything — he was looking for, no matter how many instruments were on the tape. He could pick out anything, from a guitar lick to a single note.

The Wall of Sound, however, is more than just musical sound, it is the sound of all the people who created it. It is built with all of our hearts and souls."

**Steve Douglas, Saxophonist**

"Phil's sessions were special because all he recorded were hits. The music was different, unique. Phil was the first guy to go into the studio with an unfinished tune. Generally it was all written out before you went into the session, but Phil would start with the written arrangement then move off of it. He would do two things at once; he would work the sound and the arrangement.

Everything was loud. Phil would keep the playback volume shatteringly loud. I remember the recording of "Zip-A-Dee-Doo-Dah," Billy Strange's guitar was so loud that Phil turned off his microphone — so what you hear on the single is the sound that bled through everybody else's microphone.

Phil had a very calculated, workmanlike approach to a record. He put a lot of care into every song. He would work the band hard, some grumbled but most didn't — we all wanted to work with him. There was a high level of comraderie."

The control room at Gold Star was perhaps the best-sounding room I've ever heard. We had these great Altec D3 monitors. Phil would always have the music blasting. I didn't mind one bit. Whenever anybody new came into the room, Phil would want to impress them so he'd turn it up yet another notch. I knew all they'd be able to hear would be a mess of noise because nobody can get their ears set to that level so quickly. (Around the studio, Stan and Dave and the other guys started to refer to their levels by a "Larry." Most guys would go half a "Larry." You could never get past 3/4 of a "Larry.")

**PHIL-ANTHROPIST**

Phil always knew what he wanted and, although he never put his hands on the board, he could have worked it if he wanted to. I never lied to him about what I was doing because he always knew what was happening. I saw guys try to bullshit him but it never worked. Phil trusted me on the console so that he was free to do whatever he was going to do — but he was always aware.

I remember during the recording of "And Then He Kissed Me," I was trying to get more level because Phil wanted things louder and louder. What I decided to do was to send the original to two tracks and double up the level of sound. Later, I would just keep one of the tracks and use the other track for voices and strings. What had happened was that the echo also got doubled. It was okay when I had both tracks, but when I erased the other track I was left with essentially twice the echo hearts out and we'd fill every available space on that tape with it. There were other aspects to the sound, which were very much part of the physical structure of Gold Star itself, but this was the basic building block of Phil's "wall."

I used to feel sorry for the guitar players because they played the entire session. Phil would start off by saying "Let me hear the guitars." The acoustic guitars were always first. He'd have them play off of a lead sheet and have them try different things. Then he would add the piano, then the basses, then the horns, and then back to the guitars. That's the way he'd build what he heard in his head, and only Phil knew what that was.

There were times that I had guys with microphones that weren't even plugged in. I'd tell Phil that he could

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*Suggested retail price for the Model 24/24 Megas Studio console. Other prices will vary somewhat based on specific configuration and features.

send them home because I didn't have any more open inputs. But Phil would say, "This is the sound. They stay." And he was right. There was a presence on the tape. Whether it was body absorption or whatever, they were a part of what was making the sound happen. That's something that today's electronic musician can learn. The human body adds something to the sound.

**MAKING TRACKS**

We'd always experiment, just trying out different things to see what worked. I never used EQ until I needed it to bring out an effect. What we had then was very basic anyway. We had curves of 3, 5 and 10 kHz, and they could be moved up or down in increments of 3 dB. We used balance, or a different mic, more than we did EQ.

The thing Phil hated most was that Motown was always able to get a certain sound out of their drums that he could never get. They even nailed the drums to the floor so they'd never lose it. In our room, the drum's sound changed according to what key the song was in. If the guitars were playing open strings, then there was more level coming out of the guitar itself, which meant that the mics were not as loud, which meant I didn't get as much leakage from the drums into the guitar mics. If they were playing with closed strings, the opposite would happen. Everything always sounded different, but I think that was great because there were more different sounds in the wall than just the one you could identify.

Part of Phil's genius is his ability to surround the vocals with this incredible rhythm track and not let them be swallowed by it. We only had three tracks: the orchestra went on the first, the strings on another and the vocals on the third. It would be easy to lose the vocals inside the wall of sound, but Phil never did. I never worked with anyone else who had that ability.

He had a unique way of working. When I had to mix something down, Phil would leave the room and I'd be left alone to mix. When I had something I liked, I'd call Phil back and he would critique it. Then he'd leave and I'd start mixing again. There were two great things about working this way. One was I didn't have him looking over my shoulder the whole time and the other was there was always a fresh ear listening.

**FOR THE RECORD**

When "River Deep, Mountain High" failed to do well in the States (ca. 1966), Phil took it pretty hard. He sort of rubbed people the wrong way and I know a lot of people who wanted him to fall flat. I mean, I love Phil — but our relationship didn't start out as love. Everyone who worked with Phil liked him, but there were plenty of people in the industry who were thrilled at seeing his string of hits end. It left all of us with a bad taste. I was leaving to go to A&M Records at the the start of the following year anyway, so with everything put together I guess Phil decided it was time to close up shop.

We worked together after that. We did some songs with the Checkmates and the Ronettes over at A&M Records. Later, in 1979, we recorded with the Ramones back at Gold Star. That was the last original recording we ever did together. (Gold Star closed in 1984, after 33 and a third years of operation.) It wasn't a good time for Phil and we'd be there all night and it wasn't any fun — for Phil, the Ramones or for me. Phil and I got into a big dragged out argument one night and the following evening I had a heart attack. I was smoking and drinking a lot of coffee at the time, so I don't really know if the fight had anything to do with the attack, but I know Phil felt guilty even though we never spoke of it.

After that, he really withdrew into himself and I'm really sorry for any part I played in making him pull further into himself than he already had. We didn't speak again until 1986, when he began inviting me and some other people to his house to watch championship fights on his big screen TV, and I didn't really work with him again until the making of the "Back to Mono" box set.

**BACK ON (ONE) TRACK**

It was great to work with him again — better than old times. He was a different person and it was great to get back to the music. I couldn't believe how great it sounded. It was almost as exciting as when we originally recorded them.

Mono's sound content held up amazingly; those singles have as much depth as if they were recorded in stereo. The one basic advantage of mono is that it maintains the integrity

*continued on page 83*

**Bill Medley, Singer, member of the Righteous Brothers**

"What most impressed me about Phil is that he knew what a hit song was. I once asked him why he owned a record company. He replied 'Because I know what the public wants and I can give it to them.'

Phil could also recognize a hit voice — he was the first to utilize my baritone abilities. It is always good to work with a producer who knows what he wants, except when what he wants goes against your style. Phil knew what we could do and used our singing abilities to their fullest potential.

He was the genius of that time — and he still has a lot to say."

**Stan Ross, Engineer/ Co-Owner of Gold Star Studios**

"Gold Star was a special place. My partner, Dave Gold, designed the equipment, the studio and the echo chambers. The control room didn't have sit-back seats, they were more like four foot stools. They made for better communication between artist and engineer.

We built the chambers in 1950 for the purpose of giving a nice echo on horns and other instruments. They became very successful and, although our studios were relatively small, the echo made the rooms sound three times larger than their actual size.

The electronic echo that is used today doesn't compare to acoustic echo."

**Shelly Yakus, Engineer**

"Every record that Phil Spector produced was more than a record — it was an event. These events changed the perception of what records should be. They heightened the awareness of what you could do with a record.

Other people in the industry heard what he was doing and knew that whatever they did after that would have to be just as good or better. He made the industry go further than it had and raised the standard for making records.

Phil showed us we could do it."


BY STANLEY CLARKE

# LA'S TOP SCORER

***STANLEY CLARKE STRIPES GOLD IN FILMS AND WITH HIS NEW SLAMM DUNK LABEL AT HIS LOS ANGELES THREE-CAR RECORDING STUDIO***


When I see a picture moving it's natural for me to come up with some music to accompany it. Without music, pictures seem emotionally dead. Even if it's already emotional, I can make it more so. Or, if I want, I can change the mood entirely. I can make a scene of someone walking through a grocery store, behind the spaghetti, scarier than hell. That's probably why I get such a rush from film scoring that nothing else can match. If I want to, I can even turn that spaghetti into some crazy menacing alien — just with sound.

I remember one of my first jobs, an episode I did for *Tales From the Crypt* on television. When the finished film came in to my studio and I ran it, there was nothing scary about it at all. I thought to myself, "These guys have got to be joking. They think this is scary?" But then when I put on that initial low drone, it came to life. Film scoring is like creating life, which is kind of scary in itself. It's also a great feeling of power. Regardless of what the director has done, you have the power to redirect it for the better, or for the worse. Music is more than just the final touch to a movie. You can make or break it.

Photos by Ed Colver

**Scoring** *Boyz N the Hood* was probably the best project of my career. In addition to my natural love of film scoring, I had a lot of personal interest in the subject matter. There was, and is, an important message in it, one I wanted to make sure was conveyed properly to the audience. I especially didn't want it to appear, or sound, stereotypical. With all those car chases and gun blasts it very easily could have gone that way. I needed to draw out the real emotions, the really powerful emotions, that were in the script.

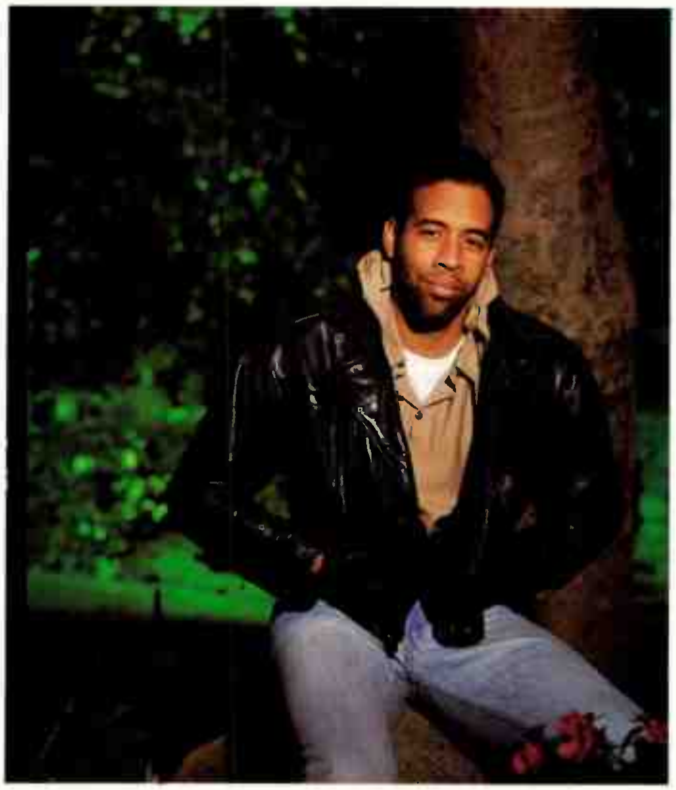
Basically, the film has two main musical styles, funky modern street music and what I call "almost orchestral" music. It was the juxtaposition of these two very different sounds that created the special, dramatic and intense feeling on the soundtrack. It made people go, "Wow! — What is it?" It created a real nice vibe, one that came through even to me when I went to a theater to see it. That sound made the movie a whole lot deeper.

I do virtually all of my basic film scoring in my project studio. When I moved out to L.A. a few years ago, I bought a house with a large three-car garage in back, which was destined to become my studio. During those days I was still pretty much just a bass player. My equipment consisted of numbers of basses and numbers of bass amps. Not much more. Now when I look at my studio, it's wild to see how far I've come in so short a time.

Like many people in this business I started out with a basic Tascam system for demos — a 12-channel mixing board and an 8-track tape recorder. It was great for demos, but being a perfectionist, I started to want more. And more. I ended up filling up that garage quickly with an Otari MTR-90 and a TAC Scorpion board (I'm thinking about upgrading to a Mozart) and tons and tons of outboard gear. Just racks of that kind of stuff. So much I can't even keep track of it. There's no use listing it all here. If you want to know what I've got, just go to your pro audio dealer and see what he's got. I've got the same. I especially like the Aphex gear, especially their Dominator. The Drawmer gates are great too. I have quite a few noise gates so that our recordings come out very clean. And those new Lexicon reverb units are as good as anything I've ever heard.

The film scoring work was what made me really organize my studio. I realized that the room could no longer be run and set up as loose as it had been. I had to make it as precise an instrument as possible. I had to master it like I had mastered my bass. Suddenly, everything had to work together. It also all needed to be locked to picture, so I installed TV monitors and synchronizers.

Basically, I use the garage for writing the score and because of this I ran into another problem: I like to write alone. I needed an engineer but



I didn't want one hanging around all the time. So the studio then had to be redesigned, or, rather, custom-designed, so that stupid me could work the thing.

As for how I work, I pretty much follow the same routine, at least on film-scoring jobs. I get a tape that's striped with SMPTE. I have everything synchronized and locked up and the master 3/4-inch tape essentially runs the show. I just start writing and "print" to Performer on my Mac. (I also use Cue.) All the MIDI stuff, everything, starts there. For the live stuff, I make notes and record that later. After a couple of days I bring in the engineer to start laying stuff down and to start transferring my sequences over to tape. If we need to, we're ready to go to a commercial facility to add orchestration or to mix.

*Boyz N the Hood* had a very large string section. I started by playing the basic string parts into the computer, just as a demo, so I could hear what the stuff sounded like, and to serve as a guide for when I would write it down later on paper for the orchestra. (I use a gigantic sound stage where John Williams does his stuff.) Still all the basic ideas and all the preproduction happened at home.

**I know** now that building the home studio was one of the best moves of my life. I encourage every musician to have one, absolutely. When I first came to Los Angeles, if I just wanted to change a bass line or overdub an electric bass, just tune-up stuff, I'd have to get in my car, book a studio, pay an incredible amount of money and have the tapes sent ahead of me. You're looking at a grand (at least) right there. Whereas with the project studio, especially if you're lucky enough to have 24 tracks or just the basic guts of a commercial studio, you can go in there any hour or the day or night. My studio has saved me tons and tons of money and time.

Of course, putting it all together over the last five years has been an education in itself. I feel like I've gone through med school. When I step back and look at all I've done, all the detail work, all the manuals I've read, I'm amazed at what I've learned. Just trying to figure all this stuff out is a full-time job. The up side, though, is that I now feel more professional as a musician. I know I could leave L.A. for any other major music center, like New York or London or Paris or Tokyo, and I'd be able to walk into a studio and deal with what's there to have to deal with. And that's a nice feeling. It's a vital part of the creative process that today's musician has to understand.

**All that** hard work has also prepared me for being a record company exec — my most recent career move. I'd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*Boyz N the Hood* used a very large string section. Still, all the basic ideas and all the preproduction happened at home.



wanted to start my own record company since I was 30, but, back then, when I told people this dream they just said, "Yeah, yeah, Stanley. Sure." This time, though, they came to me.

One day my wife told me the execs at CBS were coming over to the house. My first thought was, "Well, they're finally going to let me go — but they're being polite about it." I've been with CBS for forever, so some of these guys I've known since they were in the mailroom. When they came over, though, they said, "Stanley, how'd you like to have your own label?"

It was my son who came up with the name Slamm Dunk. (I've been a big basketball fan and player all my life, so it was a natural.) The label's first releases will be sometime this year. Right now (November '91) I'm still getting people inked. I expect, though, that I'll release three to four artists a year. And I'm not going to produce them all. I'm going to be like a real record company exec. I wasn't, in other words, looking for more producing to do. Frankly, it's not my favorite activity. But I did want the opportunity to give a chance to some instrumentalists, who will be the focus of the label.

I owe my own career to a small label. In fact, I owe it to one guy, Nat Weiss, who got me started as a soloist. He came into a club in New York one night and said, "Hey, man, you have charisma or something." And I said, "Yeah, great. Tell me more." Then he asked me if I wanted to make my own record, which at that time was not usual for a bass player. I really had the bass player attitude then, too — to sit back and wait for the guitar player to call you. I made the record and it did very well for a first LP, and everything else just continued from that.

So now I've progressed to a point where I sit around waiting for producers to call me. After playing bass live, though, film scoring is my favorite thing to do. In fact, I've really got to watch out that it doesn't take over the rest of my life and career. It's such a rush to see those pictures come to life. It's definitely an addictive experience and I'm just going to have to be careful. EQ

**Stanley Clarke** *has released thirteen solo albums in addition to his many collaborations.*

# THE COMPLETE COMPRESSOR/LIMITER

## How to get that "in your face" sound without squashing the sonics

IF SPECIAL EFFECTS DEVICES LIVE IN THE AUDIO PROCESSOR PENTHOUSE, THEN COMPRESSOR/limiters are downstairs patrolling the lobby. "Comp/limiters" are the blue-collar workers of audio processing — the sound security squad that keeps audio signals under control. We all need and use them, but somehow always take them for granted. In this issue, however, *EQ* gives credit where credit is due by spotlighting the overworked, unheralded compressor/limiter.

Years ago, simply riding a fader was the best way to keep levels under control. But as technology evolved and multiple mics and inputs became the norm, gain riding became a much more difficult task, sending a lot of recording engineers packing and off to accounting school.

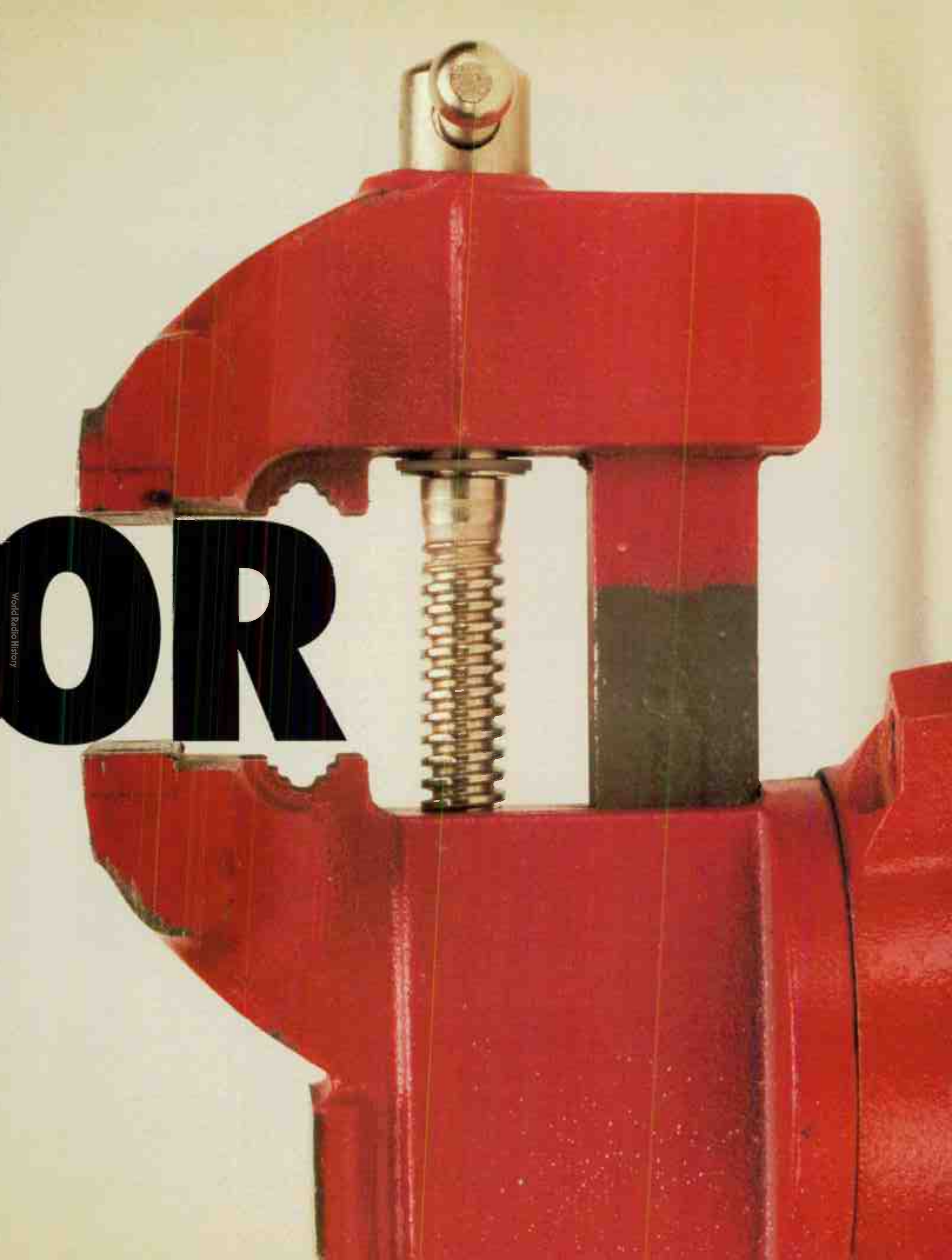
Today we have compressor/limiters to tame the dynamic range of the musical signal. Set properly, a compressor will squeeze the music's dynamics into a "useable" range — above the noise floor and below the point of distortion. (Remember: dynamic range is the difference between the loudest and softest volume levels.) However, overcompression of an input signal, unless done for special effect, can squeeze the life out of your music's output signal, making it sound thin and flat. Overcompressing can generate the equivalent of music played at one steady level — music without heart and soul.

Used judiciously, comp/limiters can be invaluable tools. (Incidentally, "limiting" can quickly be defined as heavy compression ratios of 10:1 and above.) First, though, each of us has to understand why we need one, when to use one and how to use one. That's where the accompanying articles come in. Use of a comp/limiter may vary depending on which division in the audio profession you're in. Live sound mixers, broadcasters, recording engineers and musicians all use one, but the reason for its use, the device's position in the signal chain and the degree of compression will likely vary.

Many of the models included in the accompanying comparison chart differ, some greatly, in terms of features and price. That's why it's important to know your needs before you go shopping. For some, one 2-channel deluxe model will fulfill all needs. For others, especially those in the live and studio sector, four or five basic single-channel models are far more effective. Furthermore, several of the models include noise gates, expanders, filters and more. Don't immediately discount these because of a desire to keep your life simple. Think about growing into such models. You may not need the extra functions today, but what about in six months? It might be worth spending a few added bucks on a multiple-function unit now rather than having to buy a whole new unit later.

The chart lists a group of basic features/functions for which to look. There are many more, including I/O metering, MIDI, sidechain monitoring and de-essing level, to name a few. However, start by studying the following pages. Then call or write to the manufacturers — tell 'em *EQ* sent you — for additional info. Until next time, keep an eye on those output levels.

*—Hector La Torre*


OR

# HOW TO BUY

## COMPRESSOR LIMITER

BY KEVIN MCCANN ELDER

Rane DC24

The first step in buying any piece of equipment — and especially a compressor/limiter — is to identify your needs. This helps you establish the proper operating level for the device. Different applications and environments require the compressor to work at different nominal operating signal levels or voltages, with different requirements as far as the maximum signal levels that can be accommodated.

When recording to analog tape, the compressor is used to prevent tape saturation. In digital recording, the compressor is employed to prevent digital clipping, which occurs when the digital system runs out of bits. In radio and television broadcasting, compressors are used as limiters to prevent overmodulation of the carrier signal. In live sound, compressors are used to hold the level of an individual channel or a subgroup. They can also be used to limit the overall signal going to the power amplifiers in order to prevent speaker-destroying square waves that result from power amp clipping. Some power amplifiers and digital electronic crossovers have built-in soft limiters to prevent this type of clipping. If you have amplifiers or crossovers with these features, then you should be using your compressor in a dedicated application.

Boss Pro CL-50

**Less Than Zero.** Most budget studio gear operates at a nominal signal level of -10 dBV or 0.316 volts, which is the 0 VU (volume unit) on the tape machine. On the larger multitrack tape machines, the 0 VU calibration is +4 dBm or 1.23 volts. The tape level zero on most cassette tape machines is generally 165 nanowebers, but on professional open-reel machines tape level zero can be 185 to 200 nanowebers. In live sound reinforcement, the average operating level of each channel is usually calibrated to 0 dBV or 1 volt (RMS); however, the output of the mixer itself may be averaging 3 to 5 volts with peaks going well above 10 volts. Generally compressor/limiters designed to operate at the -10 dBV nominal level are less expensive. They may even have a maximum output capability of +10 dBV, but they may not be able to drive a long signal line in a sound reinforcement application.

Sony MU-L021

**Balanced I/O You's.** In a small project studio, where the recording equipment is generally in the same room as the studio, it may not be necessary to have balanced signal inputs and outputs because of the short cable runs. In a live sound application, if the compressor is used in a channel or sub insert patch, it may not need a balanced input or output. However, if the project studio is in an environment that's subject to radio frequency (RF) or electro-magnetic interference (EMI), then balanced inputs and outputs are required. Balanced outputs are also important in those studio and live situations where the compressor is going to drive a long signal line. Some can drive long lines such as 50 meters of balanced signal line better than others. The specification to look for is Input Common Mode Rejection (CMR) that's rated in dB. Most studio applications require a CMR of 80 or 100 dB.

**Stereo or Mono?** In the studio, if one compressor is assigned to a track that

## KNOW YOUR LIMITATIONS

**A compressed glossary**

*Attack*. The time, in milliseconds, it takes for gain reduction to begin once the input signal rises above the threshold.

*Compression ratio*. A comparison, in decibels, between changes in volume in the input (numerator) and output (denominator) signals, once the input signal level rises above the threshold.

*Compressor/limiters*. Rack gear that decreases the dynamic range of program material by making the loudest moments softer. Those parts of an input signal that rise above a user-defined volume threshold are reduced by a pre-defined ratio. Therefore, these devices output a "scale model" of an input signal where scale is defined by the ratio.

*Limiting*. Limiting is an extreme amount of compression. Any compression ratio of 10:1 or greater is limiting. When limiting, the threshold is usually set comparatively high to stop only extreme peaks in program material.

*Release*. The speed, in milliseconds, with which gain reduction of the output signal ceases once the input signal falls below the threshold.

*Threshold*. The level, measured in volts or decibels, above which gain reduction begins.

—*James Mason*

**TCD-D10 Pro II**

**Portable DAT Recorder**

The TCD-D10 Pro II is the smallest professional DAT recorder from Sony. Yet, while it weighs only 4 lbs. 7 oz., the TCD-D10 Pro II is no lightweight when it comes to performance.

Built to withstand the rigorous demands of field work, the TCD-D10 Pro II allows you to stay in the digital domain from acquisition to studio. It also features absolute time (A-time) recording/playback which places a continuous time code on tape, allowing you to locate recorded segments faster and more easily.

Plus, A-time is compatible with SMPTE time code DAT recorders like our PCM-7000 Series. There's even an improved digital I/O and LCD multi-display with a combination of safety/warning indicators to help insure fail-safe operation. And when combined with one of Sony's high quality microphones, you're fully equipped to meet the most demanding challenges in the field.

**KEY SPECIFICATIONS**

DYNAMIC RANGE: MORE THAN 85 dB

FREQUENCY RESPONSE: 20 Hz-22 kHz

THD: 0.06%

I/O: ANALOG—MIC/LINE BAL, DIGITAL—AES/EBU

ACCESSORIES: BATTERY (X2), CHARGER, REMOTE, AC SUPPLY, CASE

▼ SHOWN WITH OPTIONAL SONY ECM-MS5 STEREO MICROPHONE.


**DPS-D7**

**Digital Hyper Delay**

If you want to take your creativity in exciting new directions, Sony's DPS-D7 Digital Hyper Delay is the way to go.

Featuring seven sophisticated algorithms, there's virtually no limit to the number of unique and complex digital delay effects you can create. The DPS-D7 incorporates an 18-bit oversampling A/D and 1-bit HDLC D/A converter system with digital filters for excellent linearity, ultra low noise and wide dynamic range.

Second generation LSI's allow for high-speed 32-bit digital signal processing. And with its large graphic display and help button for assistance on any function, the DPS-D7 is always simple to use.

**KEY SPECIFICATIONS**

DYNAMIC RANGE: MORE THAN 94 dB

FREQUENCY RESPONSE: 10 Hz-22 kHz

THD: <0.0035%

ANALOG I/O: BALANCED +4 dBs (+24 dBs MAX.), UNBALANCED −10 dBs (+10 dBs MAX.)

MEMORY CAPACITY: 100 FACTORY PRESETS, 256 USER LOCATIONS

**DPS-R7**

**Digital Reverb**

If you want to add even more power and versatility to your audio system, Sony's DPS-R7 is right on the money.

Offering two discreet channels of advanced digital reverb effects, the DPS-R7 is an invaluable tool for the audio professional. As with the DPS-D7, the DPS-R7 employs HDLC D/A converters for superior sound reproduction as well as high-speed 32-bit digital signal processing, which deliver sophisticated, multiple reverb effects.

It also includes 100 factory presets as well as 256 memory locations for your own presets. In addition, the DPS-R7 features an ingenious "data wheel" and large graphic display for easy operation.

**KEY SPECIFICATIONS**

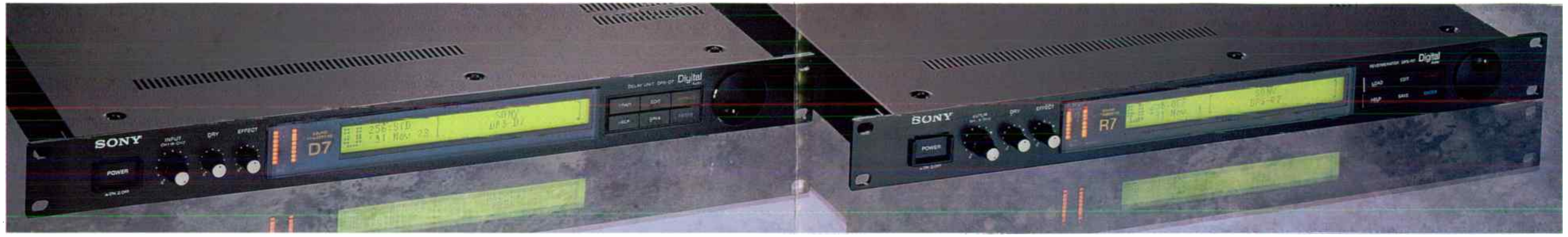
DYNAMIC RANGE: MORE THAN 90 dB

FREQUENCY RESPONSE: 10 Hz-18 kHz

THD: <0.004%

ANALOG I/O: BALANCED +4 dBs (+24 dBs MAX.), UNBALANCED −10 dBs (+10 dBs MAX).

MAXIMUM SIMULTANEOUS EFFECTS (TEN): 4 PRE EFFECTS (2 PER CHANNEL), 2 REVERB (1 PER CHANNEL), 4 POST EFFECTS (2 PER CHANNEL)







**PCM-2700**

**Studio DAT Recorder**

Taking advantage of Sony's latest innovations in digital technology, the PCM-2700 is the first affordable professional 4-head DAT recorder.

Featuring Sony's advanced HDLC (High Density Linear Converter™) System, the PCM-2700 delivers superior sound quality. The PCM-2700 also employs a 4-motor direct drive transport to insure tape stability, accuracy and reliability. Its 4-head design provides off-tape monitoring to verify your recordings.

There's even a duration adjustable digital auto fader for fade-in and fade-out times as well as an A-time search function for rapid access to any recorded A-time location—all giving you the utmost in professional performance.

**KEY SPECIFICATIONS**

SIGNAL TO NOISE RATIO: MORE THAN 90 dB

FREQUENCY RESPONSE: 20 Hz-22 kHz

THD: <0.045%

ANALOG I/O: +4 dBs (+24 dBs MAX.) ADJUSTABLE

PARALLEL REMOTE: TTL COMPATIBLE, D-SUB 37

**PCM-2300**

**Studio DAT Recorder**

As Sony's most affordable professional DAT recorder, the PCM-2300 is ideally suited for a wide variety of applications where high quality recording and playback are necessary.

Like the PCM-2700, the PCM-2300 incorporates the latest conversion devices—1-bit delta Σ A/D converter and HDLC 1-bit D/A converter—for outstanding sound quality. The PCM-2300 also incorporates a sophisticated 3-motor transport design for solid reliability. And in 32kHz long-play mode, it delivers twice the normal recording and playback time—a full four hours.

Plus, its analog and digital I/O's provide a wide range of flexible interfacing possiblities.

**KEY SPECIFICATIONS**

SIGNAL TO NOISE RATIO: MORE THAN 86 dB

FREQUENCY RESPONSE: 20 Hz-20 kHz

THD: <0.05%

ANALOG I/O: +4 dBs (+24 dBs MAX.) ADJUSTABLE

SUPPLIED ACCESSORIES: WIRED/WIRELESS REMOTE, 19" RACK MOUNT, POWER AND REMOTE CABLES

**CDP-2700**

**Compact Disc Player**

The CDP-2700 compact disc player delivers a multitude of professional features for a very compact price.

Like all Pro Standard equipment, the CDP-2700 is rugged and reliable while delivering superb sound quality. Ideal for on-air applications in radio broadcasting and sound sweetening in video post, the CDP-2700 includes important features such as variable speed playback, fader stop/start control from a mixing console and an auto cue function for instant start.

And because its digital output conforms to both the AES/EBU and IEC-958 formats, the CDP-2700 directly interfaces with other professional equipment for flexible system expandability.

**KEY SPECIFICATIONS**

DYNAMIC RANGE: MORE THAN 110 dB

CROSSTALK: 100 dB

THD: 0.04%

VARI-SPEED RANGE: ±12.7% (0.1% STEPS)

D/A CONVERSION: DUAL 18-BIT 8X OVERSAMPLING

# Everything About Our New Digital Audio Equipment Is Designed To Sound Better Than Ever.

SONY

# Including The Price.

## Introducing The New Pro Standard Digital Audio Series From Sony

If you've been looking to expand your capabilities with a new professional DAT recorder, CD player or signal processor, Sony quality is now within your reach.

Featuring some of the most highly advanced, great sounding digital audio products in the world, the Pro Standard™ Series covers a wide spectrum of the needs of today's audio professional. Making it easier than ever for studio recording, post-production and radio broadcasting to benefit from the advantages of digital.

Because the Pro Standard Series is so advanced, it allows you to push the limits of creativity like never before. And because it's designed and built for professional use, you can always count on it to stand up to your toughest demands.

The Pro Standard Series. It's just what you'd expect from The Leader in Digital Audio.™ Sony.

PRO/STANDARD DIGITAL AUDIO SERIES

## HDLC. One Of The Great Technologies Behind Our Great Sound.

Representing a breakthrough in high-performance D/A conversion, Sony's new High Density Linear Converter™ (HDLC) System defines a new level of performance in sound quality.

More specifically, HDLC, a single-bit technology, recreates an analog wave form from digital data by representing it as a rapid series of pulses, overcoming differential non-linear distortion and glitch-induced distortion as well as zero-cross distortion inherent in previous technologies.

The HDLC system also employs Sony's proven 45-bit noise shaping digital filter, circumventing base band dynamic range deterioration and assuring a high degree of oversampling accuracy. All of which translates into the purest, most accurate sound reproduction. All from Sony.

To experience the power of HDLC, along with the other advanced technologies behind our new digital audio products, call 1-800-635-SONY, ext. 914, for the Pro Standard dealer nearest you.



SONY

always contains the same voice or instrument, stereo operation is not necessary. In a live sound application, the same thing applies if the compressor is dedicated to a channel patch for, let's say, the kick drum or any other voice or instrument assignment. If stereo operation is necessary, then the linked operation is necessary so that a burst of level above threshold in one channel won't cause the stereo image to shift abruptly. Linked operation entails using a matrix or sum of the left and right signals to trigger the common threshold setting.

**Variable Attack/Release.** A variable attack control may not be necessary when a compressor is used to hold the level of a channel or subgroup in a live application, or to prevent tape overload or clipping of an A/D input. Models offering program-dependent control of the attack timer will suffice in these cases. If, on the other hand, you'll be using it to create a desired effect, you will need control over the attack time. But be careful, attack settings that are intended for special effects are very critical, and improper adjustments can brutalize the sound.

ART MDC 2001

Variable release capability may not be as critical in a dedicated application. However, when used as a special effect signal processor, a variable release is another parameter you may want to control. If the release is too fast, the threshold is triggered with each half cycle swing of the bass line and if the release is too slow some short term pianissimo detail loss may occur.

Generally, for most compression applications, attack times that are moderately fast and release times that are moderately slow yield the best results.

**Side Chains That Bind.** A side chain capability is a feature that can be very effective in a dedicated compression application. It can be used to cause one signal to be compressed by the presence of another signal above threshold. This is used in commercial broadcasting when an announcer does a voiceover with a musical bed. This application is called "ducking" and can also be quite effective

*continued on page 66*

## HEAVY HITTERS

**You've envied the latest and greatest sounds created in the top studios too long — here's how your project studio can sound as good**

With the right compressor/limiter, you can greatly improve the processing power of even a budget console. The growing list of compressor/limiter manufacturers that offer their advanced circuits for rack applications includes: Solid State Logic, Neve, Focusrite, GML, API and Drawmer. If you're looking to take that step up, and are willing to foot the bill, any of these models will lead your sound up that stairway to project studio heaven.

Can't afford an SSL? Logic FX G384 ($3950) from Solid State Logic comes fully equipped with balanced inputs, balanced outputs, side chaining and variable attack and release times.

COMPRESSOR LIMITER

Other helpful options such as Auto Fade and Gain Make-Up add to the allure of this powerful piece of equipment. It's the same compressor you'll find on any of the G Series consoles. If the Neve sound is what you're after, consider their 33609/C ($3850), a compressor/limiter that offers side chaining, two-channel operation and balanced inputs and outputs. This model can be switched into stereo mode or linked together with other Neve modules for multi-channel applications.

Then there is the compressor/limiter that George Massenburg made famous, the 8900 from GML ($5000). This dual-channel unit combines highly accurate log convertors, true RMS detectors and other modern analog processing techniques in a stable "feed-forward" topology. It features side chaining, linked operation and variable release and attack times.

In the stratosphere of high-end rack compressors, there are several other sonic treatments to consider. The ISA 130 from Focusrite ($2900) is a single-channel unit that features all the options you'd expect from Rupert Neve's console design, including balanced inputs, balanced outputs, three side chains and a proprietary VCA design. Drawmer's M500 ($2299) also serves as a significant sonic upgrade to any studio environment with its linked operation, variable release and attack times and two channels of operation. And API has enhanced its 525 into the 525b.

Are they worth the investment? Only you and your accountant can decide, but remember that many are available for rent in major markets. And any one of them can turn even the most modest studio environment into a platinum mine. —*Jon Varman*

# HOW TO USE

**BY JAMES MASON**


Klark Teknik DN504

Knowing when to use compression is one of the keys to getting a good sound. Here's a quick checklist of problems that you may be able to solve by reaching for your trusty comp/limiter.

•**The vocals are masking the band.** If a vocal performance is dynamic, the soft moments will be masked by the backing tracks unless the volume of the vocals are raised to the point where a soft voice is clearly audible. If this is done, then the danger is that, during loud passages, the vocal will mask the band or many of the musical instruments. Compression can be used to reduce the masking.


Peavey CDS2


Alesis 3630

•**The guitars sound uneven.** Bass players employ different techniques such as picking, plucking, slapping and popping strings, all of which may result in an unacceptably large dynamic range. Electric guitarists also employ a variety of techniques that may require compression to smooth out a performance. Switching from power chords to a clean rhythm sound may cause extreme volume changes. Here you'd use compression to smooth out the instruments' peaks.

•**The tape is oversaturated.** By controlling the levels of peaks, the overall program can be brought closer to optimum tape saturation level, thereby more effectively masking the noise floor.


Ashly CL52E


dbx 165A

•**The monitors are being overdriven.** A reduction of dynamics will protect speakers and other hardware. By not allowing a signal to rise above a preset level, compressors will prevent speakers from being damaged by sudden peaks in the overall program level.

*Knowing when to use compression is one of the keys to getting a good sound in the project or commercial studio.*

# SOME LIKE IT SMART.


Some like it hot. Fortunately, our new Intellifex™ effects processor has the brains to give you both.

Intellifex is a true 24 bit intelligent DSP effects processor with a 64 times oversampling "high purity" Sigma-Delta A/D converter. While most manufacturers use only one data converter, Intellifex uses three true 16 bit data converters, for maximum linearity. The 24 bit data path is maintained throughout the processor and memory circuits to maximize the dynamic range. In fact, Intellifex delivers a remarkable 104 dB dynamic range to put noise well beyond the audible level. And our new software programmed digital HUSH™ circuitry insures this absolute sonic integrity will not be compromised by typically noisy input signals.

This advanced signal path technology warms up your sound with stunningly beautiful eight voice chorusing, thick stereo delays, reverbs, pitch shifting and more that leaves the run of the mill, muddy sounding, overstuffed effects boxes cold. Our emphasis on professional quality rather than quantity led reviewer Craig Anderton to comment: "Rocktron's Intellifex doesn't try to be a jack of all trades, instead devoting all its resources to impeccable time-domain processing."

Take for instance our 8 voice chorusing. Intellifex uses all eight delay voices for an extremely lush sound, and provides each voice with independent control for delay, pan, voice level, rate and depth control. Or the extremely effective 4 voice pitch shifter that gives you four separate pitch shifts plus the original. Separate voice controls cover pitch shift, pan, voice level and delay over a three octave range.

Intellifex even knows when not to muscle in on your playing. Our Delay Ducking feature lowers the delay level when an input signal is present and then returns to full volume, allowing delays to repeat out, when you stop playing.

All this and more with an ease of programming that is clearly a case of mind over matter. So satisfy your need for higher intelligence and audition the amazing new Intellifex at your local Rocktron dealer today.

It may be the smartest choice you'll ever make.

ROCKTRON
STUDIO SERIES

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



CIRCLE 43 ON FREE INFO CARD

COMPRESSOR LIMITER

# FACT SHEET & COMPARISONS

| BRAND | MODEL # | PRICE | BAL. IN | BAL. OUT | # of CHAN. | S.C. | L.O. | V.A.T. | V.R.T. | FREE LIT. # |
|---|---|---|---|---|---|---|---|---|---|---|
| API | 525B | $1195 | • | • | Single | • | • | | • | 131 |
| ART | MDC 2001 | $499 | • | • | Dual | • | • | • | • | 101 |
| ARX Systems | Afterburner* | $599 | • | • | Dual | • | • | • | • | |
| Alesis | 3630 | $299 | | | Dual | • | • | • | • | 103 |
| Aphex | 9301 | $599 | • | • | Single | | • | | | |
| | 720 | $1350 | • | • | Dual | | • | | • | |
| | 320 | $1350 | • | • | Dual | | • | | | |
| Ashly | CL-100 | $170 | • | • | Single | • | • | • | • | 105 |
| | CL-52E | $500 | • | • | Dual | • | • | • | • | |
| BSS Audio | DPR-402 | $1400 | • | | Dual | • | • | • | • | 106 |
| Boss (Roland) | CL-50 | $240 | | | Single | • | • | • | • | 107 |
| CTI Audio (CAD) | CGM-2 Champ | $599 | • | • | Dual | • | • | | • | 108 |
| | 1002 | $150 | • | • | Dual | • | • | | • | |
| DBX**** | 903 | $449 | • | | Single | • | • | • | • | 109 |
| | 160XT | $459 | • | • | Single | • | • | • | • | |
| | 165A* | $999 | • | • | Single | • | • | • | • | |
| | 166 w/dual gate | $629 | • | | Dual | • | • | • | • | |
| DOD/DigiTech | DOD 866 | $300 | | | Dual | • | • | • | • | 110 |
| | AudioLogic 660 | $350 | • | • | Dual | • | • | • | • | |
| | AudioLogic 266 | $500 | • | • | Dual | • | • | • | • | |
| Drawmer | DL241 | $699 | • | | Dual | | • | • | • | 111 |
| | M500 | $2299 | • | • | Dual | • | • | • | • | |
| Focusrite | ISA 130 | $2900 | • | • | Single | • | • | • | • | 112 |
| Furman | LC-3A | $269 | • | • | Single | • | • | • | • | 111 |
| | LC-X** | $369 | • | • | Dual | • | • | • | • | |
| GML | 8900 | $5000 | | • | Dual | • | • | • | • | 114 |
| JBL/Soundcraft | LA-4 | $730 | • | • | Single | | • | • | • | |
| | 1176LN | $830 | • | • | Single | | • | • | • | |
| | 1178 | $1390 | • | • | Dual | | • | • | • | |
| | 7110 | $495 | • | • | Single | • | • | • | • | |
| Klark-Teknik | DN500 | $1195 | • | | Dual | • | • | • | • | 116 |
| | DN504 | $1350 | • | | Quad | • | • | • | • | |
| Neve | 33609/C | $3850 | • | • | Dual | • | • | • | • | 117 |
| Orban (AKG) | 412A | $525 | • | • | Single | | | • | • | 118 |
| | 422A | $680 | • | • | Dual | | | • | • | |
| | 414A | $800 | • | • | Single | | • | • | • | |
| | 424A | $1150 | • | • | Dual | | • | • | • | |

| BRAND | MODEL # | PRICE | BAL. IN | BAL. OUT | # of CHAN. | S.C. | L.O. | V.A.T. | V.R.T. | FREE LIT. # |
|---|---|---|---|---|---|---|---|---|---|---|
| Peavey/AMR | CDS-2 | $250 | • | | Dual | • | • | • | • | 119 |
| | CDS-2000 | $299 | • | • | Dual | • | • | • | • | |
| Rane | DC24 | $549 | • | • | Dual | • | • | • | • | 120 |
| RSP Techlogies | Model 300A | $379 | • | • | Single | • | | • | • | 121 |
| | Model 360 | $499 | • | • | Dual | • | | | | |
| | Model 2200 | $699 | • | • | Dual | • | | | | |
| Samson | Behringer | $480 | • | • | Dual | • | • | • | • | 132 |
| Solid State Logic | Logic FX G384 | $3950 | • | • | Dual | • | | • | • | 122 |
| Sony | MU-L021 | $799 | • | • | Dual | | • | • | • | 123 |
| SoundTech | ST200CL | $350 | • | • | Dual | | | • | • | 124 |
| Summit Audio*** | TLA-100A | $1400 | • | • | Single | • | • | • | • | 125 |
| | DCL-200 | $2700 | • | • | Dual | • | • | • | • | |
| Symetrix | 425** | $580 | • | • | Dual | • | • | | • | 126 |
| TubeTech*** | CL-1B | $1830 | • | • | Single | • | • | • | • | 127 |
| Valley | Dynamite 2** | $425 | • | • | Dual | | • | | • | 128 |
| | 440 | $799 | • | • | Single | | • | • | • | |
| Vestax | SL-201 | $440 | • | • | Dual | • | • | • | • | 129 |
| Yamaha | GC2020B Mark II | $395 | • | • | Dual | • | • | • | • | 130 |

S.C = Side Chains, L.O. = Linked Operation, V.A.T = Variable Attack Time, V.R.T.= Variable Release Time. * Compressor/Limiter/Enhancer ** Compressor/Limiter/Expander *** Tube Compressor **** VAT and VRT refer to manual front panel adjustments



CIRCLE 18 ON FREE INFO CARD

When your music starts out, it sounds something like ~~this~~ the way this looks. That's why, before you let even the least discriminating audience hear it, you might want to send it into a


BOSS SE-50 *Stereo* **Effects** Processor. **THERE'S** *nothing* **like this** on the market **for the** PRICE. *It* has 19 ***different*** built-in EFFECTS —*including* **Vocoder** and **Rotary**—eight *of which* you CAN use **at ONCE.** And *you* CAN process **two** SEPARATE **signals** INDEPENDENTLY. Which means **your SOUND** will *verge* on ***PROFOUND*** *by the time* it gets to your


CL-50 Compressor/Limiter. Use it to even out all the levels. You don't need to bother with a compressor, a noise gate, or a limiter, because the CL-50 does all three. So by the time your sound's done here, it's ready to head straight for a


GE-21 Graphic Equalizer, where you can futz with levels until your fingers go numb. Tweak a little more treble in, lose a little lower midrange. Get all 21 frequency bands to sound perfect, then spank it on into your

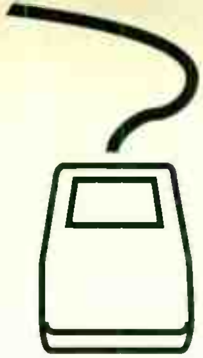

NS-50 Noise Suppressor. It'll clean up your sound without affecting the integrity of the original signal. So what you crank out is something you really want all those fans of yours to hear.

BOSS®

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

CIRCLE 60 ON FREE INFO CARD

# The Four Front



## A basic workshop on how to shop for what works

**BY CRAIG ANDERTON**

TO SOME PEOPLE, talking about computers verges on talking about religion. So let's cut through the evangelism: no matter which of the four main computer families you commit to — Amiga, Apple, Atari or IBM — you'll be able to find sequencers, sample editors and editor/librarians. Often, what you want to do other than music will determine which computer to buy.

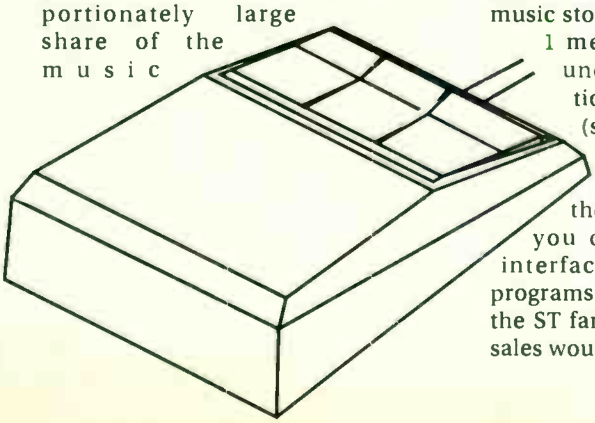
### COMMODORE AMIGA

The Amiga has the least amount of music software support (hard disk recording is particularly hard to find), but for video work it's an extremely cost-effective machine. Products such as digital titling and animation software, live action digitizers, and the extraordinary NewTek Video Toaster for special effects have earned the Amiga a substantial niche market.

The Amiga's a good bet for video production houses that need music capabilities, but for whom music is not the primary focus. It's also a great "home" computer — it's a knockout game machine and has enough "personal" software that you can use it for correspondence, finances, etc. Another strong point is that the Amiga was designed for multi-tasking from the ground up and offloads much of its work to custom chips, increasing its efficiency. Current models are the 500, 2000 and 3000 — the low, mid, and high-end computers respectively.

### APPLE MACINTOSH

The Macintosh has nowhere near the installed base of IBM PCs (IBM itself dwarfs all other computer companies combined), yet the Mac has a disproportionately large share of the music market. Also, Apple has urged developers to stick to the Mac's developer guidelines, resulting in programs where once you've learned one, you've (sort of) learned 'em all. While IBM's Mac-like Windows 3.0 operating system has taken away some of the Mac's thunder, Apple countered with new operating system software (System 7.0), which improves what was already considered the most user-friendly interface in the business.



The downside is cost. Until the recent introduction of a low cost line (the Mac Classic, lc, and si), the Mac family had a reputation of being unreasonably high-priced, as well as subject to arbitrary upgrade policies. Unfortunately, the popularly priced Classic doesn't really have the horsepower for most Mac music software.

### ATARI ST AND TT

Atari owns the European music market (and some of the European desktop publishing market as well). As a result, Atari has concentrated on its overseas markets much more than the US, which accounts for the computer's low stateside profile in areas other than music.

However, that may change. Rumor has it that Atari will launch an aggressive marketing campaign into music stores based on the 1040 STE, a 1 meg computer that lists for under $800 with high-resolution black-and-white monitor (street prices are often considerably less). It's easily upgraded to 4 megs, and the built-in MIDI ports mean you don't need an additional interface. There are thousands of programs available internationally for the ST family; presumably greater US sales would let loose more software on this side of the pond. Waiting in the wings: a 32 MHz/68030-based Atari, the TT, which is just starting to ship in this country.

Most of the types of programs available for the Mac are available for the Atari, including sample editors and hard disk recorders. Atari partisans claim that Atari software can exceed the performance and feel of Mac software, although Mac owners would probably argue with that.

### IBM PC

Big Blue *is* the computer market. Its market share and size make it a formidable company on any level, and thanks to cheap "clones" that beat IBM at its own game, the PC has become a worldwide standard. Business, professional and scientific software is abundant for the PC.

Music software support is not quite as plentiful as for the Mac, but all the basic programs (and some exotic ones) are currently available and the rate of development is picking up. What endears the PC to many users, however, is price. Peripheral and software markets are so competitive that there is a lot of price-cutting, and everything seems to cost less than for the Mac. The jury's still out on Windows, the PC's graphics interface. It requires serious computing power and many users think the Mac has a smoother interface anyway. Nonetheless, this development has overcome one of the PC's liabilities — the cryptic text-based operating system — and makes PCs much easier to use.

### GENERALIZATIONS

For pro music applications, the Mac reigns. The Amiga is strongest in video. In terms of cost-effectiveness, the Atari is the way to go. For the greatest amount of overall software support and reasonable prices, the PC shines. In some cases, your application will dictate your computer. One cost-cutting hint: buy a computer stripped and equip it with third-party peripherals. EQ

*Craig Anderton has been using computers since the late 70s, which may explain why his house looks like a floppy-disk museum.*

# TOTALLY TUBULAR

**Vintage and new units warm up even the funkiest digital sound.**

"Unforgettable, that's what you are. . . " This was a refrain we've all been hearing on the radio, from Natalie Cole's smash duet with her father Nat King Cole — the latter having laid down his vocal track in 1958. The superb quality of classic recordings such as this one still, fortunately for us, stand up today. But how did they achieve such quality with the primitive analog equipment of the day?

COMPRESSOR
LIMITER

Talent, of course. Also, most of the recordings were done live-to-stereo, thus eliminating the tape degeneration inherent to multiple overdubs. To avoid degeneration, three-track ATRs were sometimes


Yamaha GC2020B II

# HOW TO BUY

*continued from page 59*

when employed to prevent the guitar from stepping all over the vocalist in a heavy metal band.

**Soft Knee Bends.** Compressors offering soft knee compression are said to be warmer than those offering compression strictly based on the level of the threshold. In this mode, some small amount of compression is being applied to the signal at all times, but the ratio changes automatically with the signal level. Small signal levels have very little compression and as the signal level increases the amount of compression increases accordingly. Compression controls adjust the relationship between the signal level and the ratio of compression.

**Output Control.** If a comp/limiter is used as a dedicated compressor and is patched into the mixer via the "Pre" patch point of a channel, you probably won't need an output control. Since a channel level control is after the "Pre" patch point, it acts as the output level control of the compressor. If the compressor is dedicated to a submix and is patched into the submaster patch point (with most mixers this patch point is post the submaster slider), then increasing the submaster only succeeds in hitting the compressor harder. And the level of the signal, if already above threshold, ultimately remains the same. This is when you need an output level control.

These are the most common features available on generic compressors. There are, of course, signal processors that also feature noise gates, clippers, de-


essers, exciters, expanders and slope control. Just remember: the more controls on the unit, the longer it will take to set up for a different application. With a good dual trace oscilloscope and the proper time, these more sophisticated products can be optimized for each and all applications. If you don't have the application, resources, knowledge, or talent to justify this degree of calibration, then one of the simpler, more straightforward compressors will probably serve you best.

**Manic Compression.** The compressor is best utilized when used as a scalpel rather than as a shotgun. Identifying the main causes of headroom loss in the mix, and going after those channels individually with individual compressors, will result in the best possible signal quality. In a multitrack recording studio, where you have the luxury of working with individual tracks, a very expensive compressor with all of the above-mentioned features and functions may be the most valuable tool. In live sound reinforcement, however, it's preferable to have five $400 compressors that can be dedicated to only those channels or subgroups requiring compression than to have one $2000 compressor in your rack. Trying to do it all with one compressor is like declaring, "Ready! Fire! Aim!"

*Be careful — attack settings intended for special effects are very critical, and improper adjustments can brutalize the sound.*

used to allow vocals to be added after the orchestra recorded live. Miking and acoustics were also major factors. And there was still another factor — those old compressor/limiters. The warmth of Nat's voice on "Unforgettable" is just that — warm.

So how do you add some of that old-fashioned comp/limiter warmth to today's studio sensations? According to Paul Bruno of Audio Visual Resources in Boston, the Fairchild 670 comp/limiter, a veritable Holy Grail, is currently in great demand. "It goes way back, about forty years," he says. "Its market value is about $13,000 to $15,000, because there aren't that many of them left and they're really good and warm. They've got huge knobs on them, making them look like something you'd see in an old Frankenstein movie."

*continued on page 86*

# Input and Output to the Test

## The Most Surprising Result of the Electric Lady Studios D/A Shootout? A Lot of Converts to Digital Converters

**BY CRAIG "HUTCH" HUTCHISON**

A/D and D/A conversion is a mystical science. There's plenty of hype. Plenty of subjectivity. And plenty of price differences. How do these small, esoteric audio manufacturers filter sounds between the analog and digital dimensions so smoothly while the world's leading electronics giants — who manufacture millions upon millions of digital products — are dumbfounded by the process? What's an erstwhile audio engineer to do?

For two days this fall Electric Lady Studios played host to an elaborate converter shootout in order to find out whatever we could about these outboard conversion boxes. This study turned out to be the largest independent survey of converter listening tests ever conducted, involving 200 musicians, producers, recording and mastering engineers as well as several designers and manufacturers who took part in blind listening tests of seven A/D, five D/A and seven A/D/A packages.

**ON THE RACK**

Auditioners were given an opportunity to "torture test" each converter using a variety of sources under different conditions. They were given the choice of using live drums, grand piano, CDs, DATs and half-inch analog tape. Some used noise generators while a few even brought reverb units to better test low level performance. They were allowed to test each unit as they saw fit.

The lineup was as follows: The dedicated A/D converters submitted were the DCS 900, the Manley A/D converter and a couple of Pygmy units, one of which was a production model AD-I and the other a prototype with a new chip set. The A/D portion of three other units were tested including the Drake PD-5050, the new Neve HRC-1 prototype and the converter section of Lexicon's latest digital effects box, the Lexicon 300. The D/A contestants were the Manley tube D/A, the Wadia PRO-32, the Drake, Neve and the Lexicon 300 again, when its input/output modes were changed.

In another test situation some units were paired: the two Manley units and the prototype version of the Pygmy with the Wadia. For comparison's sake, we also ran four DAT machines (Sony 2500, Sony 7030, Panasonic SV-3700, Panasonic SV-3500) in record/pause to compare their A/D/A combinations to the outboard alternatives.

We were especially proud of the studio environment in which we ran our tests. Far from the average project room, of course (and a far cry from the room in which Jimi Hendrix once ruled), but nonetheless as audio-immaculate an environment as we think exists anywhere in the world.

All analog input sources were patched through a Focusrite console, whose line driver outputs had no problem handling all the A/D converter analog inputs. Separate drivers were used for balanced and unbalanced units and were then compared to the source to confirm audible equality.

Each converter was patched into a custom-built switcher box through their AES/EBU ports. This box was

**Off the wall: The Electric Lady mural at Electric Lady, N.Y.C.**

equipped with 12 separate balanced data receivers, one of which fed 12 balanced digital line drivers. The switching was simple LSTTL logic remotely controlled with a small box at the console. These line drivers fed the AES/EBU data input of the D/A converters and a Sony DRU meter was used to monitor levels. The analog outputs of the converters were patched back to the two-track monitor section to select individual D/As and the DAT decks. At this point it was also easy to compare the original analog source. Finally, this fed any of four separate speaker systems.

The monitors included the big two-way ELS wall-mounted cabinets. These are George Augsperger designed with TAD components and were powered by four Manley 350 watt tube amps. Alternately, one could listen on three near-field systems including Meyer HD-1s, Tannoys with Hothouse H500 amps and Yamaha NS-10s. Jerry Garszva, a fifteen-year veteran of digital recording, was impressed with most of the units; however, he was most impressed with the Augsperger monitors. So much for the purpose of the shoot-out.

**SOME FINAL OUTPUT**

Despite the final results (see sidebar) that we compiled from questionnaires submitted by our listening subjects, we came across quite a few unexpected conclusions. One was that digital inputs vary more widely than was originally thought. The Pygmy and Lexicon boxes had no sync inputs for slaving multiple boxes, while the others had a variety of these. Some had SDIF-2 and SPDIF inputs and word sync, but on different makes there were different connectors. The Wadia never glitched no matter how it was fed, leading many to conclude that it's

> Auditioners were given an opportunity to "torture test" each converter using a variety of sources under different conditions.

the only choice in a live broadcast situation. The Manley D/A occasionally, and loudly, protested when making a sampling rate change and a mute button was put on it to deal with this.

The Neve was the only D/A unit that required the flip of a toggle switch with a change of incoming sample rate or else it would have whined. Both the Neve and the Drake seemed excessively sensitive to marginal EBU data, while the Wadia and the Manley were immune. The Neve had a toggle to bypass the digital inputs and we found no difference in normal audible quality. (In all fairness, this was a first generation unit that was introduced at AES that week.)

Another trend we noticed was that while certain types of music made the decision easier or harder, almost no one varied their choices, even when using reverb tails and low-level noise or loud cymbals and CDs. We also noticed random results based

Phil Ramone photos by Michael Bloom

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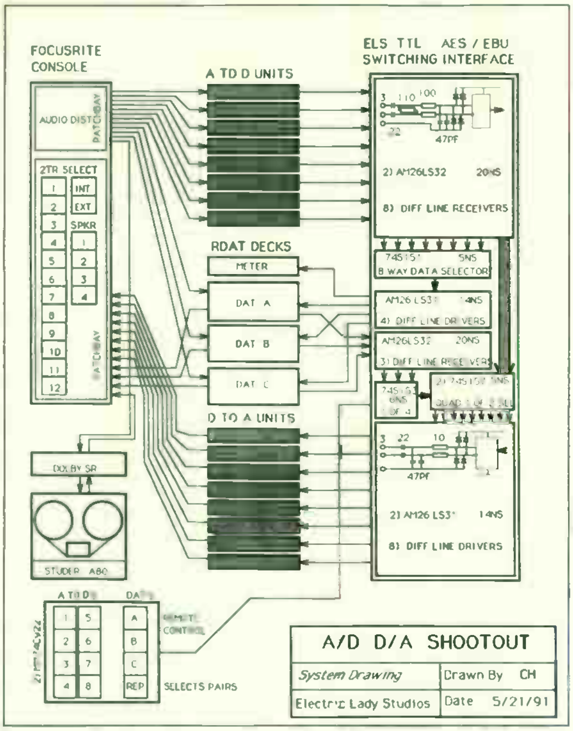



**Lady's First: The technical layout for the digital converter shootout**

on descriptions of frequency response. This was unexpected but then they all claim very flat responses.

We also noted that engineers looking for accurate time and phase effects such as imaging and clarity picked the Manley units. Among those who gave Manley the nod were George Karas and Josh Chervokas as well as Sal Greco and his entourage from Paisley Park. Those looking for amplitude effects such as linearity and smoothness almost always picked the Pygmy/Wadia combination.

Hit-making New York engineers Angela Piva, who was recently nominated for twelve Grammys, and Roey Shamir put the units through the most unique tests. They brought in a Quantec reverb and had live drums fed to the Quantec as they mixed in a copious amount of the returns — simply to find out which units did the least damage to the tail of the reverb. The Pygmy A/D was the easy winner for their test. And Andre Perreault performed lengthy tests with live and CD sources and also had a unique test where he used pink noise at levels below 50 dB to hear low-level smoothness. He also firmly chose the Pygmy/Wadia pair.

About 90 percent of the engineers picked one or both of these converters regardless of any other factors. Obviously these two packages won the popularity contest and we called it a tie. In comparison, the DAT machines all finished dead last, with even the Sony 7030, which is smooth sounding and extra clean, offering some sort of bass boost when it was compared to any of these outboard converters. In other words, we all learned the advantages of adding one of these boxes to even some of the better DAT machines on the market.

In the A/D tests, the Pygmy and Manley were most popular. The DCS and Lexicon also received good reviews from the testers. The Lexicon 300 was the winner for several of the toughest testers when coupled with the Manley or Wadia D/A. The Drake was fifth in popularity, followed by the latest Pygmy prototype and finally the Neve. Keep in mind that the differences were slight and less audible than D/A tests and combined tests.

The Manley and Wadia won the D/A tests and once again it was close. The Drake also did well, while the Neve had problems with glitches and whines. We avoided the Lexicon due to its automatic input switch.

About 20 percent of the testers were only concerned with the test that paired A/D and D/A and 75 percent started their process with paired converters. The 20 percent was almost always choosing the Pygmy/Wadia or Manley pair. No one picked the DAT machines but of those four, the new Sony 7030 was chosen the best sounding of the lot.

**TESTING ONE, TWO, THREE, FOUR . . .**
Several notes regarding this test have to be kept in mind. If we were testing and evaluating versatility, the Neve would have won, followed by the Drake; the Lexicon would be in a class by itself and the DCS was admittedly an older unit. Lou Reed was also on hand. Though he's very respected as an artist but lesser known as a serious audiophile and he A/B tests virtually every element of the recording process. He unknowingly combined the Lexicon A/D with the Manley A/D much to his pleasure. His laughing comment, "Now this is the benefit of a double blind test. I'd never gues I'd choose a Lexicon. I don't use their reverbs but this is impressive."

Lastly, we noticed trims for the A/D converters were often not obvious without a manual. This is even more true with D/As. We wished all had front-panel, labeled, screwdriver adjust trimpots like a Sony 2500 DAT. We also noted gains drifted a bit and had to adjust levels several times.

The end result was that nearly 200 engineers left with a better idea of what it takes to evaluate these vital audio devices. And since price proved not to be as pivotal an issue as many previously believed, the importance of some form of outboard converter in any digital studio environment — regardless of budget — became readily apparent. The message: don't spin digital tape (or hard disk) without one. EQ

## THE RESULTS

**Combined Pairs**
1. Manley A/D ($7000) and Manley tube D/A ($7000)
2. Pygmy AD-1 ($4000) and Wadia Pro-32 ($3500)
3. Lexicon Model 300 ($4800)
4. Drake PD-5050 ($8000)
5. Neve HRC-1 ($NA)

**A/D Converters**
1. (tie) Pygmy AD-1
1. (tie) Manley A/D
1. (tie) DCS 900 ($7000)
2. Lexicon Model 300
3. Drake PD-5050
4. Pygmy Prototype ($NA)
5. Neve HRC-1

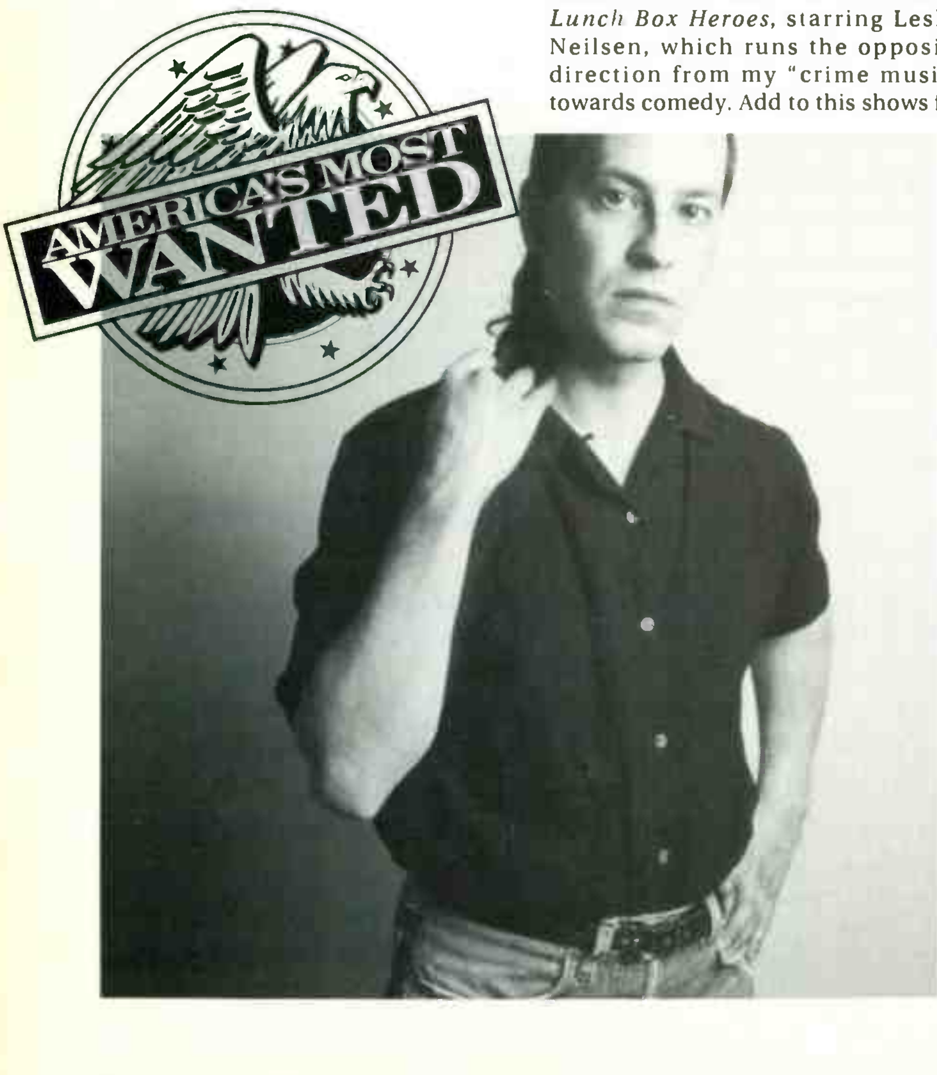
**D/A Converters**
1. (tie) Manley tube
1. (tie) Wadia Pro-32
2. Drake PD-5050
3. Lexicon Model 300
4. Neve HRC-1

The Sony DABK 7030 ($8000) and 2500 ($NA), and the Panasonic SV-3500 ($NA) and SV-3700 ($NA) DAT recorders were also tested.

# Scoring Like Gang Busters

From *Americas Most Wanted* to *Lunch Box Heroes*, I'm finally free to compose it my way

**BY MICHAEL JOSEPHS**




Composing music for television is certainly a lot different than scoring a film—or even recording an album. It's an art form unto itself; the medium, the possibilities, as well as the schedules, the pressures, and the time and financial constraints are totally unique.

My soundtracks for NBC, CBS, PBS and Fox Television shows keep me on my toes and working hard. With Fox Televisions' *Americas Most Wanted*, I had the task of creating a unique sound and aural feel for a new show on a fledging network, all on an intense weekly schedule timed to the post-production process. The show is a success now and has become part of our pop-culture. More recently, I've finished a new show for NBC called *Lunch Box Heroes*, starring Leslie Neilsen, which runs the opposite direction from my "crime music" towards comedy. Add to this shows for CBS and PBS that won a couple Emmys this year, other shows for Fox and record work, and it all conspires to keep me on the tube.

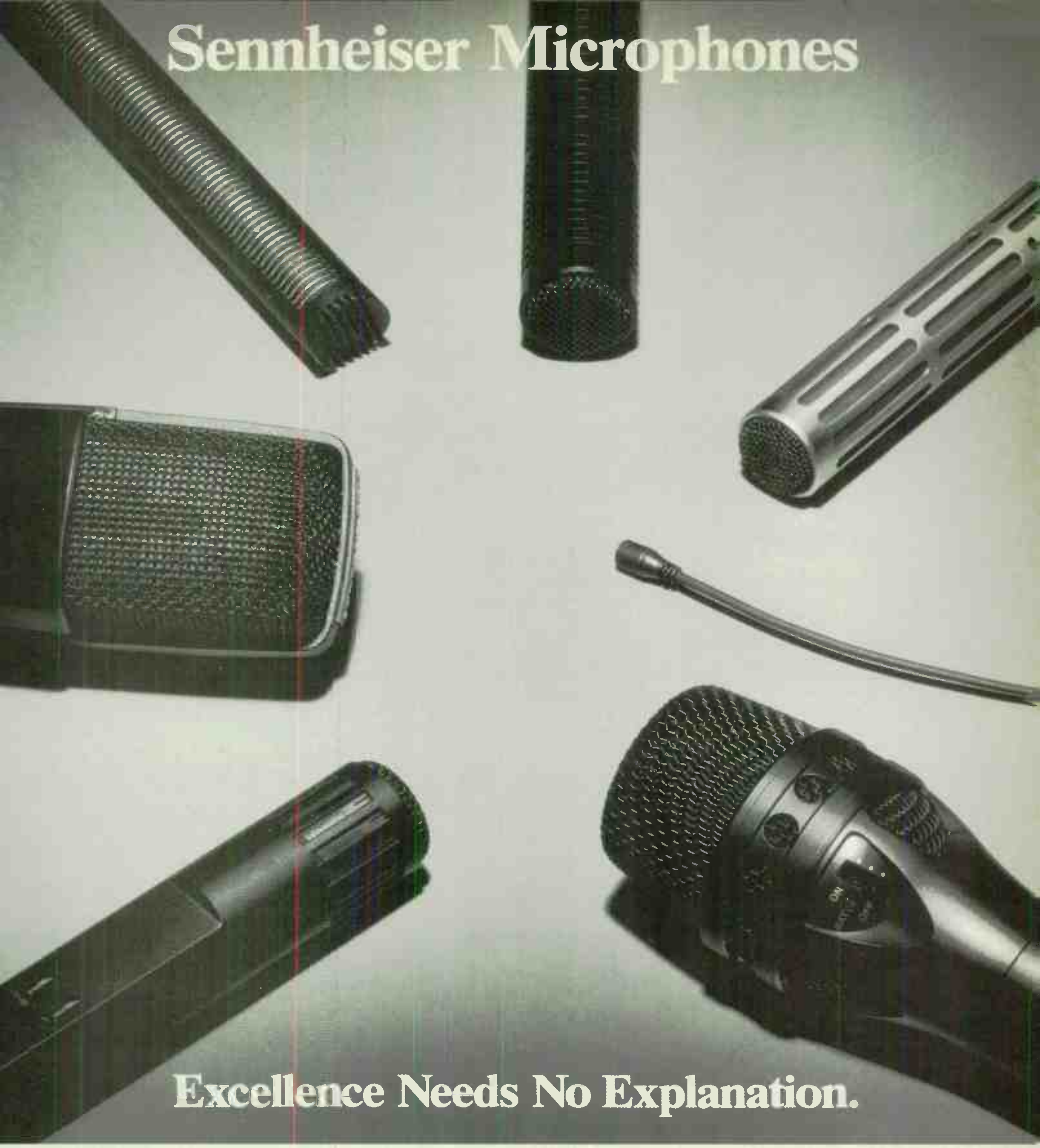
### SCORING DRIVE

Of course, the act of scoring is itself a very personal and subjective thing. There are a million ways to score a scene, and you're being paid for your expertise in choosing what's right for you and for the picture. This is where your own style and harmonic sense are called upon, as well as your background and experience. Instrumentation is a choice of taste (or budget). I could score a scene with full orchestra or with one guy snapping his fingers into a tape recorder; it's how you look at a scene, your rhythmic and harmonic sense, your sense of placement and color that makes your music unique. Budgets sometimes dictate important choices. Well, just use a synth or sampler, right? Sorry, that's just not for me. I use electronics all the time, but I'm really sick of synths, MIDI and sequencing. I'd rather use sounds for what they do best, not for something that makes them sound marginal and compromised.

### BREATHING ROOM

This philosophy makes for an interesting scoring proccess. When I first get the videotape, I "spot" the reel, take notes and start making decisions right away. At this point, most people would be inclined to put up the picture, bring out the rack, turn on the computer and lock to timecode. I'm most effective when I put my ideas straight onto paper or into a tape recorder with a piano. I feel that if you put your ideas straight into technology, you get away from the kernel of the idea.

I also like going direct to 24-track. I almost never use a sequencer. The only thing I might lock up when I'm writing and recording is drums through Performer. I'll go in later and record live drums or combine some live drums with the MIDI stuff. I prefer to use a click, and track direct with piano, or my Ensoniq EPS or Akai S-1000 if its an electronic thing. I like to


Sennheiser Microphones
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**The same sounds and even the same reverbs in a lot of work makes it all sound like musical wallpaper.**

manipulate textures in real time as I go to tape. As you perhaps can guess, I become a madman at this point; punching this in, moving that, mulling over this part. But it's the way I like to work.

Once I've laid some things down and the caffeine loses its effect, I go back and layer over it using my rack, which has Roland, Akai, Yamaha and lots of Ensoniq stuff. Then I'll have whatever live musicians I'm going to have in, or more frequently we'll recut the whole thing if it's a totally live or orchestral cue, using the original tracks as a guide. I lock to timecode with TimeLine or Adams Smith Zeta-Three. I lock the audio to the video rather than vice versa, as most people will do. Even in this day and age, studios still don't seem to understand timecode. I get a lot of attitude, even from top facilities, yet I still constantly end up with serious code problems that are inexcusable unless I am "difficult" (i.e. demanding).

### MIX TRICKS

The most important phase for me is the mix. Maybe its my background, but I seem to make most of my biggest decisions then. I make sure that everything is accessible and accounted for, the only things I have to mess with are the drum sounds. For drum samples I'll try anything, though the usual suspects seem to be the Eventide H-3000 for sampling and the Akai MPC-60.

I keep the previous phase technologically simple because the mix is where it happens for me. Putting parts in, taking them out, dynamics and EQ, and creating ambience are all part of the mix. One of the things I try to do when I'm composing, and especially when I'm in the mix, is to create a unique aural ambience. I think that if a director's images and concept are so unique, you should create a unique aural space for the images as well.

I also do this to get away from the sounds that everybody else is using. The same sounds and even the same reverbs in a lot of work make it all sound like musical wallpaper. Fortunately, it seems that a lot of scoring appears to be going back to live instruments. It's not enough to do stuff in MIDI, sequence and so on and say it's great. That's not musical, it's just filling space. Although the technology, I admit, is tempting.

Quite often, I'll mix to another 24-track to give them more choices on the mixing stage. I'll give them a few sets of stereo pairs. This is where a good console comes into play. All the standard ones we use (SSL, Neve, API, even the Peavey one I own myself) all have excellent panning, buss and routing schemes. You can really go to town and still come up with three or four discrete pairs, and the director will have more options in the final mixes. Top-notch outboard gear is essential, with enough to cover all the sub-mixes; I go for real natural sounding reverbs, with the Lexicon 480-Ls and Ensoniq's new DP-4s favorites.

### A NEW GROOVE TUBE

Because of the constraints of TV sound, the mix here is crucial. Every engineer I run into blows their own horn by saying they have their own perfect and unique techniques for mixing for TV. I'm sick of this; there are no simple answers. No one can tell what's going to happen with the audio signal when its broadcast. I'm always learning and rethinking the way music will come across on TV, trying different sounds, combinations, the way I mix and monitor. I use Dorrough meters so I can tell where relative levels would be.

TV audio has gotten better, but you've got to be careful. Something as basic as compression will take you out of whack. Things like the Aphex Compellor and Dominator are great, but most of the time I can hear the pumping pretty blatantly. Drums and bass are uniquely tough; Jan Hammer and Mike Post have built whole styles around bass sounds that work on TV.

Composing for TV is very difficult and time consuming, but the main thing for me is always making music in the most positive sense — regardless of the constraints put on you. You can never afford to lose sight of quality. EQ

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# Good Timing

## Upgrading dumb FSK sync to SMPTE synchronization

BY KARL MOELLER

SMPTE SYNC boasts a standardized format, compatibility with the video/post pro world, and the ability to let a SMPTE-synced sequencer autolocate to any position on tape within a second or two after pressing the recorder's play button. So, recognizing all these advantages, you've finally decided to upgrade from your "dumb FSK" sync system (FSK stands for Frequency Shift Keying, a process that encodes sync data as varying tones) to a SMPTE sync box, with SMPTE-aware sequencer software.

The only problem is, what happens if you want to remix or replace tracks on some older sequenced pieces that have "dumb" FSK sync? Here's an easy way to replace your old sync track with SMPTE, gain SMPTE's magic shuttle-to-anywhere capability, and sync new gear to old tapes.

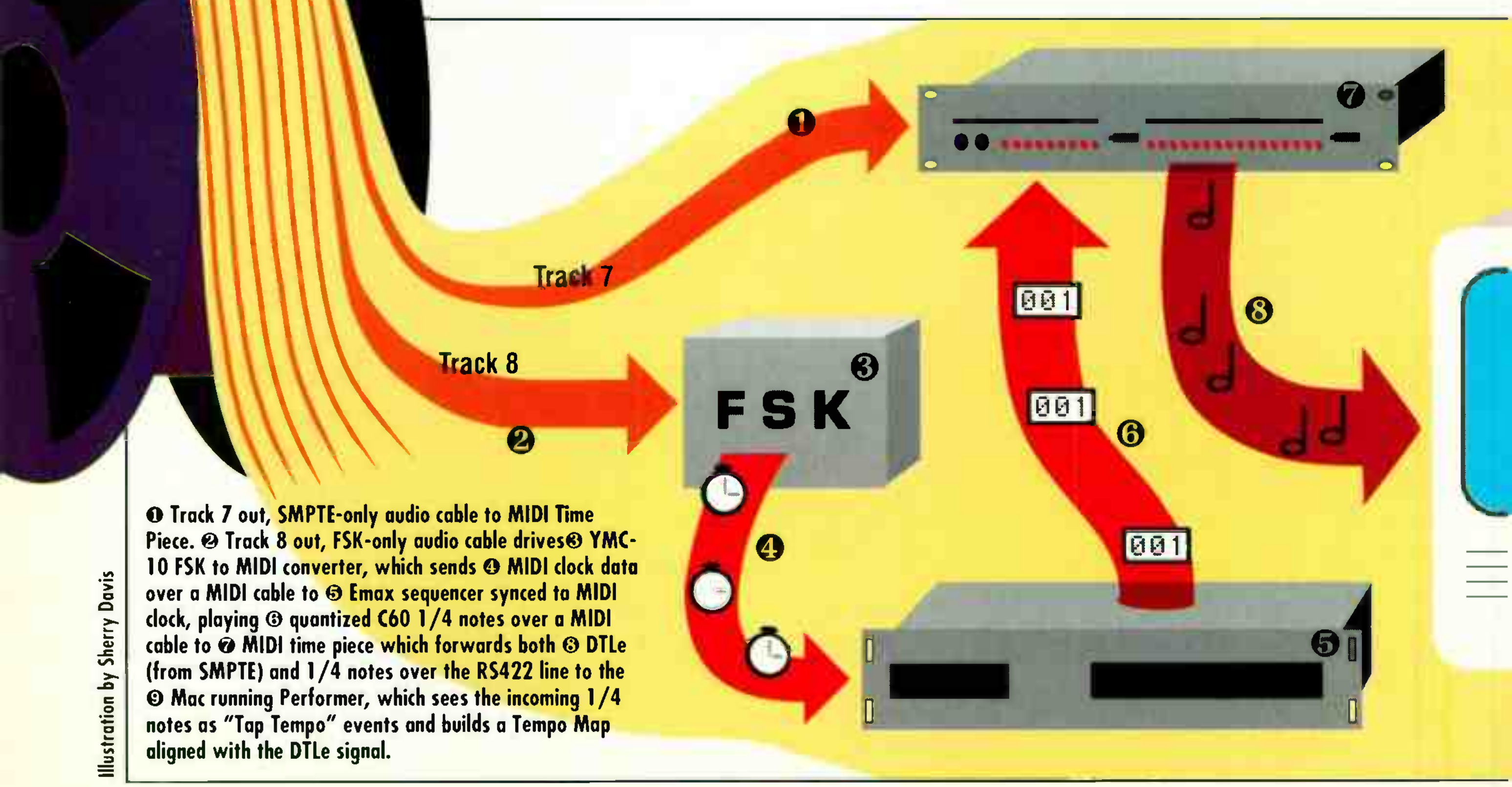
### THE PROBLEM WITH DUMB FSK

With dumb FSK sync, all timing signals are alike. If you start in the middle of a sync track, there's no way to tell whether the tape is on the first timing pulse, 43rd pulse, 100th pulse, etc. Dumb sync recognizes where the song starts (because the sync signal appears) and where it ends (because the sync signal goes away), but that's it. Like many studio owners, I've used dumb FSK sync to synchronize an eight-track deck and Mark of the Unicorn's Performer, a Mac-based sequencer.

With dumb sync, you lose time rewinding to the beginning of a song, which you have to do whenever you want to lay down a part — even if you just want to record the last few seconds. To get around this and other problems, I upgraded to Mark of the Unicorn's MIDI Time Piece for the SMPTE sync capabilities (read/write 24, 25, 30, and 30 drop-frame, with conversion to MIDI Time Code and MOTU's Direct Time Lock protocol), MIDI cable routing and channelizing software, and the deck accessory (included) that allows for on-screen adjustments of all parameters.

### OVERVIEW

The dumb FSK-to-SMPTE conversion process relies on Performer's Tap Tempo feature, which allows Performer — while synced to SMPTE — to record a Tempo Map based on clicks or taps you give it. (See the Performer manual chapter titled "Tap Tempo While Slaved To Tape.") Designed for use with already-recorded acoustic




❶ Track 7 out, SMPTE-only audio cable to MIDI Time Piece. ❷ Track 8 out, FSK-only audio cable drives ❸ YMC-10 FSK to MIDI converter, which sends ❹ MIDI clock data over a MIDI cable to ❺ Emax sequencer synced to MIDI clock, playing ❻ quantized C60 1/4 notes over a MIDI cable to ❼ MIDI time piece which forwards both ❽ DTLe (from SMPTE) and 1/4 notes over the RS422 line to the ❾ Mac running Performer, which sees the incoming 1/4 notes as "Tap Tempo" events and builds a Tempo Map aligned with the DTLe signal.

Illustration by Sherry Davis

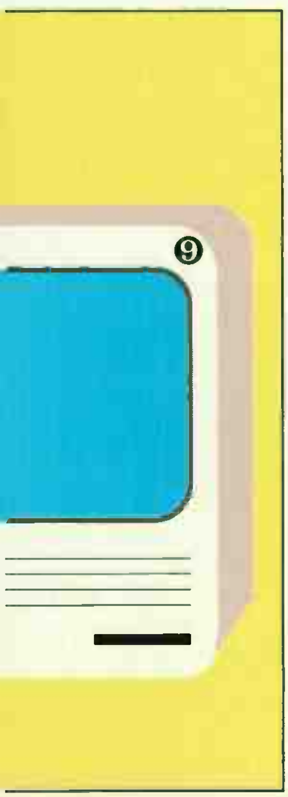
tracks to which you want to synchronize a sequencer, the taps could be any appropriate incoming MIDI event such as MIDI note-ons, tapped in time to the music.

The basic idea is to stripe the multitrack with a second sync track for SMPTE, and sync Performer to that. Meanwhile, any outboard sequencer that can sync to the dumb FSK track (I used the one in my Emax sampler) could send Tap Tempo events to Performer and create a new tempo map. To keep the tracks from bleeding through into the audio tracks, my goal was to keep the two sync tracks in tracks 7 and 8.

**THE PROCESS**

Here's how I did the upgrade; refer to the diagram for patching info.

(1) Create a reference sequence. Since the Emax could sync to the dumb sync, I needed an Emax sequence to provide the tap tempo events. So, I created a one-bar, looped Performer sequence consisting of 100 percent quantized, quarter-note middle C note events. This data was transmitted on MIDI channel 1 to the Emax sequencer, which was slaved to MIDI clock and set to record incoming data on channel 1. After Emax had recorded a suitably long sequence, I saved it to disk.




(2) Stripe track 7 with SMPTE. In my setup, track 8 was always reserved for the dumb sync track. If track 7 was empty, I'd record SMPTE there (be careful about levels, though; you don't want to bleed into the dumb track). If track 7 was occupied by an easily reproduced sequenced part, I would note what it was and stripe SMPTE over it. If track 7 had an irreplaceable acoustic track, I'd find another track containing a sequencer instrument, note what it was, bounce track 7's acoustic track to that track, then stripe SMPTE on track 7.

(3) Set up the MIDI Time Piece to read SMPTE. Send DTLe sync to the Mac running Performer.

(4) Load the old Performer sequence, and set up Performer to slave to incoming DTLe. Also set it to generate a Tap Tempo Map while slaved to tape, based on incoming MIDI note event 60 (middle C) on Channel 1.

(5) Set up the dumb FSK box to send MIDI clocks to the Emax sequencer. This allows Emax to sync to tape.

(6) Load the Emax sequence from disk, and slave Emax to incoming MIDI clocks. Patch the Emax MIDI out (which is sending quarter notes on Channel 1) to a MIDI Time Piece MIDI input.

(7) Roll tape. The dumb FSK tone goes to the FSK sync box, which sends MIDI clocks to the Emax sequencer, which starts sending out the quantized Middle C events. Simultaneously, the MIDI Time Piece sees incoming SMPTE and sends DTLe to Performer, which automatically starts, and begins recording a Tempo Map based on the Tap Tempo quantized note events it gets from Emax.

(8) When the song is complete, save the Performer file with its new Tempo Map. Since the dumb FSK track on the 8-track is no longer needed, I lock the MIDI Time Piece to track 7 and use Jam Sync to generate SMPTE, which gets recorded on track 8. Track 7 is then erased to serve as a guard track, or replaced with whatever sequenced part had been on it originally.

So that's what worked for me. If you have Performer (or any other computer-based, SMPTE-syncable sequencer with a Tap Tempo feature), along with a second sequencer that can sync to your dumb FSK track, this technique may be just the ticket for upgrading your tapes from dumb FSK to "smart" SMPTE. EQ

# Listen Up!

## Choosing & Using Near-Field Recording Monitors

**BY RALPH JONES**

Not so many years ago, engineers and producers relied exclusively on large, soffit-mounted control room monitors for tracking and mixing. Completed recordings were checked by playing back on cheap home stereos or car radio speakers, with the assumption that these were more representative of what the average guy on the street was listening to.

If the side sounded OK on these speakers, the thinking went, then it would sound fine on the radio, too. For convenience, studios began setting up small, inexpensive speakers right on the console meter bridge so that engineers could simply switch to them when they wanted to check their work. Engineers began increasingly to rely on these speakers.

Eventually, it became common to track on control room monitors, but mix on small speakers. Because the latter still were generally low-quality devices with limited frequency response, mixes were checked on the control room monitors to be sure that the extreme low- and high-frequency balance was correct. But certain engineers, convinced of the benefits they found in using small monitors, began to substitute audiophile speakers purchased in hi-fi stores.

Today, it's increasingly common to see entire album projects brought to completion using small, high-quality loudspeakers alone, with the large control room monitors serving only in an occasional high-level playback for the musicians.

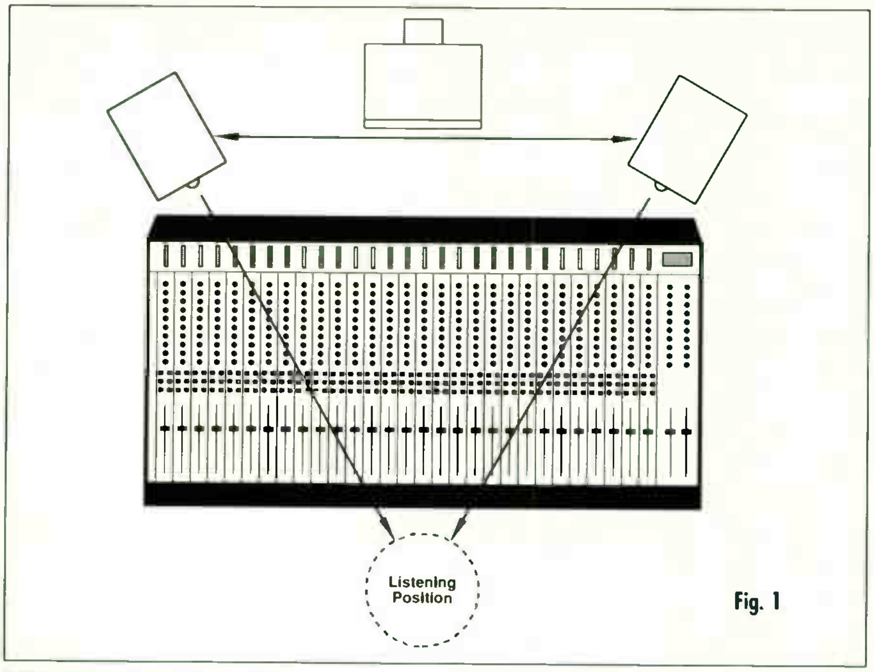



Fig. 1

### WHY NEAR-FIELD MONITORS?

The respected loudspeaker designer, Ed Long, was the first to explain why this reversal of roles took place. Since the small monitors are situated close to the engineer's position, Long observed, their direct sound reaches his or her ears at a much higher level than sound that reflects from the walls and ceiling.

Their output consequently remains relatively free of coloration and muddying effects from the environment, so they tend to reveal more detail. (The low level at which they usually operate also reduces ear fatigue.) In other words, the recording engineer sits in what is called, in acoustics, the "near field" of the small loudspeakers: that area where the direct sound predominates.

Long coined the term "near-field monitor" to describe speakers used in this manner, and suggested that the best results could be obtained with high-quality professional devices.

### SELECTING A NEAR-FIELD

Particularly if it is to be the primary recording monitor, a near-field speaker must exhibit flat, full-range frequency response and linear phase characteristics. Although it will be placed close to the mixing position, it should be capable of reproducing moderately high peak levels, so that musical transients are not distorted. It ought to be reasonably small so as not to obstruct sight lines into the studio, and ideally should be provided with some protection to minimize component failures and consequent downtime.

A number of companies offer small loudspeakers that satisfy these requirements to varying degrees, and the manufacturer's data sheet can supply basic information to narrow down your choices. When examining the specifications, pay particular attention to such factors as low-frequency response and amplifier power handling (most small loudspeakers are limited in these areas).

It also pays to take a look at what engineers that you respect are using. If you find that a particular individual's recordings consistently impress you with their sonic qualities and ability to translate on different speaker systems, the odds are good that he or she is using accurate, reliable monitors. By the same token, recordings that are overbalanced in a particular frequency range (very bright highs, for example) are evidence of deficiencies in the monitors.

Ultimately, your ear will tell you whether or not you can relate to the qualities of a particular monitor. Listen for such aspects as accurate and clear bass reproduction; smooth, well-balanced highs with no harshness; "air" in the reverb tails; and stable, realistic imaging. The very best monitors will render sounds with almost microscopic detail, permitting you to make fine adjustments of equalization

**It's increasingly common to see entire album projects brought to completion using small, high-quality loudspeakers.**

and balance when you track and mix.

**PLACEMENT**

The way that near-field monitors are placed in the studio can substantially affect their tonal balance and imaging characteristics. As a rule, you want to avoid loading effects from walls (which can boost low frequencies unnaturally) and reflections from nearby surfaces.

Keep both speakers equally distant from walls, and place them to form an equilateral triangle with the mixing position, as shown in Figure 1. You want to be in the near field of the speakers, so keep them no more than about five feet away. Aim the cabinets so that the mix position is on their main axes.

Where large reflective objects such as video monitors must be located between the speakers, position them as far behind the baffles as is practical (dotted object in Figure 1). Console surface reflections also can severely affect sound quality, causing destructive interference that appears as "combing" (frequency response ripples spaced like a comb's teeth).

Figure 2 depicts two possible monitor positions and their associated console reflection paths. When the monitor is placed directly on the meter bridge, destructive reflections occur at the listening position. Elevating the monitor (either by standing it up so that the tweeter is higher above the console, or using blocks to raise it upward) directs reflections away from a standing listener, but doesn't help a seated one.

Figure 3 illustrates an alternative placement using speaker stands set up behind the meter bridge. As shown, this placement is best for both standing and seated listeners, since reflections don't reach either position.

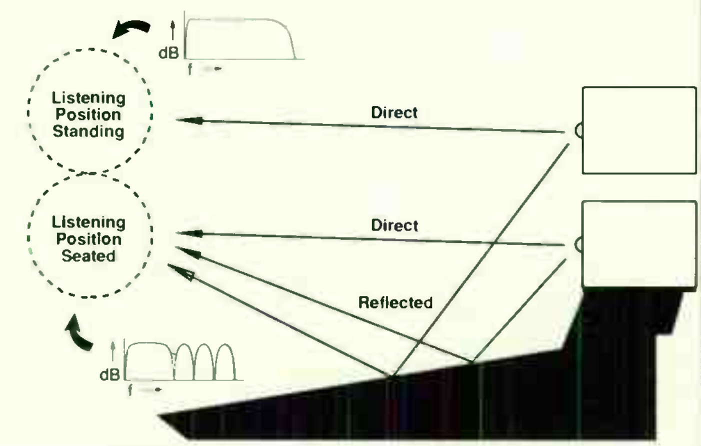



Fig. 2

You can use a mirror to check for possible reflection problems in your studio. Lay the mirror on the console surface and sit in the mixing position, then adjust the position of the speakers until you can no longer see their tweeters reflected in the mirror. Then, have an associate place the mirror on adjacent walls and equipment surfaces while you observe from the mixing position. Wherever you can see the tweeters, either reposition the equipment or, in the case of immovable objects and walls, hang absorptive material such as thick curtains or acoustic foam.

**WORKING WITH NEAR-FIELDS**

While it's true that we all know how to use speakers, some basic caveats

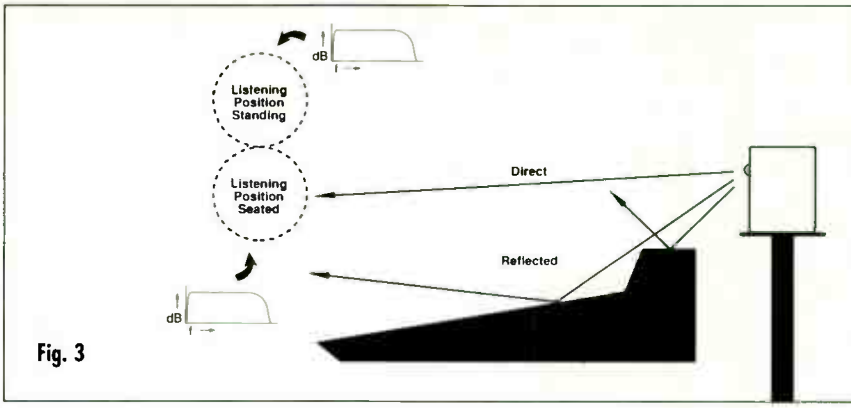



Fig. 3

apply to near-field monitoring.

First of all, it's usually a bad idea to run near-fields at very high levels. Being placed relatively close to your ears, near-fields with significant output capability can cause rapid ear fatigue. Some types of speakers also produce greatly increased distortion at higher levels (this is especially true of coaxial monitors, which are very susceptible to intermodulation distortion), and this will present a false picture of the tonal balances in your mix. Also, clipping the monitor amplifier on peaks may make a kick drum sound great to you, but it won't accurately reflect what's going to tape.

Second, be aware of the low-frequency limitations, if any, of the speaker that you're using. In a session that I observed some time ago, the engineer used a program equalizer on the console stereo bus to boost the lows in his mix. The result sounded fine on the control room monitors, so he set about making tape dubs through the board while monitoring on the near-fields.

> You can use a mirror to check for possible monitor reflection problems in your studio.

But he forgot to pull out the program EQ. The near-fields took a plunge below about 120 Hz, so he couldn't hear that he was boosting the lows a second time as he transferred the master to cassette. When the clients took their tapes home, just about all they heard was bass, and it took some effort to convince them that their master tape was, in fact, okay.

### A SOUND INVESTMENT

That story illustrates just how important good monitors are, and the extent to which deficient monitors can compromise your work. Ironically, one still finds studios with million-dollar consoles, quarter-million-dollar multitracks, a fortune in vintage tube mics, and a $250 pair of nearfield monitors! All that money invested in the finest equipment available, yet the engineer is making decisions using the sonic equivalent of a cheap black-and-white television.

In my personal half-inch, eight-track studio, the mains and near-fields represent an investment nearly equal to all of the other equipment combined — and the near-fields cost as much as two synths. You may think that's going overboard, but I've produced masters in my home that have gone directly onto the soundtracks of feature films, with no equalization at the dubbing stage. I'm using the same semi-pro gear everybody else uses, but my monitors make all the difference. Plus, they're fun to listen to.

So, don't skimp on your speakers. Good monitors won't be obsolete by the next NAMM show, and a reasonable investment in the best that you can afford will pay rich dividends in the quality of every track you cut. EQ

*Co-author of the "Yamaha Sound Reinforcement Handbook,"* **Ralph Jones** *is also a performing artist, film composer, recording engineer/producer, technical author and advertising/marketing consultant.*

## SPECTOR

*continued from page 48*

of the sound. The sound that you put on the tape will be the sound you come back to whether its been a day, a week, a month or thirty years.

Phil changed the industry with more than just his music. He created a little record company [Philles] that could make things happen and he was also the first producer to put an engineer's name on the label. He had a highly developed social conscience and insisted that the people who contributed be acknowledged. You would never have heard of me or Jack Nitzsche, who was Phil's arranger, if Phil hadn't done that. I still believe that because Phil gave credit to others, it diluted some of the things that are more creditable to him than to Jack or myself. We were tools in his hands and practically everything that came out of those sessions came out of Phil's head. I don't think that a mixer or engineer can create a hit. It's the producer or songwriter who creates the hit; the most an engineer can do is call attention to it.

I still hear some of Phil's techniques today. It's not really all that surprising, anybody who made good records has left his influence. Bruce Springstein has credited Phil for influencing him. It's not necessarily copying, but contemporary artists are taking Phil's techniques and enlarging on them — which is what should be done.

Phil is talking about going back into the studio and recording, and if he wants me to go in with him I will, because I know that it's not going to be in the same way that groups are recorded today. You can overkill with technology. But there's also so much you can do. I'd love to be involved with Phil's projects today and see what kind of hits he could put out — mono or otherwise. EQ

### REBUILDING THE "WALL"

**The Making (or Remaking) of "Back to Mono," the Phil Spector Box Set**

Phil Spector's "Back To Mono" (Abkco Records) box set was more than a nostalgia trip for Spector and engineer Larry Levine. The collection was the result of a careful blend of new digital technology and vintage sound equipment.

All of the analog-to-digital transfers, sequencing and equalizing of the original masters were done at Frankford/Wayne Mastering Labs in New York. According to the facility's mastering engineer, Tom Steele, the majority of the work involved transferring the original masters to Sony PCM 1630 digital. Keeping with Spector's "Back to Mono" theme, all of the remastering was done in good-old hi-fi monaural.

To preserve the authenticity of the recordings (some over thirty years old), an Ampex 351 tape machine, the model the original material was recorded on, was used for playback during the transfer process. A "vintage" machine to say the least, the 351 was hard to find, and locating mono heads for it was even harder. Steele dusted off some other vintage gear for the project including several Pultec and Fairchild compressor/limiters. Most of the 25- to 30-year-old masters, also a bit dusty, were originally recorded in either full-track mono or 2-track mono on Ampex tape. According to Steele,.they were all still in remarkably good condition.

Steele also says that Larry Levine's presence was the key to the process. "He was both the recording engineer and the mastering engineer for Phil Spector back then, and it was really a big advantage that he was playing the same role again," Steele explains. "He knew the tapes and their producer intimately."

Levine also helped Spector adapt his expectations to the era of digital audio. For example, in most cases, Spector didn't like the sound that resulted from the transfer to digital. "When you equalize in digital the sound stays too clean and you don't hear any changes," Levine says. "We had to take the digital back to analog and put it through a tube amp to soften the sound. And then we went back to digital with it."

Though the clash between old and new was problematic at times, the results were more than positive. "I think with today's equipment you can hear more of the textures of the 'Wall of Sound' than you could back then," claims Levine.

Interestingly, improving the sound of the original recordings wasn't one of their goals. "We just wanted to recreate the music the way it was recorded," adds Steele. "We went in thinking that the old stuff was great to begin with. From what we've heard back about the box set already, it looks like we were right."

*—Jon Ruzan*

# PEAVEY

*continued from page 113*

All these controls over the tone of your samples not only make up for the limitations of the envelope generators, but take things in a direction I wish more instrument designers would follow. Envelope generators are basically a stiff mechanical approximation of the way the sounds of most traditional acoustic instruments behave. They were invented to overcome the uninteresting square, saw and pulse waves found in early synthesizers, and are frankly pretty old. The design of the SP, on the other hand, puts more of the responsibility for shaping the sound under the fingers (and toes) of the musician. It will probably require a little more time to program, as well as to learn how to use key pressure, mod wheel and pedals, but the SP gives you the opportunity to change the shape and colors of your sounds live, in real time, just the way most acoustic musicians do.

The SP's filter is very good sounding. Although its resonance isn't programmable, it is set broad enough so it imparts no obvious coloration to the sound. When I filtered samples of acoustic instruments like strings, the filter only cut off highs as I intended, with no noticeable distortion.

One last cool feature that's particularly useful for drum sounds is the wave start modulation quantization. This lets you set a number of start points in a single wave and then assign each start point to a different velocity. The idea is to splice together a number of different samples end-to-end into a single wave. So if you combine four samples of a drum played at four different volumes, for example, you can play any one of the four by pressing the same key with different velocities, with no additional programming or using up different layers.

On the global level, you can switch or crossfade layers, detune them and modulate the detune by many different controllers, and control polyphonic note allocation priority. This level also offers overall pan control, output assignment and velocity settings.

### WRAP

I was peaved to find the SP won't let you save just one wave or tone or preset, but only a full bank (although you can add individual waves and tones to a bank). This can be dangerous. If you remove a wave, accidentally or on purpose, from a bank in memory, then save the bank, as I did, you've lost the sample from both RAM and disk.

I also wish there were more outputs, and at least one more envelope stage. But other than these fairly minor gripes, I found the SP a very good-sounding instrument that offers ambitious programmers the potential for creating new signature sounds. The processing here is clever and uncommon enough to let you do things you don't hear all the time. While some of the more sophisticated programming can get tricky, the SP's operating system is generally logical and easy to navigate. The manual is quite good, but it's only a reference; the promised tutorials while not yet available for this review, will be by the time you read this.

Particularly considering its low price, expandability and first-rate storage and loading capabilities, the SP offers a musician more than just an introduction to sampling.

*—Tim Tully*

# SKINS GAMES

*continued from page 31*

the number of mics and/or stands a club might have, for example, the drummer can determine how many microphones he'll wish to use to obtain a better representation of his kit. In this world full of turmoil, variables and mediocrity, it's nice to be able to have some control over a portion of your life. That's why I carry my own mics around with me (even if I'm not traveling with my own complete drum kit). Again, they are, in effect, a "pick-up" for the instrument, *part* of the drum set; and they connect me to my audience as I hope and intend. I recommend that more musicians count microphones as part of their own equipment, whatever their mic choice. EQ

***Former drummer for Weather Report, Peter Erskine has also recorded four albums with the Jazz/Fusion group Steps Ahead (serving as co-leader and composer). He has recently begun composing original music for theater and dance.***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## TUBULAR

*continued from page 67*

The famous Pultecs, while not rare, are growing in demand and value. The EQP-183 is at the top of the list, according to Bruno, and it sells for about $3000. The EQH-2, second in demand, commands a little less. Bruno said that most of his clients want the old tube-type compressors for use on rock vocals and, interestingly, there's an increasing demand for analog sound on the once digital-only dance music scene.

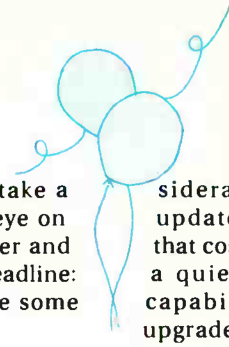
Teletronics' LA-2A and LA-3A are popular pieces at The Toy Specialists in New York, according to Bill Tesar. "Beyond that, occasionally people want to use a Fairchild 670. It's more of an esoteric piece," he explains. Tesar feels that such equipment, including the Pultecs to some extent, are inherently noisy by today's standards, but "people just like the sound that they give. For whatever noise advantages there are, there are also aesthetic advantages that people are drawn to." Tesar says that Guns 'N Roses rents a lot of his vintage units .

James Sabella, owner of Sabella Studios in Roslyn Heights, N.Y., operates a 24-track, Neve- and Studer-equipped studio with all the trimmings: "Right now, a favorite is the LA-175 from the Sixties. A lot of guys like it for bass and also for guitar." Sabella also emphasizes the popularity of the Pultecs. "The [Pultec] EQP-1A is very popular because it goes to 16k," he says. The EQH-2 and the midrange MEQ-5 are next in demand.

And for those who want something new that retains the classic warmth of the old, Summit Audio and Tube Tech are two manufacturers producing contemporary comp/limiters built around tube technology. The DCL-200 from Summit Audio incorporates vacuum tubes and reliable op-amps to achieve its high-tech classic characteristics. Plenty of digital recordists are adding this device to their system front ends to add warmth to their sound. Tube Tech's CL 1B, on the other hand, uses semiconductors only as part of the sidechain, while its attack/release select switch makes it possible to use its fixed or variable attack/release circuits separately or combined.

It's nice to know that not everything done today (or yesterday) will be gone tomorrow. EQ

# Snappy New Year



## Kick Off 1992 By Cleaning Up Your Act.

**BY RICHIE MOORE, Ph.D**

Something about this time of year makes me feel like cleaning. I know that everyone else does it in the spring. But personally I like beginnings to come along at logical times — like the beginning of a new year.

Winter is a great time to address those chores that are continually put off around the studio, to check on various odds and ends that can make your life in the studio a little easier. That's why every year around this time I make a resolution to take a technical inventory, keep an eye on things, and get a piece of paper and start making a list under the headline: "Technical To Do's." Here are some highlights on my current list.

### WHAT'S YOUR VERSION?

How current is your firmware? At various points in the history of a piece of gear, the manufacturer issues a new revision to the firmware. This usually contains updates that fix or enhance the operation of the gear. They're usually sent to the dealer, but sometimes they don't make it down the chain to you. Most of the time the upgrade requires the replacement of an EPROM, which every once in a while requires that you cut a trace and add a jumper.

For example, you may already own a Yamaha SPX-90. It's a good piece of gear, but the SPX-90II is considerably better. There's an update available from Yamaha that costs about $150. The result is a quieter unit with enhanced capability. Unfortunately, this upgrade is not for the technically faint of heart. The SPX has no sockets for ICs, and it's a double-sided board. You have to unsolder some chips and install some new ones. Most other pieces of gear, fortunately, have sockets for the EPROMs.

When is the last time you checked the firmware revisions for the various boxes in your rack? I suggest creating a database with all the pieces of gear and the current revision software. Remember, if the device uses vacuum tubes, it probably doesn't have software.

You can usually see what version you own when a device is first booted up. Sure you have to read quickly because the number doesn't stay in the window very long. Or you can open it up, and read what the version is on the top of the chip. During the past year, Otari, Sony, Mark of the Unicorn, Opcode, TimeLine and Adams-Smith, just to mention a few, released new firmware for their products. Why buy new when you can make current something you already own?

Illustration by Milton Reyman

### TALE OF THE TAPE

This is also a good time for the annual tape transport inspection. For months in *EQ*, I talked about setting up your transport. Now's the perfect time to put all that information to good use.

Check for garbage (slime, goo, small coral reefs) around the guides and capstan. This indicates that the guides may not be parallel. It can also indicate that you have been using badly slit tape. Clean the guides really well with a lint free cloth and some methanol. On the top and bottom of the guides there is sometimes a little indentation. Use a business card or a thin credit card to reinforce the cloth. Applying a moderate amount of pressure, remove the oxide build-up in this area.

Also check the pressure roller to

# PERESTROIKA

## 50 Years Ago . . .

Allied war planes laid siege to Berlin forcing Georg Neumann and his company to flee for their lives. They set up shop in a small town far from the bombing and, once again, started production. But history would again overtake them. They were inside the Eastern Block and the Iron Curtain was soon to fall.

## Time Passed . . .

We knew little about their fate. But with the dawn of Perestroika, we decided to trace this branch of the NEUMANN family tree. To our surprise, the company had flourished, supplying high quality condenser microphones to Soviet Bloc broadcasters.

*But more to our surprise, they are still manufacturing the M 7 capsule, the capsule that made the original U 47 the glorious sounding instrument it was.*

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Dirt and oil on the console surface eventually work their way down into the pots, faders and switchers.

determine the wear. When the rubber hardens and gets shiny, it should be replaced. This is also a good time to check the capstan. Make sure it's clean and hasn't become too smooth. Both the capstan and the pressure roller need to have a little rough surface on them to grab tape properly. You might also take the opportunity to lubricate the capstan if so required.

Check the tape tension and the brakes. As with anything mechanical, they can become loose. Make sure the tape is riding evenly across the guides and heads. Make sure the turntable height is correct and the tape enters the guides and heads without skewing. Brakes last a long time, especially in these days of servo motors and dynamic braking. You must clean them with acetone to remove oil and dirt.

Last, but not least, check the heads for wear. Get some of that wonderful Otari head ink, and check the wear pattern. Make sure that the pattern is symmetrical from top to bottom. This might also be a good time, if you have not done it lately, to send the heads for evaluation to either JRF Magnetics or Saki Magnetics. This will give you a benchmark about the life of the heads.

### CONSOLE-ATION PRIZE

The console should be totally cleaned. This is a cosmetic thing that helps extend the longevity of the console. Dirt and oil on the console surface eventually work their way down into the pots, faders and switchers. Find some relatively safe type of Freon to clean all the dirt off. Blowing the dirt with air just makes it settle elsewhere. Q-tips dipped in the solution also help scrub in hard to reach places. If it hasn't been done in a while, pull the modules out of the frame (with power off, of course) and vacuum all the dirt out. If a module has gold-printed pc fingers on it, use a pencil eraser to remove any oxidation before you reinsert the modules.

### CLEAN MACHINES

Amps and power supplies attract dust like magnets. The build-up of dust can cut down the efficiency of these devices and cause electrostatic problems. This is a good time to rent a tank of CO2 or liquid nitrogen and blow the units out. Take them outside to do this. No sense in just moving the dust around. Be sure to wipe down all the small fans. Dust encrusts itself to these devices.

This is also a good time to replace burnt out lights in the console and tape machines. Set aside some time to do this. Sure this is a cosmetic thing. The result is a more professional looking setup.

When you change the bulbs, check all the VU meters for calibration. Put a 1 kHz sine wave through all your devices and measure the levels coming back and compare them to the meter settings and the voltage going in. This might give you an idea of a potential problem.

Don't forget about the batteries. These invisible power supplies are what make your memory nonvolatile. Every device that remembers settings when it's turned off has a battery in it to keep the RAM from losing the current parameters. They eventually go bad; usually a year and a half from purchase. Make a note of which devices remember settings. Make up a priority list and go at these one at a time. Measure the voltage present versus what the battery says it should be. If it's less than 75 percent of the rated voltage, replace it. Otherwise, you may lose power and lose your valued settings. It's cheaper in time and therapy sessions to replace the battery.

Of course, there's plenty else you can add to your list. If there's time left, the carpet can always get cleaned, or the control room window can be washed. But don't go too crazy. You wouldn't want to mess with all those microscopic particles that contribute to your studio sound (except for the pile of coffee cups in the sink). EQ

# CUBASE TASTE TEST

Make no mistake, Cubase for the Macintosh™ is the 'freshest' Desktop MIDI Recording System currently available.

STEINBERG'S Mac Cubase offers many new features not previously available: a dedicated drum editor; user definable groove quantize; graphic remote control/automation of virtually all MIDI hardware; and MIDI Time-Piece™ compatibility. STEINBERG'S unique MIDI real-time operating system (M•ROS) lets you view and edit your music in any of four edit windows, including standard notation, as you record it.

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Take the taste test, **"Buy it, Boot it, Use it, Love it or your money back!"***...Cubase, the freshest Sequencer available.

*This guarantee is valid through participating dealers only. Please verify prior to purchase. A full refund, less 10% handling and reshelving charge, will be remitted upon receipt of the complete (including both installs) Cubase software package in undamaged condition and within the 30 day period. This offer expires on December 31, 1991.

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CIRCLE 69 ON FREE INFO CARD

# The Joy of Six

## A Half Dozen Projects to Get You Back to the Bench

BY CRAIG ANDERTON

Photo by Ed Aiona

Who said every one of these projects has to be wild, weird or bewildering? What about some simple, useful projects you can assemble in an evening (or better yet, fifteen minutes). Here are some accessories that are so easy to build, and so utilitarian, you'll probably find at least one project that inspires you to head into the workshop.

### WALL WART TAMER

Wall transformers cut down on hum and simplify UL approval, but they usually block adjacent sockets on a barrier strip. Also, leaving them plugged in when no current is being drawn can shorten the transformer's life.

The wall wart tamer consists of a length of two-conductor zip cord and three parts that can attach to zip cord without any soldering: AC receptacle (GE Quick Clamp Connector, part no. GE1710-21D, or equivalent), on-off switch (ACE Hardware Quick Connect Cord Switch, part no. ACE 31089, or equivalent), and plug (ACE Hardware Quick Connect Plug, part no. ACE 31099 or equivalent). Clip the receptacle and plug to opposite ends of the zip cord, and the switch in between. Plug the transformer into the receptacle, and the plug into your barrier strip. Use the switch to turn the AC off when the unit isn't in use.

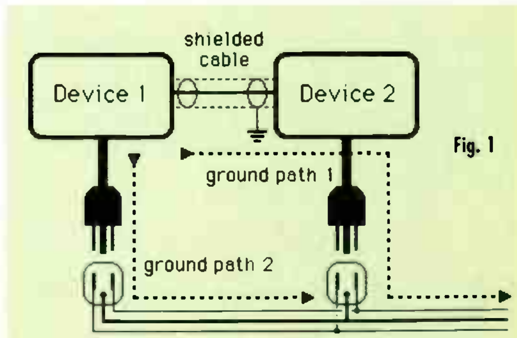


Fig. 1

### MIDI CABLE EXTENDER

When one MIDI cord doesn't quite reach, grab a second cable and this extender. Build it by placing two female MIDI jacks (Radio Shack no. 274-005) back to back so that their pins butt against each other, solder the pins together, then wrap electrical tape around the soldered pins for insulation. For a more permanent extender, mount the two jacks in a small metal box and run wires between the pins.

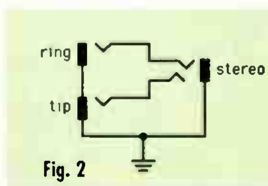


Fig. 2

### GROUND LOOP LIFTER CABLE

People often "solve" ground loop problems by lifting a unit off ground with a three wire-to-two wire AC adapter plug. Wrong! This sacrifices the protection the ground wire provides; if a short occurs in the unit or the chassis becomes "live," the ground wire will no longer be able to shunt the voltage to ground and will pop a fuse or circuit breaker to eliminate the unsafe condition.

The correct solution is to break the ground loop elsewhere. Fig. 1 shows a typical situation where two paths to ground cause a ground loop.

If you break the ground shield on the shielded cable audio patch cord connecting the two units, you'll break the ground loop. To do this, make a patch cord where the shield connects at only one plug. This prevents a direct ground connection through the cord, but since the shield connects to ground through one of the plugs, the cable's inner conductor is still protected from hum and noise.

### STEREO/MONO BREAKOUT BOX

Using stereo jacks for insert points saves space on mixers, but is the tip connection the input or output? What about the ring connection? What about devices with a single stereo output jack that you want to feed into a mixing board with mono jacks? The Stereo/Mono Breakout Box (Fig. 2) is the answer.

You'll need two quarter-inch mono phone jacks (Radio Shack no. 274-252) and one quarter-inch stereo jack (Radio Shack no. 274-312). Wire one mono jack hot lead to the stereo jack tip connection, the other mono jack hot lead to the stereo jack ring connection, then connect all the grounds together. Mission accomplished: run a stereo cord to the insert jack, and patch the mono jacks to your signal processor in and out connections.

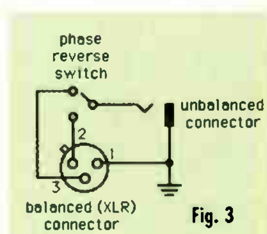


Fig. 3

### BALANCED/UNBALANCED ADAPTER

When you have an unbalanced out or in and a balanced out or in and you need to hook the two together, what are you gonna do? Build the goodie in Fig. 3.

The unbalanced connector will usually be a quarter-inch mono phone jack (Radio Shack no. 274-252) and the balanced connector, a 3-pin XLR type (Radio Shack no. 274-013). The phase reverse switch (Radio Shack no. 275-603) accommodates those manufacturers who are not yet aware that there's an international standard, IEC 268, that specifies pin 2 of an XLR connector as being the "hot" signal pin, and instead wire pin 3 as hot.

**WORLD'S SIMPLEST MIDI PATCH BAY**

It's not programmable or transposable, and doesn't even have a microprocessor, but this patch bay can make it easy to patch MIDI connections.

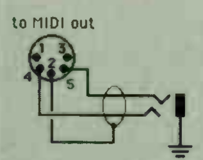

Fig. 4

All you need is a rack panel, a bunch of stereo jacks (Radio Shack no. 274-312) to mount on the front panel, and a bunch of MIDI plugs (5-pin DIN types; Radio Shack no. 274-003). Remember, MIDI has only two signal lines; the third connection is simply a ground to provide shielding. Fig. 4 shows how to wire a stereo jack to a MIDI plug designed to patch into a MIDI out or thru connection. Fig. 5 shows the wiring to a MIDI plug designed to plug into a MIDI in connection.

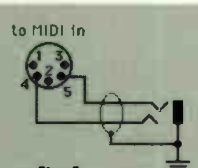

Fig. 5

The front panel stereo jacks now provide your various MIDI patch points, and all you need are standard, shielded stereo patch cords to repatch your MIDI gear connections. (You might want to build the MIDI monitor circuit in the October 1991 *EQ* into the patchbay so you can check for MIDI activity.)

There, that wasn't too difficult, was it? Happy soldering, and remember — never solder with your shorts on! See you next time. EQ

# Hearing Is Believing

## There's Something Far Deadlier In the Audio Industry Than a Tin Ear — Tinnitus

**BY MARTIN POLON**

What do we have here? A new column called Fast Forward? A place where you can read about what is going to happen in the *business* of recording audio in the project studio and the mainstream studio? Yes, to be sure. But before we get the tape rolling, let's take a brief look at the problem of hearing damage for those who use audio in the studio environment. Other topics also beg to be covered first but this issue must take precedence because "People are getting hurt out there."

### AS BAD AS IT SOUNDS

Hearing damage is an issue frequently overlooked by those working in project studios. Owner-operators and others involved in project studios tend to work themselves harder and with less precautions than employees of large studio complexes. And there is almost never a visit by a government health and safety inspector to establish reasonable levels.

"Reasonable Limits" is the key phrase to consider in setting audio monitoring levels. An enormous body of medical research has established beyond a doubt that any kind of continuous exposure to sound pressure levels in excess of 85 decibels places certain individuals at risk of suffering permanent and irreversible hearing damage. (A level of 85 decibels of sound pressure is the equivalent of that found in the cabin of a 747 jumbo jet. Many studio monitoring environments routinely exceed 90 dB levels.)

While it's true that some persons will have a higher threshold of damage than others, it's impossible to predict who these people might be. So one can only assume that 85 decibels is the right place to start dealing with a damage potential. The exposure problem is cumulative over time and the greater the levels encountered the greater the risk of severe damage.

The medical consequences of hearing damage can include the onset of a condition called tinnitus. This disorder takes various forms but has been characterized by many permanent sufferers as "having a jet engine in each ear revving up for takeoff — twenty-four hours a day seven days a week — fifty-two weeks a year." Permanent tinnitus, caused by cumulative and consecutive hearing damage, is almost never reversible and can cost those afflicted their sleep, their ability to concentrate and often their further contact with audio.

### SAFETY FIRSTS

The first step in keeping the studio environment safe is to purchase a sound pressure level (SPL) meter. Although we will usually not recommend a specific product for purchase in this column, in this case it seems appropriate to indicate that a relatively precise instrument is available from Radio Shack stores for under $50. More expensive meter units are available and some large studios have tried audio dosimeters designed for the factory floor to measure total exposure.

A very satisfactory control room "fix" is to use an acoustically isolated mount or suspension system for an inexpensive meter facing the monitor speakers near ear level. The meter face can be enlarged with an inexpensive plastic "fresnel" magnifier lens for distant viewing or a small lamp circuit can be added to the SPL meter output jack to fire at a set level such as 85 or 90 decibels.

### LISTEN UP

Many audio practitioners (and project studio users are no exception) fear the establishment of safe listening levels but most frequently register surprise at how generous safe listening levels can be. The only "fly in the ointment" is for those who have already sustained some damage from previous exposure and compensate by "cranking up the juice." It's important to remember that several "name" musicians from top rock groups who used to joke about sound levels being set "either at off or loud" now make public service films about the dangers of hearing damage.

One not-so-famous but steadily employed studio and back-up musician described his current life as a less than desirable state of affairs: "After fifteen years of cranking it up, I seem to have blown my ears away. I cannot hear normal conversations more than a few inches away from my head due to the damage. The tinnitus means I cannot ever be in a completely quiet place or the roar will drive me completely over the edge. I sleep with a rainfall machine in my bedroom and I keep a TV on all day in the office. I can't handle music any more. I have to keep 'masking noises' available or I start to come apart. Some gig, huh?"

Many individuals employed in the audio industry have started to carry small, inexpensive hearing protectors. The units will provide enough attenuation to duck levels down 20 or 30 dB in the ear, which is enough attenuation to prevent damage but not enough to interfere with monitoring of amplified sound. But the bottom line may depend on the outside lifestyle issue. Little is accomplished by monitoring levels in the studio and then jumping into your 105 dB audio environment in your car for the drive home. Ditto headphones with personal stereos. And watch out for motorcycles and shotguns! The ears you save are very likely to be your own! EQ

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# Stage (Monitor) Right


## A quality stage monitoring system for a band on the run might be more in reach than you think

**BY J. D. SHARP**

There comes a time in the life of many bands when they reach a level of venue that requires a monitor system — but they're not quite up to the plateau where a system will be provided as part of a rider. Monitor systems below the level of world-class touring bands are somewhat of a neglected category. Whereas the musician has hundreds of guitars, dozens of amps, a plethora of microphones and a wretched excess of keyboards to select from, when it comes to monitor systems the choices are far more circumscribed. The limiting factor is the monitor mixer; there are only a couple of viable products to recommend. A monitor mixer differs from a main mixer in that its sole reason to exist is to create independent mixes from the same source, which is a "split" of the main board's inputs. Typically such a mixer will offer eight or more pre-fader sends per unit, EQ of some sort, and an output section that facilitates easy auditioning of each mix by the monitor engineer. Luckily, the choice that's offered is a good one, with two solid contenders:

Yamaha's MC2408M ($3995) has been around for a while, and it's perfect for the job. It provides 24 inputs, eight monitor mixes, insert points for each input and master bus, a useful phase switch and channel on/off switches which allow the engineer to kill a problem input without changing fader levels. EQ is a three-band with sweepable midrange. Both Talkback and Communication modules are built in. (Talkback gets you into the monitor mixes, while Communication accesses the headphones of the crew.) A Cue Out lets the engineer send a selected mix to an additional monitor speaker, which is typically a wedge monitor or the same type used on stage, so the engineer can hear exactly how the mix sounds to the artist. Illuminated VU meters ease the task of watching levels.

The new kid on the block is the Soundcraft Spirit Monitor ($5650), shipments of which have commenced recently. It's the newest member of the successful Spirit series, which also includes recording and live mixing boards, and is available in 16- and 24-input configurations. Not surprisingly (since they had the Yamaha to look at) a couple of enhancements are offered, including sweepable EQ in both lows and mids (shelving highs) and a 100 Hz high-pass filter, a very useful feature for blocking subsonics and bass rumble from entering the monitor mix. There's also a variable high-pass filter in each of the eight outputs for further control. The Spirit Monitor borrows a feature from Soundcraft's upscale monitor mixers: a linear fader is fitted to each channel (in addition to the eight rotary monitor mix controls) that allows the monitor engineer to assemble his or her personalized mix for the "wedge" output.

The other new release is Studiomaster's Stagemaster ($3750). It's a 16-in, eight-out configuration. Two features are unique: It includes a splitter (more on this in a moment) which can save hundreds of dollars and can be expanded in increments of eight inputs ($1065 per expander) up to forty inputs. Additional features include three-band EQ with sweepable mids and lows; EQ bypass; channel and output mutes; insert points all around; and 48 volt phantom microphone powering.

When it comes to choosing the right mixer, consider your needs. The Spirit has the overall edge in features and compactness. The Yamaha scores on layout and intelligibility. (Tilted up VUs are easier to look at than flat LED ladders, and the Soundcraft's output section has two outputs per strip, where Yamaha's run right across the mixer. This takes up more room but makes it easier to grab the right fader.) Other points in Yamaha's favor are the Communications output, and the most inputs for the buck. Studiomaster's unit makes sense if your needs may expand down the road; the built-in splitter is also both a convenience and a money saver. You be the judge. If more inputs are needed, consider more upscale offerings from **Soundtracs, Soundcraft** and others.

If the cost of a separate monitor console is way out of line, consider a main console that has lots of sends (six to eight is a healthy range), so you can still have several effects and perhaps four monitor mixes at the same time. Or try to find a used **Peavey MD Monitor** board; this discontinued model provides six independent mixes for a song!

### SPLITSVILLE

With the console out of the way, it's time to make moves on the areas that offer a broader choice. But first we must mention the splitter. This is a metal box that contains a collection of one in, two out transformers. It's used to isolate the main mix from the monitor, and to provide a feed to the monitor console. If you have a 24-input setup, you'll need a 24-input splitter. These are custom-built units, often attached directly to a multipair snake, and are available from the larger cable companies like **Proco** and **Whirlwind**, among others. Shop around for the best deal, and pay attention to the quality of the transformers. (Cheap ones will degrade bass response and will add distortion.)

### SPEAKER YOUR MIND

There are quite a few monitor speakers available, since nearly every producer of reinforcement cabinets makes at least one model. Here the choice is dictated by several requirements: budget, maximum volume needed and size. First consider size. Larger monitors are more efficient and thus will get louder for a given power amp input; they'll also develop more low end, which may or may not be desirable. A drummer, for instance, may want to hear the full impact of the kick drum, but the vocalist may be somewhat content to hear somewhat less thump. Big monitors also take up stage real-estate; make sure they fit in the venues you play. Overall volume also affects what type of monitor to go for. If you need lots of volume, specify a monitor that can penetrate the chaos of a live stage performance. If you're doing folk music, a more high-fidelity unit is called for. Be aware that a monitor that sounds great (that is, like a hi-fi) in a music store may not cut through on stage.

Having said all this, here are some choices. For cost-effective, yet high performance, try **EV's FM-12C** ($500). It's unimpressive-looking molded cabinetry cleverly allows versatility in angling and placement. They'll also handle 200 watts long term. For more volume, consider **EV's FM-1502ER**

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($898) and FM-1202ER ($726). (You can tell the size of the woofer by looking at the model number.) The FM-1502ER handles up to 300 watts continuous and 1200 watts short term, with 102 dB efficiency and 65-20,000 Hz frequency response. Peavey's 1245M ($450) and 1545M ($480) are a great value; they cut through in even the most adverse conditions, handle a ton of power and are notably less expensive than both EV and JBL MR models. Speaking of JBL, their MR802 (12-inch; $575) and MR805 (15-inch; $650) floor monitors come in a bit less than the EVs, albeit with slightly less power handling capacity and efficiency. It's also worthwhile to check out the offerings of Yamaha, Bag End, PAS, EAW, Klipsch and others; you'll be surprised how competitive most of these products are.

Side-fill monitors are useful for drummers, and for small venues where floor monitors will take up too much room. Typically, a fairly high-dispersion system is used (90-120°) so that not too much energy is projected past the musicians and into other microphones. Any efficient two-way or three-way system can do the job. JBL's MR835 ($775) comes to mind, offering a compact three-way configuration with 15-inch woofer, 8-inch cone midrange, and an effective titanium compression driver. Of course every other manufacturer has a competitive two- or three-way offering in this range.

### HIGH EQ

EQs are a necessity. You can't go wrong with Rane's ME15 (dual 15-band; $369) or ME30 (mono 1/3rd octave; $359) graphics; these have the advantage of balanced ins and outs, and the Constant-Q design leads to minimum interaction between bands, so you get exactly the action you want when a control is moved, and no more! Or try the Rane ME-60, with two 1/3rd-octave EQs in a two-space chassis ($649). For cost effectiveness check out the Alesis MEQ-230 ($249), which combines two 3-band, 1/3rd octave EQs into just one rack space and lists for just $249. Consider also the newly updated Feedback Exterminator from Sabine, which can give you a few extra dB of gain when all else fails, with minimal detectable alteration of the signal.

### GETTING COMPED

Some cost-effective limiting is also in order; try a few Symetrix SX208 dual compressor/limiters ($299). Each one offers two channels, yet fits into a half rack format, so you can cram eight mixes' worth of dynamic control into just two spaces! A dbx 900 rack ($759) loaded with 903 Compressor/Limiter modules ($449 ea.) is a worthwhile, if somewhat more expensive, alternative.

> Big monitors also take up stage real-estate; make sure they fit in the venues you play.

### POWER TO THE PEOPLE

Power amplification is almost silly to talk about, with so many excellent choices. But there are some guidelines to make your shopping easier. Remember that if you're going to run several monitors from the same amp, you'll need to pay attention to impedances, and get an amp that performs well when hooked to 2 ohm loads. (That's not very much of them.) It's preferable to use more amps (you'll need one channel per monitor mix, anyway) and keep impedances at 4 ohms or higher for stability. There's basically no such thing as too much power; you don't need to use it to over-drive the speakers, but rather to keep transients and peaks clean. (Distortion blows a component far faster than clean power.) Also, look for balanced inputs; if you can keep your whole signal chain balanced there's far less likelihood of interference. Remember that cheaper amps, for the most part, have less sophisticated protection circuitry and fewer output transistors to share the work load.

A well-tuned stage monitor system is one of the most valuable assets a band can have. After all, if you can't hear what you're doing it's hard to get into the feeling, and even harder to get the audience there as well. EQ

## NICHOLS

*continued from page 114*

the flags to say consumer. An annoying byproduct of this feature is that the SV-3700 will go into record mode and actually start recording for about ten seconds. The machine will then lock up completely. You can't move the tape in any direction, the only thing you can do is eject it. (IN THE MIDDLE OF YOUR MASTER!) This will reset the transport and let you try again. For those of you who are interested, the SV-3700's consumer brother, the Technics SV-DA10 performed exactly the same on the consumer side tests.

I got an SV-DA10 for my wife's demo studio, but had it modified to record analog in at 44.1 kHz. I never figured that one out either. You're supposed to be allowed to digitally copy a CD at 44.1 kHz, but you can only record analog at 48 kHz. Smart.

### DAT INS AND OUTS

Well, basically you've got the idea of what to expect when trying to do anything with digital copies. Some of the other problems we ran across were:

•The Tascam DA-30: ignores all SCMS bits when recording digitally through the AES/EBU interface, but honors them if you try to record consumer information. Everything the DA-30 records has the SCMS ID-6 set to 11. This means that the tape is the final copy and you can't make another copy. Contact the Tascam repair station for a fix.

•The Sony D-10 Pro: If you try to play back a DAT tape that has the SCMS code set to 11, no audio will come out of the unit. What was supposed to happen was that the digital out was supposed to be blocked so you couldn't make a digital copy of the tape. What they did instead is block the digital data to the analog convertor. Don't worry. The Sony repair stations have a fix.

•The new JVC DAT machines and Sony 7000 series machines do not record auto start IDs when you're recording from the digital inputs, even though they do when recording analog.

•Here is an interesting fact: all machines that sent data out as consumer format did not set the checksum properly in the data block. The receiving device would always display "CRC Error" if so equipped. This has nothing to do with audio errors, however, only errors in the information block.

•We ran across a few DAT machines that have high pass filters in the circuit when copying tapes digitally. This means that the copy will not be a clone of the original. We haven't weeded them all out yet, but I'll let you know.

Glenn Meadows has coined a couple of names for these non-consumer/non-professional machines. Pro-sumer was the first one, but I like the alternate, Con-fessional DAT machines. It has a familiar ring to it, don't you think? EQ

# Advertiser Index



## Coming Next Issue

EQ

- Peter Gabriel's sonic-cocoon — The Workroom at Real World in Bath, England.
- The complete guide to amplifiers — or, how to really fill out your project studio.
- A new beat for Roger Nichols —the current drum machine situation.
- Craig Anderton shows you how to outsmart your own hard-disk with some super-duper shortcuts (not short-circuits!)
- Dr. Richie Moore applies his Rx-ray eyes to the problem of doctoring old audio tape.
- Exclusive hands-on reviews of Opcode's Track Chart, and Mackie's CR 1604 mixer.

Plus

Mic splitters, notebook recorders, the Fostex automation system and how to make your patchbay pay

# QSC

*continued from page 109*

frequency (ultrasonic) oscillations. These could destroy high frequency drivers without ever being audible. QSC claims a worst case slewing rate of better than 20 volts/microsecond. Short circuit protection is handled by an Output Averaging circuit that will limit the output to 25 percent of full power if a short circuit is detected. The EX series of amplifiers has circuitry that enables the protection mode to recognize the difference between a very low impedance load (>2 ohms) and a short circuit on the output. The possibility of DC voltage appearing at the output of the amplifier is prevented by a relay that is energized when the DC voltage is detected and grounds the output to prevent the voltage from destroying loudspeaker drivers. All are very practical requirements in live sound applications, but accomplished here in a very elegant manner.

As well as the cooling fan inlet, the rear panel has the Open Input Architecture input module, output connectors and a captive 6-foot 16/3 SJT grounded AC power cord. The ground pin on the AC power cord is connected to the audio ground. The unit is rated to draw 12 amps at 120 volts AC but can draw up to 32 amps at peak power output.

### FEATURES PRESENTATION

There are several very handy features included in the connector panels. The Open Input Architecture module allows for future input interfaces to be easily retrofitted to an installed amplifier in just a few minutes. This could include anything from crossover/limiter modules to digital audio inputs or computer control. The module is removed with four Philips screws and connects to the main PC board via a latching 26-pin header connector. The module shipped with the amplifier I tested allowed the choice of a barrier strip or XLR type input connection. Inside on the module PC board it was possible to swap the polarity of the XLR connectors to pin 3 "hot" and to add an input transformer, the Jensen JE 8043 or Triad TY-144P. There's no direct provision for lifting the input ground and if this grounding scheme (source grounding) is required then it must be done at the input connector.

It is also inside on this module that the operating mode is set using rocker switches. This is an excellent place for switches that should only be operated when the amp is off and can often be switched by accident on other amps. The amplifier can be switched from either two-channel operation (stereo), parallel mono (sending one input signal to both channels or sending a signal in one connector and loop through to the next amp from the other input connector) or bridged mono mode, which couples the amplifier's output stage together, to deliver up to 1600 watts into 4 ohms. Although the manual discusses this power/load combination, to meet with UL approval, the amp's rear panel states that an 8 ohm load is the minimum for the bridged mode and 4 ohms is the minimum load for stereo (two-channel) operation. The switch and jumper positions are clearly indicated on a label inside the module to allow field personnel to make these changes quickly and accurately. There is also space on the PC board to install resistor pads to allow the sound system gain structure to be preset internally and leave the front panel controls set to 0 dB.

> **The EX 1600 has the best combination of output connectors I have seen on an amplifier.**

The Open Input Architecture module system could allow for many interesting possibilities in compact system design and control when the modules become available. QSC had designed these modules for computer control and monitoring of the amplifier status but will soon also offer other types of modules. Other manufacturers have similar systems in some of their models. I really like being able to preset limiters and crossovers within the amps for a system that is much more compact and less likely to be mis-adjusted in the field than one that uses outboard units to achieve these functions.

Inside the amplifier are the main PC board and a few small PC boards mounted at 90 degrees to it. Everything seems to be thoughtfully constructed and securely mounted, including the semi-toroidal power transformer near the front panel, to create a very road-worthy unit.

The amplifier has the best combination of output connectors I have seen on an amplifier. There are the old standard five-way binding posts, which are positioned to allow a dual-banana connector to be used for either two-channel operation or bridged-mono output. There are also Neutrik Speakon NL4MP connectors, one for each channel and a third that can function for dual output cables or the bridged-mono output connection. This is a very versatile system of connection. The Neutrik Speakon connector is both robust and inexpensive, two features that were lacking in previous loudspeaker connectors, and they are also capable of handling large diameter cable and have contacts with high current ratings. Their main disadvantage is that a separate Neutrik NL4MM connector is required to extend cables. The wiring of the Speakon connectors follows the NL4 conventions and is also displayed near each connector. The amplifier is supplied with an excellent manual that covers everything you would need to know about using the unit.

### THE SOUNDS OF SILENCE

The audible qualities of the EX 1600 compare very favorably to many other amplifiers, including a few that lack the QSC level of amp and driver protection. I compared this amp to a number of other brands including some that are famous for being used in critical monitoring applications. The EX 1600 was just as colorless and quiet as any of the other amps, its low frequency response was subjectively more extended and subsequently slightly less "tight" sounding when directly compared to some amps. I used the amp with a number of different loudspeakers from hyper-critical Tannoy and UREI studio monitors to the very efficient EV MT4 sound reinforcement system. The EX 1600 never complained even when driving four 18-inch subwoofers in parallel from one channel at full power. The unit is capable of driving many of the common reinforcement loudspeaker systems while still providing a little headroom. There are obviously situations where the output power of larger amps such as the EX 4000 may be required but in the main the EX 1600, at 600 watts per channel into 4 ohms, will be quite sufficient. This is an amplifier that can work hard in the recording studio or sound reinforcement system and not even break into a sweat.

*—Wade McGregor*

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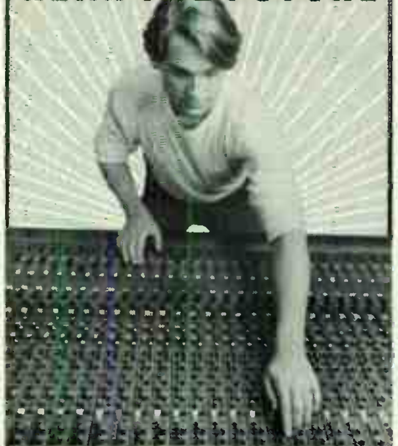

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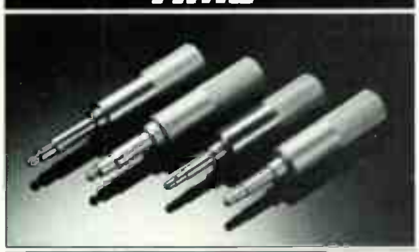

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# IN REVIEW

# DigiTech Vocalist VHM5

EQ LAB REPORT

**MANUFACTURER:** DigiTech, 5369 South Riley Lane, Salt Lake City, UT 84107; tel. 801-268-8400.

**APPLICATION:** Synthesizes extremely natural sounding harmonies (up to five parts total) from a monophonic voice input, without Munchkin/Darth Vader effects.

**SUMMARY:** The Vocalist is dedicated exclusively to solo voice processing and does that one application extremely well. It has a high "fun factor," but is nonetheless a serious, professional tool.

**STRENGTHS:** Excellent sound quality, relatively easy to use, small size, comprehensive MIDI implementation.

**WEAKNESSES:** Only works with voice. Requires some understanding of music theory if you really want to master it.

**PRICE:** $850 **EQ FREE LIT. #:** 135

DIGITECH HAS PURSUED PITCH-shifting technology more aggressively than any other company. In conjunction with IVL Technologies of Victoria, Canada (which does design work to DigiTech's specs), DigiTech has created units for general applications (see the IPS 33B review in the April 1991 *EQ*) as well as devices specifically for guitar, bass and, now, vocals. The Vocalist's main claim to fame is being the first pitch shifter to overcome the Munchkin/Darth Vader effects usually associated with pitch transposition — but that's not all there is to the story.

**FACTS AND SPECS**

The Vocalist is designed exclusively for generating up to five-part harmonies from a single voice input — no polyphonic inputs or other instruments need apply (although I have heard of sax players who use the Vocalist). It's packaged in a small, tabletop case (with an external AC adapter) whose small size belies the high technology inside. Inputs are XLR mic in and 1/4-inch phone line in, with a +4/-10 dB operation switch. Outputs include a line out that carries the dry signal, headphone jack and right and left outs (each carries two harmony lines, along with a variable amount of the input signal; for mono operation, the left carries all lines). There's also MIDI in, out, and thru to allow not just for program changes but to trigger/control the harmonies being generated.


The front panel consists of sliders for input signal, vocal level and amount of effect, along with 21 large, positive action "rubber top" pushbutton switches. There are two displays, LED for program number and LCD for programming. Programming uses the standard process of selecting a program to modify or create, selecting parameters, and altering parameter values. If you've programmed other DigiTech devices, you pretty much know how to program this one.

The Vocalist offers "intelligent" harmonization that follows musical harmony rules instead of just creating a parallel harmony. There are two main harmonization schemes: chord harmony, which generates any of eight common chord types (major, major 7, minor, etc.) in response to an incoming note, and scalar harmony, where you specify a key and scale type from six options (chromatic, major, minor, blues, whole-tone, and diminished).

Some of the 128 factory preset harmonies have titles like "Beach Boys," "Mamas and Papas," "Barbershop," etc. to give you an idea of the intended application; you also have 128 user-editable presets. You can input the chord or scale information from the front panel via 12 keys (arranged with the same spacing as a standard C-B keyboard), by analyzing notes present at the MIDI input (a very powerful technique), or by programming preset numbers in advance as a 32-step song list and stepping through changes with an optional footswitch.

Although harmonization is the star of the show, there are several other nifty options: pitch correction (which "quantizes" your voice to the nearest semitone), chorus, portamento, pitch randomization, and vibrato. The latter two are particularly useful in creating realistic vocal effects. You can also cue yourself with a brief pitch reference by pressing one of the keyboard keys. The "vocoder" effect is not what you're expecting but it's cool anyway: trigger the Vocalist from a MIDI keyboard, and your voice shifts to the notes you play.

**APPLICATIONS**

•**Project studios**. If you're running out of tracks for vocal harmonies or don't have backup singers on call, the Vocalist is the ticket. It's possible to get really lush-sounding, big-group vocals even out of solo performers. Those who sing along with virtual tracks direct to DAT to avoid using a multitrack tape machine will now be able to get great harmonies in real time. Of course, the Vocalist can process taped vocals just as easily.

•**Live**. If you want to fill out a tune, this is a singer's dream come true. Bands that use sequencers or MIDI keyboards will be particularly tickled that the Vocalist can analyze the chord notes and create correct harmonies. (Live, I often sequence a "phantom" bass part that cannot be heard, but sets up the required harmony in real time.)

•**General vocal processing**. Although the Vocalist lacks EQ and reverb, the chorus, vibrato, and pitch correction options make this box useful even when it's not doing harmonies.

•**Vocoding**. While not really comparable to dedicated units, for quick vocoder-like effects the Vocalist does a credible job.

•**Automated mix functions**. The Vocalist harmony lines can respond to MIDI continuous controller messages, so you can automate fades, swells, and so on during mixdown. MIDI controllers can also control harmony detuning, vibrato delay, vibrato speed, vibrato depth, pitch randomization and portamento speed.

**FINDINGS**

The eight-word bottom line: "To know the Vocalist is to love it." Under close scrutiny you can tell it's still a machine, but there's no question that the Vocalist is an order of magnitude beyond any other device at creating natural-sounding harmonies, even with wide transpositions. Sing "Amazing Grace" with the Gospel patch selected, and you'll wonder where you grew the extra four heads. The effect is spectacular.

My favorite chord selection method is MIDI control. Just feed in a sequencer track or play a chord progression on a MIDI keyboard, and solo voices blossom into rich harmonies. Using MIDI also bypasses much of the learning curve; the Vocalist will adapt the harmonies to the chords you play — you can't get any simpler than that. About the only thing I miss from the IPS 33B is the ability to delay harmonies, but you can always do this with a delay line at the Vocalist's harmony outputs.

The Vocalist is one of those rare devices that's instantly likable and almost instantly usable, but with enough depth to keep you busy long after the initial giggling and amazement has worn off. This is as close as it gets to turning the human voice into a polyphonic instrument.

*—Craig Anderton*

# QSC EX 1600

EQ LAB REPORT

**MANUFACTURER:** QSC Audio Products, Inc., 1926 Placentia Ave., Costa Mesa, CA 92627; tel. 714-645-2540

**APPLICATION:** The QSC EX 1600 is in the middle power range of the new EX series. Capable of up to 1600 watts (into four ohms, bridged), it requires only two rack spaces and incorporates some very sophisticated protection circuitry.

**SUMMARY:** The unit is capable of driving many of the common reinforcement loudspeaker systems while still providing a little headroom. This is an amplifier that can work hard in the recording studio or sound reinforcement system without even breaking into a sweat.

**STRENGTHS:** High quality amplification in two rack spaces.

**WEAKNESSES:** Optional input modules are not yet available.

**PRICE:** $1498

**EQ FREE LIT. #:** 136

POWER AMPLIFIERS, AT THEIR BEST, form a transparent link between a line level audio source and loudspeaker. Amplifier models are attributed sonic signatures and are often surrounded by the same type of subjective assessments that are used to describe their counterpart at the other end of the signal chain, the mic pre-amp. Power amps are often the active electronic device nearest the transducer and with typical gain range of between 20 and 40 dB, the power amp is second only to the mic pre-amp in adding level to the signal chain.

There are basic criteria that an ideal power amplifier should meet — add the appropriate gain to the signal and otherwise be entirely inaudible while also providing reliable operation under all load and signal conditions. In live sound applications, we require high output power, often running multiple drivers connected by long cables. We also need features to protect the loudspeaker components; these may include; limiting just before clipping occurs, filtering out RF and sub-sonic signals, and detection of fault conditions within the amplifier. Many amplifier designs have provided these protections but some have suffered, audibly, to achieve them.

The QSC EX 1600 is in the middle power range of the new EX series of amplifiers. It is capable of up to 1600 watts (into four ohms, bridged) but requires only two rack spaces. It also incorporates some very sophisticated protection circuitry to prevent this considerable output power from causing system problems.

**THE PARTICULARS**

The EX 1600 is compact but at 44 pounds it would not qualify as an ultralight amplifier. The nearly 18-inch deep steel rack mount case includes the ability to add optional rear supports for portable racks. The case includes a ventilation system that fans air in from the rear and out at the top of the front panel. This allows amplifiers to be densely packed into a rack without needing to leave space for air flow between units. The use of a very compact semi-toroidal AC power transformer, which contains it's hum field better than other types, also contributes to being able to put more amps and processing in less rack space.

The front panel has minimal controls, a power switch and two detented gain controls. Powering up or down will mute the output until the amp has stabilized. This will prevent the amplification of transients from equipment prior to the amp causing component damage. The input gain controls are labeled in dB of attenuation and the detents match the marking and levels very accurately. This allows settings to be repeated accurately and as these are not stepped attenuators, it's possible to set gains between the detents if required. I measured the tracking of the input gain to be within 0.6 dB of the labeling and gain match between channels of better than 0.3 dB.

The reduced operational AC current of this class G circuit is provided by a two-step, high-efficiency design.

The EX 1600 uses a FET-switched. two-stage DC supply that switches between output devices using +/- 57 volt to +/- 97 volt supply rails depending on the demand of the input signal.

There are also LEDs to display the output level (-30 dB, -10 dB), limiter operation and protection mode. This display uses flashing LEDs to show the onset of clipping distortion, at which point the limiter circuit begins to slightly reduce amplifier gain internally. When the LED indicator is fully displayed, a full 15 dB limiting is occurring due to excessive input levels. This allows the amplifier to operate without reducing the maximum headroom and still limit the output level and distortion products produced when it is driven into clipping.

The cooling fan does not run until the amp exceeds typical room temperature. In many applications, such as control room monitoring the amplifier's cooling fan might never turn on. The only acoustic noise typically produced by this amp is the just perceivable hum of the AC power transformer. The amp's cooling fan will increase speed as the amp's temperature rises above 50°C, until reaching 80°C, where the amp will begin to protect itself by limiting gain. If the temperature rises beyond 85°C, then the amplifier will mute the input. This condition is expected to occur only when serious problems exist. The fan didn't turn on for several minutes while driving an EV MTL-4 bass bin at full power from one channel of the amplifier (but, was that ever LOUD).

### OUTER LIMITER

The limiter limits the distortion products enough to protect drivers from excessive high frequencies but is audible enough to make the operator (mixing engineer) aware that full power has been exceeded. The amplifier also incorporates protection against dangerous input signals, such as those that include significant levels of RFI or high

*continued on page 102*

# Peavey SP

**MANUFACTURER:** Peavey Electronics Corp., 711 A Street, PO Box 830, Meridian, MI 39302.; tel. 601-483-5365.

**APPLICATION:** The SP (Sample Playback) is designed to work with the SX, Peavey's 16-bit sampling unit ($349). Together, the two make a highly affordable sampling and playback combo that still offers serious specs.

**SUMMARY** An inexpensive, one-rack-space sample playback unit with expressive programming and SCSI interface, Peavey's new sample playback unit takes its flexible architecture philosophy to new frontiers.

**STRENGTHS:** Very good sound.

**WEAKNESSES:** Only four outs and only three-stage envelopes.

**PRICE:** $999 **EQ FREE LIT. #:** 137

For the past few years, Peavey has been touting a philosophy of software-based design in its instruments. The idea is that upgrading software is easier and less costly than designing new hardware, so upgrades can get into customers' hands faster and less expensively. So far, Peavey seems to be as good as its word, already having put a major new system into the DPM 3, and spinning off lower-cost little brothers and cousins of the flagship synth. With the release of the SP, a single-rack-mount, programmable sample playback unit, the company moves the flexible-architecture philosophy to new frontiers.

The SP is designed to work with the SX, Peavey's 16-bit sampling unit ($349). Together, the two make a highly affordable sampling and playback combo that still offers serious specs. With 16-bit sampling at rates up to 48 kHz, the SX brings good sounds to the party; the question is, can the SP get the crowd on its feet when it's time to dance? Peavey says the SP is intended to give musicians a low-cost introduction to sampling, but while it's easy to see where costs have been cut, the unit has some top-notch specs, excellent sound and some intriguing features to boot.

**BASICS**

The SP can load and play 16-bit samples with 16-voice polyphony, and can play up to 16 multitimbral parts over 16 MIDI channels. It comes with two megabytes of memory, and is expandable to 32 MB, using the same inexpensive, off-the-shelf SIMMs the Apple Macintosh uses. It has a high-density, 1.44 MB floppy disk drive; MIDI in, out and thru; two 25-pin SCSI ports and four audio outs.

While the SP and SX were designed to complement each other, you do have an option: the two units are sold separately, and the SP doesn't need the SX. It can also load samples from a number of other sources. Putting the sampling and the playback functions into two different units is an idea as good as making synthesizers without keyboards, and can only save the musician money. (On the other hand, given the prices of the SX and SP, even combined they constitute about the most affordable 16-bit sampling package available.)

The SP's other sources of samples include floppies, any MIDI device that supports the MIDI sample dump standard, and SCSI devices. MIDI sample dump devices include the SX, Peavey's DPM 3, and many other samplers and computer-based sample-editing programs such as Alchemy or Sound Designer. For the cost of an SCSI cable, the SP's SCSI ports not only provide access to the same devices as MIDI — with transfer rates of about fifty times as fast — but also to such external storage devices as hard disks and the popular Syquest removable hard-disk drives. Since the SP has two SCSI ports, it fits nicely into a setup that would include the SX, as well as a computer and storage device.

**SOUND ARCHITECTURE**

The SP organizes its sounds in a clear and logical way. Each sample is called

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a "wave." To do analog-synth-style programming to a wave, you put it into a "tone," which you then map into "zones," covering specifiable ranges of the keyboard. Up to four maps'-full of zones can be arranged in a layer, and two layers make a preset. You can switch among different maps and layers by crossfading, switching, and using a number of MIDI controllers. This structure, combined with an extensive set of note allocation setups, provides the possibilities for the SP to create some sophisticated patches.

Once a sampled wave is brought in, there are several options controlling its playback. You can edit start and end points, trim the wave and create (or import) a loop. The SP, by the way, only allows one kind of loop — a "forward" loop — and only one per sample. It has no crossfade or other kinds of loops. To help smooth the loop, though, the SP will automatically set loop points at zero crossings and has a loop fraction control to fine-tune the loop length. While these help, for truly invisible loops, I still see no alternative to a good, computer-based sample-editing program.

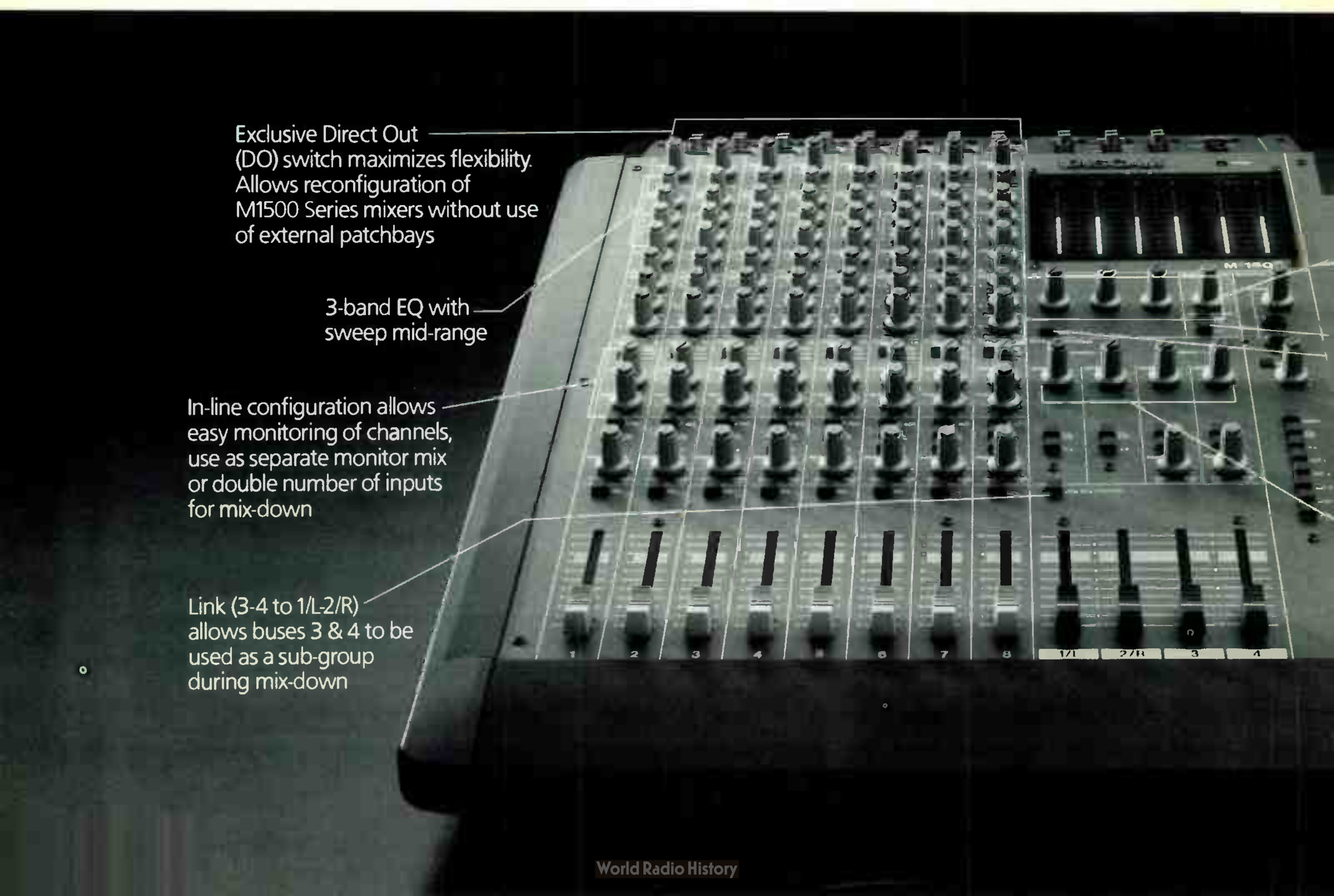
Unfortunately, as of this writing there is no Alchemy driver for the SP, but Passport promises to have drivers by the January NAMM. This means you cannot now edit a whole bank of sounds on the Mac, then send them to the SP in one shot. You have to send one, wait, send the next and so on. One 170K sample took one minute, 41 seconds to load via MIDI. SCSI takes significantly less and should be considered de rigeur by anyone doing serious sampling (but still works in the one-at-a-time mode).

The SP has a nifty setup for combining two mono waves into one stereo sound, including ways to adjust start points, and a stereo merge function that's quite easy to use. You can also tune a wave in increments as fine as a cent.

In general, samples I brought in sounded very good on the SP, and I found the unit to transpose remarkably well. I even played some samples as much as two octaves below their original pitch without hearing clock noise, and transposing upwards produced as little Munchkinization as I can recall on any sampler.

**PROGRAMMING**

Once waves are trimmed and looped, you put each into a "tone" to take advantage of the SP's extensive programming features. A tone has an editable, digitally controlled oscillator, low pass filter, amplitude, two



envelope generators and an LFO.

The SP has two envelopes which you can assign to amplitude, filter, pitch or pan and set to one of two modes: gate and trigger. In trigger mode, pressing a key causes the envelope to go through its stages, regardless of when or whether you release the key. The gate mode is the more familiar, where a note will decay according to the envelope's decay setting until you release the key, when the sound will decay according to the envelope's programmed release time.

Curiously, the envelopes have only three stages: attack, decay and release. There's no programmable sustain level, so the envelope will either decay not at all (infinite sustain), or decay all the way to zero. You can't make them decay to a point in between and then sustain there. Despite that limitation, however, there's a lot of juice here, in the form of three modulation schemes that do a lot.

First, the amplitude and filter cutoff point can be modified by any of the SP's MIDI control sources: mod wheel, pressure or "aux control" — your choice of three MIDI controllers you assign globally. Second, you can choose from this same set of MIDI controls to modulate the depth and speed of the LFO which, in turn, can modulate amplitude, filter cutoff, DCO pitch, and Pan. Finally, the DCO and Pan can each be dynamically modified by a complex interaction of the MIDI controllers, the envelopes and the LFO. For the DCO and Pan, the envelope, velocity, LFO and aux controls can be put in scaled or summed mode. In summed mode, the values of all controllers are added together to affect a given parameter. In scaled mode, the MIDI controllers attenuate a parameter's basic setting. Each mode gives you a noticeably different feel in controlling a note. In particular, the complex ways you can control panning in the SP can give an otherwise turgid sample a lively, dynamic feel.

Adding a dimension to LFO modulation are the eight shapes to which you can set the LFO wave: triangle, square, saw tooth, sine, clipped sine, sample and hold, random square and grunge. The last three in particular can create some entirely unusual effects, including an excellent, digitoid growl I got by assigning grunge to the DCO pitch and bringing it in with the mod wheel. The LFO can also be bipolar (oscillating up and down across an axis point) or unipolar (moving only on one side of the axis).

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# More Dis About DAT!

Photo by Ed Colver

Never believe anybody about anything! Well, unless it's something you heard from me

**BY ROGER NICHOLS**

I'm still having problems transferring digital information across standard AES/EBU or SPDIF interfaces between standard DAT machines. The only machine that will copy anything from anywhere that I've run across is the Fostex D-10. Last week I went over to Masterfonics to test some of these wayward interfaces using their new Audio Precision test gear with digital I/O. Glenn Meadows of Masterfonics in Nashville and I found a lot of interesting information.

As a little refresher, the DAT format dictates that there won't be any DC voltages recorded as information on the tape. DC voltages usually come from an offset voltage that's applied to the analog-to-digital convertor to lower the zero-crossing noise apparent in low level signals. This is usually implemented as either a 2 Hz hi-pass filter in the digital domain, or a DC servo circuit that subtracts out the DC component whenever the audio signal is dormant for a couple of seconds. SCMS bits are the Serial Copy Management System that determines whether or not you'll be allowed to make a copy of your DAT tape. Also, the SPDIF interface for transfer of digital information specifies that a consumer machine must look at the actual data rate of the information to decide whether to record the incoming signal at 48 or 44.1 kHz. The sample rate flag is not supposed to be supported in the consumer interface, only the professional version. One more thing, the physical interface for SPDIF is supposed to be unbalanced, 0.5 volt peak-to-peak, and terminated in an RCA pin type phono jack. The AES/EBU professional interface uses an XLR type connector (I still think the guy who decided it should be an XLR should be taken out back and shot), is a balanced signal (RS-422), and has a level of 1.5 volts peak-to-peak.

Of course there are more specifications involved in the DAT format than I've mentioned here, but these are the only ones that we'll be concerned with here. The rest will be dealt with as I gather more incriminating evidence.

**ON TO THE GOOD PART**

We piled up several DAT machines and various other pieces of digital gear at one end of the room, and placed the Audio Precision and the Lexicon LFI-10 format tester and convertor at the other end. It felt like we were reenacting the Spanish inquisition with electronics.

The machine I wanted to test first was the Akai DD-1000 optical disk recorder. For almost a year now, I've noticed that a signal that contained overs, all 16 bits full, did not contain overs after going to the DD-1000 and back again. According to the Audio Precision, everything going into the DD-1000 is multiplied by 0.9999389639, as close as I can tell. This means that you cannot turn off the DSP when recording digitally into the DD-1000. This math also affected the noise-floor measurements. They were -91 dB instead of the -96 dB that should be expected with a 16-bit device. I told Akai of the problem and they said that the designer of the DD-1000 would get in touch with me. I hope they get it straightened out, 'cause I like the way the DD-1000 sounds.

The first DAT machine to enter the gauntlet was the Panasonic SV-3700 professional. The bit tests were good. The machine records and plays back exactly what you put into it through the digital interface. This includes DC voltages that were recorded on other tapes. This is a good feature. It means that when you do a backup of data from Sound Tools to DAT, all the computer data that's sent before the audio will be faithfully recorded and reproduced by the machine.

The digital interface of the SV-3700 is another story. This machine, which I'm told has become the de facto standard for DAT machines, can't really decide whether it's a consumer machine or a professional machine. The professional features are only available if you record digitally via the balanced XLR input. There's a switch on the back of the unit to select which physical input is operational. The catch here is that the signal entering the SV-3700 via the XLR input must in fact be an AES/EBU formatted signal. If the in-coming data is flagged as consumer, then the SV-3700 will not record the data unless it enters through the consumer connector. This means that you cannot record digitally from an Akai DD-1000 optical-disk recorder, because, as we found out, even though the signal comes out of an XLR connector at RS-422 levels, it's a consumer format signal. The only way to get from the DD-1000 to the SV-3700 is to use an adapter cable, XLR to RCA, and go into the consumer input.

The consumer input. What a mess. The SV-3700 will not record a signal coming in the consumer jack unless the sample rate flag matches the actual sample rate. But the sample rate flag is not supposed to be there in the consumer format. This means that it was really designed to record professional data that has cheated and set

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